



Grandstream Networks, Inc.

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How to Connect the GXE5000 and a HT503

Typical Scenario:

Sometimes the user might find a situation where they want to add a HT-503 not only as an extension but also as an external PSTN trunk. This is common scenario where we want to grab a PSTN line from another PBX or a PSTN line in a remote location, but we don't want to invest too much money on a FXO gateway.

There are two ways to configure the GXE with the HT-503: We can register it directly or we can make it work as a SIP trunk.

### Method #1: How to connect the HT503 to the GXE-502X by register mode:

Configuring the HT-503:

Both the FXO and FXS ports can register to the GXE5000.

Grandstream Device Configuration	
STATUS	BASIC SETTINGS
<p><b>Account Active:</b> <input type="radio"/> No <input checked="" type="radio"/> Yes</p>	
<b>SIP Server:</b>	<input type="text" value="192.168.10.1"/> (e.g., sip.mycompany.com, or IP address)
<b>Outbound Proxy:</b>	<input type="text"/> (e.g., proxy.myprovider.com, or IP address, if any)
<b>SIP Transport:</b>	<input checked="" type="radio"/> UDP <input type="radio"/> TCP <input type="radio"/> TLS (default is UDP)
<b>NAT Traversal (STUN):</b>	<input checked="" type="radio"/> No <input type="radio"/> No, but send keep-alive <input type="radio"/> Yes
<b>SIP User ID:</b>	<input type="text" value="110"/> (the user part of an SIP address)
<b>Authenticate ID:</b>	<input type="text" value="110"/> (can be identical to or different from SIP User ID)
<b>Authenticate Password:</b>	<input type="text" value="●●●"/> (purposely not displayed for security protection)
<b>Name:</b>	<input type="text" value="HT503FXS"/> (optional, e.g., John Doe)

Simply register each port as if it was an extension of the GXE-502X

Configuring the FXO port of the HT503:

#### FXO disconnect method section:

First we should confirm which method the PSTN line is using. If the PSTN line is using current disconnect (typical case in North America), then we should enable current disconnect and disable PSTN disconnect tone detection. The default current disconnect threshold is 100ms, but if you start experiencing dropped calls then you should raise this value in 100ms intervals.

If the PSTN disconnects using the tones method, then enable it and disable the current disconnect method.

The tone disconnect method is widely used everywhere else in the world

The North American busy tone value is “f1=480@-32,f2=620@-32,c=500/500” but these tone vary from country to country, you can look up for the settings for your country at [www.3amsystems.com](http://www.3amsystems.com) or you can download the information for multiple countries from: <http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf>

You should set “Number of Rings” option to 1  
You should set “PSTN Ring Thru Delay” option to 1

### Channel Dialing:

We will use the two stage mode, so we should set the option “Stage Method (1/2)” to 2, and also we need to set the option “wait for Dial-Tone” to Yes.

#### FXO Termination

*Enable Current Disconnect:*  No  Yes (Default Yes. If set to yes, enter threshold below)

*Current Disconnect Threshold (ms):*  (50-800 milliseconds. Default 100 milliseconds)

*Enable PSTN Disconnect Tone Detection:*  No  Yes (Default No)

(If set to yes, the following tone is used as the disconnect signal)

*PSTN Disconnect Tone:*   
(Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...])  
(Allowed Range: freq = 0 to 4000Hz; vol = -40 to -24dBm)  
(Default: Busy Tone: f1=480@-32,f2=620@-32,c=500/500;)

*AC Termination:*

*Number of Rings:*  (1-10. Default 4)  
(Number of rings for a PSTN incoming call to FXO port before FXO port picks up)

*PSTN Ring Thru FXS:*  No  Yes (Default Yes)  
(If set to yes, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)

*PSTN Ring Thru Delay (sec):*  (1-10 seconds. Default 4 seconds)

#### Channel Dialing

*DTMF Digit Length (ms):*  (40-127 milliseconds, Default 100 milliseconds)

*DTMF Dial Pause (ms):*  (40-127 milliseconds, Default 100 milliseconds)

*First Digit Timeout (sec):*  (1-20 seconds. Default 10 seconds)

*Inter-Digit Timeout (sec):*  (1-15 seconds. Default 4 seconds)

*Wait for Dial-Tone:*  No  Yes (Default Yes - dial upon dial-tone)

*Stage Method (1/2):*  (Default 2 - 2 stage dialing)

### Inbound Call Setting:

In the basic settings page, we need to set the option “Unconditional Call Forward to VoIP” correctly, so that the incoming call through the FXO port of the HT-503 can be forwarded to the GXE-502X correctly.

For the user ID we can use \*\*\*\*01 which is a star code for the auto-attendant, next type in the IP address of the SIP server and the destination port should be written as 5060 (default).

	User ID	Sip Server	Sip Destination Port
<i>Unconditional Call Forward to VOIP:</i>	****01	@ 192.168.10.1	: 5060

### How to dial in and out:

For outbound calls, the user in the GXE502X can dial the HT-503’s FXO extension number, after you get dial tone, you can dial the number in the PSTN network. So it is done in a 2 stage manner.

For inbound calls, the user can dial into the PSTN line’s number, for example 626-555-4444 and then he should reach the IVR of the GXE502X, and then he can dial the internal extensions of the GXE502X.

### Method #2: The HT503 and the GXE502X connect through SIP Trunk DID mode (FXO needs to register)

#### How to configure HT503:

- 1, The FXO port of the HT-503 registers to the GXE502X
- 2, How to configure the FXO port:

Follow the same steps as above to configure the PSTN disconnect method

This time we will use 1 stage dialing, so we should set the option “Stage Method (1/2)” to 1, and we also need to set the option “wait for Dial-Tone” to No.

Also in the basic settings page, we have to configure the “Unconditional Call Forward to VoIP” field. Fill in the blank option “User ID” with a number; this number can be any number, but you should note that this number should also be used as the DID number which you will use to configure the GXE502X. We will use “000” in this example

	User ID	Sip Server	Sip Destination Port
<i>Unconditional Call Forward to VOIP:</i>	000	@ 192.168.10.1	: 5060

Important: You should configure the local sip port of the FXO port to 5060 and configure the local port of FXS port to 5062. So basically exchange the SIP ports.

## How to Configure the GXE5000:

Create a SIP trunk: Click on the add button to register a SIP trunk.

1. SIP Server URL: Fill in with the IP address of the HT503
2. Dial Prefix: Fill in with a prefix that is not in use.
3. Register Active: Set to No.
4. DID Switch: Set to “Yes” and click add button to add DID numbers. Fill in the DID number with number you just set in the HT503, so in this example type in “000”

Register Active	<input type="radio"/> Yes	<input checked="" type="radio"/> No
CBCOM Encryption	<input type="radio"/> Yes	<input checked="" type="radio"/> No
Incoming Calls Routed by	Request URI	
Account ID As From Name	<input checked="" type="radio"/> Yes	<input type="radio"/> No
Call Routing Profile	<p><b>Available List</b></p> <ul style="list-style-type: none"> <li>Internal Call</li> <li>PlayVoiceMenu</li> <li>Inbound English</li> <li>General outbound</li> </ul>	<p><b>Selected List</b></p> <ul style="list-style-type: none"> <li>General Inbound</li> </ul>
DID Switch	<input checked="" type="radio"/> Yes	<input type="radio"/> No
	DID Number 000	Add
		Delete

The user can set the inbound call answer option by selecting the inbound call routing profile that will take the incoming call.

## How to dial:

For outbound calls, the user can dial dial prefix + PSTN number

For the inbound calls, the user can simply dial into the PSTN line that is connected to the FXO port of the HT-503