Grandstream Networks, Inc.

Configuring UCM6100 Series with GXW410X

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OVERVIEW

This document describes basic configuration to interconnect UCM6100 series and GXW410X. In this example, we will be using a GXW4104. 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 UCM6100 series IP PBX with the GXW410X.

- Method 1: Register the GXW410X to the UCM6100 directly.
- Method 2: Configure GXW410X as a SIP peer trunk.

⚠ **Warning:**

- When the UCM6100 series is interconnected with other GXW410X, it is NOT recommended to turn on "Allow Guest Calls" under the UCM6100 web GUI->**PBX**->**SIP Settings**->**General**. Turning on this option will allow unauthenticated calls coming through the UCM6100 series. Please be aware of the security concerns when using this option.
- When using the IVR in UCM6100 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 UCM6100'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".

Connect UCM6100 to GXW410X Using Peer SIP Trunk

Create IVR on UCM6100

On the UCM6100 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 UCM6100 IVR will be able to reach the internal extensions registered to the UCM6100. Also, you can assign the "Key Pressing Event" to different destinations.

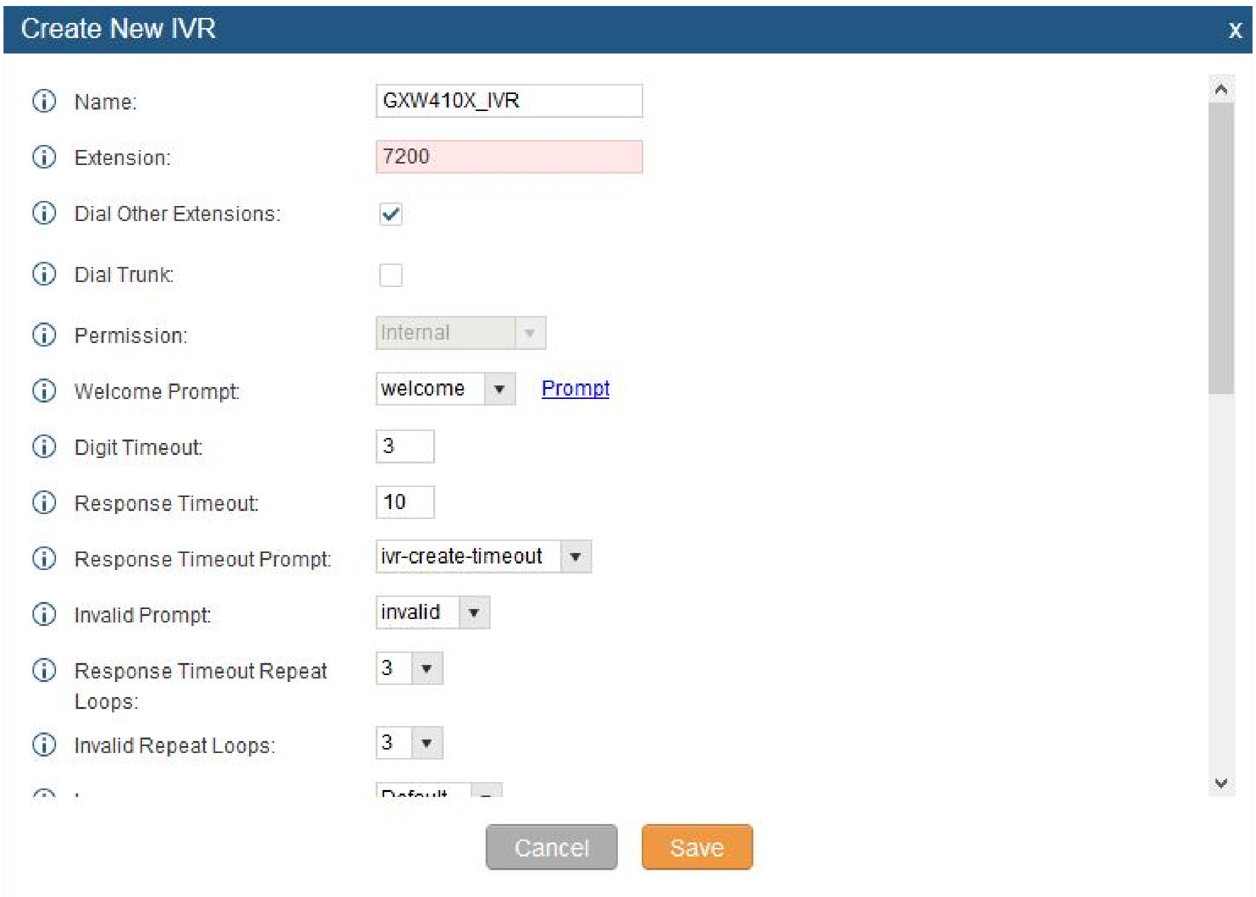

Figure 1: Method 2 - Create IVR 7200 on the UCM6100

Create Peer SIP Trunk on UCM6100

On the UCM6100 web GUI, create a peer SIP trunk under **PBX**->**Basic/Call Routes**->**VOIP Trunks**. In this example, the GXW410X IP address is 192.168.40.137.

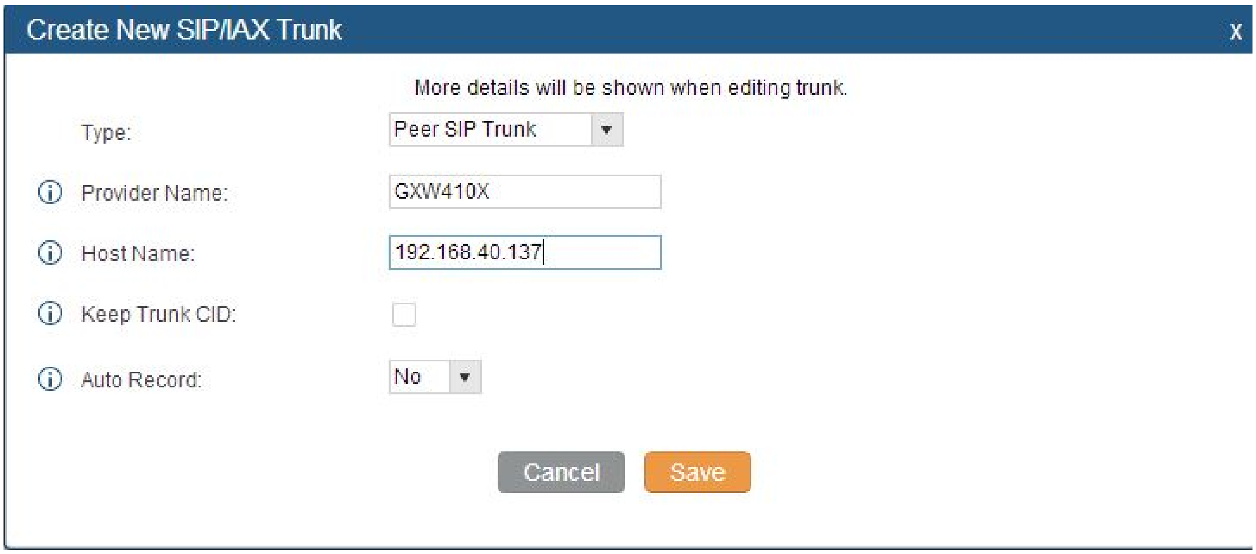

Figure 2: Method 2 - Create Peer SIP Trunk on the UCM6100

Configure Outbound Rule on UCM6100

On the UCM6100 web GUI, go to **PBX**->**Basic/Call Routes**->**Outbound Routes** to create a new outbound rule. This would allow the extension on the UCM6100 to reach numbers in PSTN network via the peer SIP trunk we just configured.

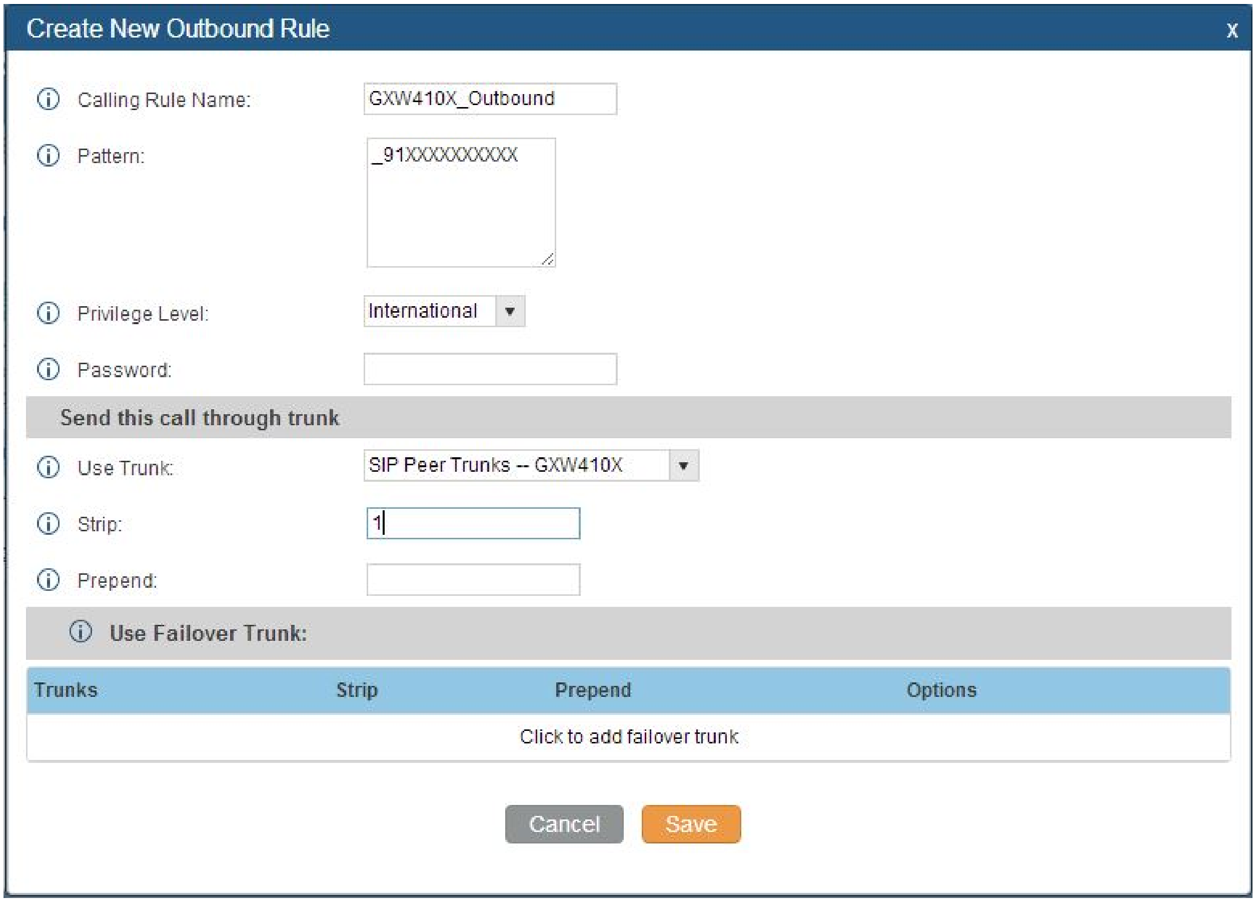

Figure 3: Method 2 - Configure Outbound Rule on the UCM6100

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 UCM6100

On the UCM6100 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 GXW410X call forward setting.

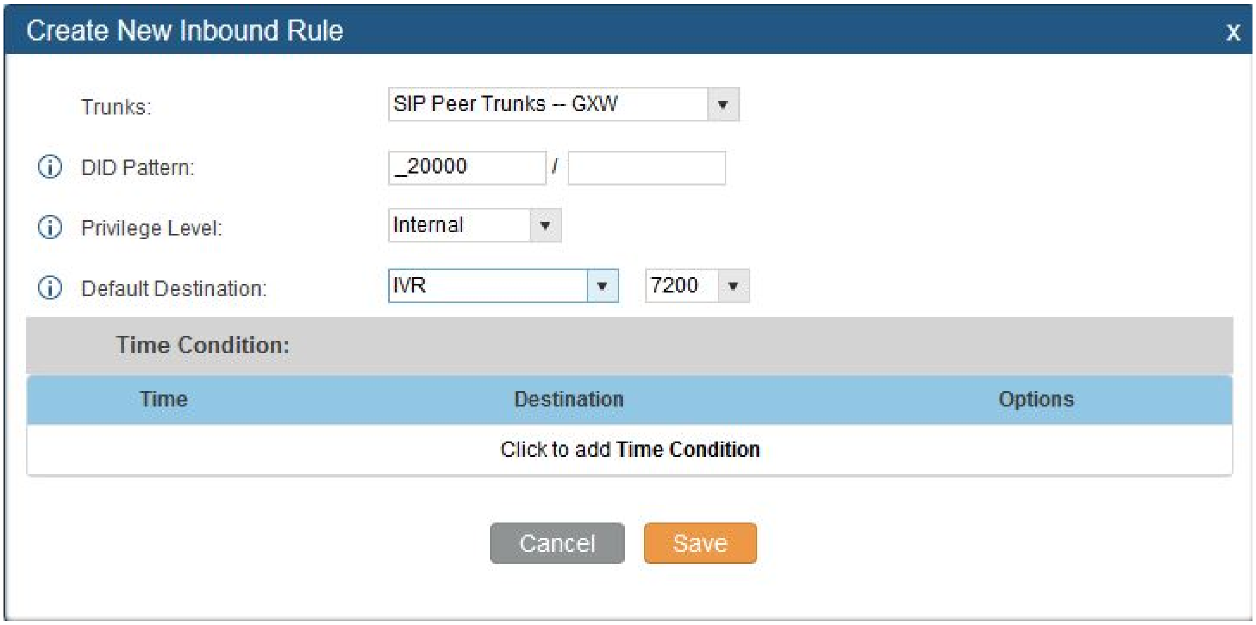

Figure 4: Method 2 - Configure Inbound Rule on the UCM6100

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

Configure FXO Port on GXW410X

1. Connect the PSTN line to the GXW410X FXO port.

2. On the GXW410X web GUI, go to the Accounts page and insert the IP address of the UCM that you are peering with.

   In this example, the UCM6100 IP address is 192.168.40.207.

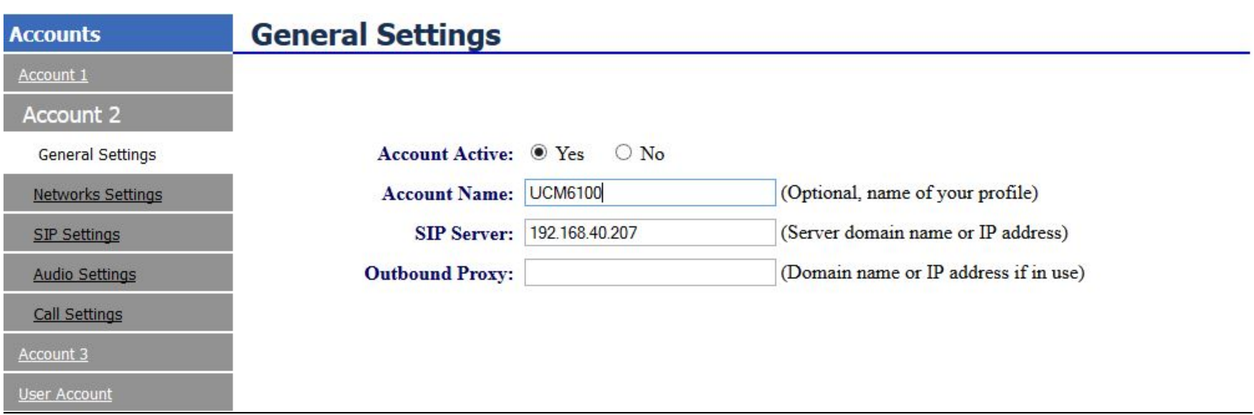

Figure 5: Method 2 – Configure FXO Port on GXW410X: Registration

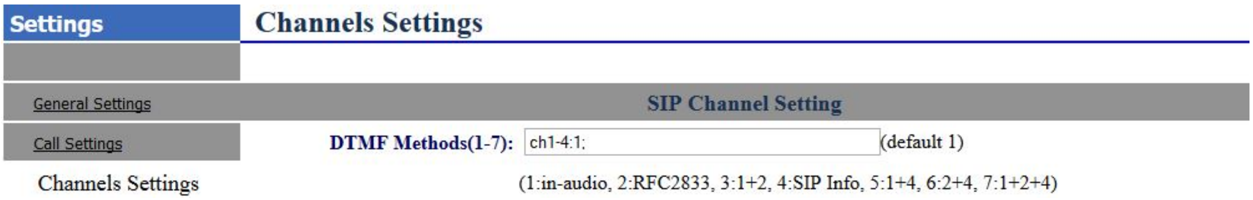

Figure 6: Method 2 - Configure FXO Port on the GXW410X: DTMF Method

Since we are going to use IVR when the call is forwarded to the UCM6100, the UCM6100 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

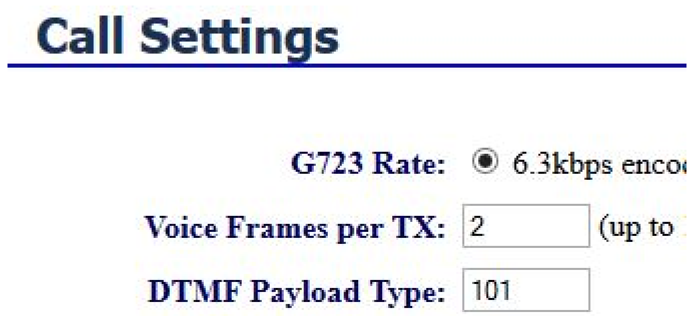

Figure 7: Method 2 - Configure FXO Port on the GXW410X: DTMF Payload Type

- Set the DTMF Payload Type to 101. This value can be found under SETTINGS → CALL SETTINGS

There are a few necessary changes to be made in FXO termination section. This can be found in the FXO LINES settings page.

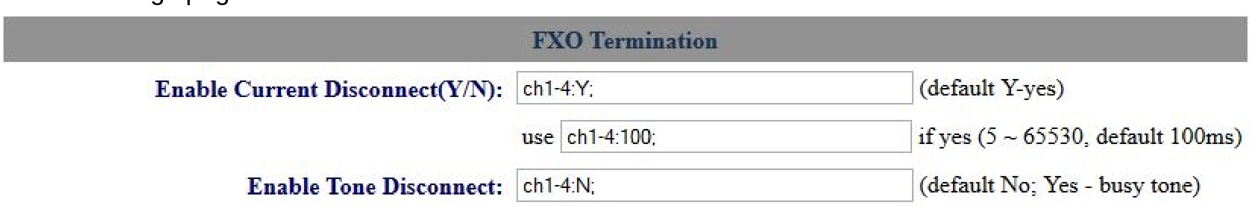

Figure 8: Method 2 - Configure FXO Port on the GXW410X: FXO Termination

- 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" 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.

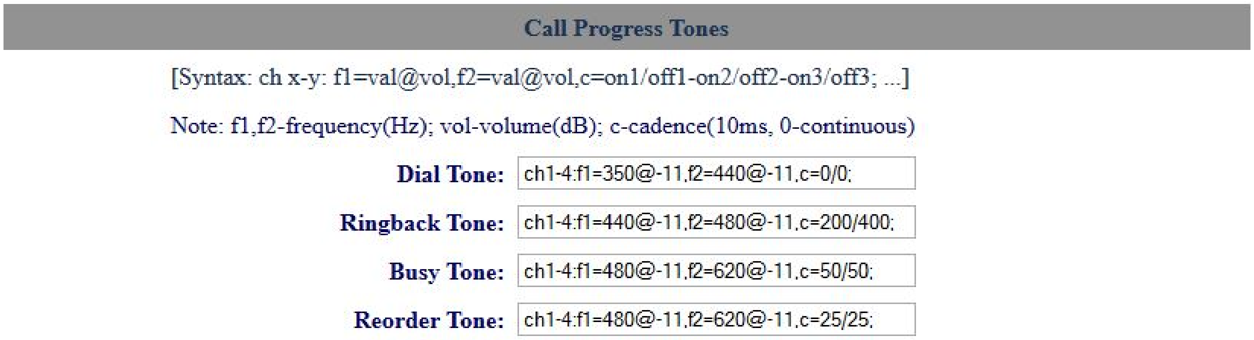

Figure 9: Method 2 - Configure FXO Port on the GXW410X: Call Progress Tones

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 or download the information from http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf.

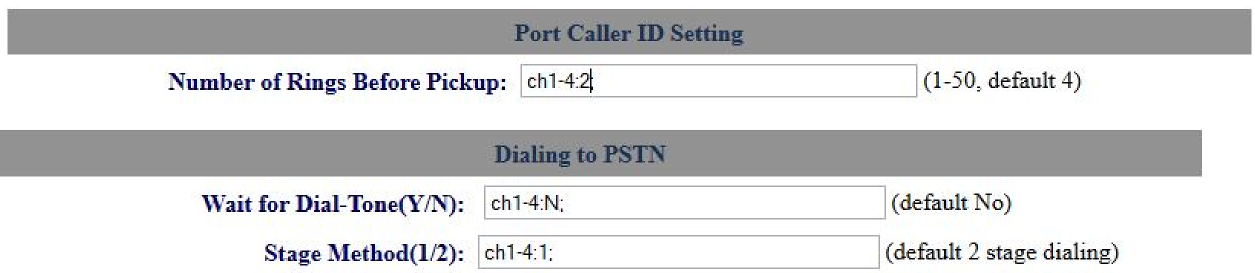

Figure 10: Method 2 - Configure FXO Port on the GXW410X: FXO Termination

- 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.

Figure 11: Method 2 - Configure FXO Port on the GXW410X: Channel Dialing

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 UCM6100 inbound route dial pattern. In this example, we will use the SIP server for profile 1 (p1).

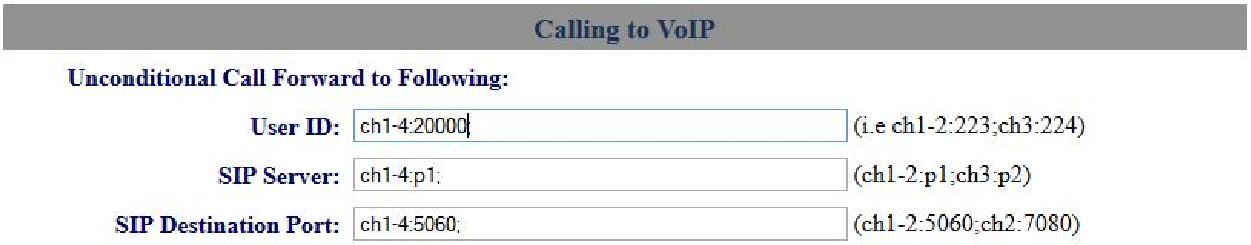

Figure 12: Method 2 - GXW410X: Call Forwarding

How to Dial

Once the GXW410X and the UCM6100 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 UCM6100 can dial prefix + PSTN number to reach outside numbers in PSTN network, as defined in UCM6100 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 UCM6100. The IVR on UCM6100 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 UCM6100.