FIBERME Communications LLC.

Configuring FCM630A Series with HT813

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OVERVIEW

This document describes basic configuration to interconnect FCM630A series and HT813. This is typically applied to the scenario where users would like to add a HT813 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 in a remote location and use the carrier service on another remote office, in this case this guide will help you implement this configuration.

There are two ways to set up the FCM630A series IP PBX with the HT813.

- **Method 1:** Register the HT813 to the FCM630A directly.
- **Method 2:** Configure HT813 as a SIP peer trunk.

The following illustration show the typical setup that will be used in this guide:

Figure 1: Typical Architecture

⚠ Warning:

- When using the IVR in FCM630A, please be aware that if "Dial Trunk" option is turned on in IVR settings, the callers 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".

METHOD 1: REGISTER HT813 TO FCM630A

Create Extension on FCM630A

On the FCM630A web GUI, create two extensions under Extension/Trunk→Extensions. These two extensions are used for HT813 FXS and FXO registration.

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

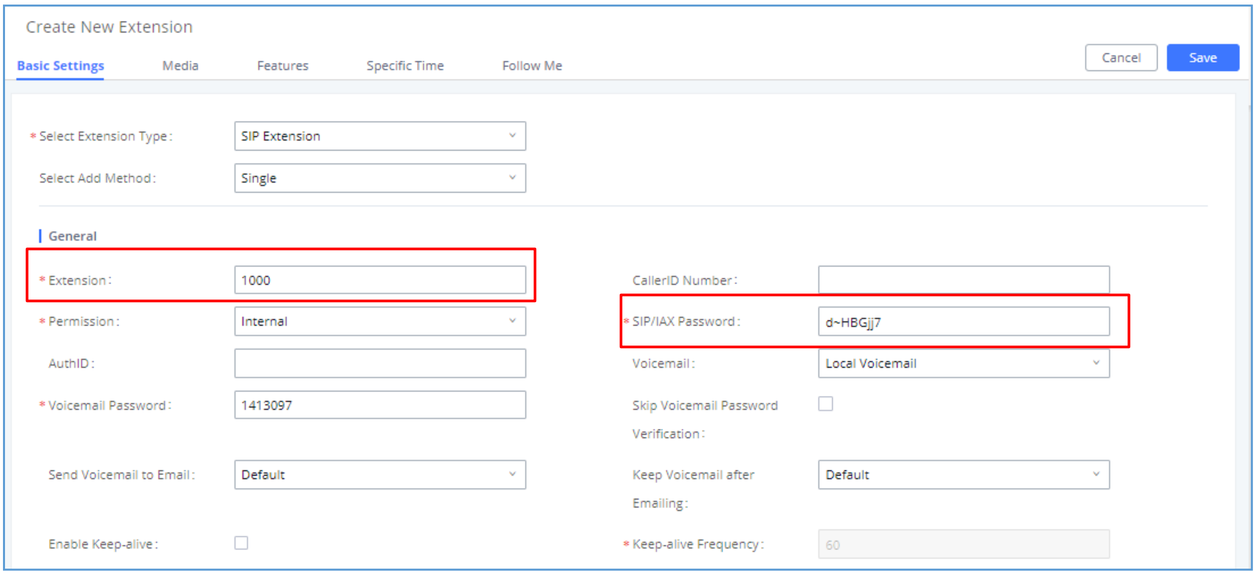

Figure 2: Create Extension 1000 on the FCM630A

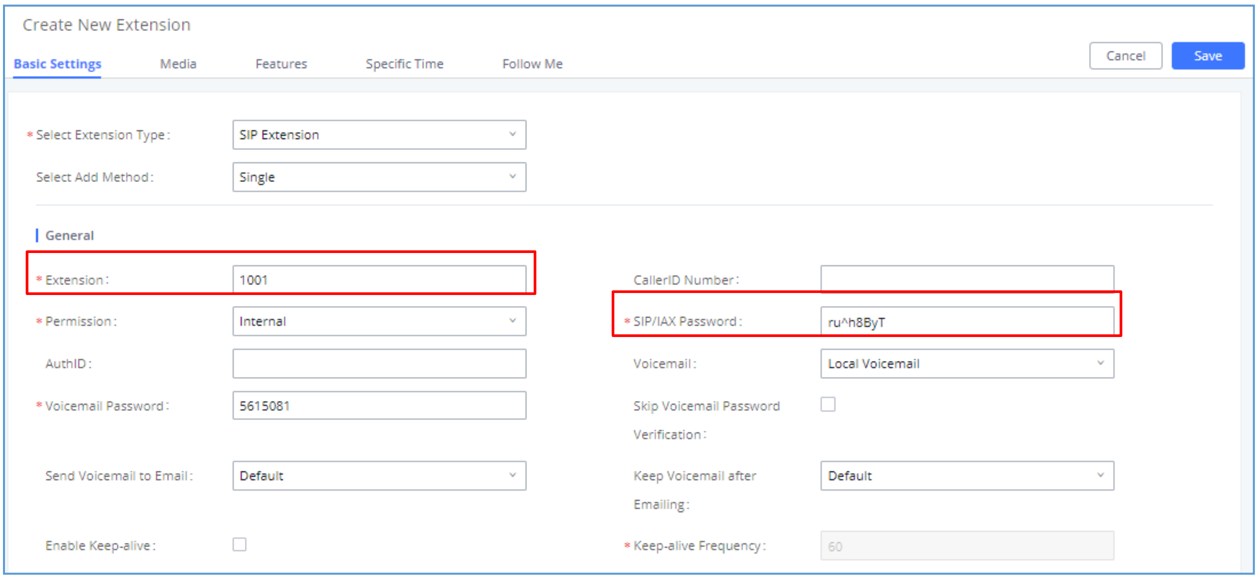

Figure 3: Create Extension 1001 on the FCM630A

Create IVR on FCM630A

On the FCM630A web GUI, create an IVR extension under **Call Features→IVR**. This is to receive the calls forwarded from the HT813.

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

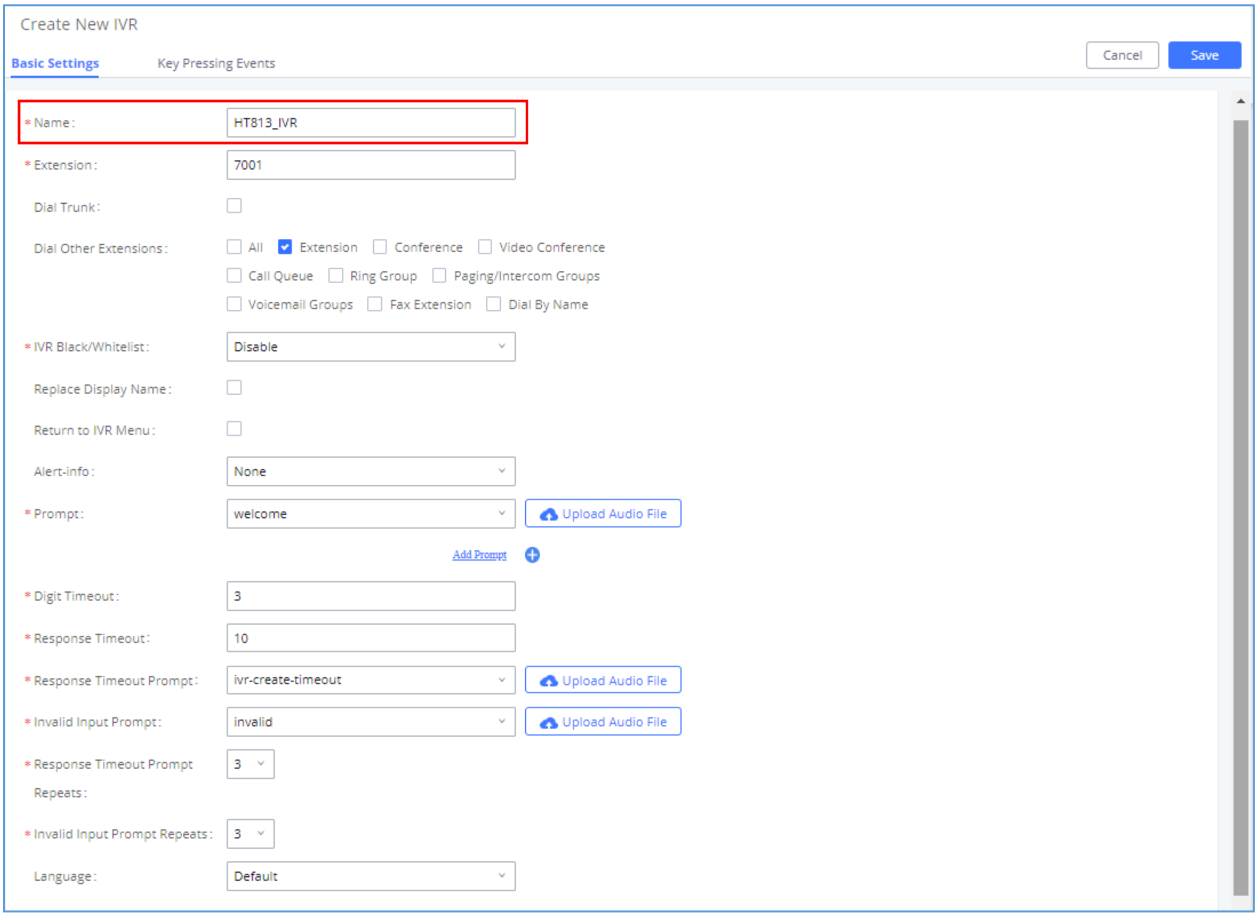

Figure 4: Create IVR 7000 on the FCM630A

Configure FXS Port on HT813

1. Connect an analog phone to the HT813 FXS port.
2. On the HT813 web GUI, go to FXS Port setting page, configure to register the FXS port to the FCM630A extension 1000. Please refer to the highlighted settings in the following figure.

   In this example, the FCM630A IP address is 192.168.5.190.

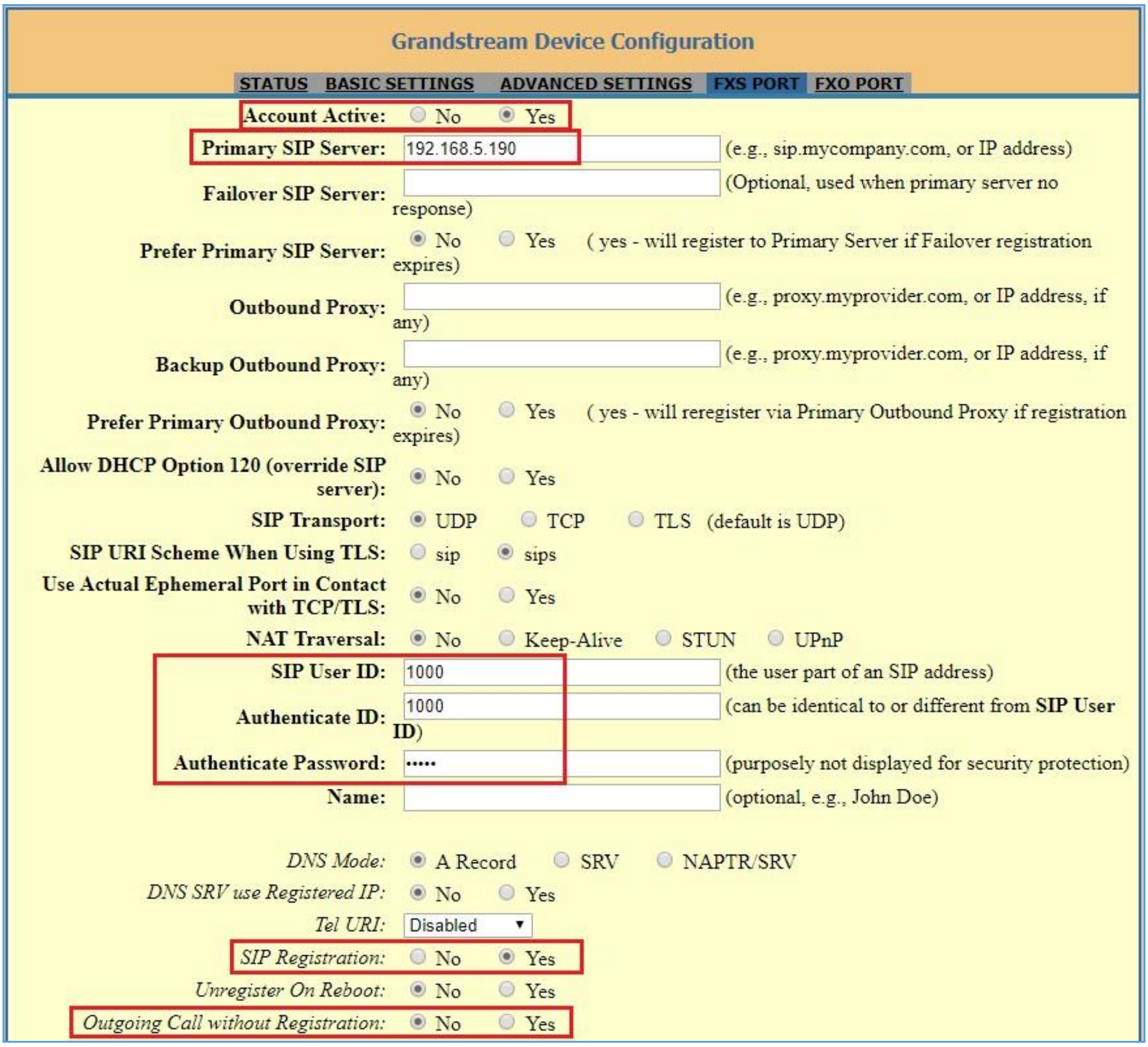

Figure 5: Configure FXS Port on the HT813

Configure FXO Port on HT813

1. Connect the PSTN line to the HT813 FXO port.
2. On the HT813 web GUI, go to FXO Port setting page, configure to register the FXO port to the FCM630A extension 1001. Please refer to the highlighted settings and other necessary settings in the following figures.

   In this example, the FCM630A IP address is 192.168.5.190.

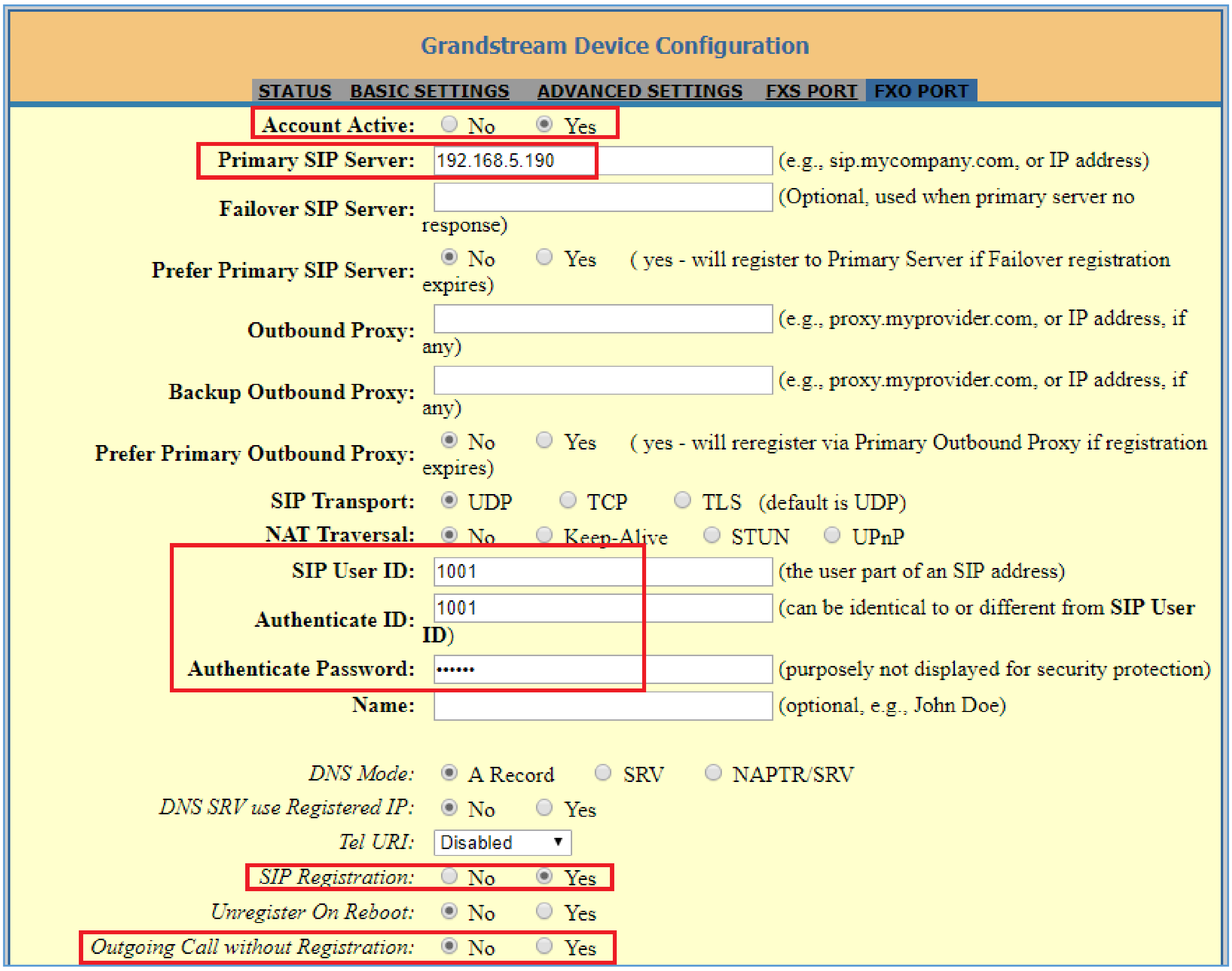

Figure 6: Configure FXO Port on the HT813 - Registration

Since we are going to use IVR when the call is forwarded to the FCM630A, the FCM630A will need to be able to detect the DTMF digits. Configure the HT813 FXO port DTMF settings as below as an initial setup.

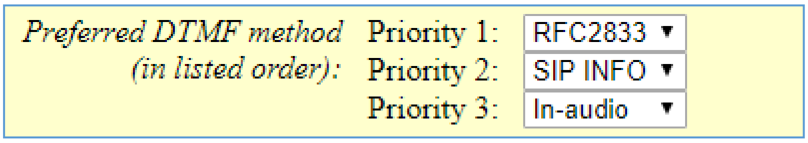

Figure 7: Configure FXO Port on the HT813 - DTMF Settings

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

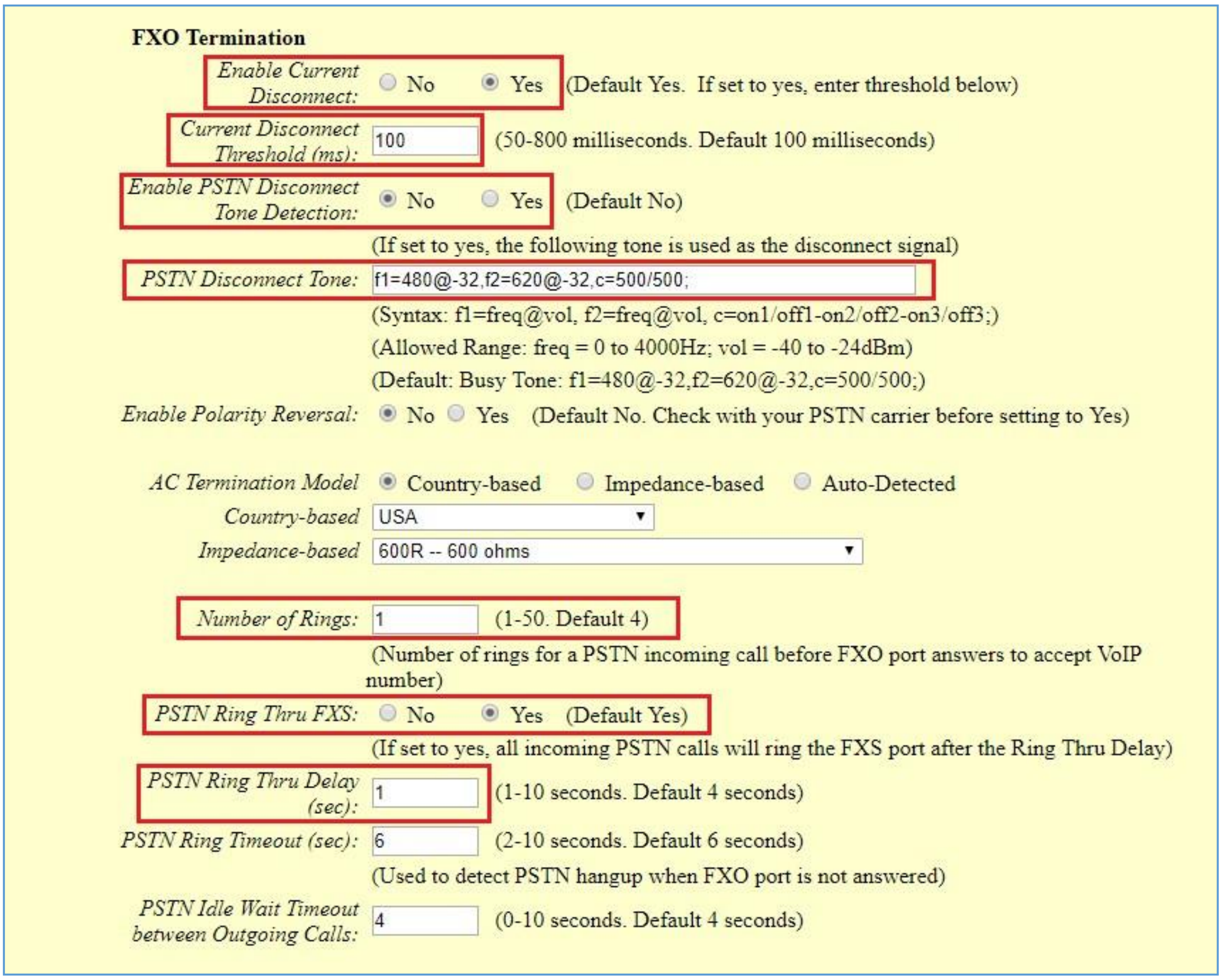

Figure 8: Configure FXO Port on the HT813 - FXO Termination

- 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 thenyou 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. 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.

- Set "Number of Rings" option to 1. If you happen to experience caller ID issue, you may set it to 2 or 4.

- Set "PSTN Ring Thru FXS" to "No" if you prefer not to ring the FXS port on incoming PSTN calls 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 to2.
- Set the "Stage Method (1/2)" to 2 for 2-stage dialing.

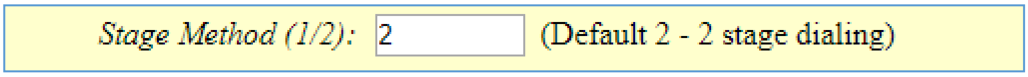

Figure 9: Configure FXO Port on the HT813 - Channel Dialing

Configure Unconditional Call Forward on HT813

On the HT813 web GUI, go to Basic setting page, configure "Unconditional Call Forward to VOIP" to the IVR extension on the FCM630A. In this example, the FCM630A IP address is 192.168.5.190.

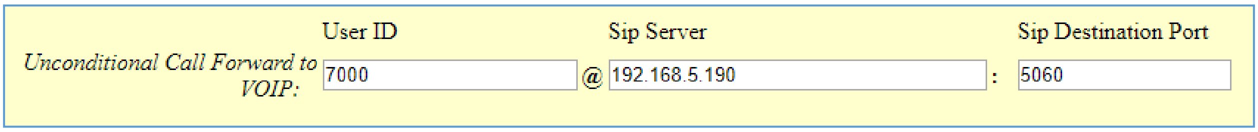

Figure 10: HT813 Basic Settings

How to Dial

Once the HT813 and the FCM630A 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 FCM630A can dial the HT813'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 HT813). 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.

METHOD 2: CONNECT FCM630A TO HT813 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.

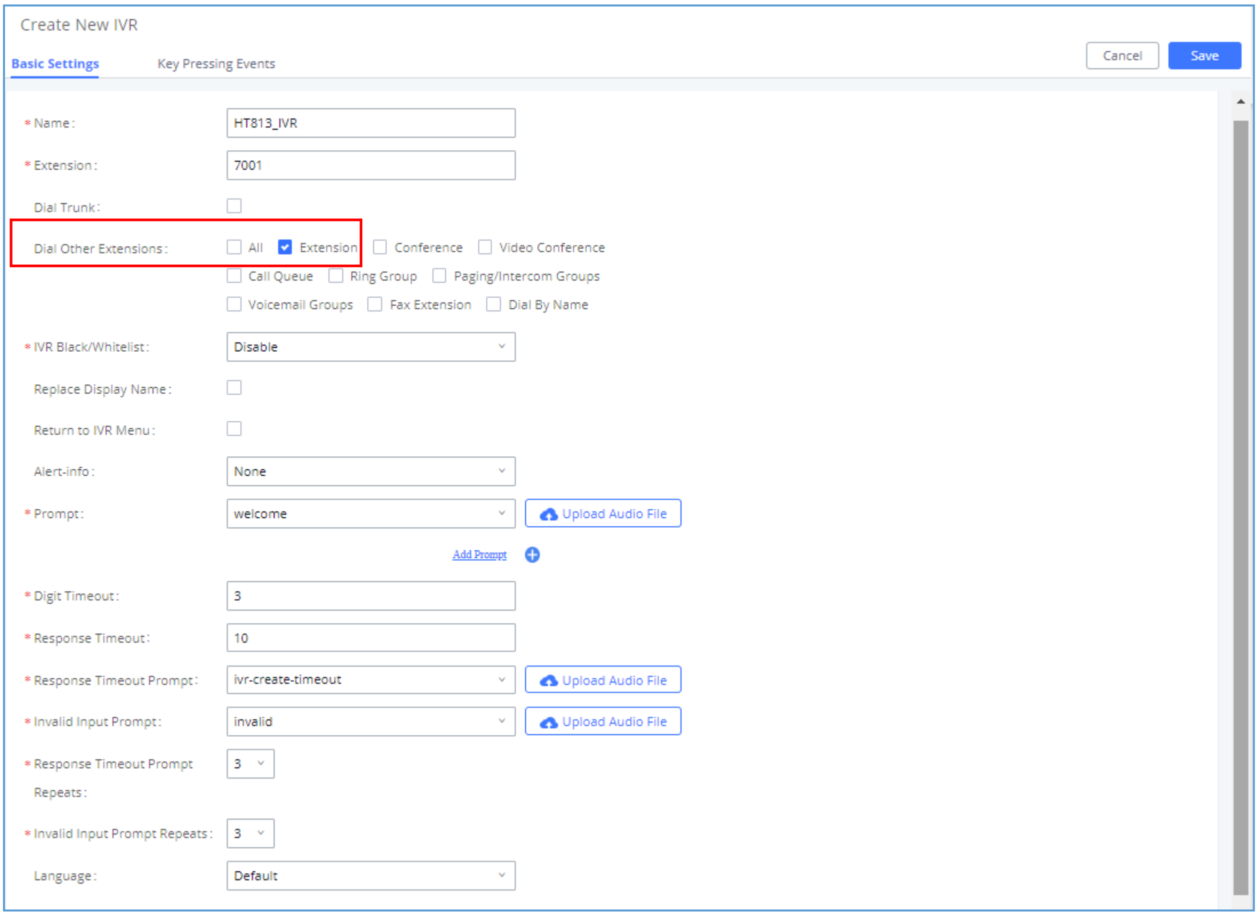

Figure 11: Create IVR 7000 on the FCM630A

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 HT813 IP address is 192.168.5.144.

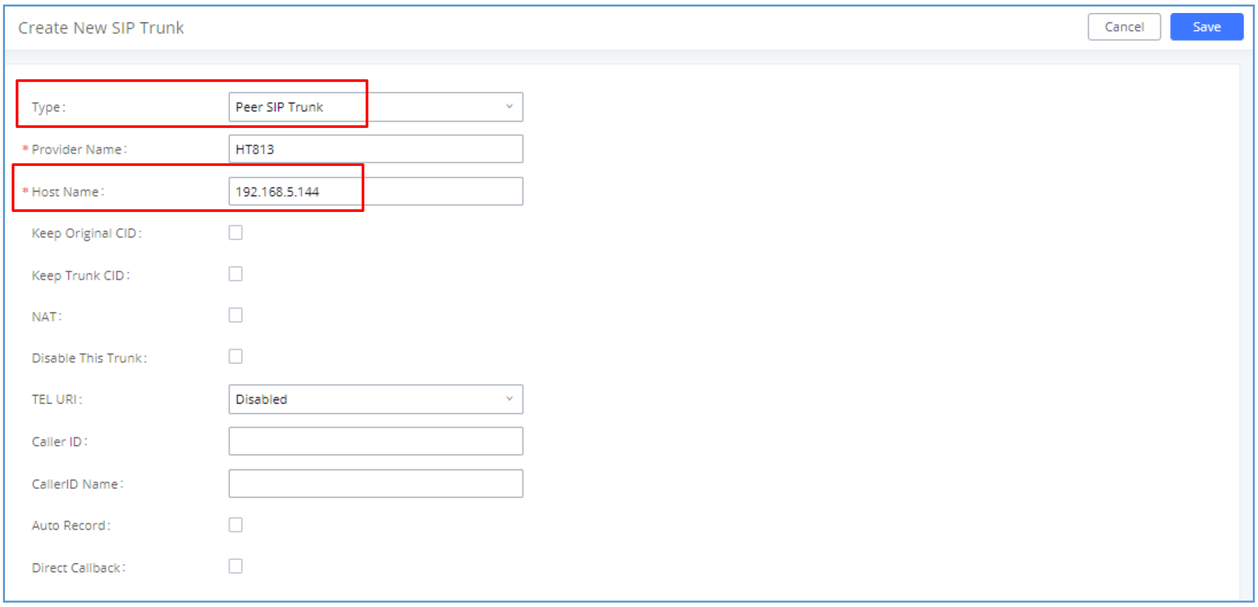

Figure 12: Create Peer SIP Trunk on the FCM630A

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.

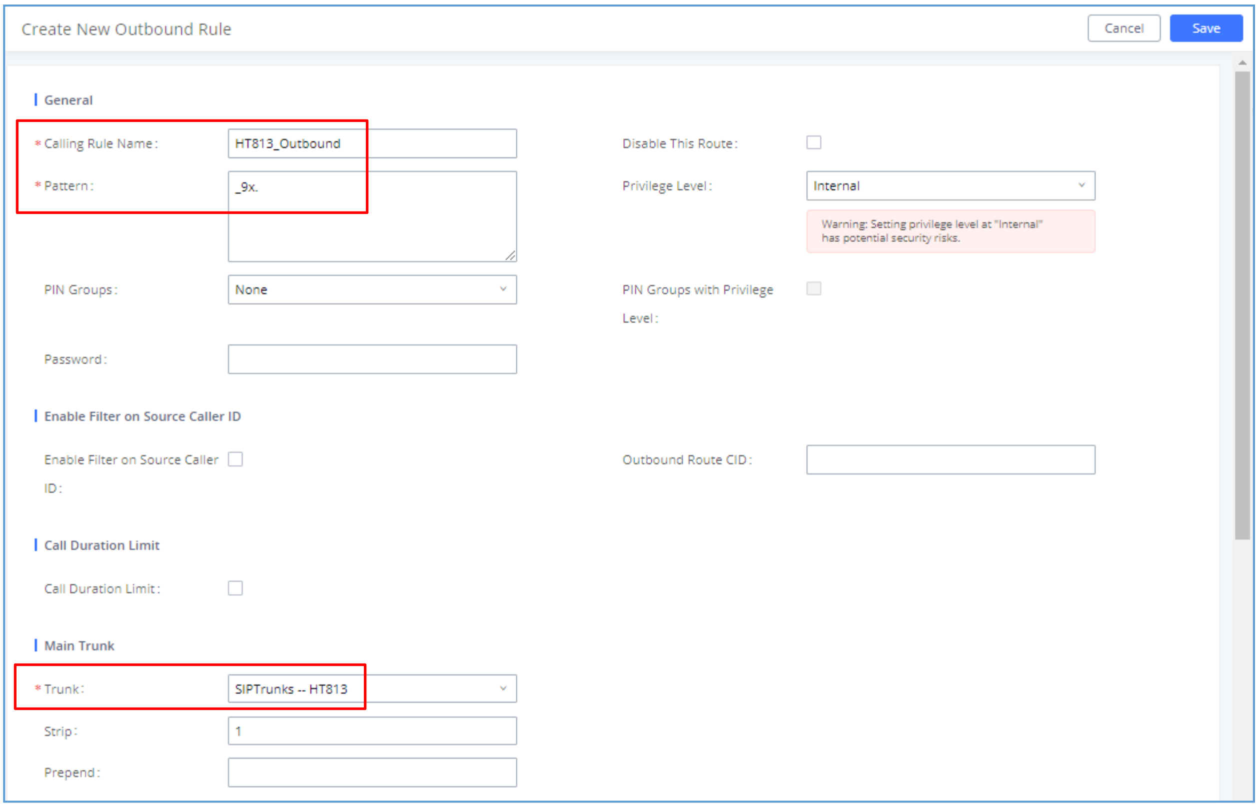

Figure 13: Configure Outbound Rule on the FCM630A

In this example "9x.", 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 Routes** to create a new inbound rule.In this example, we create the DID as 20000, which will be used in the HT813 call forward setting.

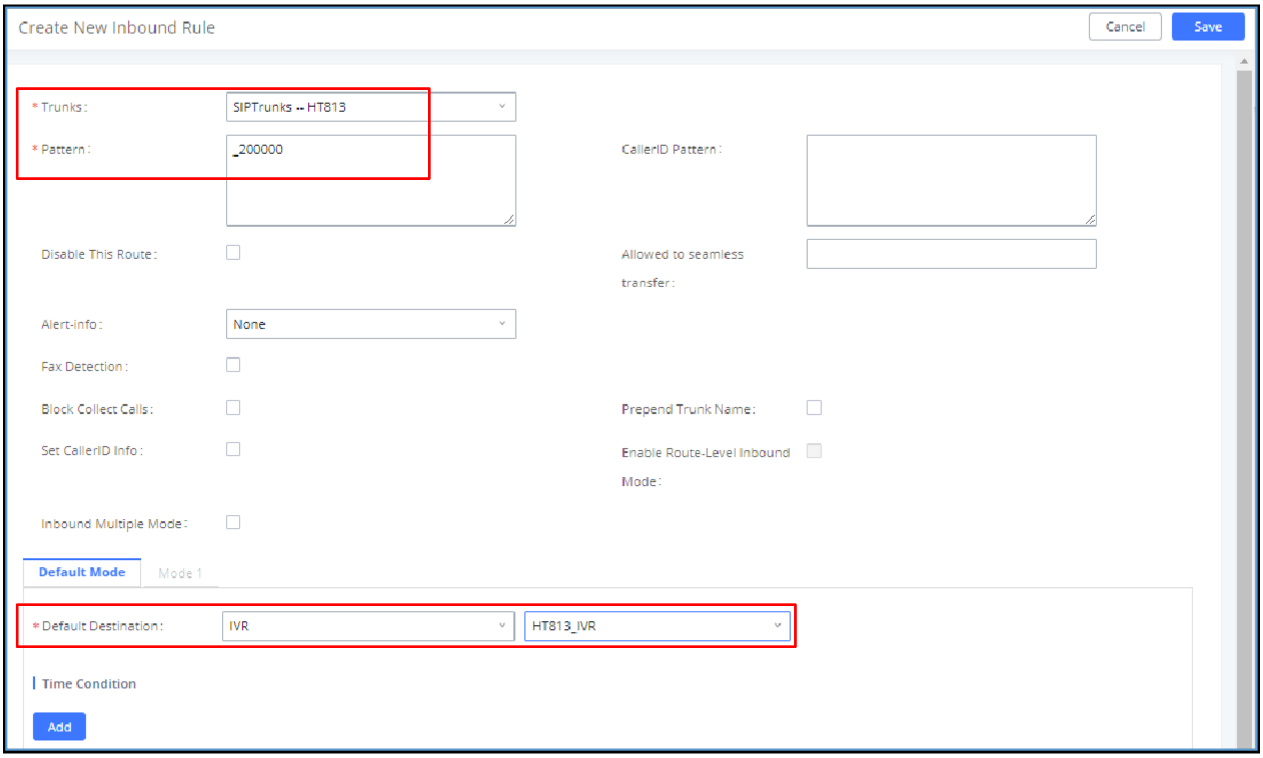

Figure 14: Configure Inbound Rule on the FCM630A

The default destination is configured to IVR.

Configure FXO Port on HT813

1. Connect the PSTN line to the HT813 FXO port.
2. On the HT813 web GUI, go to FXO Port setting page, configure the FXO port to send signaling SIP messages to the FCM630A's IP address. Please refer to the highlighted settings and other necessary settings in the following figures.

You can set anything you want on the SIP user ID, authentication ID and username. We choose 1111in our example.

In this example, the FCM630A IP address is 192.168.5.190.

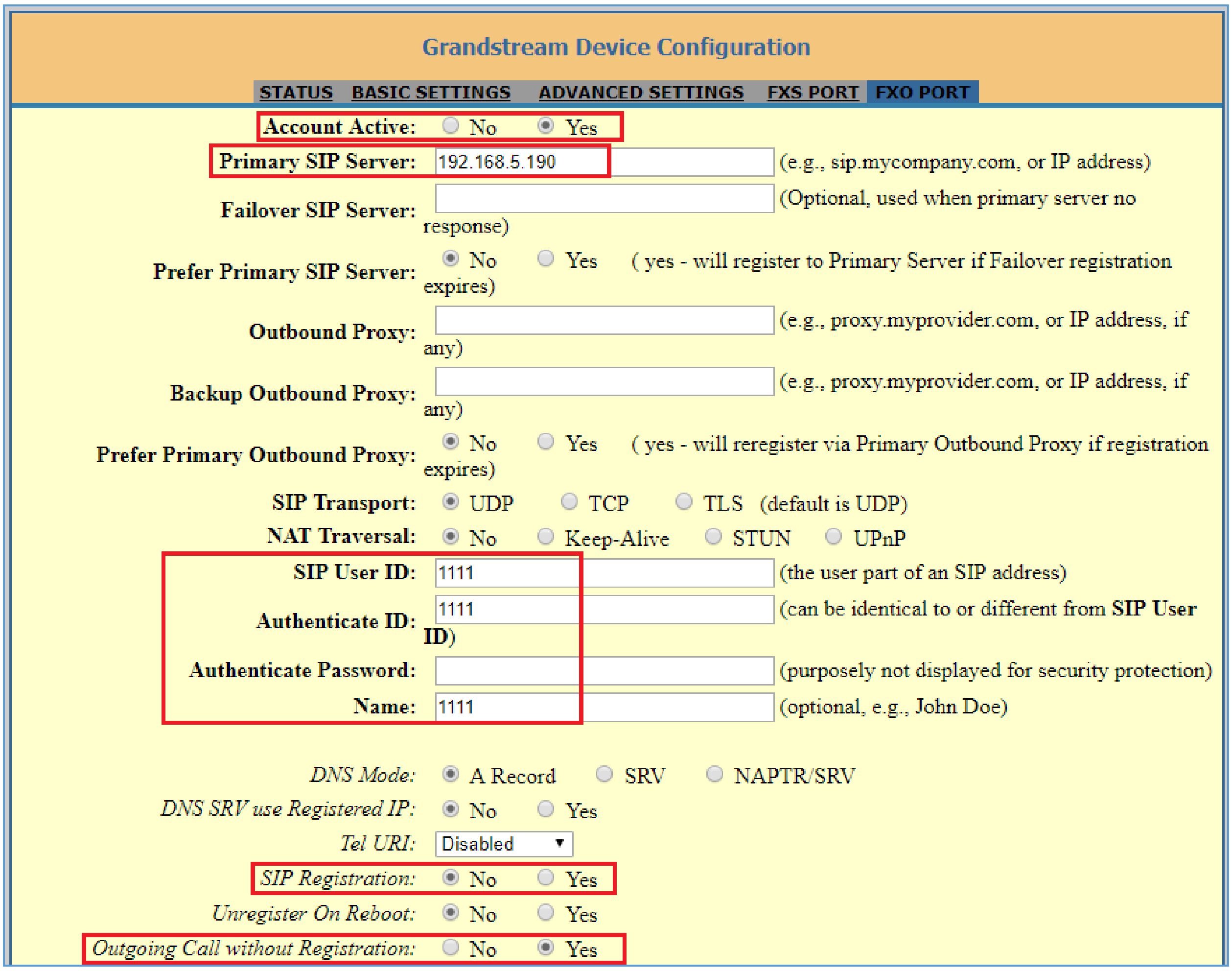

Figure 15: Configure FXO Port on the HT813 - Registration

Since we are going to use IVR when the call is forwarded to the FCM630A, the FCM630A will need to be able to detect the DTMF digits. Configure the HT813 FXO port DTMF settings as below for an initial setup.

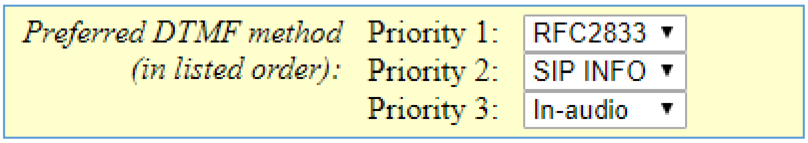

Figure 16: Configure FXO Port on the HT813 - DTMF Settings

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

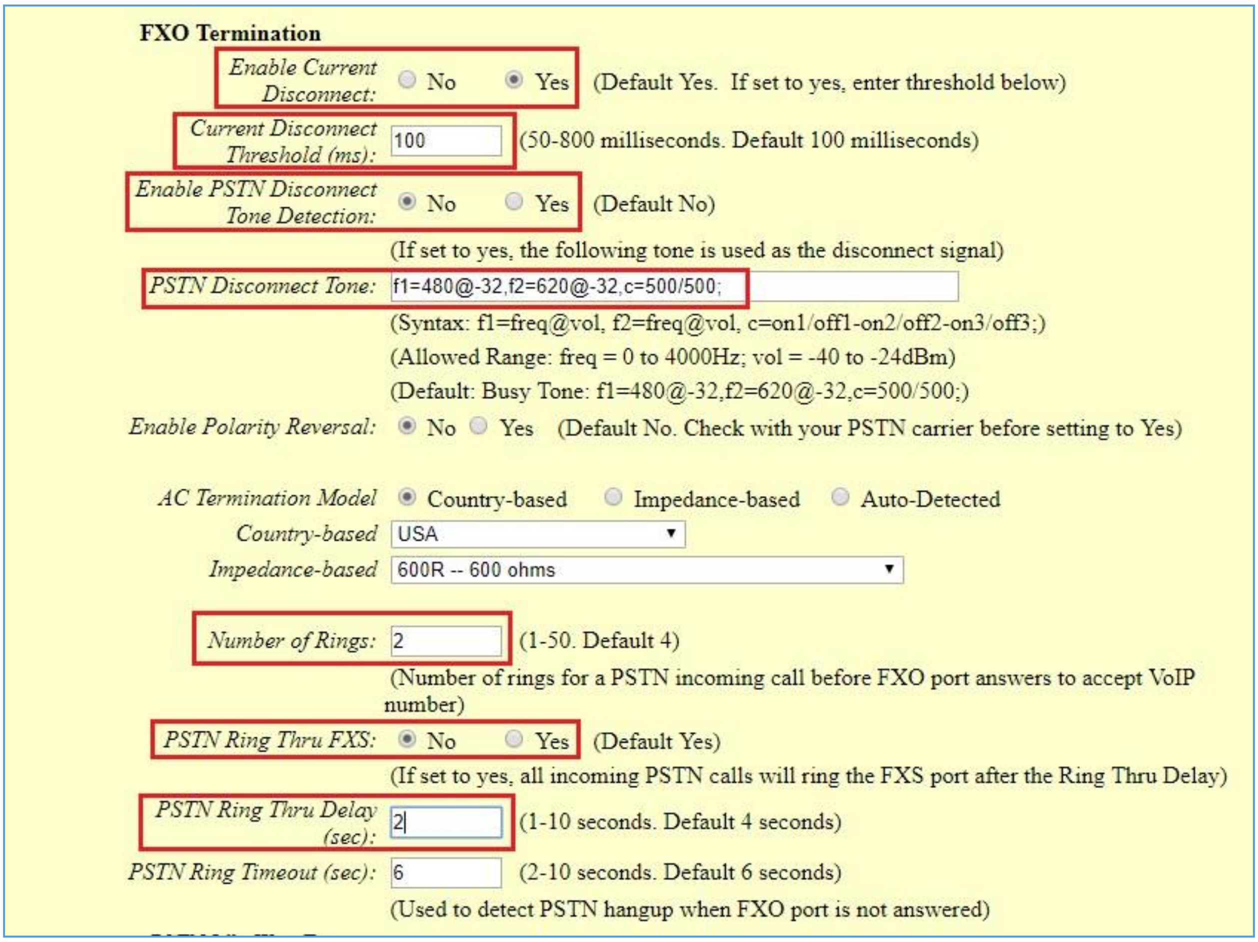

Figure 17: Configure FXO Port on the HT813: FXO Termination

- 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 thenyou 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. 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.

- 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.

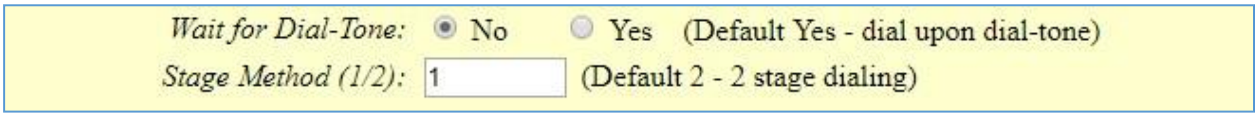

Figure 18: Configure FXO Port on the HT813 - Channel Dialing

Exchange SIP Port Settings for FXS and FXO on HT813

- On the HT813 web GUI, go to FXO setting page, configure the "Local SIP Port" to be 5060. (The default setting is 5062.)

- On the HT813 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 HT813

On the HT813 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 FCM630A inbound route dial pattern. In this example, the FCM630A IP address is 192.168.5.250.

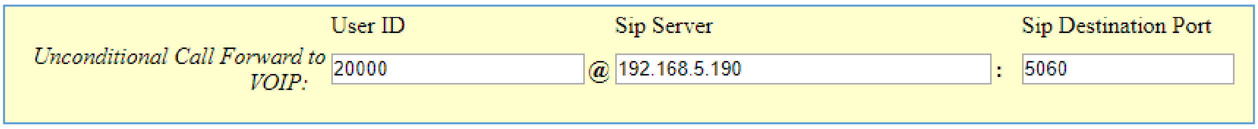

Figure 19: HT813 Basic Settings

How to Dial

Once the HT813 and the FCM630A 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 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 HT813). And then he/she will reach the IVR of the FCM630A. The IVR on FCM630A would allow the user to further enterextension number or key pressing digit to reach the desired destination. The inbound call will go through the inbound route set up on the FCM630A.

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