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ThinkTel HPBX and Fax config with Grandstream HT81x – Configuration Guide

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Contents

Overview

Welcome to Grandstream HT81x series analog telephone adapter provisioning guide. This document covers the basic steps required to activate your new or existing Grandstream HT81x ATA with a single or multiple ThinkTel HPBX lines.

Please keep in mind that connecting any ATA to a single or multiple HPBX lines the user will have PBX and/or HPBX phone features. These features include:

- Directory or corporate directory features.
- Automated button provisioning,
- Call pickup, etc.

The provisioning guide will cover the following steps:

- Configuring the SIP ATA with ThinkTel HPBX line.
- Configuring the SIP ATA with ThinkTel HPBX line for faxing.

We will be connected the HT81x using the WAN port for our implementation and configuration example.

Requirements

Before you start

1. The installer/support representative will require the configuration credentials.
    a. Username: for the HPBX line.
    b. Password: for the HPBX line.
    c. SIP Domain Name:
        i. HPBX: This would be the provisioning URL.
2. An analog phone for testing inbound and outbound calling.

Connecting the HT81x

The HT818 is designed for easy configuration and easy installation, to connect your HT818, please follow the steps below:

Scenario 1: Connecting the HT81x using the WAN port

When connecting HT81x using the WAN port, it will act as simple DHCP Client.

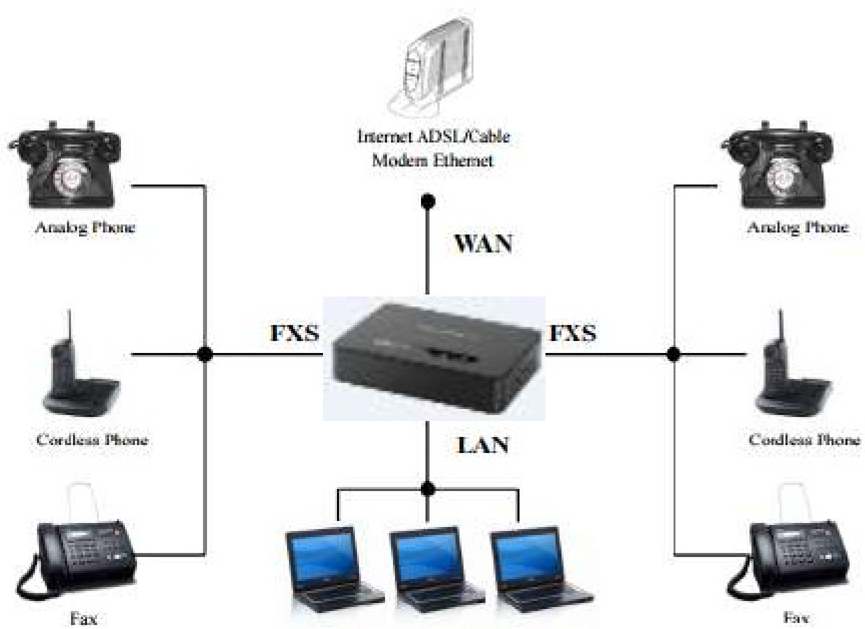
- Insert a standard RJ11 telephone cable into the p hone ports and connect the other end of the telephone cable to a standard touch-tone analog telephone.
- Connect the WAN port of the HT81x to a router, switch or modem using an Ethernet cable.
- Insert the power adapter into the HT81x and connect it to a wall outlet and make sure to respect the technical specifications of the power adapter used.
- Power, WAN and Phone LEDs will be solidly lit when the HT81x is ready for use.

Scenario 2: Connecting the HT 81x using the LAN port

When connecting the HT81x using the LAN port, it will act as a router and DHCP serving addresses, the devices connected with HT81x LAN will pull DHCP addresses from your HT81x.

- Insert a standard RJ11 telephone cable into the p hone ports and connect the other end of the telephone cable to a standard touch-tone analog telephone.
- Connect a computer or switch to the LAN port of the HT81x using an Ethernet Cable.
- Insert the power adapter into the HT81x and connect it to a wall outlet and make sure to respect the technical specifications of the power adapter used.
- Power, LAN and Phone LEDs will be solidly lit when the HT81x is ready for use.

Note: Please make sure to enable NAT Router under Web GUI → Basic Settings →Device Mode.



LED Lights	Status
Power LED	The Power LED lights up when The HT81x is powered on and it flashes when the HT81x is booting up.
WAN LED	The WAN LED lights up when The HT81x is connected to your network through the WAN port.
LAN LED	The LAN LED lights up when The HT81x is connected to your network through the LAN port.
Phone LED 1- 16	The phone LEDs indicate status of the respective FXS port-phone on the back panel: • **OFF** - Unregistered • **ON (Solid Blue)** - Registered and Available • **Blinking every 500 ms** - Off-Hook / Busy • **Slow blinking** - FXS LEDs indicates voicemail

Obtain HT81x IP Address via Connected Analog Phone

HT81x is by default configured to obtain the IP address from DHCP server where the unit is located. In order to know which IP address is assigned to your HT81x, you should access to the "**Interactive Voice Response Menu**" of your adapter via the connected phone and check its IP address mode.

Please refer to the steps below to access the interactive voice response menu:

- Use a telephone connected to phone ports (FXS) of your HT81x.
- Press *** (press the star key three times) to access the IVR menu and wait until you hear "Enter the menu option ".
- Press **02** and the current IP address will be announced.

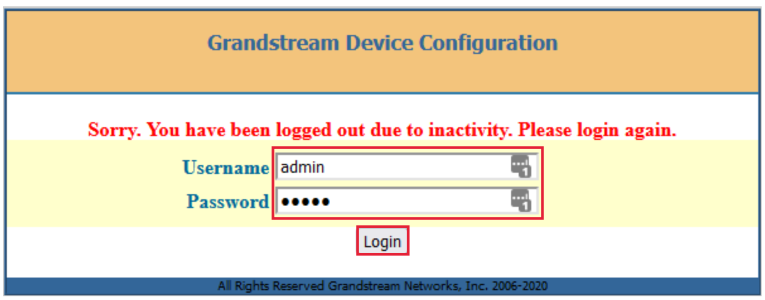
Upgrading the HT81x

Status Page

When the IP address is obtained using the interactive voice response menu of the HT81x. Navigate to the device via the web GUI IP address.

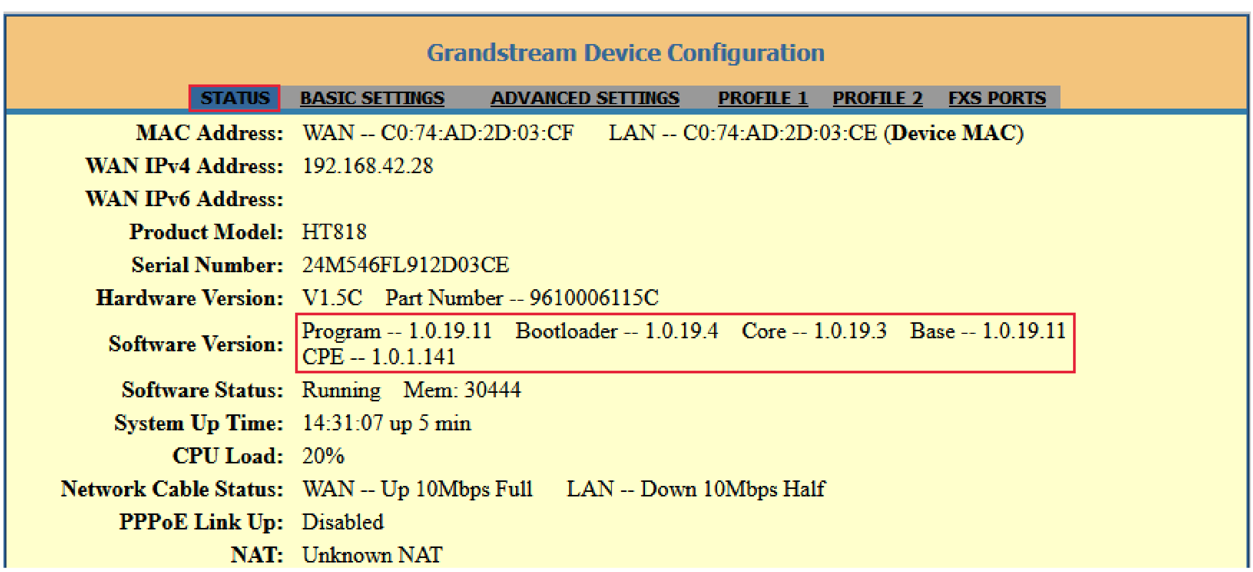
The default credentials are:

- Username: admin
- Password: admin



Log into the HT81x ATA with the default credentials if this is a new device, if it is an existing device use the preconfigured credentials to gain access to the device menu. Once logged into the HT81x ATA the user will be transported to the device stats page.

Once on the status page, look for the Software version of the device, once located navigate to the Grandstream support page and then find the Gateways and ATA's section and/or search for your device.



Once the Gateways and ATA's are located on the Grandstream support site, find your HT81x model and see if there is a new firmware release available for download.

In our example above the device is on 1.0.19.11 and the Grandstream support site has 1.0.13.1 available for download.

NOTE: Always ensure your device has the most current software/firmware installed on it to mitigate any security threats or vulnerabilities and apply any bug fixes to the device for optimum operation.

Gateways and ATA's

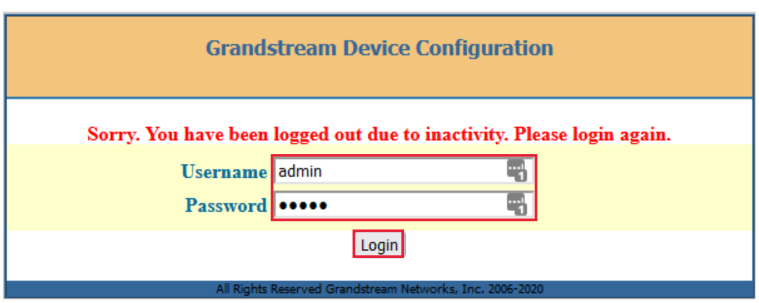
HandyTone ATA's

Model	General Firmware	HTTP Upgrade Server	General Beta Firmware
HT801	1.0.31.1 Release Notes	firmware.grandstream.com	1.0.33.4 BETA
HT802	1.0.31.1 Release Notes	firmware.grandstream.com	1.0.33.4 BETA
HT812	1.0.31.1 Release Notes	firmware.grandstream.com	1.0.33.4 BETA
HT813	1.0.13.3 Release Notes	firmware.grandstream.com	
HT814	1.0.31.1 Release Notes	firmware.grandstream.com	1.0.33.4 BETA
HT818	1.0.31.1 Release Notes	firmware.grandstream.com	1.0.33.4 BETA

To download the firmware just select the release version 1.0.31.1 (in this example) for the specific type of device you are using.

Advanced Settings Page

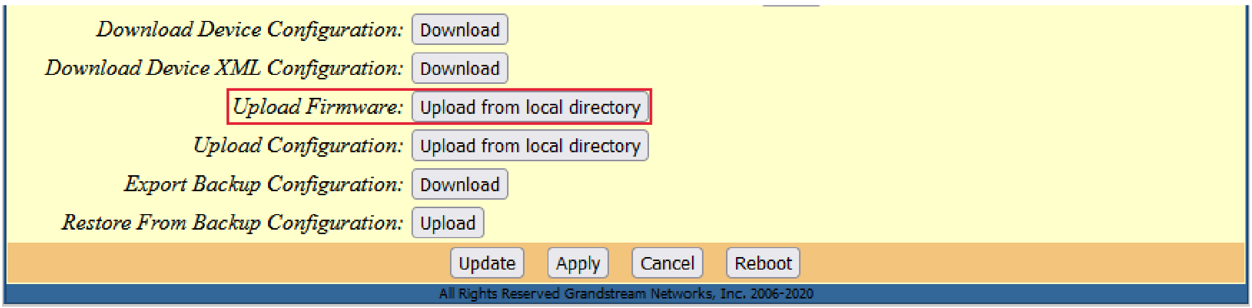
Once the firmware is downloaded navigate to the device and log back into the device.



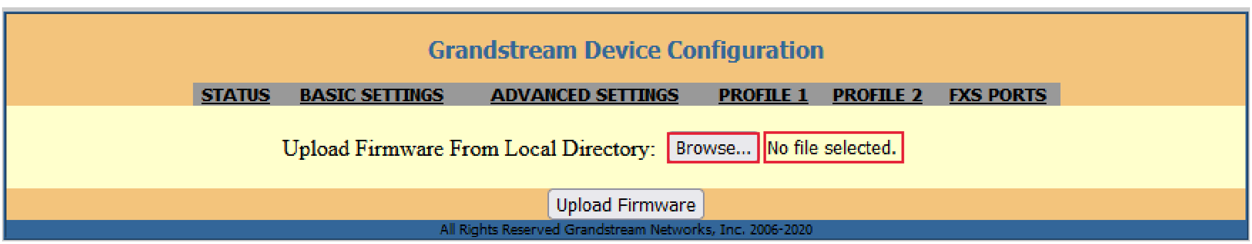
Select the Advanced Settings tab and scroll all the way to the bottom of the screen.

Navigate to the downloaded firmware file and extract the file from its compressed package. This must be done before the firmware file can be uploaded.

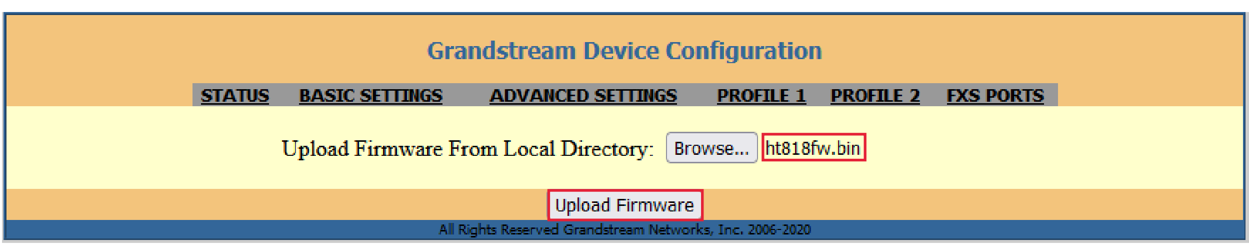
At the bottom of the screen select the Upload Firmware: Upload from local directory button.



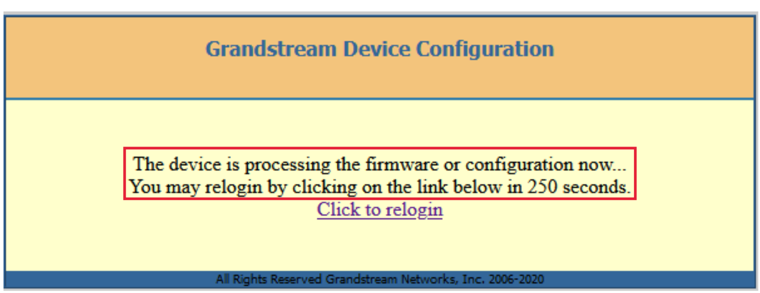
A new window opens, brows to the saved location of the newly extracted firmware file.



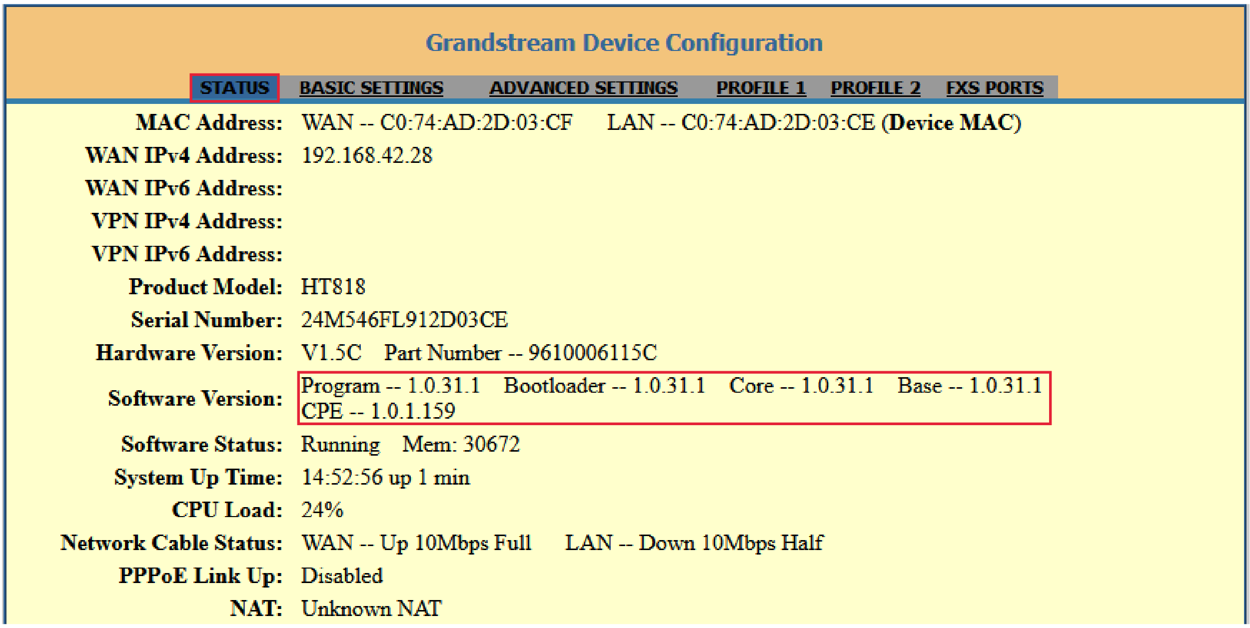
Once the firmware file is attached, select the upload Firmware button to start the upgrade process.



A new window will open informing the installer that the device is being upgraded.



This process takes about 2-5 min to complete. Once the upgrade is completed, refresh the page and log back into the device and confirm the device is upgraded.



Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS PROFILE 1 PROFILE 2 FXS PORTS

MAC Address:	WAN -- C0:74:AD:2D:03:CF LAN -- C0:74:AD:2D:03:CE (Device MAC)
WAN IPv4 Address:	192.168.42.28
WAN IPv6 Address:	
VPN IPv4 Address:	
VPN IPv6 Address:	
Product Model:	HT818
Serial Number:	24M546FL912D03CE
Hardware Version:	V1.5C Part Number -- 9610006115C
Software Version:	Program -- 1.0.31.1 Bootloader -- 1.0.31.1 Core -- 1.0.31.1 Base -- 1.0.31.1 CPE -- 1.0.1.159
Software Status:	Running Mem: 30672
System Up Time:	14:52:56 up 1 min
CPU Load:	24%
Network Cable Status:	WAN -- Up 10Mbps Full LAN -- Down 10Mbps Half
PPPoE Link Up:	Disabled
NAT:	Unknown NAT

Configuring ThinkTel HPBX Line

Profile 1 – Configure ThinkTel HPBX Lines

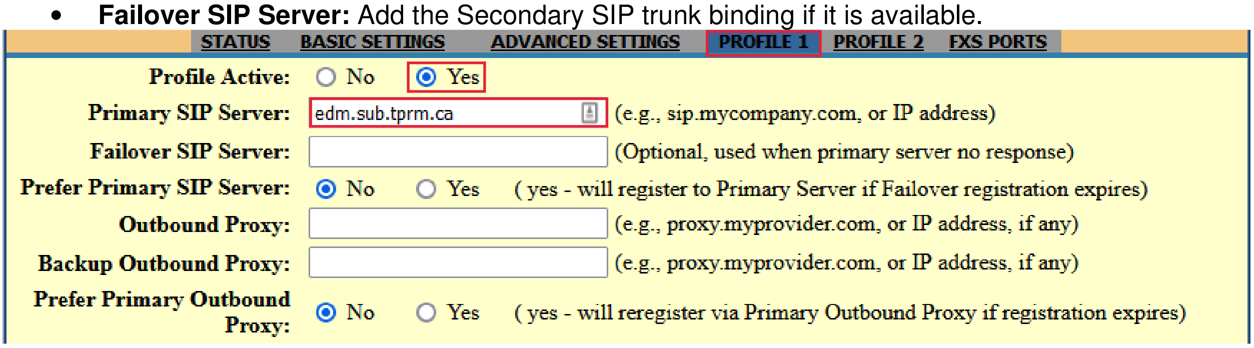
In this section we will configure the Grandstream HT81x with 2 ThinkTel HPBX lines. When configuring the ATA with HPBX lines there is no need for a configuration server as it is configured manually and the HT81x does not require any registration for the device itself, however every HPBX Line will be registered to the ThinkTel Metaswitch.

In our configuration example we will be using 3 HPBX lines and the important thing here is to have the relevant HPBX lines configured with its own preconfigured password.

In our example there is only one binding or SIP Proxy for each HPBX line and thus there is no Secondary SIP Server that will be configured.

Configuration of Profile 1:

- **Primary SIP Server:** Add the primary SIP trunk binding.
- **Failover SIP Server:** Add the Secondary SIP trunk binding if it is available.



STATUS BASIC SETTINGS ADVANCED SETTINGS PROFILE 1 PROFILE 2 FXS PORTS

Profile Active:	No Yes
Primary SIP Server:	edm.sub.tprm.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)
Backup Outbound Proxy:	(e.g., proxy.myprovider.com, or IP address, if any)
Prefer Primary Outbound Proxy:	No Yes (yes - will reregister via Primary Outbound Proxy if registration expires)

- **IP Transport:** Ensure the SIP transport protocol is set to UDP or TCP, this depends on the SIP trunk configuration.
- **NAT Traversal:** Set to Keep-Alive.

Allow DHCP Option 120 (override SIP server):	◉ No ○ Yes
SIP Transport:	◉ UDP ○ TCP ○ TLS (default is UDP)
SIP URI Scheme When Using TLS:	○ sip ◉ sips
Use Actual Ephemeral Port in Contact with TCP/TLS:	◉ No ○ Yes
NAT Traversal:	○ No ◉ Keep-Alive ○ STUN ○ UPnP ○ VPN

- **SIP Registration:** Set the SIP registration to Yes.
- **Register Expiration:** Set the expiration to your desired setting.
- **Options/Notify:** Set it to Options

Use Request Routing ID in SIP INVITE Header:	◉ No ○ Yes
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 (0-64800. Default 0 second)
SIP Registration Failure Retry Wait Time:	20 (in seconds. Between 1-3600, default is 20)
SIP Registration Failure Retry Wait Time upon 403 Forbidden:	1200 (in seconds. Between 0-3600, default is 1200. 0 means stop retry registration upon 403 response.)
Enable SIP OPTIONS/NOTIFY Keep Alive:	○ No ◉ OPTIONS ○ NOTIFY
SIP OPTIONS/NOTIFY Keep Alive Interval:	30 (in seconds. Between 1-64800, default is 30)

- **Local SIP Port:** Set the SIP Port configured on your Contact End Point in uControl.
- **Local RTP Port:** Set the SIP RTP Port or leave the default port

SIP OPTIONS/NOTIFY Keep Alive Max Lost:	3 (Number of max lost packets for SIP OPTIONS/NOTIFY Keep Alive before re-registration. Between 3-10, default is 3)
Layer 3 QoS:	26 SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) 46 RTP DSCP (Diff-Serv value in decimal, 0-63, default 46)
Local SIP Port:	5060 (default is 5060 for UDP; 5061 for TLS)
Local RTP Port:	5004 (even number between 1024-65535, default 5004)

Scroll down to the bottom of the window and select the Update button.

Update Apply Cancel Reboot

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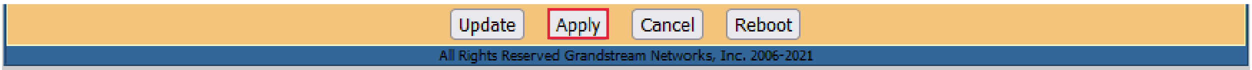
A new message will appear at the top of the window.

Configuration changed. Click Apply to save settings.

Scroll down to the bottom of the window and select the Apply button.



Fax Not Supported Over SIP

Why is Faxing Not Supported Over SIP

With recent updates faxing over sip has become more unreliable the section below will explain why faxes sometime fail over sip.

One advantage of SIP is compression, you can combine data, voice and video, broken into packets, over an SIP network in compressed form to save bandwidth and, usually, maintain acceptable quality. With a VoIP phone call, dropping packets is annoying, but the connection survives.

Faxes, however, cannot be compressed, and it takes more bandwidth to send a fax over SIP than a regular phone call. Worse, even a small amount of packet loss along the way can cause the entire fax to fail. The longer the fax, the less likely it is to be transmitted successfully. And that's only one of many problems fax has with SIP.

That is why we do not support faxing over sip and recommend placing critical fax applications on plain old telephone lines (also known as a POTS line) or making use of a cloud-based fax service that ThinkTel offers, since faxing is known to be problematic on SIP networks.

Why Fax Can Fail over VoIP/SIP.

1. **Packet delay, jitter, and loss:** While mostly reliable, VoIP (also known as SIP trunking) can delay or drop a packet over the duration of the call, by design. Important to remember that packet loss as low as 1% can cause a fax transmission to fail.
2. **Protocol incompatibility:** For example, many VoIP services use the G.729 protocol to compress voice calls; since faxes cannot be compressed, they require the standard-rate G.711 protocol. The brief gaps in fax tones that occur as the system tries to negotiate between the two protocols can cause the fax to fail.

Even Fax-over-IP protocols have inefficiency and reliability problems.

- **G .711** - Considering that the G.711 protocol is actually designed to digitize voice, converts fax tones into a digital signal at 64kbps. Packetizing the G.711 signal adds SIP overhead totaling 88Kbps, which is 38% more bandwidth than a standard voice call and 175% more than a VoIP call compressed to 32Kbps.
- **T.38** - The newer T.38 protocol was intended to transmit faxes directly over SIP (FoIP), so the fax doesn't need to be converted to an audio stream first. But T.38 must be on both ends of a network to work, and many service providers never implemented the protocol. If the fax has to traverse networks that do not support T.38, it will need to be transcoded, which can add latency, increase cost, and may cause the call to disconnect.

Why is Faxing Not Supported Over SIP

Grandstream have updated their analog gateway hardware and software to allow customers who want to keep the cost down and add a fax service at a lower cost to be able to do so, however there is still no guarantee that any fax transmissions will have a 100% success rate all the time.

Description:

The HT818 is a powerful 8-port VoIP gateway with 8 FXS ports and an integrated Gigabit NAT router. Built for users looking for a strong analog-to-VoIP converter, it features Grandstream's market-leading SIP ATA/gateway technology with millions of units successfully deployed worldwide. This powerful gateway carries exceptional voice quality in various application environments, strong encryption with unique security certificate per unit, automated provisioning for volume deployment and device management, and outstanding network performance for enterprise use.

Features:

- Supports 2 SIP profiles and 8 FXS ports
- Strong AES encryption with security certificate per unit
- Automated & secure provisioning options using TR069
- 3-way voice conferencing per port
- Exceptional voice quality with wide-band HD codec
- Supports T.38 Fax for reliable Fax-over-IP
- Supports dual Gigabit network ports
- High performance NAT router

Configuring ThinkTel HPBX for Fax

Profile 2 – Configure HPBX Fax Line

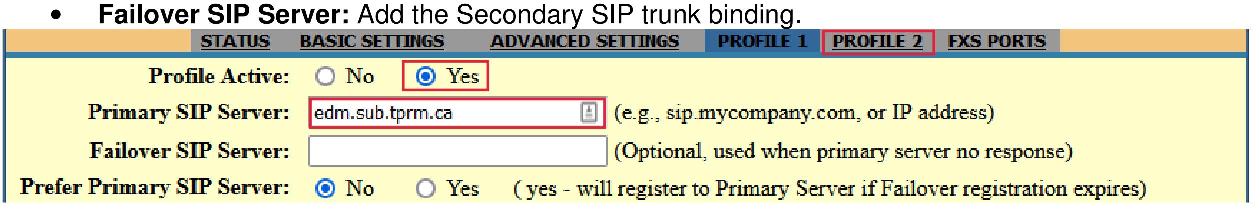
In this section we will configure the Grandstream HT81x with 1 ThinkTel HPBX line for faxing.

In our configuration example we will be using 1 HPBX lines and the important thing here is to have the relevant HPBX line configured with its own preconfigured password.

In our example there is only one binding or SIP Proxy for each HPBX line and thus there is no Secondary SIP Server that will be configured.

SIP Settings

- **Primary SIP Server:** Add the primary SIP trunk binding.
- **Failover SIP Server:** Add the Secondary SIP trunk binding.



- **IP Transport:** Ensure the SIP transport protocol is set to UDP or TCP, this depends on the SIP trunk configuration.
- **NAT Traversal:** Set to Keep-Alive.

Allow DHCP Option 120 (override SIP server):	No Yes
SIP Transport:	UDP TCP TLS (default is UDP)
SIP URI Scheme When Using TLS:	sip sips
Use Actual Ephemeral Port in Contact with TCP/TLS:	No Yes
NAT Traversal:	No Keep-Alive STUN UPnP VPN

- **SIP Registration:** Set the SIP registration to Yes.
- **Register Expiration:** Set the expiration to your desired setting.
- **Registration Expiration:** 5 min
- **Options/Notify:** Yes

Use Request Routing ID in SIP INVITE Header:	No Yes
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 (0-64800. Default 0 second)
SIP Registration Failure Retry Wait Time:	20 (in seconds. Between 1-3600, default is 20)
SIP Registration Failure Retry Wait Time upon 403 Forbidden:	1200 (in seconds. Between 0-3600, default is 1200. 0 means stop retry registration upon 403 response.)
Enable SIP OPTIONS/NOTIFY Keep Alive:	No OPTIONS NOTIFY
SIP OPTIONS/NOTIFY Keep Alive Interval:	30 (in seconds. Between 1-64800, default is 30)

- **Local SIP Port:** Set the SIP Port configured on your Contact End Point in uControl.
- **Local RTP Port:** Set the SIP RTP Port or leave the default port

SIP OPTIONS/NOTIFY Keep Alive Max Lost:	3 (Number of max lost packets for SIP OPTIONS/NOTIFY Keep Alive before re-registration. Between 3-10, default is 3)
Layer 3 QoS:	26 SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) 46 RTP DSCP (Diff-Serv value in decimal, 0-63, default 46)
Local SIP Port:	5060 (default is 6060 for UDP; 6061 for TLS)
Local RTP Port:	6004 (even number between 1024-65535, default 6004)

- **Allow Incoming SIP Messages from SIP Proxy Only:** 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
Use P-Access-Network-Info Header:	No Yes
Use P-Emergency-Info Header:	No Yes

Feature Settings

- **Disable Call-Waiting:** Yes
- **Disable Call-Waiting Caller ID:** Yes
- **Disable Call-Waiting Tone:** Yes

RFC2543 Hold:	○ No	◉ Yes
Disable Call-Waiting:	○ No	◉ Yes
Disable Call-Waiting Caller ID:	○ No	◉ Yes
Disable Call-Waiting Tone:	○ No	◉ Yes
Disable Connected Line ID:	◉ No	○ Yes

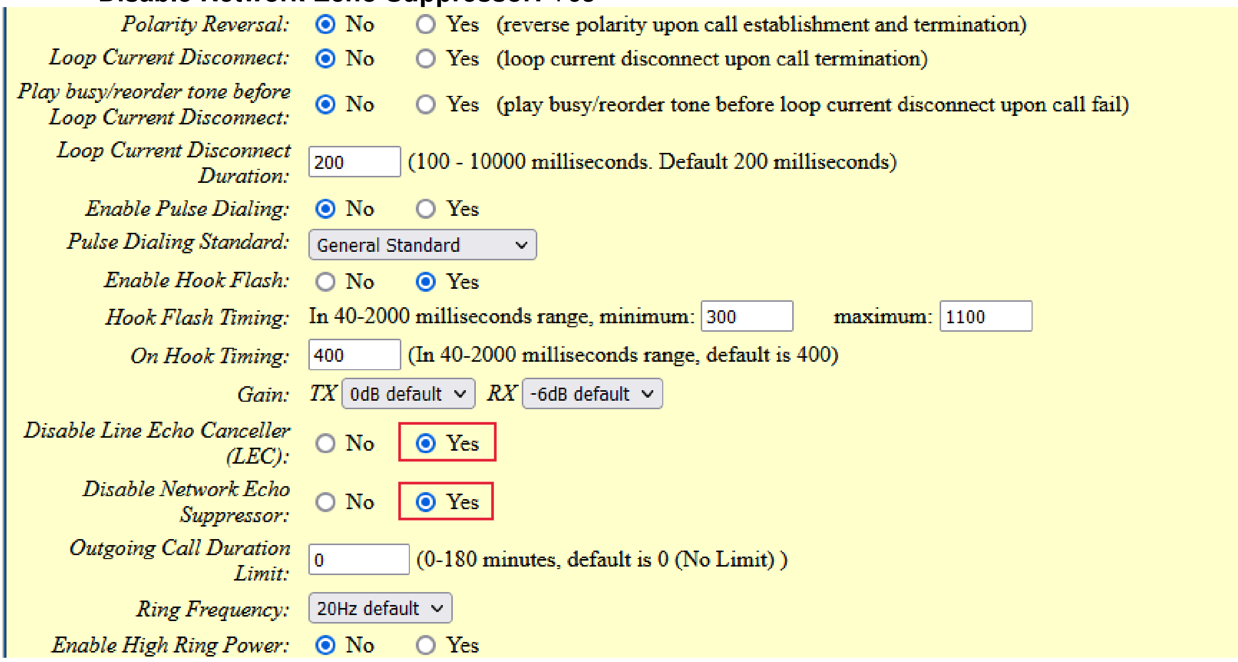
Fax Settings

- **Use First Matching Vocoder in 200OK SDP:** Yes (PCMU = G.711)
- **Fax Mode:** T.38
- **Re-INVITE After Fax Tone Detected:** Enabled
- **Jitter Buffer Type:** Fixed
- **Jitter Buffer Length:** High

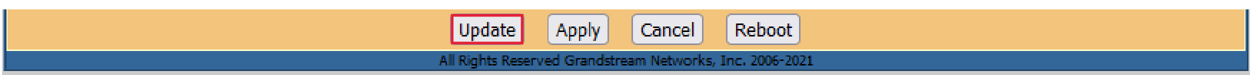
Use First Matching Vocoder in 200OK SDP:	○ No ◉ Yes
Preferred Vocoder: (in listed order)	choice 1: PCMU choice 2: PCMU choice 3: G723 choice 4: G729 choice 5: G726-32 choice 6: iLBC choice 7: OPUS choice 8: G722
Voice Frames per TX:	2
G723 Rate:	◉ 6.3kbps encoding rate ○ 5.3kbps encoding rate
iLBC Frame Size:	◉ 20ms ○ 30ms
Disable OPUS Stereo in SDP:	◉ No ○ Yes (removes "/2" from offer)
iLBC Payload Type:	97 (between 96 and 127, default is 97)
OPUS Payload Type:	123 (between 96 and 127, default is 123)
VAD:	◉ No ○ Yes
Symmetric RTP:	◉ No ○ Yes
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

SLIC Setting:	USA 1 (BELLCORE 600 ohms)
Caller ID Scheme:	Bellcore/Telcordia
DTMF Caller ID:	*Start Tone* Default *Stop Tone* Default
Disable Unknown Caller ID:	◉ No ○ Yes
Replace Beginning '+' with 00 in Caller ID:	◉ No ○ Yes
Number of Beginning Digits to Strip from Caller ID:	0 (between 0 and 10, default is 0.)

- **Disable Line Echo Canceller (LEC):** Yes
- **Disable Network Echo Suppressor:** Yes



Scroll down to the bottom of the window and select the Update button.

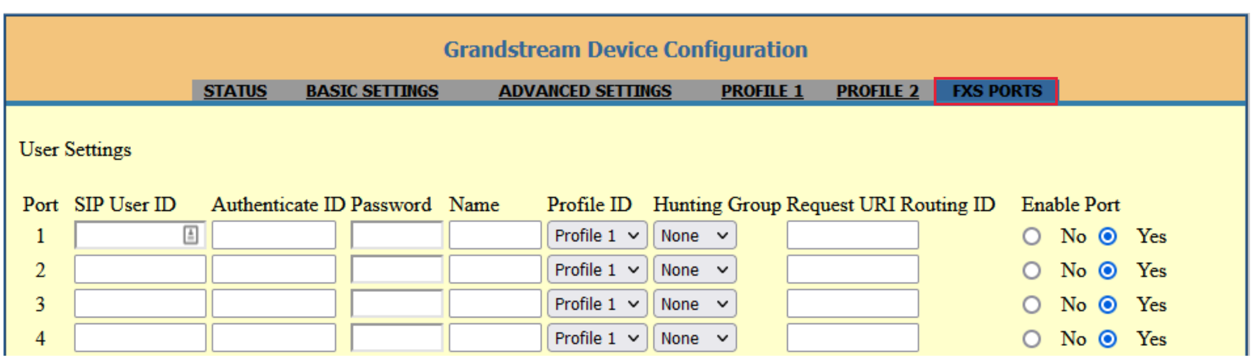


A new message will appear at the top of the window.

Apply the same changes to Profile 2. The reason this is done is to allow for the configuration of different profiles for different ports. This is helpful when configuring Fax ports and it requires different settings than the standard voice ports.

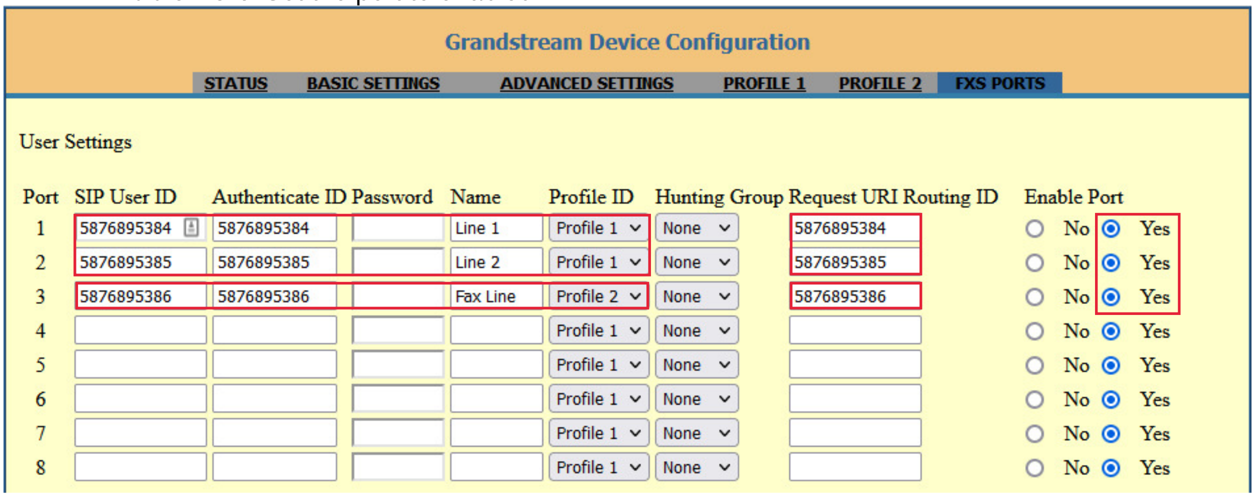
FXS Ports Page

In this section the configuration of the actual ports will be completed. Navigate to the FXS Ports tab to start the configuration of the device ports.



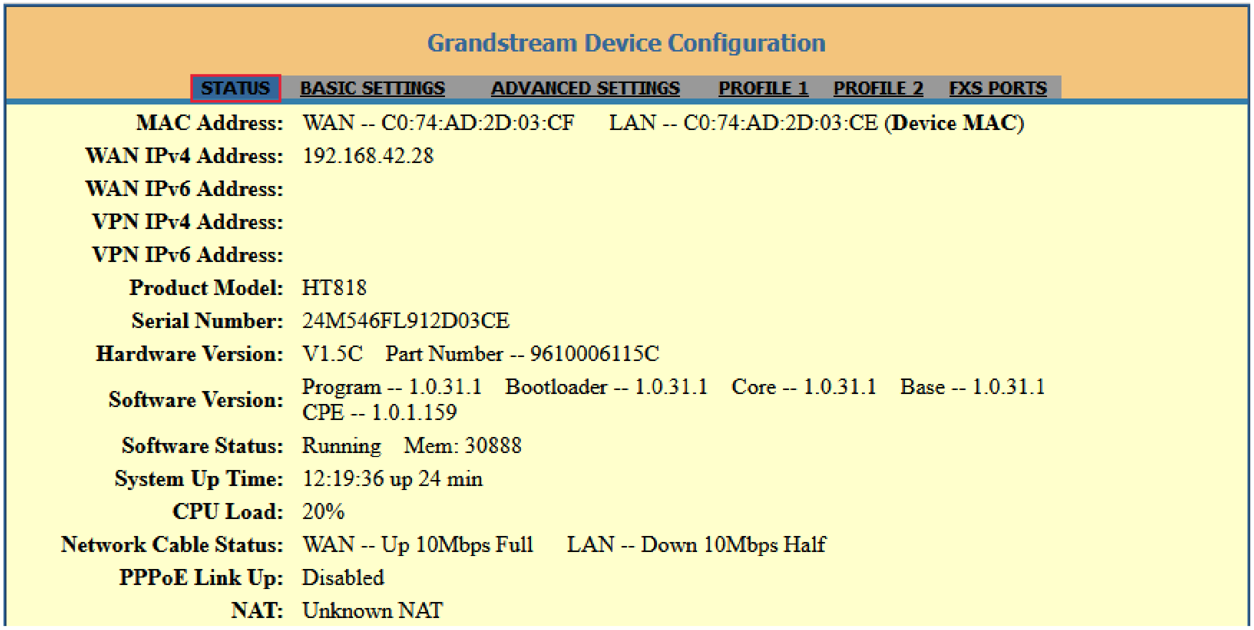
For this example, there will only be one trunk configured.

- **SIP User ID:** Enter the SIP trunk phone number.
- **Authentication ID:** Enter the SIP trunk phone number.
- **Password:** Enter the SIP trunk password.
- **Name:** Enter the name or number to display when dialing out.
- **Profile:** Select the desired profile for this port.
- **Request URI Routing ID:** Enter the SIP trunk phone number.
- **Enable Port:** Set the port to enabled.



Status Page

In this section the installer can determine the registration status of each port. From the status displayed on FXS 1, 2 and 3 it is notable that the ports have successfully registered. The ports will now be able to receive and make calls using the ThinkTel HPBX lines.



Port Status:

Port	Hook	User ID	Registration
FXS 1	**On Hook**	**5876895384**	**Registered**
FXS 2	**On Hook**	**5876895385**	**Registered**
FXS 3	**On Hook**	**5876895386**	**Registered**
FXS 4	**On Hook**		**Not Registered**
FXS 5	**On Hook**		**Not Registered**
FXS 6	**On Hook**		**Not Registered**
FXS 7	**On Hook**		**Not Registered**
FXS 8	**On Hook**		**Not Registered**

Port Options:

Port	DND	Forward	Busy Forward	Delayed Forward	CID	Call Waiting	SRTP
FXS 1	No				Yes	Yes	No
FXS 2	No				Yes	Yes	No
FXS 3	No				Yes	No	No
FXS 4	No				Yes	Yes	No
FXS 5	No				Yes	Yes	No
FXS 6	No				Yes	Yes	No
FXS 7	No				Yes	Yes	No
FXS 8	No				Yes	Yes	No

CDR File: None

SIP File: None

Provision: Not running, Last status : Downloading file from url.

Core Dump: Clean