



Grandstream Networks, Inc.

Configuring UCM6XXX Series with HT503



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OVERVIEW

This document describes basic configuration to interconnect UCM6XXX series and HT503. This is typically applied to the scenario where users would like to add a HT503 not only as a remote extension but also as an external PSTN trunk. It could be common that we prefer 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 on a FXO gateway.

There are two ways to set up the UCM6XXX series IP PBX with the HT503.

- **Method 1:** Register the HT503 to the UCM6XXX directly.
- **Method 2:** Configure HT503 as a SIP peer trunk.

Note: UCM6XXX series include UCM6100 series (UCM6102, UCM6104, UCM6108 and UCM6116), UCM6200 series (UCM6202, UCM6204 and UCM6208) and UCM6510.

 **Warning:**

- When the UCM6XXX series is interconnected with other HT503, it is NOT recommended to turn on "Allow Guest Calls" under the UCM6XXX web GUI->**PBX->SIP Settings->General**. Turning on this option will allow unauthenticated calls coming through the UCM6XXX series. Please be aware of the security concerns when using this option.
 - When using the IVR in UCM6XXX series, please be aware that if "Dial Trunk" option is turned on in IVR settings, the call into the IVR will be able to dial outbound call using UCM6XXX's trunk. The IVR's permission level will be used when making outbound calls in this case. Please select proper permission level for the IVR to control the outbound calls allowed via "Dial Trunk".
-

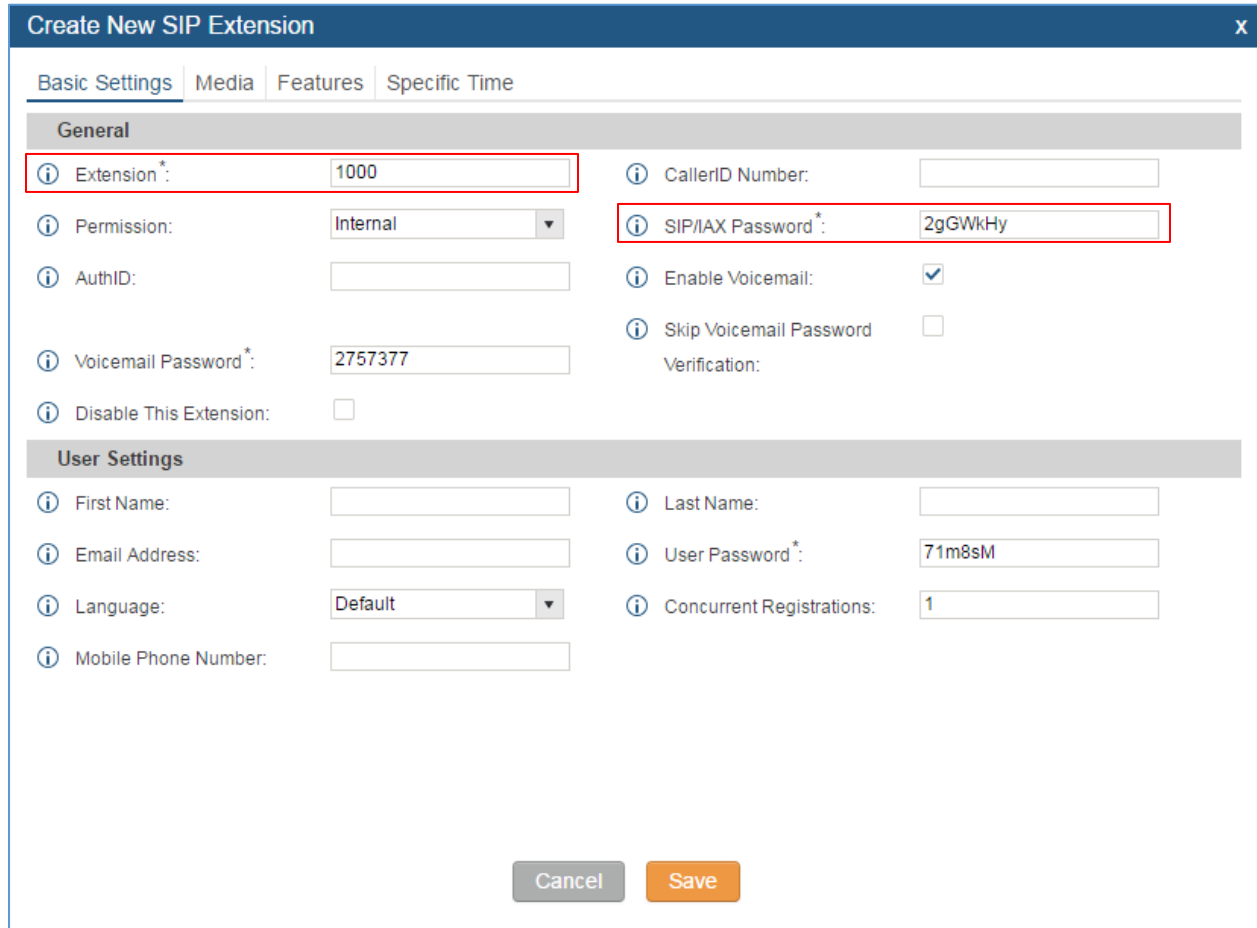


METHOD 1: REGISTER HT503 TO UCM6XXX

Create Extension on UCM6XXX

On the UCM6XXX web GUI, create two extensions under **PBX->Basic/Call Routes->Extensions**. These two extensions are used for HT503 FXS and FXO registration.

The password for the extension will be randomly generated if not specified.



Create New SIP Extension	
Basic Settings Media Features Specific Time	
General	
Extension *	1000
CallerID Number:	
Permission:	Internal
SIP/IAX Password *	2gGWkHy
AuthID:	
Enable Voicemail:	<input checked="" type="checkbox"/>
Voicemail Password *	2757377
Skip Voicemail Password Verification:	<input type="checkbox"/>
Disable This Extension:	<input type="checkbox"/>
User Settings	
First Name:	
Last Name:	
Email Address:	
User Password *	71m8sM
Language:	Default
Concurrent Registrations:	1
Mobile Phone Number:	
<input type="button" value="Cancel"/> <input type="button" value="Save"/>	

Figure 1: Create Extension 1000 on the UCM6XXX

Create New SIP Extension
X

Basic Settings
Media
Features
Specific Time

General

<div style="border: 1px solid #f00; padding: 2px;"> i Extension * : <input style="width: 90%;" type="text" value="1001"/> </div>	<div style="border: 1px solid #f00; padding: 2px;"> i SIP/IAX Password * : <input style="width: 90%;" type="text" value="1\$a1*t"/> </div>
i Permission: <input style="width: 90%;" type="text" value="Internal"/>	i CallerID Number: <input style="width: 90%;" type="text"/>
i AuthID: <input style="width: 90%;" type="text"/>	i Enable Voicemail: <input checked="" type="checkbox"/>
i Voicemail Password * : <input style="width: 90%;" type="text" value="81239909"/>	i Skip Voicemail Password Verification: <input type="checkbox"/>
i Disable This Extension: <input type="checkbox"/>	

User Settings

i First Name: <input style="width: 90%;" type="text"/>	i Last Name: <input style="width: 90%;" type="text"/>
i Email Address: <input style="width: 90%;" type="text"/>	i User Password * : <input style="width: 90%;" type="text" value="BA3VYzY"/>
i Language: <input style="width: 90%;" type="text" value="Default"/>	i Concurrent Registrations: <input style="width: 90%;" type="text" value="1"/>
i Mobile Phone Number: <input style="width: 90%;" type="text"/>	

Figure 2: Create Extension 1001 on the UCM6XXX

Create IVR on UCM6XXX

On the UCM6XXX web GUI, create an IVR extension under **PBX->Call Features->IVR**. This is to receive the calls forwarded from the HT503.

In IVR settings, if "Dial Other Extensions" is enabled, the calls forwarded to the UCM6XXX IVR will be able to reach the internal extensions registered to the UCM6XXX. Also, you can assign the "Key Pressing Event" to different destinations.



Create New IVR
X

Basic Settings
Key Pressing Events

i
Name*:

i
Extension:

i
Dial Trunk:

i
Dial Other Extensions:

Extension
 Conference
 Call Queue
 Ring Group

 Paging/Intercom Groups
 Voicemail Groups

 Fax Extension
 Dial By Name
 All

i
Replace Caller ID:

i
Alert-info:

i
Welcome Prompt:

[Prompt](#)

i
Digit Timeout*:

i
Response Timeout*:

i
Response Timeout Prompt:

[Prompt](#)

i
Invalid Prompt:

[Prompt](#)

Figure 3: Create IVR 7000 on the UCM6XXX

Configure FXS Port on HT503

1. Connect an analog phone to the HT503 FXS port.
2. On the HT503 web GUI, go to FXS Port setting page, configure to register the FXS port to the UCM6XXX extension 1000. Please refer to the highlighted settings in the following figure.

In this example, the UCM6XXX IP address is 192.168.5.250.



Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT
FXO PORT

Account Active: No Yes

Primary SIP Server: (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: (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: (optional, e.g., John Doe)

DNS Mode: A Record SRV NAPTR/SRV Use Configured IP

Primary IP:

Backup IP1:

Backup IP2:

Tel URI:

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes

Figure 4: Configure FXS Port on the HT503

Configure FXO Port on HT503

1. Connect the PSTN line to the HT503 FXO port.
2. On the HT503 web GUI, go to FXO Port setting page, configure to register the FXO port to the UCM6XXX extension 1001. Please refer to the highlighted settings and other necessary settings in the following figures.

In this example, the UCM6XXX IP address is 192.168.5.250.



Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT
FXO PORT

Account Active:	<input type="radio"/> No <input checked="" type="radio"/> Yes	
Primary SIP Server:	<input style="width: 100%;" type="text" value="192.168.5.250"/>	(e.g., sip.mycompany.com, or IP address)
Failover SIP Server:	<input style="width: 100%;" type="text"/>	(Optional, used when primary server no response)
Prefer Primary SIP Server:	<input checked="" type="radio"/> No <input type="radio"/> Yes	(yes - will register to Primary Server if Failover registration expires)
Outbound Proxy:	<input style="width: 100%;" type="text"/>	(e.g., proxy.myprovider.com, or IP address, if any)
SIP Transport:	<input checked="" type="radio"/> UDP <input type="radio"/> TCP <input type="radio"/> TLS (default is UDP)	
NAT Traversal:	<input checked="" type="radio"/> No <input type="radio"/> Keep-Alive <input type="radio"/> STUN <input type="radio"/> UPnP	
SIP User ID:	<input style="width: 100%;" type="text" value="1001"/>	(the user part of an SIP address)
Authenticate ID:	<input style="width: 100%;" type="text" value="1001"/>	(can be identical to or different from SIP User ID)
Authenticate Password:	<input style="width: 100%;" type="text"/>	(purposely not displayed for security protection)
Name:	<input style="width: 100%;" type="text" value="1001"/>	(optional, e.g., John Doe)
DNS Mode: <input checked="" type="radio"/> A Record <input type="radio"/> SRV <input type="radio"/> NAPTR/SRV <input type="radio"/> Use Configured IP		
Primary IP:	<input style="width: 100%;" type="text"/>	
Backup IP1:	<input style="width: 100%;" type="text"/>	
Backup IP2:	<input style="width: 100%;" type="text"/>	
Tel URI:	<input type="text" value="Disabled"/> ▼	
SIP Registration:	<input type="radio"/> No <input checked="" type="radio"/> Yes	
Unregister On Reboot:	<input checked="" type="radio"/> No <input type="radio"/> Yes	
Outgoing Call without Registration:	<input checked="" type="radio"/> No <input type="radio"/> Yes	

Figure 5: Configure FXO Port on the HT503 - Registration

Since we are going to use IVR when the call is forwarded to the UCM6XXX, the UCM6XXX will need to be able to detect the DTMF digits. Configure the HT503 FXO port DTMF settings as below as an initial setup.

<i>Preferred DTMF method:</i>	Priority 1:	<input style="width: 90%;" type="text" value="RFC2833"/> ▼
<i>(in listed order)</i>	Priority 2:	<input style="width: 90%;" type="text" value="SIP INFO"/> ▼
	Priority 3:	<input style="width: 90%;" type="text" value="In-audio"/> ▼

Figure 6: Configure FXO Port on the HT503 - DTMF Settings

There are a few necessary changes to be made in FXO termination section and Channel Dialing section as well.



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 Model Country-based Impedance-based (Default Country-based)

Country-based

Impedance-based

Number of Rings: (1-50. Default 4)
 (Number of rings for a PSTN incoming call before FXO port answers to accept VoIP number)

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)

PSTN Ring Timeout (sec): (2-10 seconds. Default 6 seconds)
 (Used to detect PSTN hangup when FXO port is not answered)

Figure 7: Configure FXO Port on the HT503 - FXO Termination

- 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 turn on "Enable Current Disconnect" and disable "Enable PSTN Disconnect Tone Detection".

The default "Current Disconnect Threshold" is 100ms, but if you start experiencing dropped calls then you should raise this value by 100ms intervals.

If the PSTN disconnects using the tones method, then turn on "Enable PSTN Disconnect Tone Detection" and turn off the "Enable Current Disconnect" option.

For PSTN tone detection, 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 tones vary from country to country. You may look up for the settings for your country at www.3amsystems.com or download the information from <http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf>.

- Set "Number of Rings" option to 1. If you happen to experience caller ID issue, you may set it to 2.
- Set "PSTN Ring Thru FXS" to "No" if you prefer not to ring the FXS port after the Ring Thru Delay. In the sample setup, it's set to "Yes".



- Set "PSTN Ring Thru Delay" option to 1. If you happen to experience caller ID issue, you may set it to 2.
- Set the "Stage Method (1/2)" to 2 for 2 stage dialing.

<i>Stage Method (1/2):</i> <input type="text" value="2"/> (Default 2 - 2 stage dialing)

Figure 8: Configure FXO Port on the HT503 - Channel Dialing

Configure Unconditional Call Forward on HT503

On the HT503 web GUI, go to Basic setting page, configure "Unconditional Call Forward to VOIP" to the IVR extension on the UCM6XXX. In this example, the UCM6XXX IP address is 192.168.5.250.

	User ID	Sip Server	Sip Destination Port
<i>Unconditional Call Forward to VOIP:</i>	<input type="text" value="7000"/>	@ <input type="text" value="192.168.5.250"/>	: <input type="text" value="5060"/>

Figure 9: HT503 Basic Settings

How to Dial

Once the HT503 and the UCM6XXX are set up as above, the inbound call and the outbound call will be working as described below.

- **Outbound call**
 The extension registered to the UCM6XXX can dial the HT503's FXO extension number (1001 in this example). After you get the second dial tone, you can then dial a PSTN network number. Basically, the outbound call is done in a 2-stage manner.
- **Inbound call**
 The user from outside network can dial into the PSTN line's number (connected to HT503). And then he/she will reach the IVR of the UCM6XXX. The IVR on UCM6XXX would allow the user to further enter extension number or key pressing digit to reach the desired destination.



METHOD 2: CONNECT UCM6XXX TO HT503 USING PEER SIP TRUNK

Create Extension on UCM6XXX

On the UCM6XXX web GUI, create one extension under **PBX->Basic/Call Routes->Extensions**. This extension is used for HT503 FXO registration.

The password for the extension will be randomly generated if not specified.

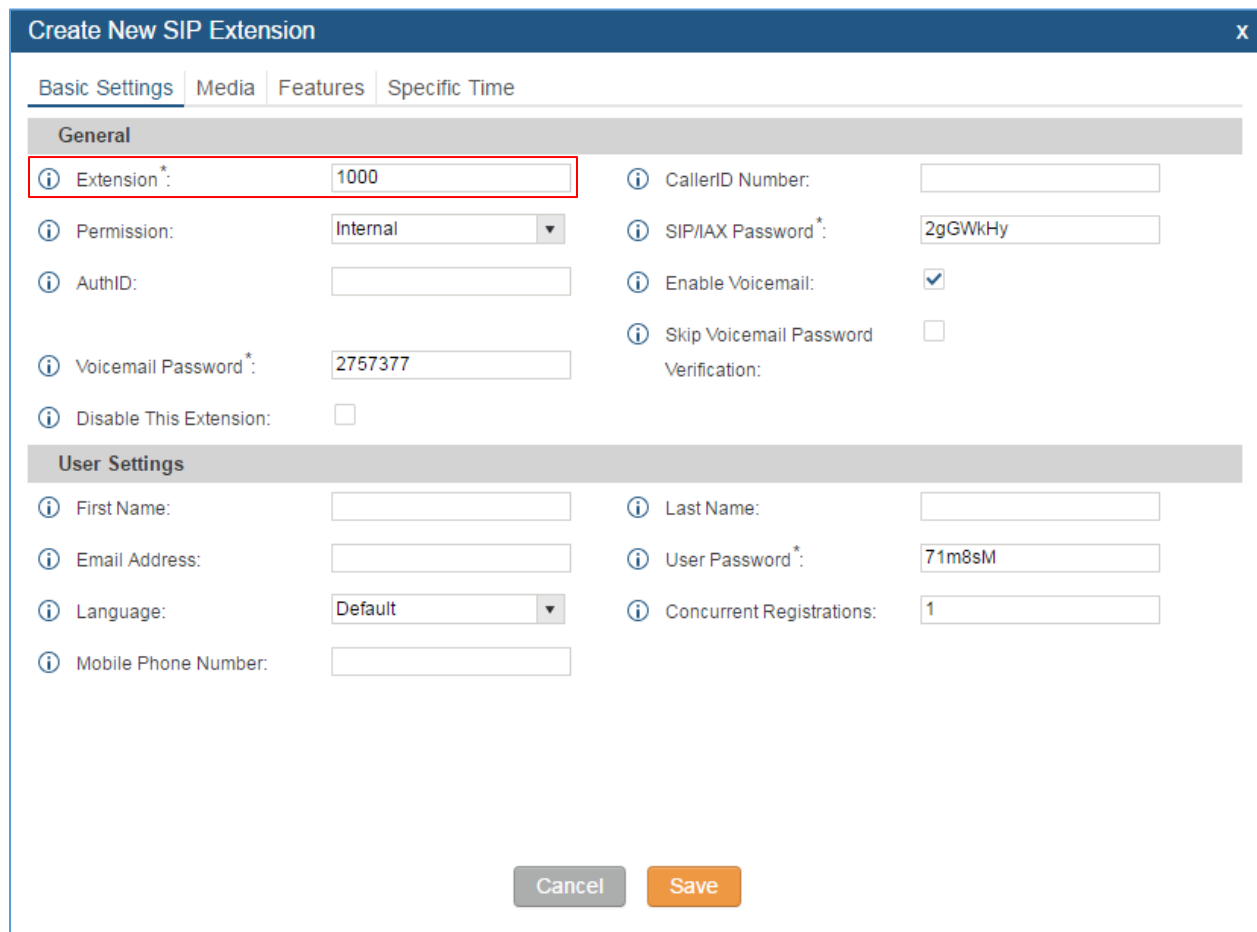


Figure 10: Create Extension 1000 on the UCM6XXX

Create IVR on UCM6XXX

On the UCM6XXX web GUI, create an IVR extension under **PBX->Call Features->IVR**.

In IVR settings, if "Dial Other Extensions" is enabled, the calls dialing into the UCM6XXX IVR will be able to reach the internal extensions registered to the UCM6XXX. Also, you can assign the "Key Pressing Event" to different destinations.



Create New IVR
X

Basic Settings
Key Pressing Events

(i) Name*:

(i) Extension:

(i) Dial Trunk:

(i) Dial Other Extensions:

Extension
 Conference
 Call Queue
 Ring Group

 Paging/Intercom Groups
 Voicemail Groups

 Fax Extension
 Dial By Name
 All

(i) Replace Caller ID:

(i) Alert-info:

None

(i) Welcome Prompt:

welcome

Prompt

(i) Digit Timeout*:

(i) Response Timeout*:

(i) Response Timeout Prompt:

ivr-create-timeout

Prompt

(i) Invalid Prompt:

invalid

Prompt

Cancel

Save

Figure 11: Create IVR 7000 on the UCM6XXX

Create Peer SIP Trunk on UCM6XXX

On the UCM6XXX web GUI, create a peer SIP trunk under **PBX->Basic/Call Routes->VoIP Trunks**. In this example, the HT503 IP address is 192.168.5.127.



Create New SIP Trunk x

More details will be shown when editing trunk.

Type:	Peer SIP Trunk ▼
Provider Name*:	HT503
Host Name*:	192.168.5.127
Keep Original CID:	<input type="checkbox"/>
Keep Trunk CID:	<input type="checkbox"/>
NAT:	<input type="checkbox"/>
Disable This Trunk:	<input type="checkbox"/>
TEL URI:	Disabled ▼
Caller ID:	<input type="text"/>
CallerID Name:	<input type="text"/>
Auto Record:	<input type="checkbox"/>

Figure 12: Create Peer SIP Trunk on the UCM6XXX

Configure Outbound Rule on UCM6XXX

On the UCM6XXX web GUI, go to **PBX->Basic/Call Routes->Outbound Routes** to create a new outbound rule. This would allow the extension on the UCM6XXX to reach numbers in PSTN network via the peer SIP trunk we just configured.



Create New Outbound Rule
X

i Calling Rule Name * :

i Pattern * :

i Disable This Route:

i Call Duration Limit:

i PIN Groups:

i Password:

i Privilege Level:
Warning: Setting privilege level at "Internal" has potential security risks.

i Enable Filter on Source Caller ID:

Send This Call Through Trunk

i Use Trunk * :

i Strip:

i Prepend:

Use Failover Trunk:

Trunks	Strip	Prepend	Options
Click to add failover trunk			

Time Condition

Time Condition	Time	Options
Click to add Time Condition		

Figure 13: Configure Outbound Rule on the UCM6XXX

In this example "91XXXXXXXXXX", 9 is the first dialing digit and it will be stripped off when the call goes out.

Configure Inbound Rule on UCM6XXX

On the UCM6XXX web GUI, go to **PBX->Basic/Call Routes->Inbound Routes** to create a new inbound rule.

In this example, we create the DID as **20000**, which will be used in the HT503 call forward setting.



Create New Inbound Rule
X

i Trunks * : SIPTrunks -- HT503

i DID Pattern * : _20000

i Disable This Route:

i Prepend Trunk Name:

i Prepend User Defined Name:

i Alert-info: None

i Inbound Multiple Mode:

i Default Destination * : IVR HT503_IVR

Time Condition

Time Condition	Time	Destination	Options
Click to add Time Condition			

Cancel
Save

Figure 14: Configure Inbound Rule on the UCM6XXX

The default destination is configured to IVR.

Configure FXO Port on HT503

1. Connect the PSTN line to the HT503 FXO port.
2. On the HT503 web GUI, go to FXO Port setting page, configure to register the FXO port to the UCM6XXX extension 1001. Please refer to the highlighted settings and other necessary settings in the following figures.

In this example, the UCM6XXX IP address is 192.168.5.250.



Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT
FXO PORT

Account Active: No Yes

Primary SIP Server: (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: (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: (optional, e.g., John Doe)

DNS Mode: A Record SRV NAPTR/SRV Use Configured IP

Primary IP:

Backup IP1:

Backup IP2:

Tel URI:

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes

Figure 15: Configure FXO Port on the HT503 - Registration

Since we are going to use IVR when the call is forwarded to the UCM6XXX, the UCM6XXX will need to be able to detect the DTMF digits. Configure the HT503 FXO port DTMF settings as below for an initial setup.

Preferred DTMF method: (in listed order)

Priority 1:

Priority 2:

Priority 3:

Figure 16: Configure FXO Port on the HT503 - DTMF Settings

There are a few necessary changes to be made in FXO termination section and Channel Dialing section.



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 Model Country-based Impedance-based (Default Country-based)

Country-based:

Impedance-based:

Number of Rings: (1-50. Default 4)
(Number of rings for a PSTN incoming call before FXO port answers to accept VoIP number)

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)

PSTN Ring Timeout (sec): (2-10 seconds. Default 6 seconds)
(Used to detect PSTN hangup when FXO port is not answered)

Figure 17: Configure FXO Port on the HT503: FXO Termination

- 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 turn on "Enable Current Disconnect" and disable "Enable PSTN Disconnect Tone Detection".

The default "Current Disconnect Threshold" is 100ms, but if you start experiencing dropped calls then you should raise this value by 100ms intervals.

If the PSTN disconnects using the tones method, then turn on "Enable PSTN Disconnect Tone Detection" and turn off the "Enable Current Disconnect" option.

For PSTN tone detection, 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 tones vary from country to country. You may look up for the settings for your country at www.3amsystems.com or download the information from <http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf>.

- Set "Number of Rings" option to 1. If you happen to experience caller ID issue, you may set it to 2. In the sample setup, it's set to 2.
- Set "PSTN Ring Thru FXS" to "No".



- Set "PSTN Ring Thru Delay" option to 1. If you happen to experience caller ID issue, you may set it to 2. In the sample setup, it's set to 2.
- Set the "Wait for Dial-Tone" to "No".
- Set the "Stage Method (1/2)" to 1.

Wait for Dial-Tone: <input checked="" type="radio"/> No <input type="radio"/> Yes (Default Yes - dial upon dial-tone) Stage Method (1/2): <input type="text" value="1"/> (Default 2 - 2 stage dialing)

Figure 18: Configure FXO Port on the HT503 - Channel Dialing

Exchange SIP Port Settings for FXS and FXO on HT503

- On the HT503 web GUI, go to FXO setting page, configure the "Local SIP Port" to be 5060. (The default setting is 5062.)
- On the HT503 web GUI, go to FXS setting page, configure the "Local SIP Port" to be 5062. (The default setting is 5060.)

Configure Unconditional Call Forward on HT503

On the HT503 web GUI, go to Basic setting page, configure "Unconditional Call Forward to VOIP" to the DID number **20000**. This is the same number configured in UCM6XXX inbound route dial pattern. In this example, the UCM6XXX IP address is 192.168.5.250.

	User ID	Sip Server	Sip Destination Port
Unconditional Call Forward to VOIP:	<input type="text" value="20000"/>	@ <input type="text" value="192.168.5.250"/>	: <input type="text" value="5060"/>

Figure 19: HT503 Basic Settings

How to Dial

Once the HT503 and the UCM6XXX are set up as above, the inbound call and the outbound call will be working as described below.

- **Outbound call**
The extension registered to the UCM6XXX can dial prefix + PSTN number to reach outside numbers in PSTN network, as defined in UCM6XXX outbound route.
- **Inbound call**
The user from outside network can dial into the PSTN line's number (connected to HT503). And then he/she will reach the IVR of the UCM6XXX. The IVR on UCM6XXX would allow the user to further enter extension number or key pressing digit to reach the desired destination. The inbound call will go through the inbound route set up on the UCM6XXX.

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