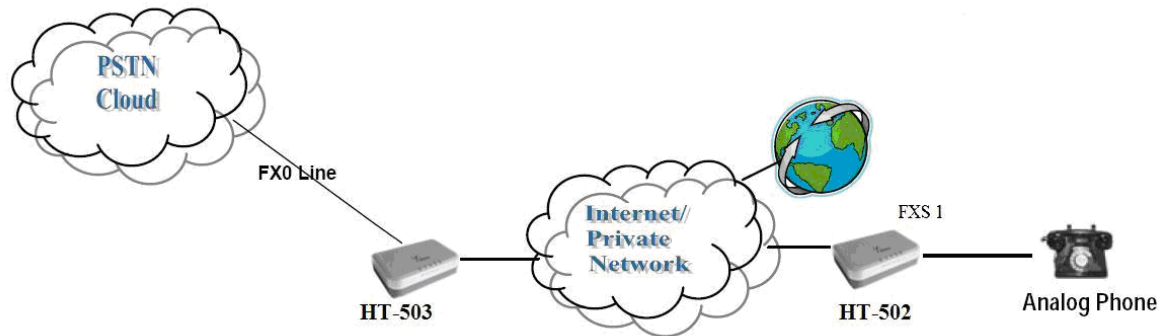


Peer to Peer HandyTone Scenario

(Extend Analog Lines Using the HandyTone 503 without SIP Server)



CONFIGURATION OF THE HT502 & HT503 SCENARIO

Configure the following settings for each of the devices using the Grandstream Web Configuration pages for each device.

HT502

FXS Port 1

- Primary Sip Server: Set to [IP_Address_of_HT-503:5062](#)
- SIP User ID: Any Number
- Authenticate ID: Any Number
- Sip Registration: No

*Sip User ID and Authenticate ID Should be the same

Note: Never Set Radon Port to Yes

HT503

FXO Port

- Primary Sip Server: Set to IP address of HT-502
- SIP User ID: Any Number
- Authenticate ID: Any Number
- Sip Registration: No
- Number of Rings: 1
- PSTN Ring Thru FXS: No
- Wait for Dial Tone: No
- Stage Method: 1

*Sip User ID and Authenticate ID Should be the same

*Stage Method 2 doesn't apply for peer to peer. It works when registered with a Sip Server. So, you can call any Extension.

Note: Never Set Radon Port to Yes

Basic Settings

- Unconditional Call Forward to VOIP: Must have a User ID (Could be Any)

NOTE: In order for this setup to work, it is extremely important that both Handy Tones (HT502 and HT503) are located on the same LAN OR have Public Static IPs. In short, the Handy Tones should be able to locate each other.