FIBERME Communications LLC.

Configuring FCM630A with GXW410X

Table of Contents

Table of Figures

OVERVIEW

This document describes basic configuration to interconnect FCM630A series and GXW410X. In this document, we are using GXW4104 as an example. The following methodology can be used for the GXW4108 as well. This is typically applied to the scenario where users would like to add a GXW410X not only as a remote extension but also as an external PSTN trunk.

There are two ways to set up the FCM630A series IP PBX with the GXW410X.

- **Method 1**: Configure GXW410X as a SIP Peer Trunk.
- **Method 2**: Register GXW410X on the FCM630A directly as an extension.

⚠ Warning:

When using the IVR in FCM630A 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 FCM630A's trunk. TheIVR's permission level will be used when making outbound calls in this case. Please select properpermission level for the IVR to control the outbound calls allowed via "Dial Trunk".

CONNECT FCM630A TO GXW410X USING PEER SIP TRUNK

Create IVR On FCM630A

On the FCM630A web GUI, create an IVR extension under **Call Features**->**IVR**.

In IVR settings, if "Dial Other Extensions" is enabled, the calls dialing into the FCM630A IVR will be able to reach the internal extensions registered to the FCM630A. Also, you can assign the "Key Pressing Event" to different destinations.

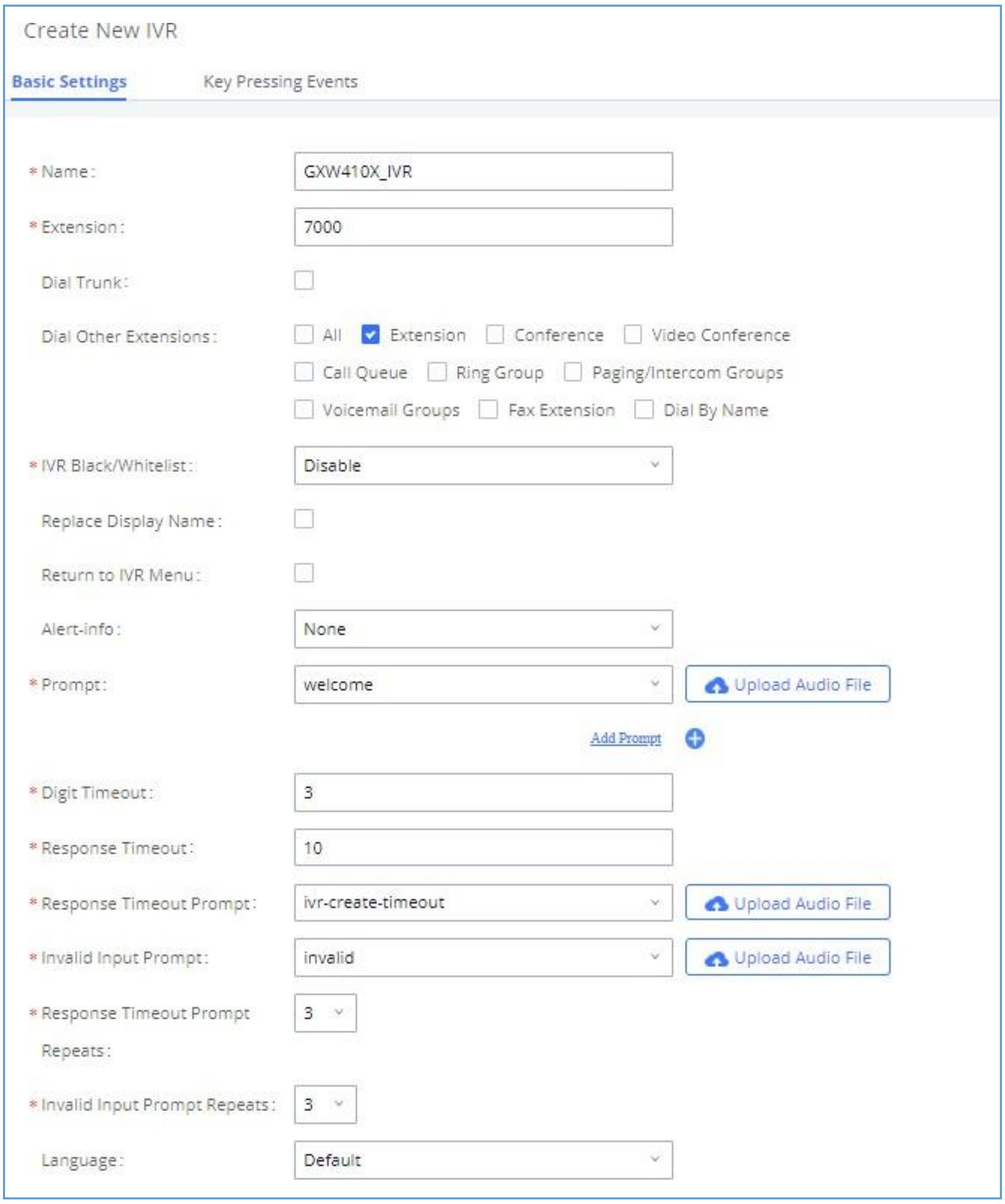

Figure 1: Create IVR 7000 on the FCM630A

Create Peer SIP TRUNK On FCM630A

On the FCM630A web GUI, create a peer SIP trunk under **Extension/Trunk** ->**VOIP Trunks**. In this example, the GXW410X IP address is 192.168.5.159.

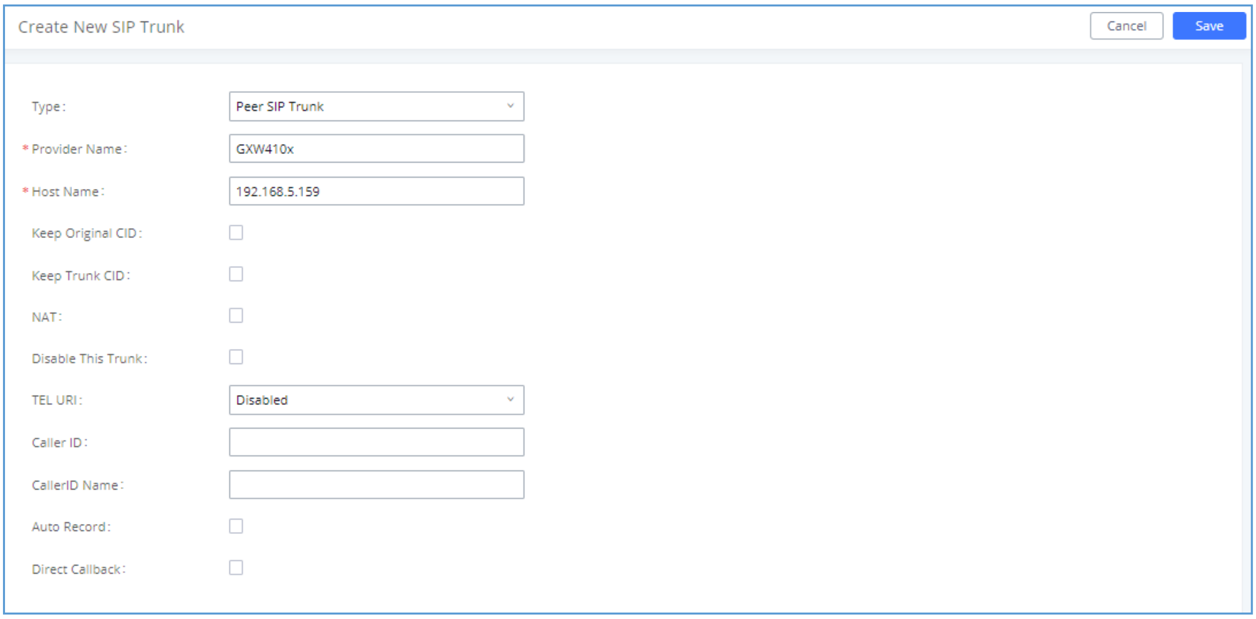

Figure 2: Create Peer SIP Trunk on the FCM630A

Configure Outbound Rule On FCM630A

On the FCM630A web GUI, go to **Extension/Trunk** ->**Outbound Routes** to create a new outbound rule. This would allow the extension on the FCM630A to reach numbers in PSTN network via the peer SIP trunk we just configured.

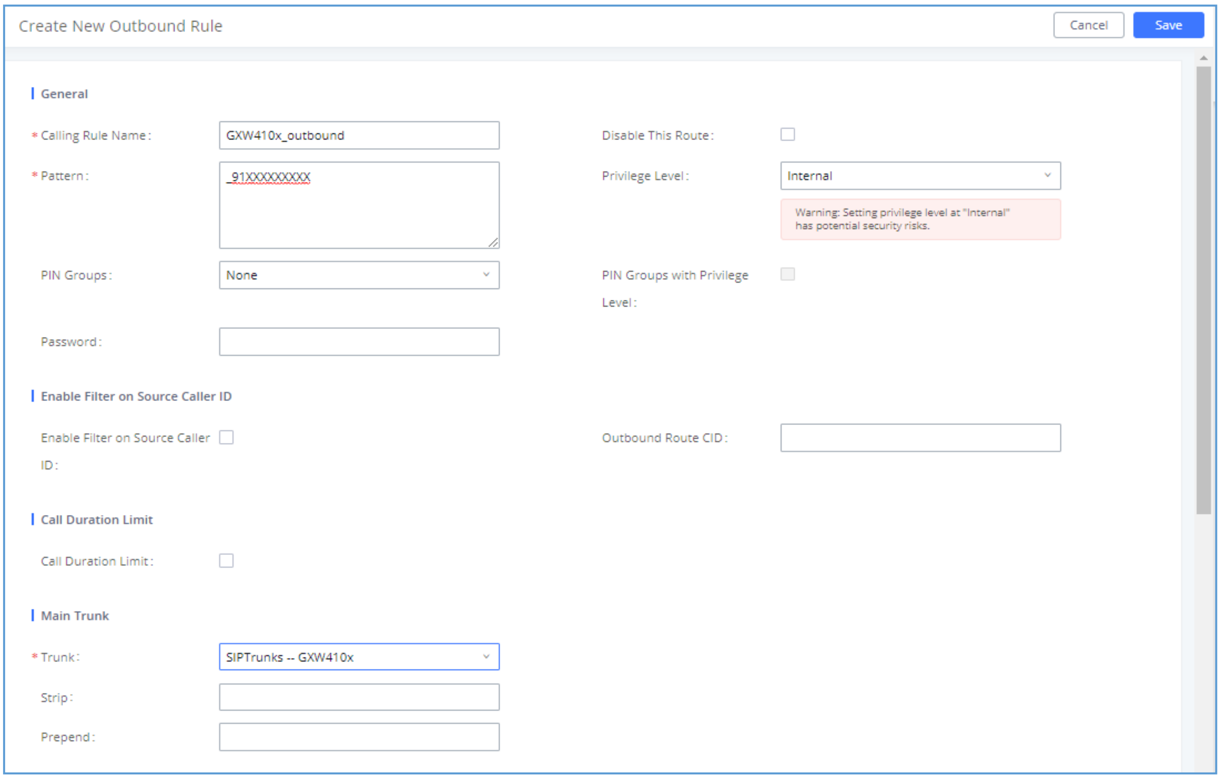

Figure 3: Configure Outbound Rule on the FCM630A

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

Configure Inbound Rule On FCM630A

On the FCM630A web GUI, go to **Extension/Trunk** ->**Inbound Rules** to create a new inbound rule. In this example, we create the DID as 20000, which will be used in the GXW410X call forward setting.

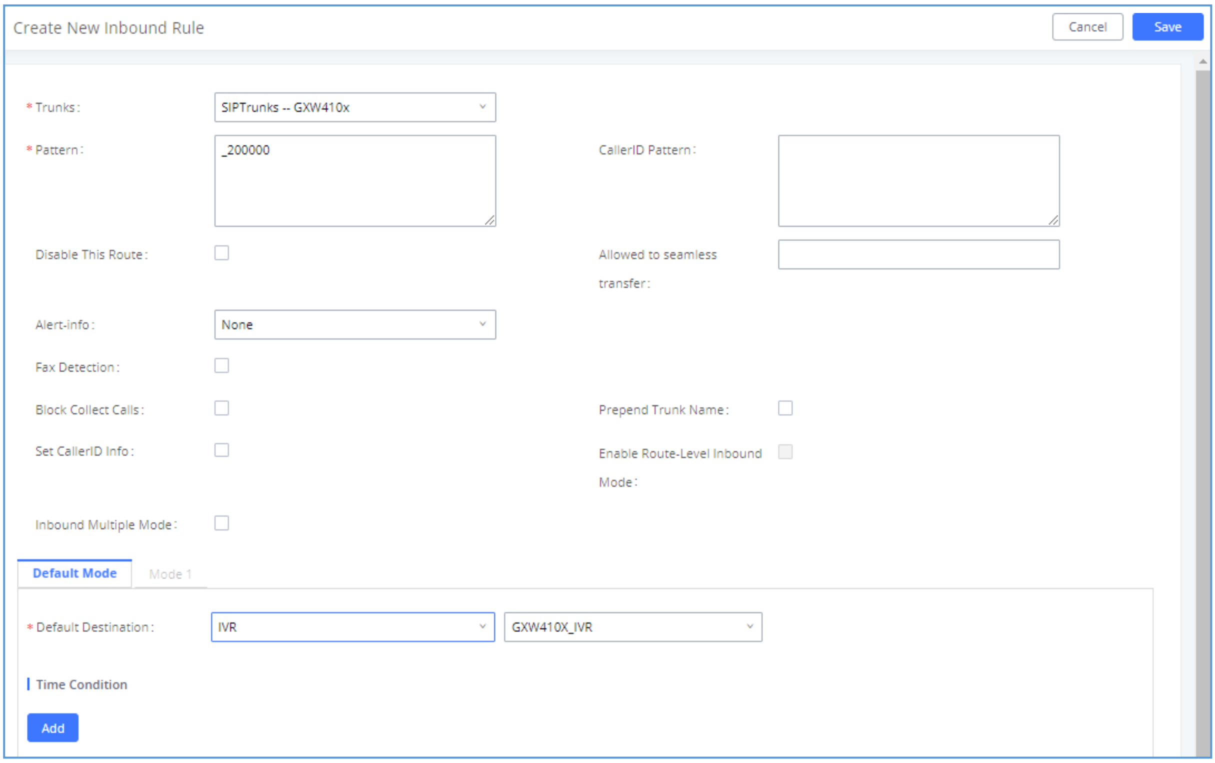

Figure 4: Configure Inbound Rule on FCM630A

The default destination is configured to IVR. Ensure to select the proper extension for the IVR.

Configure FXO Port On GXW410X When Peered with FCM630A

1. Connect the PSTN line to the GXW410X FXO port.
2. On the GXW410X web GUI, go to the **Accounts->Account X-> General Settings** page and enterthe IP address of the FCM630A that you are peering with.

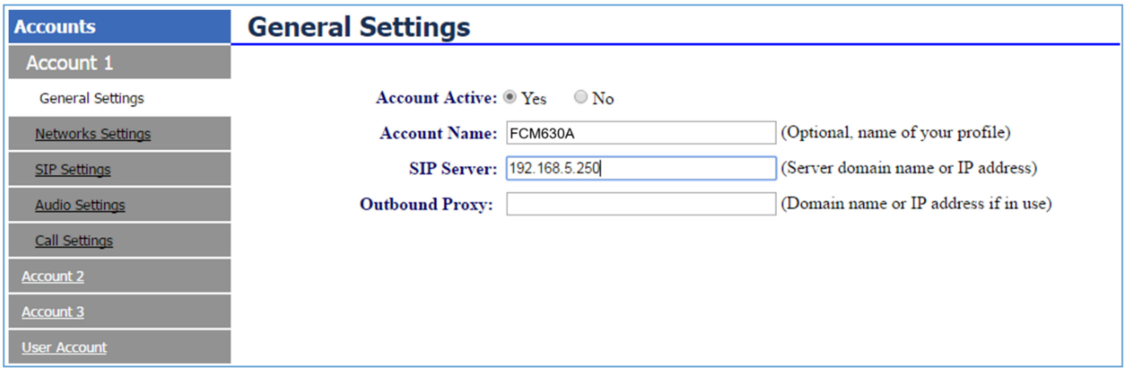

Figure 5: Configure FXO Port on GXW410X: General Settings

3. Please make sure the **SIP Registration** option under **Accounts-> Account X-> SIP Settings** is set to **No**. In the following example, FCM630A has IP address 192.168.5.250.

Figure 6: Configure FXO Port on the GXW410X - SIP Settings

Since we are going to use IVR when the call is forwarded to the FCM630A, FCM630A will need to be ableto detect the DTMF digits. Configure the GXW410X FXO port DTMF settings as below for the initial setup. This can be found under **Settings**-> **Channel Settings**.

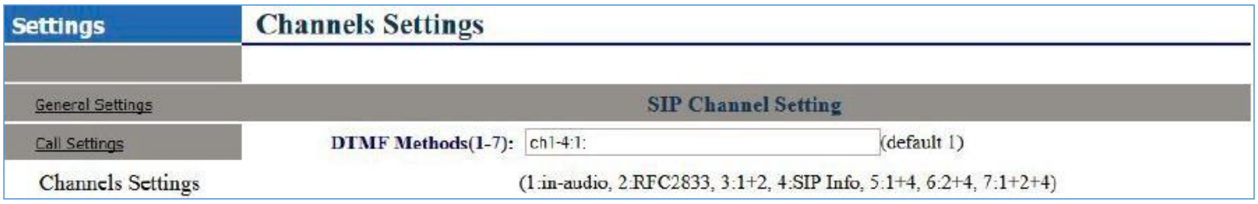

Figure 7: Configure FXO Port on the GXW410X - DTMF Method

Set the DTMF Payload Type to 101. This value can be found under **Settings->Call Settings**.

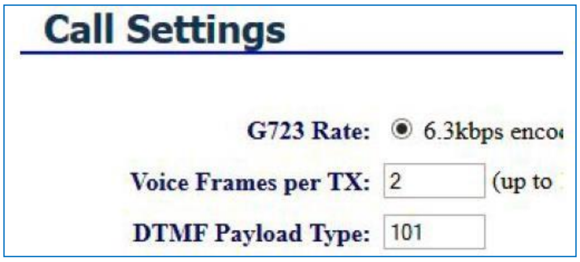

Figure 8: Configure FXO Port on the GXW410X - DTMF Payload Type

There are few changes to be made in FXO termination section. This feature can be found under **FXO Lines** settings page.

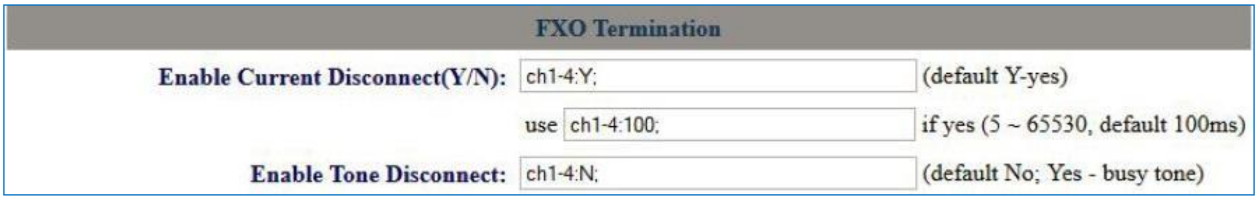

Figure 9: Configure FXO Port on the GXW410X: FXO Termination

- 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" value is 100ms, but if you start experiencing call drop 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.

Call Progress Tones	
[Syntax: ch x-y: f1=val@vol,f2=val@vol,c=on1/off1-on2/off2-on3/off3; ...]	
Note: f1,f2-frequency(Hz); vol-volume(dB); c-cadence(10ms, 0-continuous)	
Dial Tone:	ch1-4:f1=350@-11,f2=440@-11,c=0/0;
Ringback Tone:	ch1-4:f1=440@-11,f2=480@-11,c=200/400;
Busy Tone:	ch1-4:f1=480@-11,f2=620@-11,c=50/50;
Reorder Tone:	ch1-4:f1=480@-11,f2=620@-11,c=25/25;

Figure 10: Configure FXO Port on the GXW410X: Call Progress Tones

For PSTN tone detection, the tone disconnect method is widely used everywhere else in the world. TheNorth 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

Port Caller ID Setting		
Number of Rings Before Pickup:	ch1-4:2	(1-50, default 4)
Dialing to PSTN		
Wait for Dial-Tone(Y/N):	ch1-4:N;	(default No)
Stage Method(1/2):	ch1-4:1;	(default 2 stage dialing)

Figure 11: Configure FXO Port on the GXW410X - FXO Termination

- 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 the "Wait for Dial-Tone" to "No".
- Set the "Stage Method (1/2)" to 1.

REGISTER GXW410X ON FCM630A AS AN EXTENSION

Create SIP Extension on FCM630A

To manually create new SIP user, go to FCM630A web GUI-> Extension/Trunk->Extensions. Click on "Add" and a new dialog window will show for users to fill in the extension information.

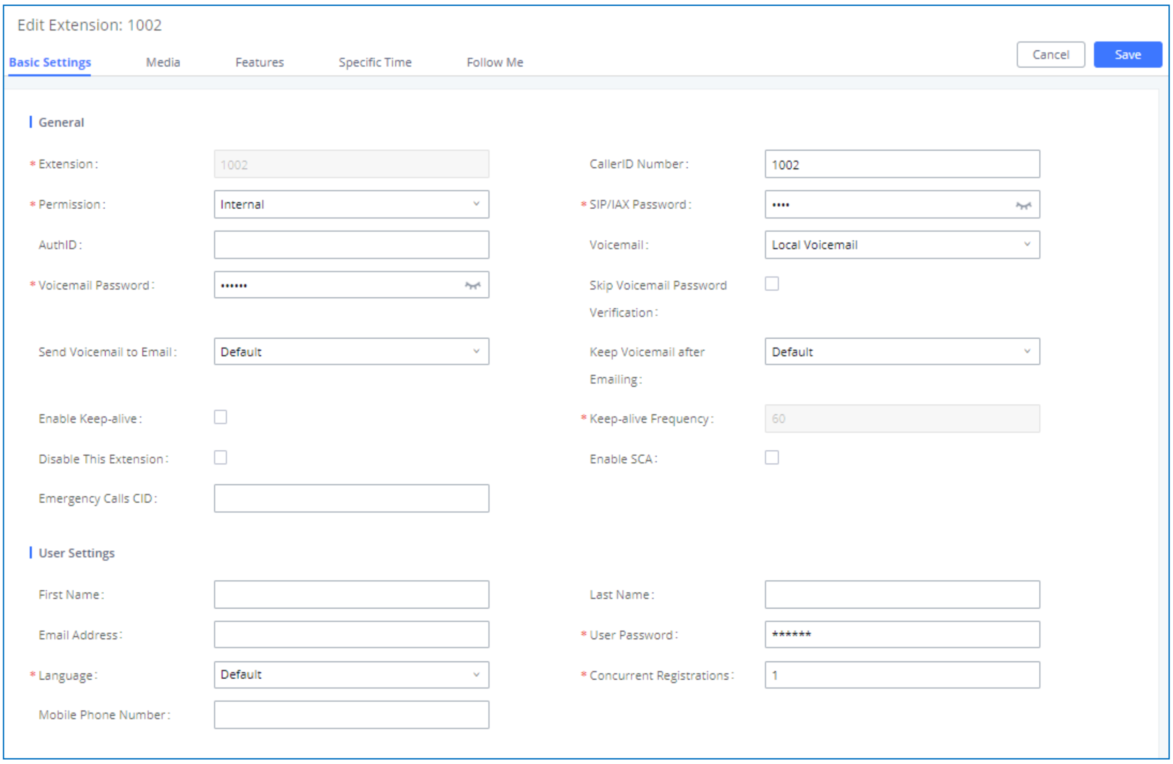

Figure 12: Create SIP Extension on FCM630A

Configure GXW410X User Setting as an Extension Registered On FCM630A

Under GXW410X web GUI->Accounts->User Account, please enter the SIP Extension information created earlier in the FCM630A. In this example, extension 1002 is used in order to register GXW410X asan extension user on FCM630A.

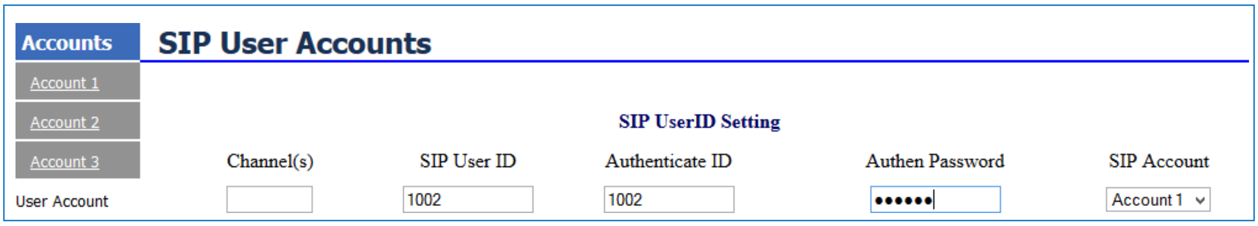

Figure 13: GXW410X User Settings

Under GXW410X web GUI, **Accounts->Account X->General Settings**, please fill in FCM630A information as explained in method 1.

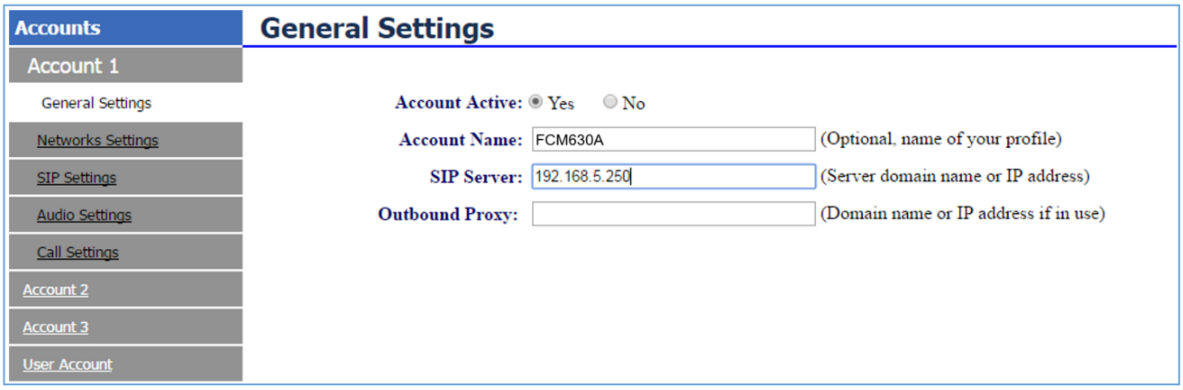

Figure 14: GXW410X User Settings: General Settings

Please make sure under **SIP Settings** tab, **SIP Registration** option is set to **Yes**, as it is required for GXW410X to successfully register on FCM630A.

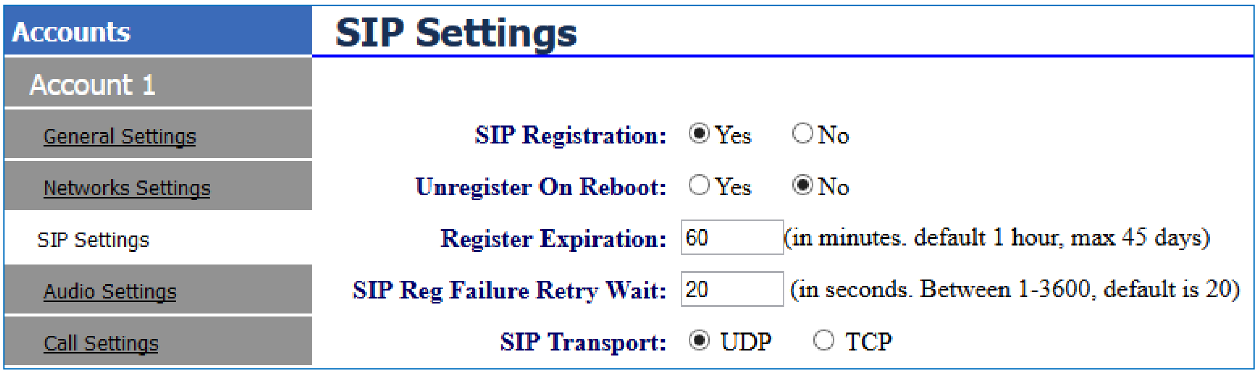

Figure 15: GXW410X SIP Settings

We can check FCM630A SIP Extension Status to see if GXW410X has been successfully registered as an extension device. The green icon indicates that GXW410X is registered on FCM630A.

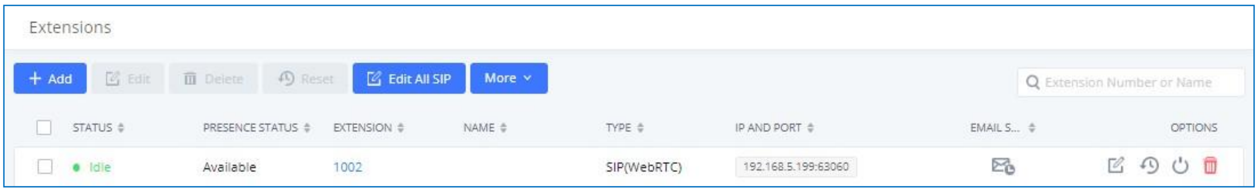

Figure 16: FCM630A - SIP Extension Status

Now GXW410X is registered at FCM630A as an extension device. Please refer to method 1 in the previous section to adjust FXO Port and DTMF settings on GXW410X.

GXW410X CALL SETTINGS

Configure Unconditional Call Forward On GXW410X

On the GXW410X web GUI, go to the Settings->Channel Settings page, configure "Unconditional Call Forward to VOIP" to the DID number 20000. This is the same number configured in FCM630A inbound route dial pattern. In this example, we will use the SIP server for profile 1 (p1).

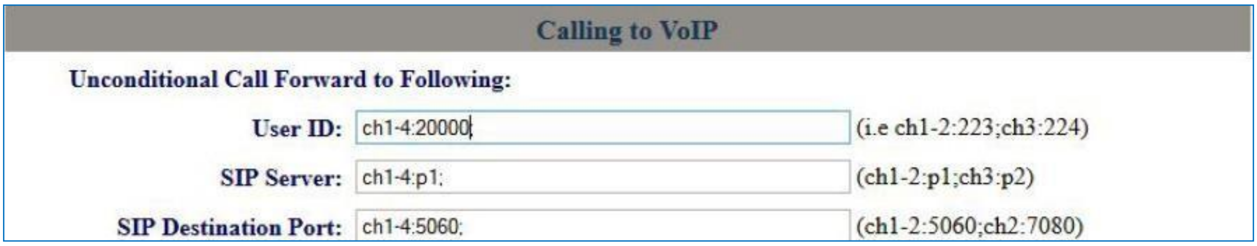

Figure 17: GXW410X - Call Forwarding

How to Dial

Once the GXW410X and the FCM630A are configured correctly, the inbound call and the outbound call will be working as described below.

- **Outbound call:**
  The extension registered to the FCM630A can dial prefix + PSTN number to reach outside numbers in PSTN network, as defined in FCM630A outbound route.

- **Inbound call:**
  The user from outside network can dial into the PSTN line's number (connected to GXW410X). And then he/she will reach the IVR of the FCM630A. The IVR on FCM630A 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 FCM630A.