



## **FIBERME Communications LLC.**

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Configuring FCM630A with GXW410X

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## 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. 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".

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# 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.

The screenshot displays the 'Create New IVR' configuration page in the FCM630A web GUI. The page is divided into two tabs: 'Basic Settings' (active) and 'Key Pressing Events'. The 'Basic Settings' tab contains the following fields and options:

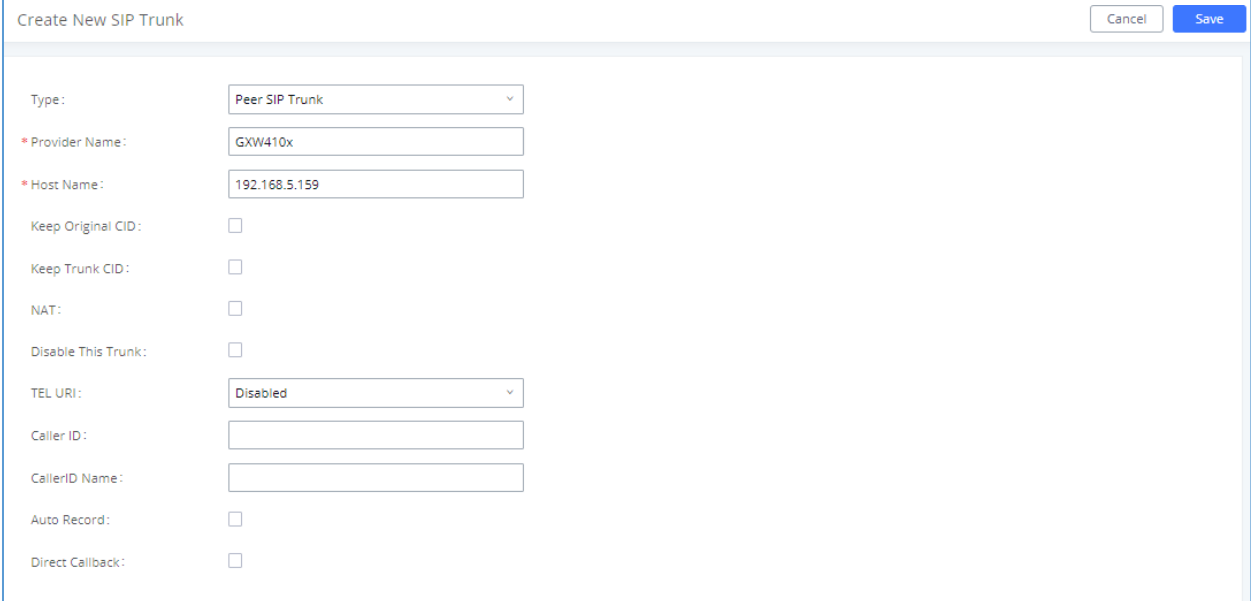
- \* Name:** Text input field containing 'GXW410X\_IVR'.
- \* Extension:** Text input field containing '7000'.
- Dial Trunk:** Check box, currently unchecked.
- Dial Other Extensions:** A group of check boxes for selecting dial destinations: 'All' (unchecked), 'Extension' (checked), 'Conference' (unchecked), 'Video Conference' (unchecked), 'Call Queue' (unchecked), 'Ring Group' (unchecked), 'Paging/Intercom Groups' (unchecked), 'Voicemail Groups' (unchecked), 'Fax Extension' (unchecked), and 'Dial By Name' (unchecked).
- \* IVR Black/Whitelist:** Dropdown menu set to 'Disable'.
- Replace Display Name:** Check box, currently unchecked.
- Return to IVR Menu:** Check box, currently unchecked.
- Alert-info:** Dropdown menu set to 'None'.
- \* Prompt:** Dropdown menu set to 'welcome', with an 'Upload Audio File' button to its right.
- Add Prompt:** A blue link with a plus icon.
- \* Digit Timeout:** Text input field containing '3'.
- \* Response Timeout:** Text input field containing '10'.
- \* Response Timeout Prompt:** Dropdown menu set to 'ivr-create-timeout', with an 'Upload Audio File' button to its right.
- \* Invalid Input Prompt:** Dropdown menu set to 'invalid', with an 'Upload Audio File' button to its right.
- \* Response Timeout Prompt Repeats:** Dropdown menu set to '3'.
- \* Invalid Input Prompt Repeats:** Dropdown menu set to '3'.
- Language:** Dropdown menu set to 'Default'.

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.



The screenshot shows a web form titled "Create New SIP Trunk" with a "Cancel" button and a "Save" button. The form contains the following fields and options:

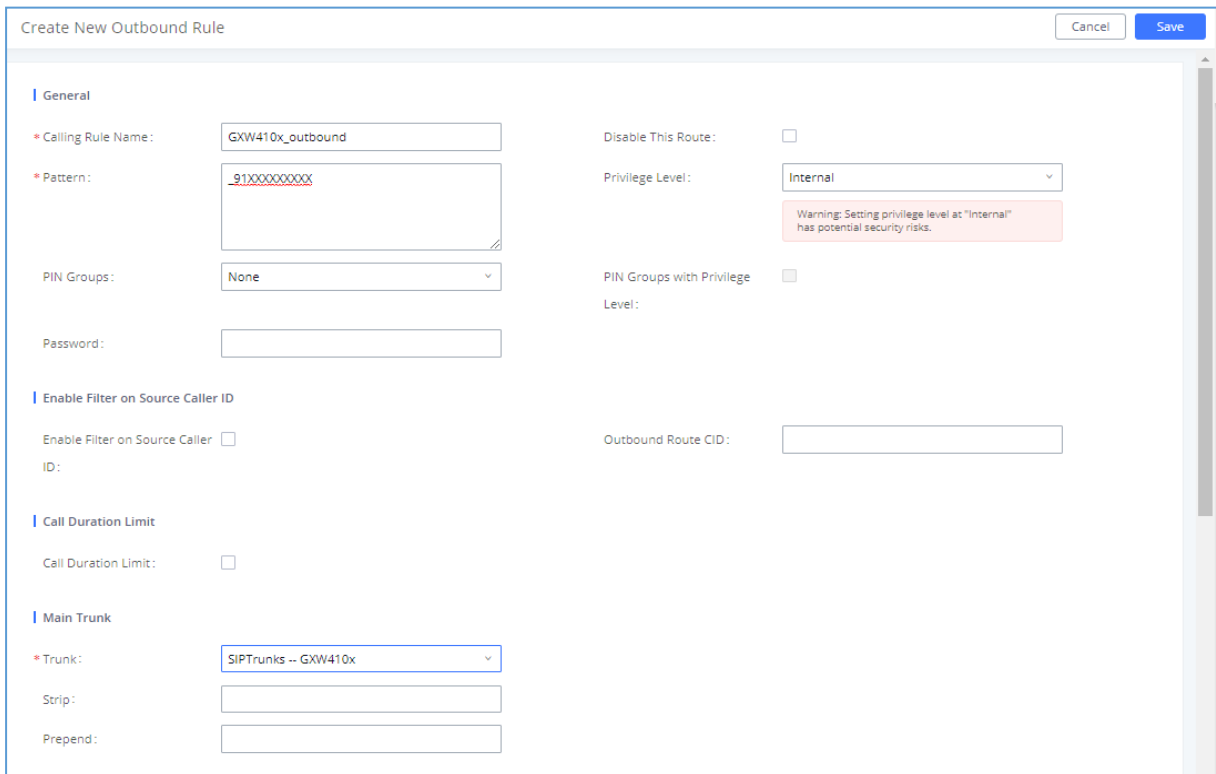
- Type: Peer SIP Trunk (dropdown menu)
- \* Provider Name: GXW410x (text input)
- \* Host Name: 192.168.5.159 (text input)
- Keep Original CID:
- Keep Trunk CID:
- NAT:
- Disable This Trunk:
- TEL URI: Disabled (dropdown menu)
- Caller ID: (text input)
- CallerID Name: (text input)
- Auto Record:
- Direct Callback:

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 enter the 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.





Accounts	SIP Settings
Account 1	
General Settings	
Networks Settings	
SIP Settings	<p><b>SIP Registration:</b> <input type="radio"/> Yes <input checked="" type="radio"/> No</p> <p><b>Unregister On Reboot:</b> <input type="radio"/> Yes <input checked="" type="radio"/> No</p> <p><b>Register Expiration:</b> <input type="text" value="60"/> (in minutes. default 1 hour, max 45 days)</p> <p><b>SIP Reg Failure Retry Wait:</b> <input type="text" value="20"/> (in seconds. Between 1-3600, default is 20)</p> <p><b>SIP Transport:</b> <input checked="" type="radio"/> UDP <input type="radio"/> TCP</p>
Audio Settings	
Call Settings	

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 able to 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**.

Settings	Channels Settings
General Settings	SIP Channel Setting
Call Settings	<p><b>DTMF Methods(1-7):</b> <input type="text" value="ch1-4:1"/> (default 1)</p> <p>(1.in-audio, 2.RFC2833, 3.1+2, 4.SIP Info, 5.1+4, 6.2+4, 7.1+2+4)</p>
Channels 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**.

Call Settings
<p><b>G723 Rate:</b> <input checked="" type="radio"/> 6.3kbps encoded</p> <p><b>Voice Frames per TX:</b> <input type="text" value="2"/> (up to 10)</p> <p><b>DTMF Payload Type:</b> <input type="text" value="101"/></p>

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.

FXO Termination
<p><b>Enable Current Disconnect(Y/N):</b> <input type="text" value="ch1-4:Y;"/> (default Y=yes)</p> <p>use <input type="text" value="ch1-4:100;"/> if yes (5 ~ 65530, default 100ms)</p> <p><b>Enable Tone Disconnect:</b> <input type="text" value="ch1-4:N;"/> (default No; Yes - busy tone)</p>

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)	
<b>Dial Tone:</b>	ch1-4:f1=350@-11,f2=440@-11,c=0/0;
<b>Ringback Tone:</b>	ch1-4:f1=440@-11,f2=480@-11,c=200/400;
<b>Busy Tone:</b>	ch1-4:f1=480@-11,f2=620@-11,c=50/50;
<b>Reorder Tone:</b>	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. 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](http://www.3amsystems.com) or download the information from <http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf>

Port Caller ID Setting	
<b>Number of Rings Before Pickup:</b>	ch1-4:2 (1-50, default 4)
Dialing to PSTN	
<b>Wait for Dial-Tone(Y/N):</b>	ch1-4:N; (default No)
<b>Stage Method(1/2):</b>	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 as an extension user on FCM630A.

Accounts	SIP User Accounts				
Account 1					
Account 2					
Account 3					
	SIP UserID Setting				
	Channel(s)	SIP User ID	Authenticate ID	Authen Password	SIP Account
User Account	<input type="text"/>	<input type="text" value="1002"/>	<input type="text" value="1002"/>	<input type="password" value="*****"/>	<input type="text" value="Account 1"/>

Figure 13: GXW410X User Settings

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



Accounts	General Settings
Account 1	
General Settings	Account Active: <input checked="" type="radio"/> Yes <input type="radio"/> No
Networks Settings	Account Name: <input type="text" value="FCM630A"/> (Optional, name of your profile)
SIP Settings	SIP Server: <input type="text" value="192.168.5.250"/> (Server domain name or IP address)
Audio Settings	Outbound Proxy: <input type="text"/> (Domain name or IP address if in use)
Call Settings	
Account 2	
Account 3	
User Account	

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.

Accounts	SIP Settings
Account 1	
General Settings	SIP Registration: <input checked="" type="radio"/> Yes <input type="radio"/> No
Networks Settings	Unregister On Reboot: <input type="radio"/> Yes <input checked="" type="radio"/> No
SIP Settings	Register Expiration: <input type="text" value="60"/> (in minutes. default 1 hour, max 45 days)
Audio Settings	SIP Reg Failure Retry Wait: <input type="text" value="20"/> (in seconds. Between 1-3600, default is 20)
Call Settings	SIP Transport: <input checked="" type="radio"/> UDP <input type="radio"/> TCP

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.

Extensions							
STATUS	PRESENCE STATUS	EXTENSION	NAME	TYPE	IP AND PORT	EMAIL S...	OPTIONS
<input type="checkbox"/>	Idle	Available	1002	SIP(WebRTC)	192.168.5.199:63060		

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).

Calling to VoIP	
<b>Unconditional Call Forward to Following:</b>	
<b>User ID:</b>	<input type="text" value="ch1-4:20000;"/> (i.e ch1-2:223;ch3:224)
<b>SIP Server:</b>	<input type="text" value="ch1-4:p1;"/> (ch1-2:p1;ch3:p2)
<b>SIP Destination Port:</b>	<input type="text" value="ch1-4:5060;"/> (ch1-2:5060;ch2:7080)

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.

