Grandstream Networks, Inc.

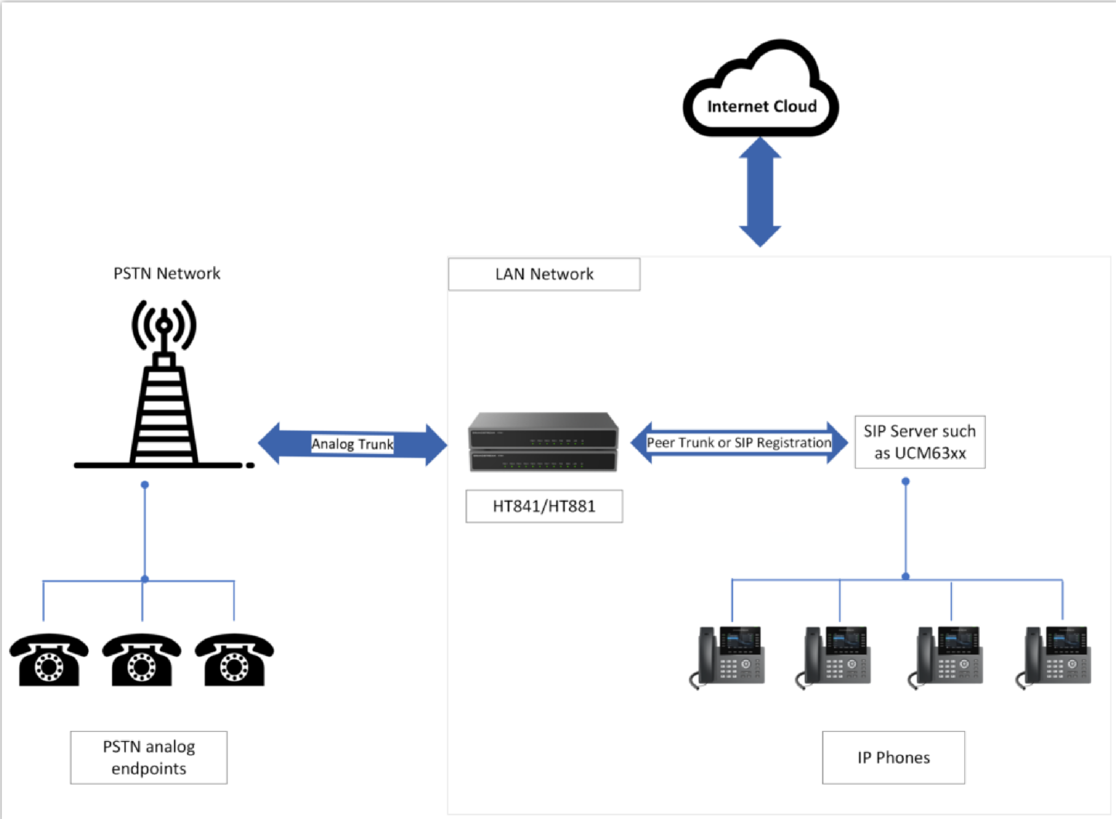
GXV3370

User Guide

Connecting UCM6XXX with HT841/HT881 FXO Gateway

Introduction

This document describes basic configuration to interconnect UCM6XXX series and HT841/HT881. In this document, we are using HT841 as an example. The following steps can be used for the HT881 model as well. This is typically applied to the scenario where users would like to add a HT841/HT881 not only as a remote extension but also as an external PSTN trunk.



HT8x1 and UCM63xx Interconnection

There are two ways to set up the UCM63XX series IP PBX with the HT841/HT881.

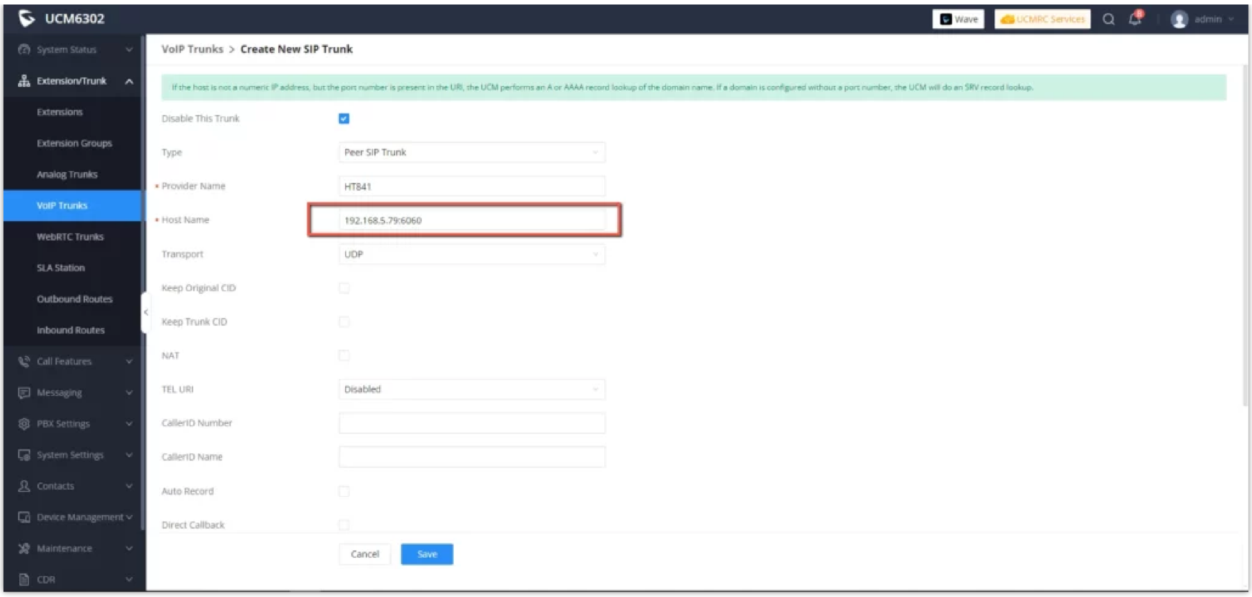
- **Method 1**: Configure HT841/HT881 as a SIP Peer Trunk.
- **Method 2**: Register HT841/HT881 on the UCM6XXX directly as an extension.

Another section of the guide will be dedicated for configuring multiple HT8x1 FXO gateways with one UCM63xx to attain more FXO ports if needed. Refer to the **section** below for more information.

Method 1: Connect UCM6XXX to HT841 using Peer SIP Trunk

Create Peer SIP Trunk on UCM6XXX

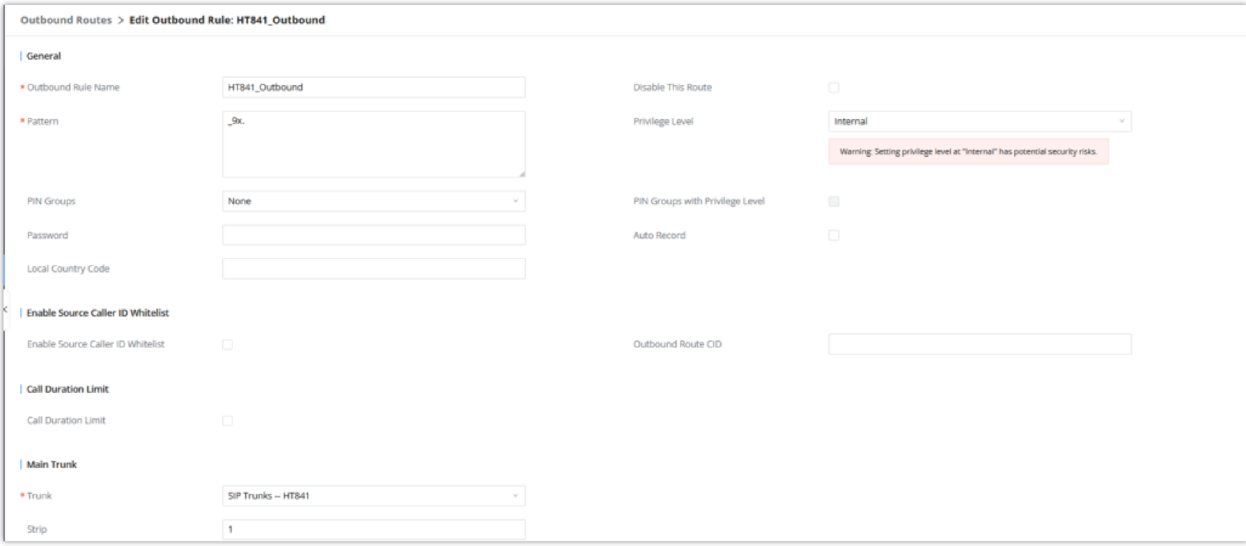
On the UCM6XXX web GUI, create a peer SIP trunk under **Extension/Trunk** ->**VOIP Trunks**. In this example, the HT841/HT881 IP address is 192.168.5.79, followed by 6060 which is the default listening port of FXO 1, on which the analog line is connected.



Create Peer SIP Trunk on the UCM6XXX

Configure Outbound Rule on UCM6XXX

On the UCM6XXX web GUI, go to **Extension/Trunk** ->**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.



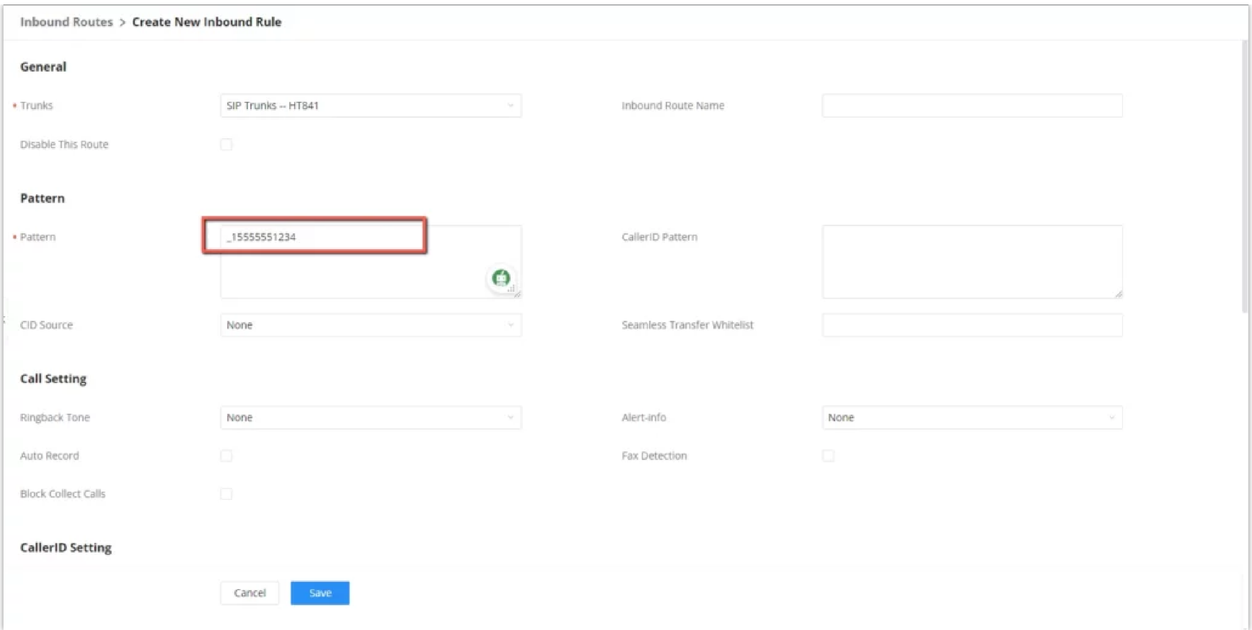
Configure Outbound Rule on the UCM6XXX

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

Configure Inbound Rule On UCM6XXX

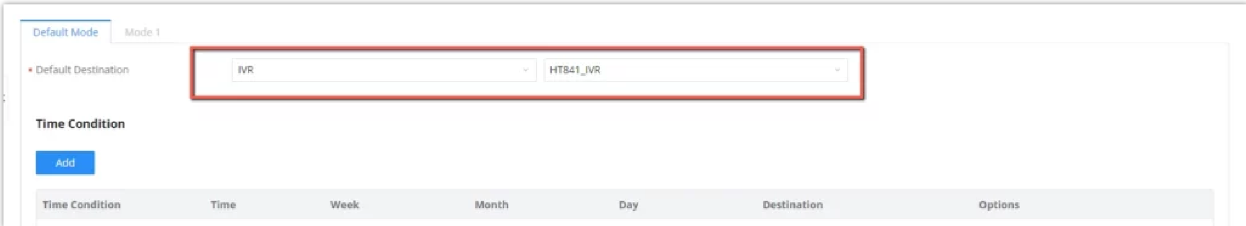
On the UCM6XXX web GUI, go to **Extension/Trunk** ->**Inbound Rules** to create a new inbound rule.

In this example, we create the DID as 15555551234, this will be used in the HT841/HT881 call forward setting.



Configure Inbound Rule on UCM6XXX

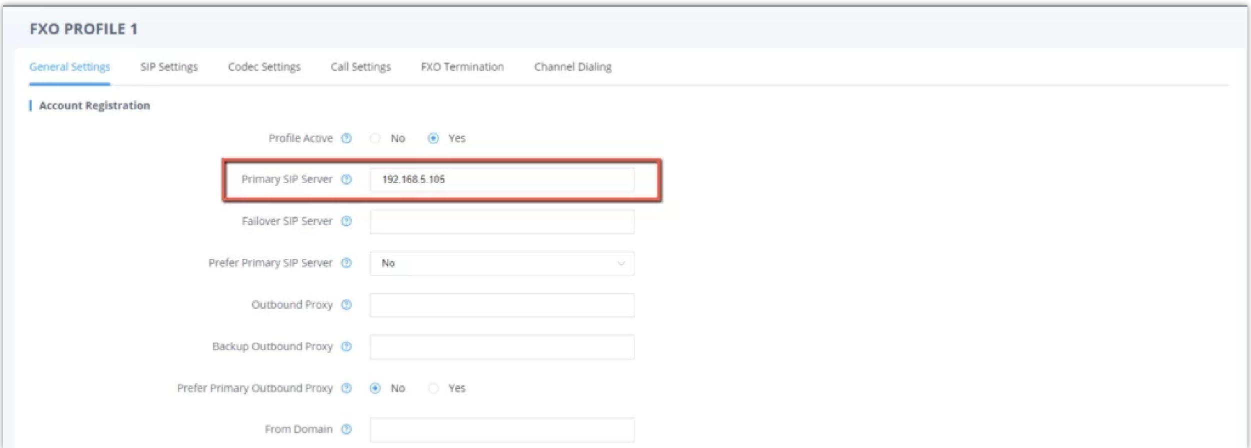
The default destination is configured to IVR.



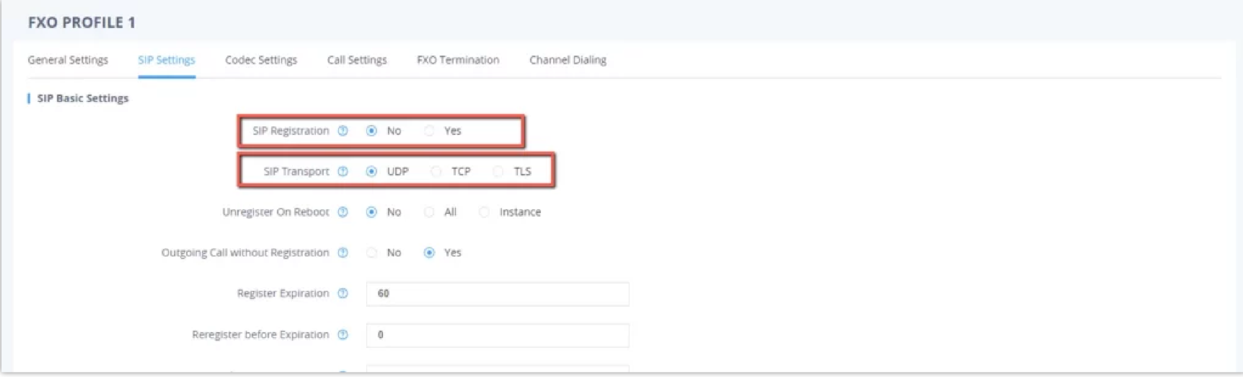
Configure Inbound Default Mode

Configure FXO Port on HT841 when Peered with UCM6XXX

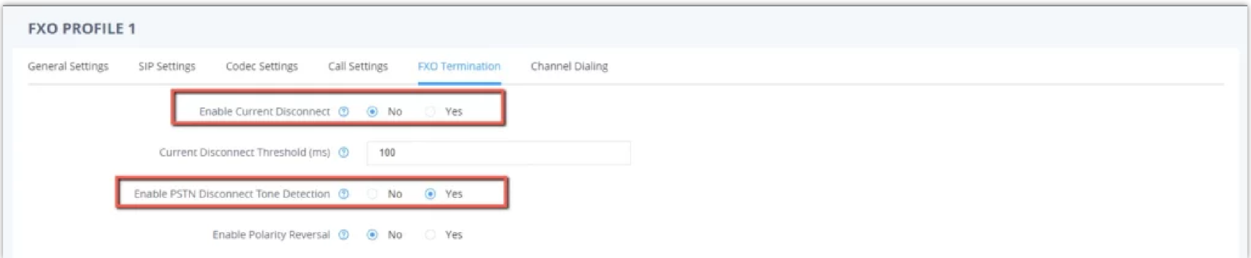
- Connect the PSTN line to the HT841 FXO port.
- On the HT841 web GUI, go to the **FXO Profile** settings page and enter the IP address of the UCM6XXX that you are peering with. In the following example, UCM6XXX has IP address 192.168.5.105



- Please make sure the **SIP Registration** option under is set to **No**.

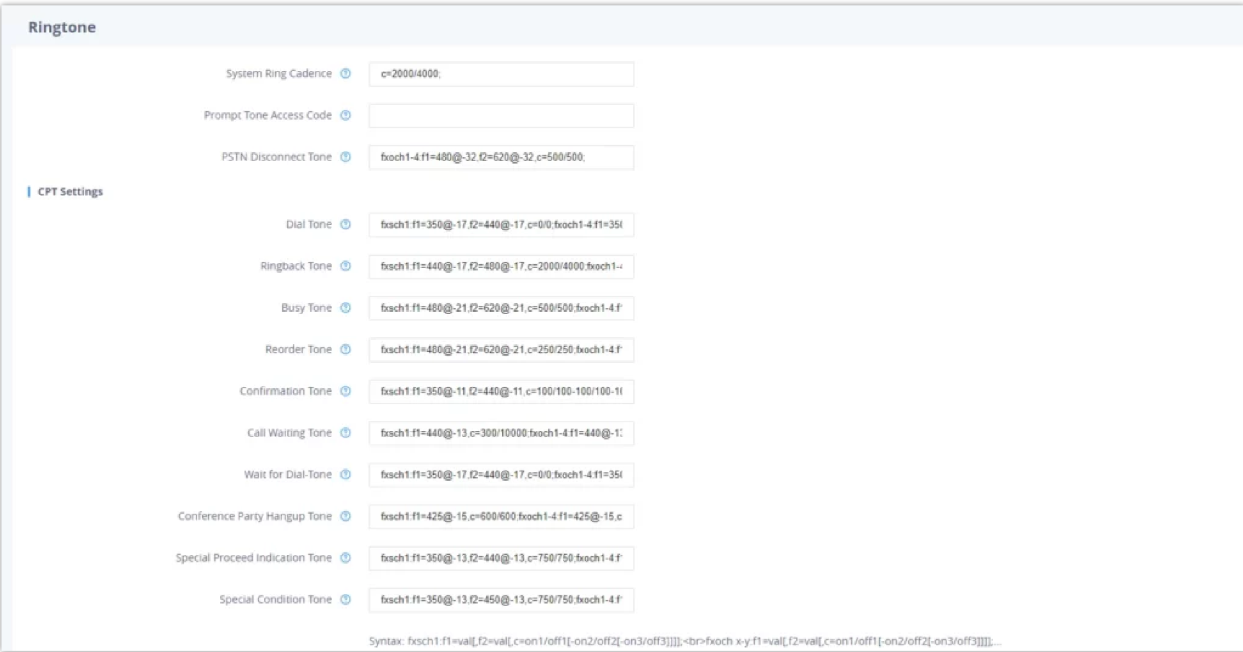


- There are few changes to be made in FXO termination section.



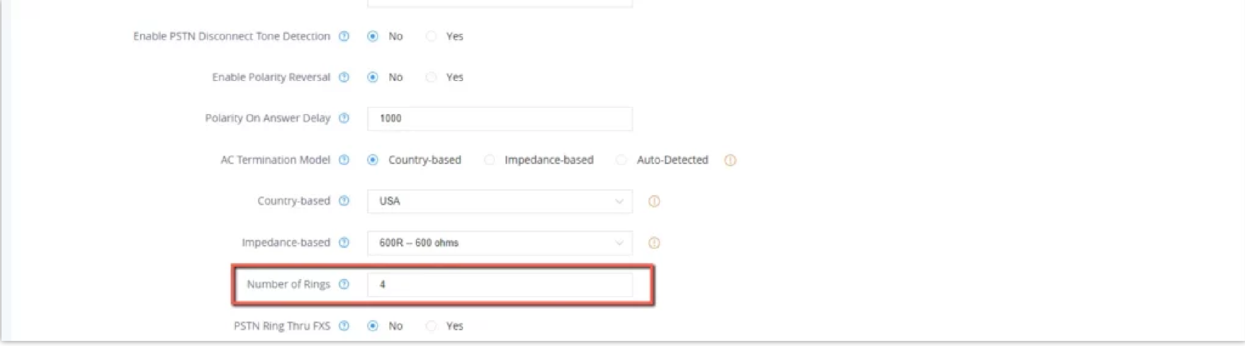
- First we should confirm which method the PSTN line is using.
  1. If your PSTN line uses current disconnect (common in North America), enable 'Current Disconnect' and disable 'PSTN Disconnect Tone Detection.' The default 'Current Disconnect Threshold' is 100ms; increase it in 100ms increments if call drops occur.
  2. 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 http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf
- Configure the Call Progress tones under **System Settings => Ringtone**, you can also keep them to the Default value as shown in the screenshot below:



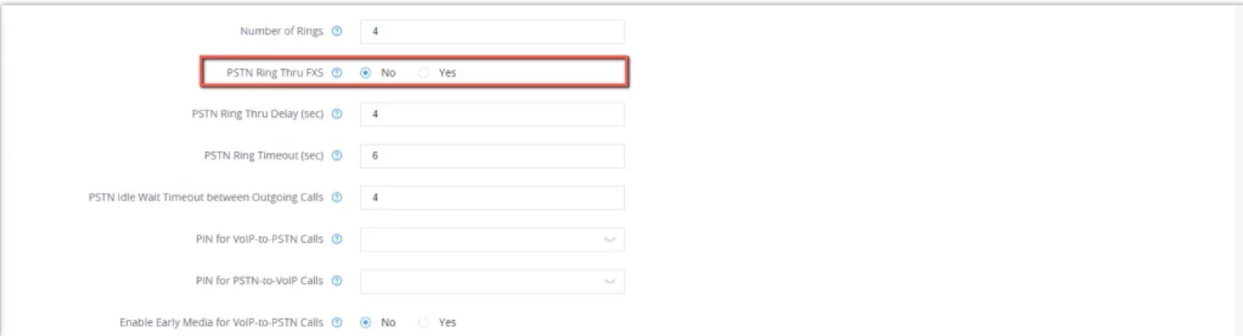
Configure FXO Port on the HT841 Call Progress Tones

- Set "Number of Rings" option to 4, it will ring 4 times before sending the call through the SIP trunk.



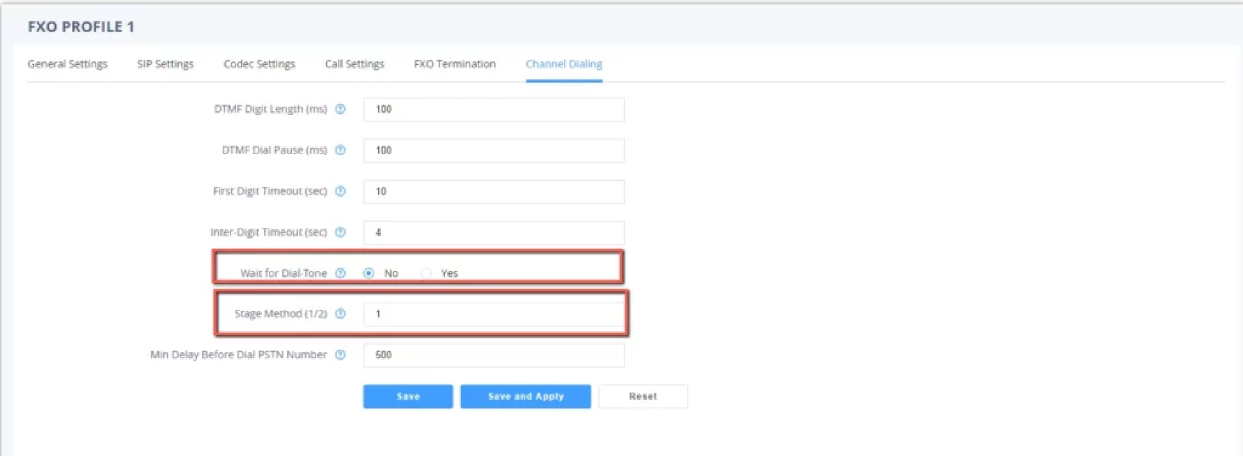
Configure FXO Port on the HT841 Number of Rings

- Set the “PSTN Ring Thru FXS” option to “No”, that way the call is routed through the defined FXO



Configure PSTN Ring Thru FXS

- Under **Channel Dialing** :
  1. Set the “Wait for Dial-Tone” to “No”.
  2. Set the “Stage Method (1/2)” to 1.



Configure FXO Port on the HT841 Channel Dialing

Note

- The formula of defining the FXO SIP ports is the following: SIP port = Profile’s base SIP port + (Port id – 1) *2, while the Port Id is defined as follows: FXS = 0, FXO1-8 = 1 – 8 for HT881 and FXO1-4 = 1 – 4 for HT841, in this case, if we set FXO profile 1 local SIP port to 6060, then FXO1 will use port 6060, FXO2 will use 6062, FXO3 will use 6064 and so on…

Configuring Unconditional Call Forward on HT841

On the HT841 web GUI, go to the **FXO Ports** page, configure “Unconditional Call Forward to VOIP” to the DID number 15555551234, this is our TEL-to-IP call.

In this example, we will use the SIP server for FXO Profile1, which is the IP Address of the UCM6xxx, and for the port number, we set it to 5060.

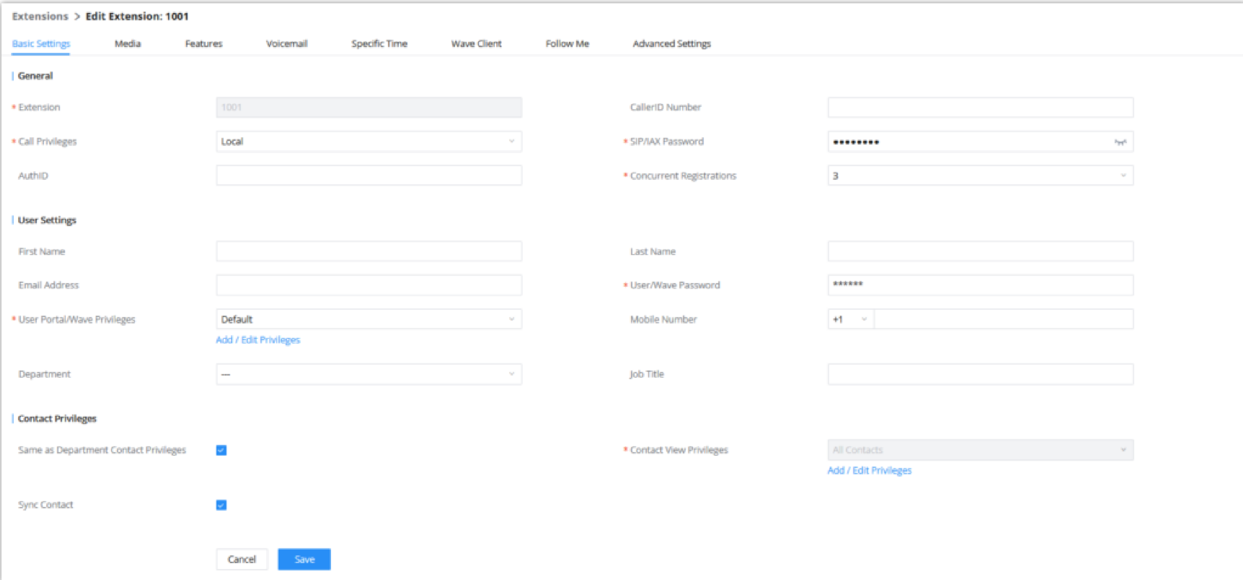
HT841 Call Forwarding

Method 2: Register HT841 on UCM6XXX as an Extension

Create SIP Extension on UCM6XXX

On the UCM6XXX web GUI, create an extension under **Extension/Trunk→Extensions**. This extension is used for HT841 FXO registration.

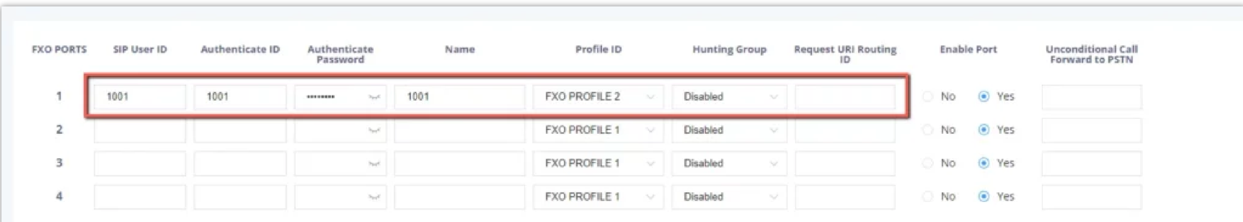
The password for the extension will be randomly generated if not specified.



Create SIP Extension on UCM6XXX For FXO port

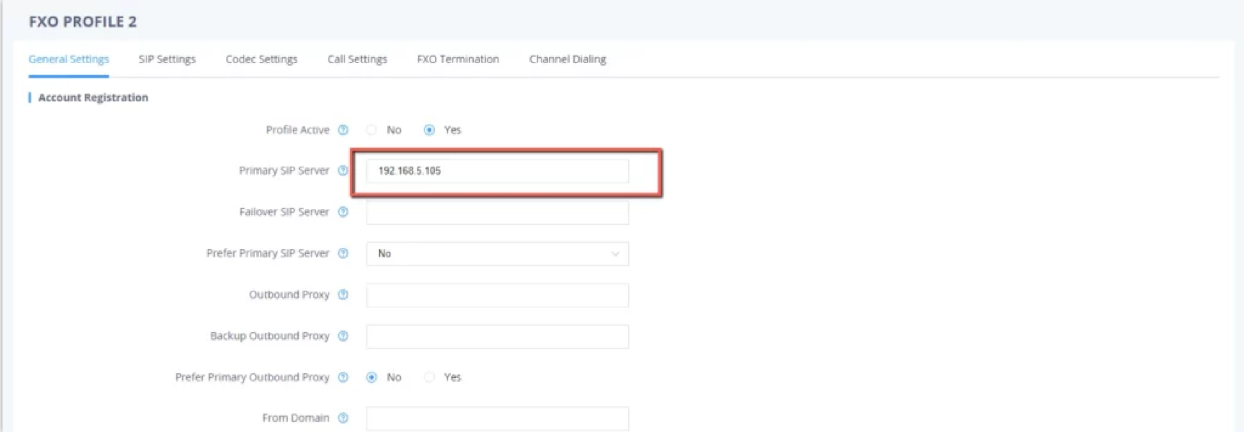
Configure HT841 User Setting as an Extension Registered on UCM6XXX

Under HT841 web GUI->**Ports**, please enter the SIP Extension information created earlier in the UCM6XXX for the FXO port, In this example, extension 1001 is used in order to register HT841' FXO port as an extension user on UCM6XXX.

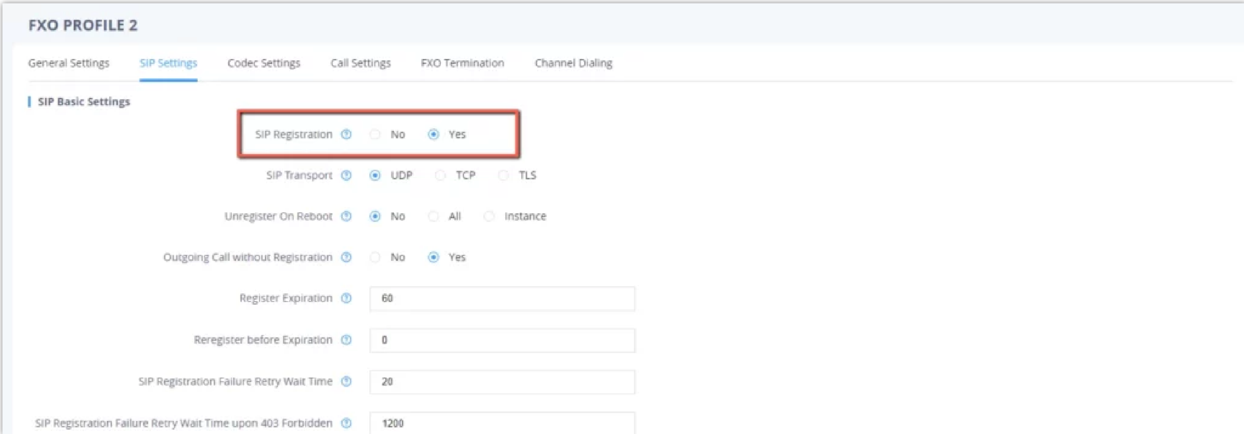


Registering extension on HT841

Under HT841 web GUI, **FXO Profile 2**, please fill in UCM6XXX information as explained in method 1.

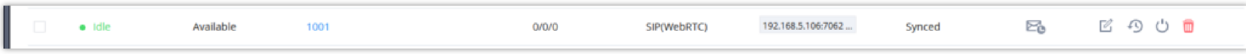


Please make sure under **SIP Settings** tab, **SIP Registration** option is set to **Yes**, as it is required for HT841 to successfully register on UCM6XXX.



HT841 SIP Settings

We can check UCM6XXX SIP Extension Status to see if HT841 has been successfully registered as an extension device for the FXO port. The green icon indicates that HT841 is registered on UCM6XXX.



UCM6XXX SIP Extension Status

Now HT841 is registered at UCM6XXX as an extension device. Please refer to method 1 in the previous section to adjust FXO Port and DTMF settings on HT841, Click **here** to access that section.

Configuring Unconditional Call Forward on HT841

On the HT841 web GUI, go to the **Ports** page, configure "Unconditional Call Forward to VOIP" to the extension number of the extension registered to the UCM63xx, this is our TEL-to-IP call.

On the HT841 web GUI, go to the FXO Ports, configure "Unconditional Call Forward to VOIP" to the IVR extension on the UCM6XXX. In this example, the UCM6XXX IP address is 192.168.5.105

HT841 Call Forwarding

Note

In order for this setup to work, it is extremely important that both the FXO gateway HT841/HT881 and the UCM63xx are located on the same LAN OR have Public Static IPs. In short, both devices should be able to locate each other.

Connecting Multiple HT841/HT881 FXO Gateways to One UCM63xx for FXO Port Expansion

In certain deployments, the number of PSTN lines required exceeds the number of FXO ports available on the UCM63xx device. In this case, multiple FXO gateways such as **HT841 or HT881** can be connected to a single UCM63xx system.

This section summarizes the required steps to deploy a **UCM63xx with two HT841/HT881 FXO gateways**, each gateway is used for a specific set of FXO lines

Device	IP	Ports
UCM63xx	192.168.6.65	8 FXO
HT881 Gateway #1	192.168.6.164	8 FXO
HT881 Gateway #2	192.168.6.78	8 FXO
Total PSTN Lines Required		**24 FXO**

Step 1 – Prepare the Network

1. Assign **static IP addresses** to all devices.

Example:

```
UCM63xx → 192.168.6.65
HT881 #1 → 192.168.6.164
HT881 #2 → 192.168.6.78
```

2. Ensure all devices are **on the same LAN** or can reach each other via routing.
3. Connect the **PSTN lines** to the FXO ports on the gateways.

Step 2 – Create SIP Peer Trunks on the UCM

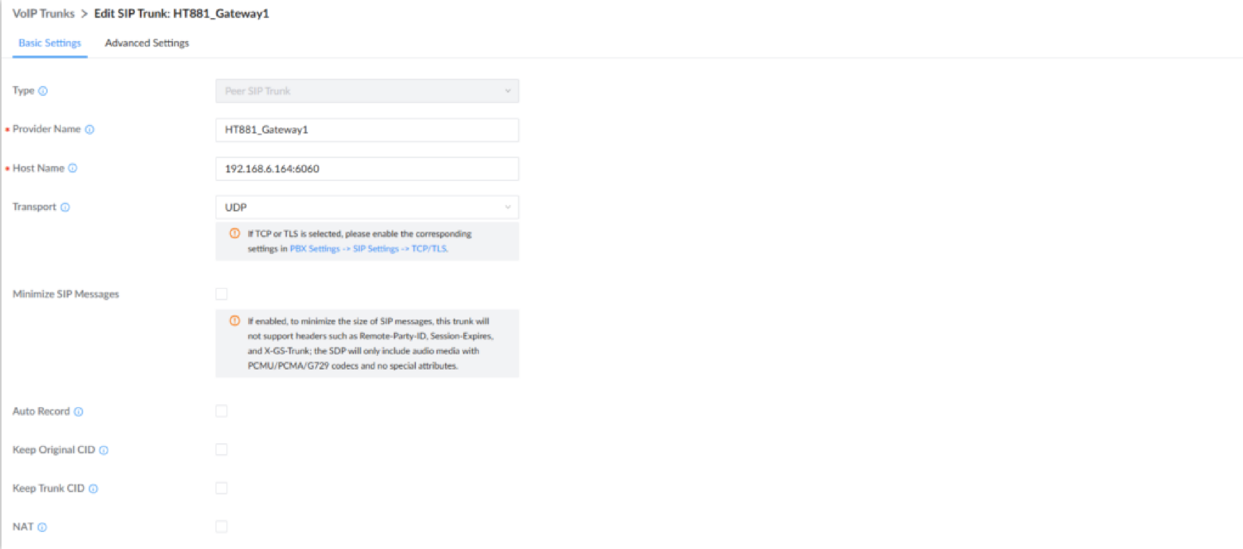
On the UCM WebUI, Navigate to:

```
Extension/Trunk → VoIP Trunks
```

Create **two trunks**.

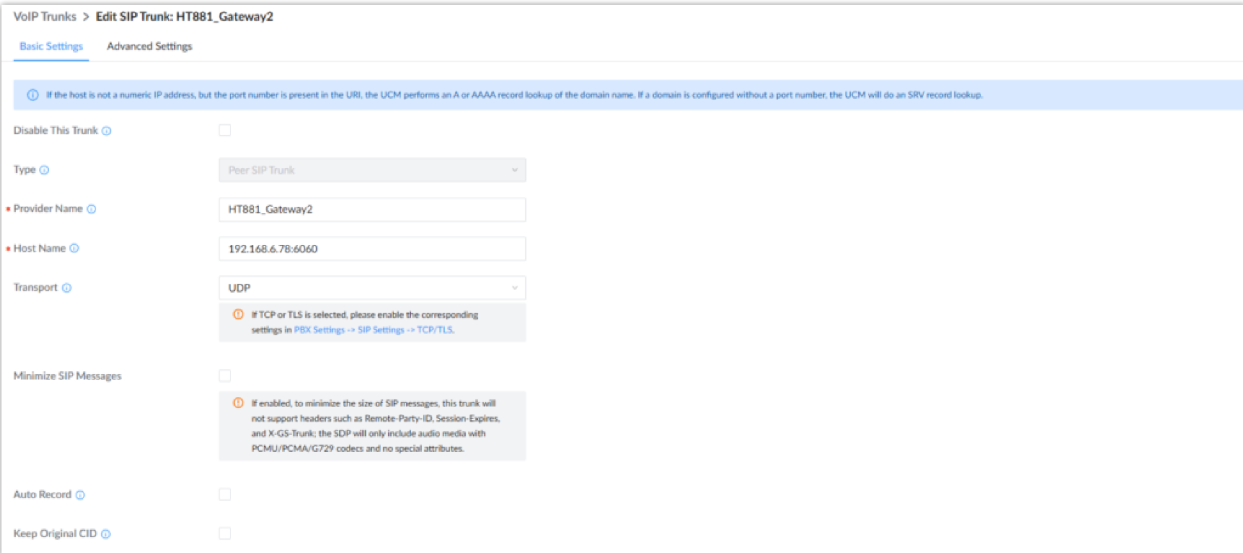
Trunk 1 – HT881 Gateway #1

Parameter	Value
Type	Peer SIP Trunk
Host/IP	192.168.6.164
SIP Port	6060
Transport	UDP

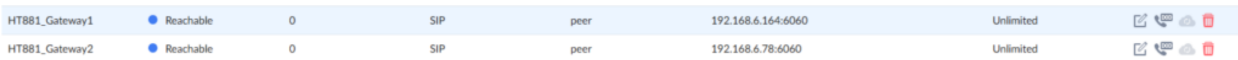


Trunk 2 – HT881 Gateway #2

Parameter	Value
Type	Peer SIP Trunk
Host/IP	192.168.6.78
SIP Port	6060
Transport	UDP



The created trunks will be marked as reachable:



Step 3 – Configure Outbound Route

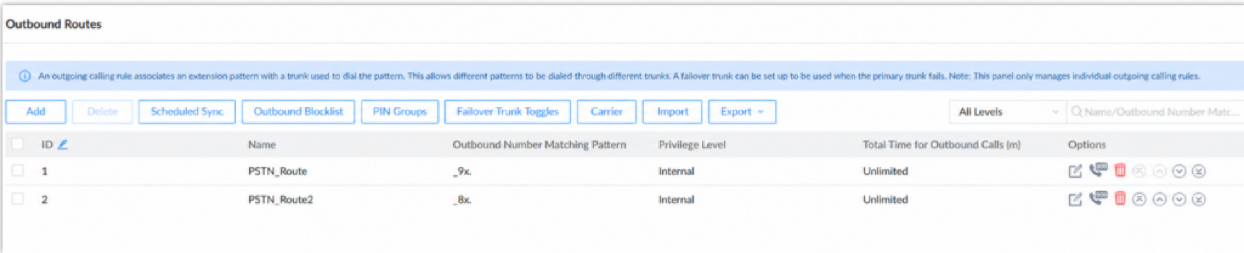
Navigate to:

```
Extension/Trunk → Outbound Routes
```

Create a route allowing internal extensions to dial PSTN numbers.

Example configuration:

Parameter	Value
Route Name	PSTN_Route
Pattern	_9x. / _8x.
Strip	1



Configure Outbound Route

Add trunks in order:

1. HT881_Gateway1 : used for PSTN_Route (_9x.) to reach FXO lines of the first gateway
2. HT881_Gateway2 : used for PSTN_Route (_8x.) to reach FXO lines of the second gateway

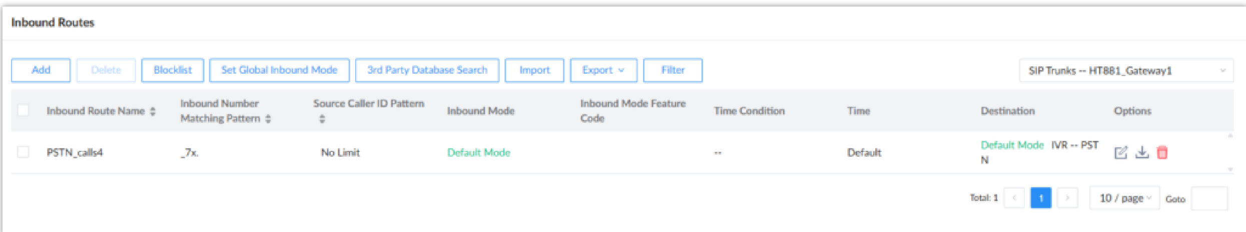
What happens here is when a user inside your call system dials the prefix 9 followed by the extension number, he will be using **HT881_Gateway1** trunk to perform an outgoing call through one of the FXO ports of the first gateway , when dialing prefix 8 followed by the extension number, he will be using **HT881_Gateway2** trunk to perform an outgoing call through one of the fxo ports on the second gateway, please use these settings carefully to route outgoing calls without disrupting the availabiliy of the fxo lines.

Step 4 – Configure Inbound Route

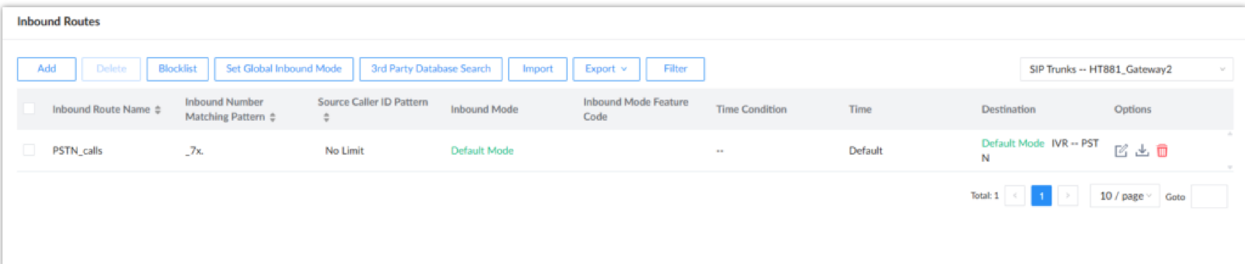
On the UCM63xx, Navigate to:

```
Extension/Trunk → Inbound Routes
```

Create a DID rule for both HT881_Gateway1 and HT881_Gateway2 trunks, and to both have the same destination, for example both configured to have DID: 7000



Inbound Route for Gateway1



Inbound Route for Gateway2

Example:

Parameter	Value
DID	7000

Parameter	Value
Default Destination	IVR

This allows PSTN calls arriving on the gateways to reach the IVR.

Step 5 – Configure HT881 Gateway #1

Access the web interface of the first FXO gateway:

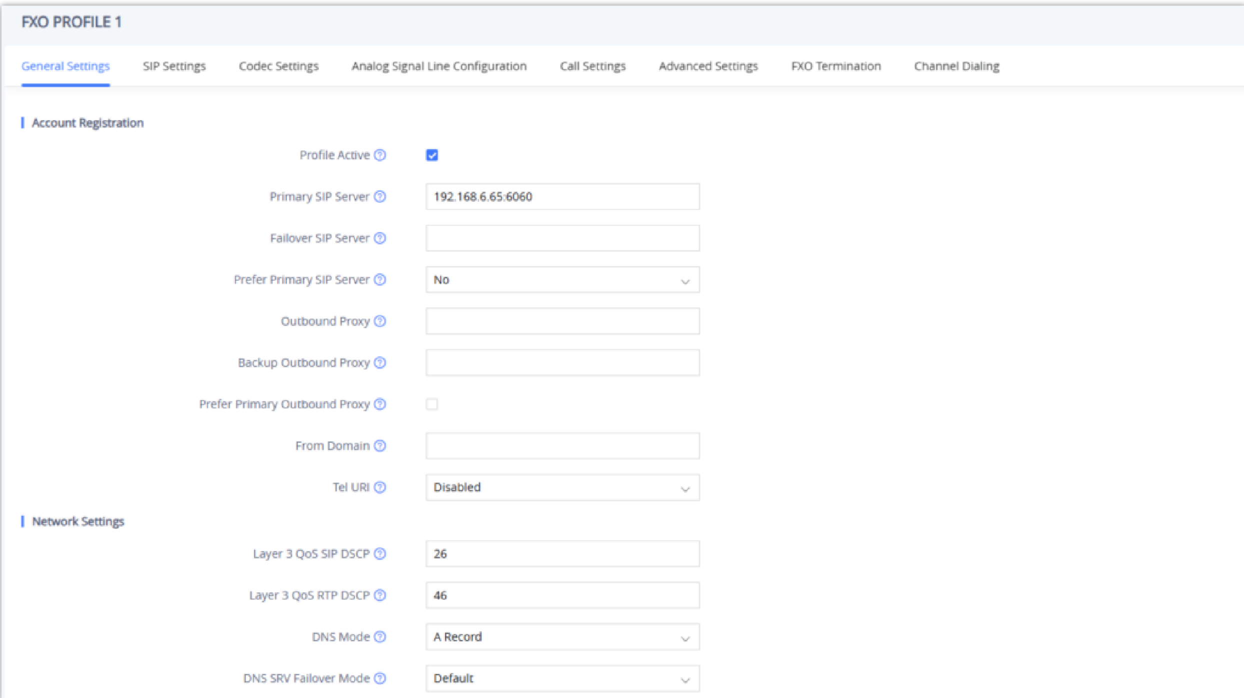
```
http://192.168.6.164
```

Navigate to:

```
FXO Profile 1 → General Settings / SIP Settings
```

Configure:

Parameter	Value
SIP Server	192.168.6.65
SIP Registration	No
Local SIP Port	6060



To Configure FXO Port Settings, Navigate to:

```
FXO Profile 1 → FXO Termination
```

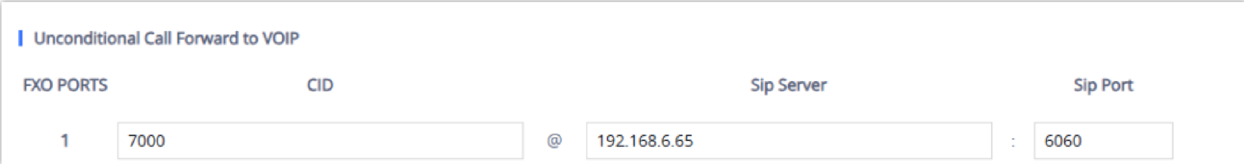
Configure:

Parameter	Value
Number of Rings	4
PSTN Ring Thru FXS	No

Parameter	Value
Wait for Dial Tone	No
Stage Method	1

After this, Configure Call Forward Under **FXO Ports**:

Parameter	Value
Unconditional Call Forward to VOIP	7000



Unconditional call forward

Step 6 – Configure HT881 Gateway #2

Access:

```
http://192.168.6.78
```

Repeat **exactly the same configuration** as Gateway #1.

Key parameters:

Parameter	Value
SIP Server	192.168.6.65
SIP Registration	No
Local SIP Port	6060

Configure identical **FXO port settings and call forwarding**.

Step 7 – Verify SIP Port Mapping

Each FXO port corresponds to a SIP port.

Formula:

```
SIP Port = Base SIP Port + (Port ID - 1) × 2
```

Example (Base SIP Port = 6060):

When deploying multiple HT841/HT881 gateways with a UCM, it is possible to use the **same base SIP port range on different gateways** (for example, 6060, 6062, 6064, 6066 on both devices). This works because the UCM identifies each gateway using the **combination of IP address and SIP port**, not the port alone.

Gateway	IP Address	FXO Ports (SIP)
HT881 #1	192.168.6.164	6060, 6062, 6064, 6066
HT881 #2	192.168.6.78	6060, 6062, 6064, 6066

How to dial

Once the FXO gateway or gateways and the UCM6XXX are set up as above, the inbound call and the outbound call will be working as described below.

- **Outbound call**

Method 1: The extension registered to the UCM6XXX can dial prefix + PSTN number to reach outside numbers in the PSTN network, as defined in the UCM6XXX outbound route.

Method 2: The extension registered to the UCM6XXX can dial HT841's FXO extension number (1001 in this example). After getting the second dial tone, you can then dial the PSTN network number. Basically, the outbound call is done in a 2-stage manner.

Example:

Method 1: The UCM-registered extension 1000 can call +1(555) 555-1234 by dialing 915555551234, with the '9' stripped and the rest of the number dialed.

Method 2: The UCM-registered extension 1000 can call the HT841 registered extension 1001, then dials the PSTN number +1(555) 555-1234

- **Inbound call**

The user from the outside network can dial into the PSTN line's number (connected to HT841/HT881). And then he/she will reach the IVR of the UCM6XXX. The IVR on UCM6XXX would allow the user to further enter the 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.

Example:

A user dials the office number +1(555) 555-1234, this is the DID number provided by the PSTN service provider, then he will be prompted with an IVR, he can wait until he is prompted to call the desired extension, then he can call extension number 405 to reach his destination.

Supported Devices

Device	Firmware Required
HT841	1.0.5.7+
HT881	
UCM6301/6302/6304/6308	1.0.23.17+
UCM6300A/6302A/6304A/6308A	

Supported Devices