Grandstream Networks, Inc.

Configuring UCM6100 Series with HT503

Grandstream Networks, Inc.

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CHANGE LOG

This section documents significant changes from previous versions of this configuration tutorial. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

Firmware Version 1.0.1.25

- This is the initial version.

OVERVIEW

This document describes basic configuration to interconnect UCM6100 series and HT503. This is typically applied to the scenario where users would like to add a HT503 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 from another PBX or a PSTN line in a remote location, but we don't want to invest too much on a FXO gateway.

There are two ways to set up the UCM6100 series IP PBX with the HT503.

- Method 1: Register the HT503 to the UCM6100 directly.
- Method 2: Configure HT503 as a SIP peer trunk.

⚠ **Warning:**

- When the UCM6100 series is interconnected with other HT503, 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".

METHOD 1: REGISTER HT503 TO UCM6100

Create Extension on UCM6100

On the UCM6100 web GUI, create two extensions under **PBX**->**Basic/Call Routes**->**Extensions**. These two extensions are used for HT503 FXS and FXO registration.

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

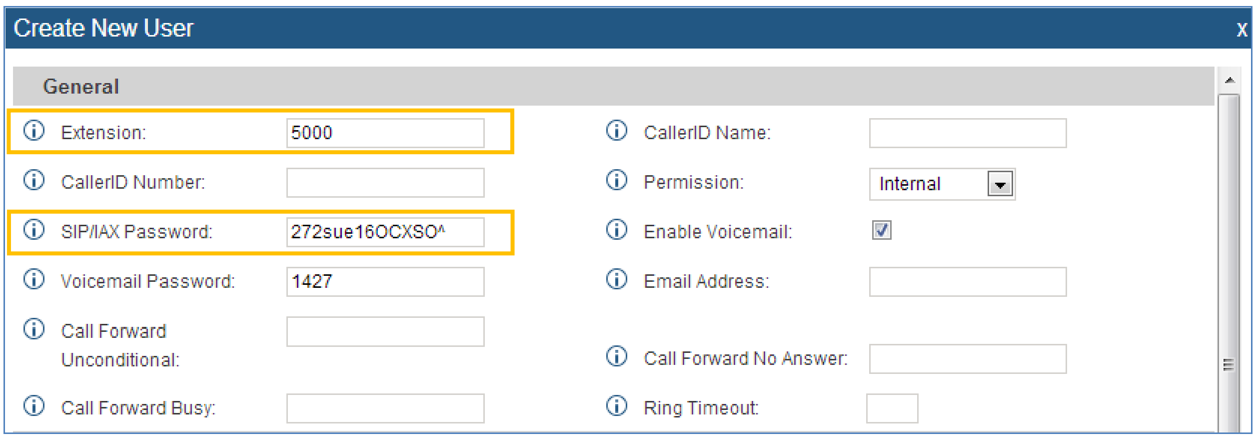

Figure 1: Method 1 - Create Extension 5000 on the UCM6100

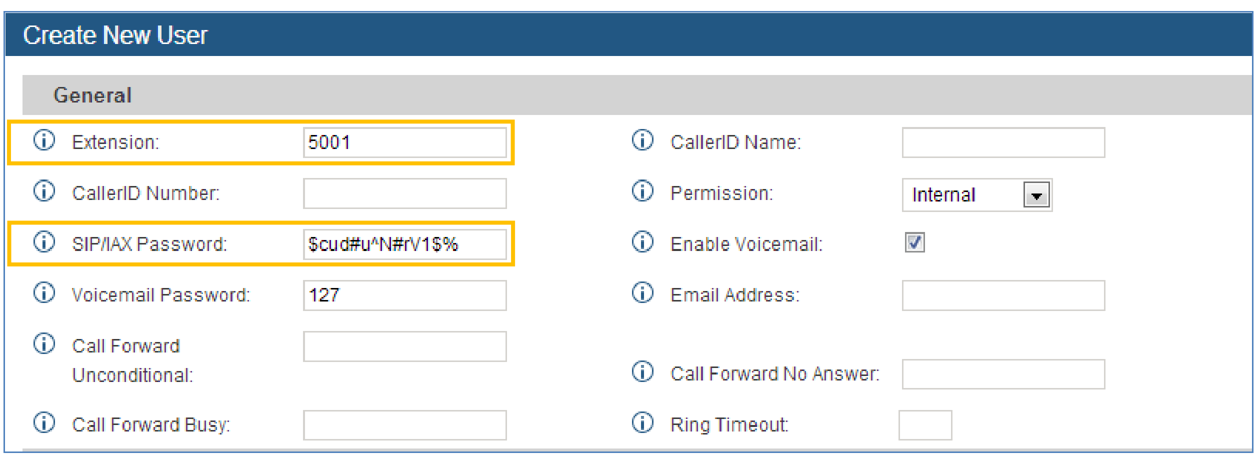

Figure 2: Method 1 - Create Extension 5001 on the UCM6100

Create IVR on UCM6100

On the UCM6100 web GUI, create an IVR extension under **PBX**->**Call Features**->**IVR**. This is to receive the calls forwarded from the HT503.

In IVR settings, if "Dial Other Extensions" is enabled, the calls forwarded to 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.

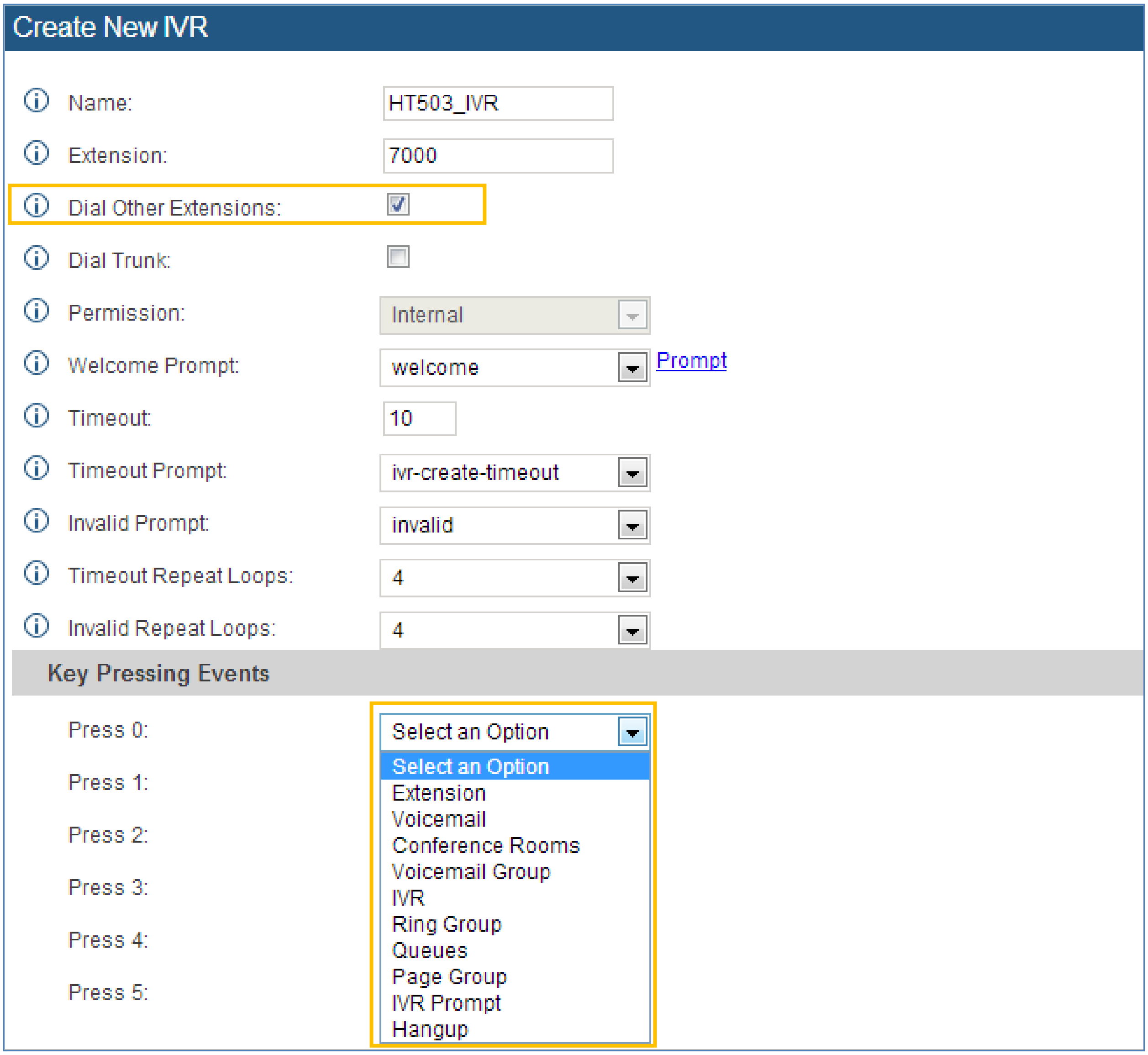

Figure 3: Method 1 - Create IVR 7000 on the UCM6100

Configure FXS Port on HT503

1. Connect an analog phone to the HT503 FXS port.
2. On the HT503 web GUI, go to FXS Port setting page, configure to register the FXS port to the

UCM6100 extension 5000. Please refer to the highlighted settings in the following figure.

In this example, the UCM6100 IP address is 192.168.40.207.

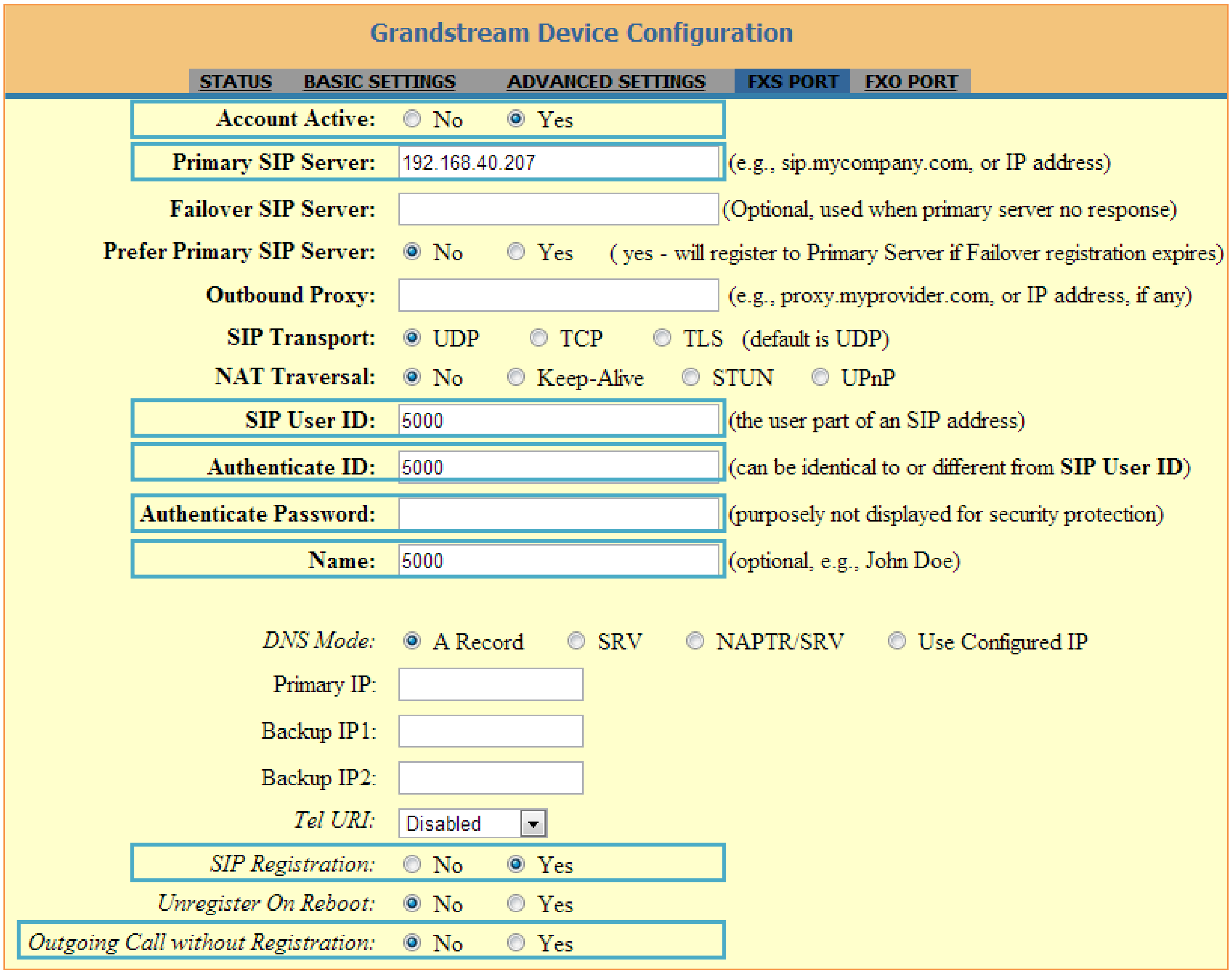

Figure 4: Method 1 - Configure FXS Port on the HT503

Configure FXO Port on HT503

1. Connect the PSTN line to the HT503 FXO port.
2. On the HT503 web GUI, go to FXO Port setting page, configure to register the FXO port to the UCM6100 extension 5001. Please refer to the highlighted settings and other necessary settings in the following figures.

   In this example, the UCM6100 IP address is 192.168.40.207.

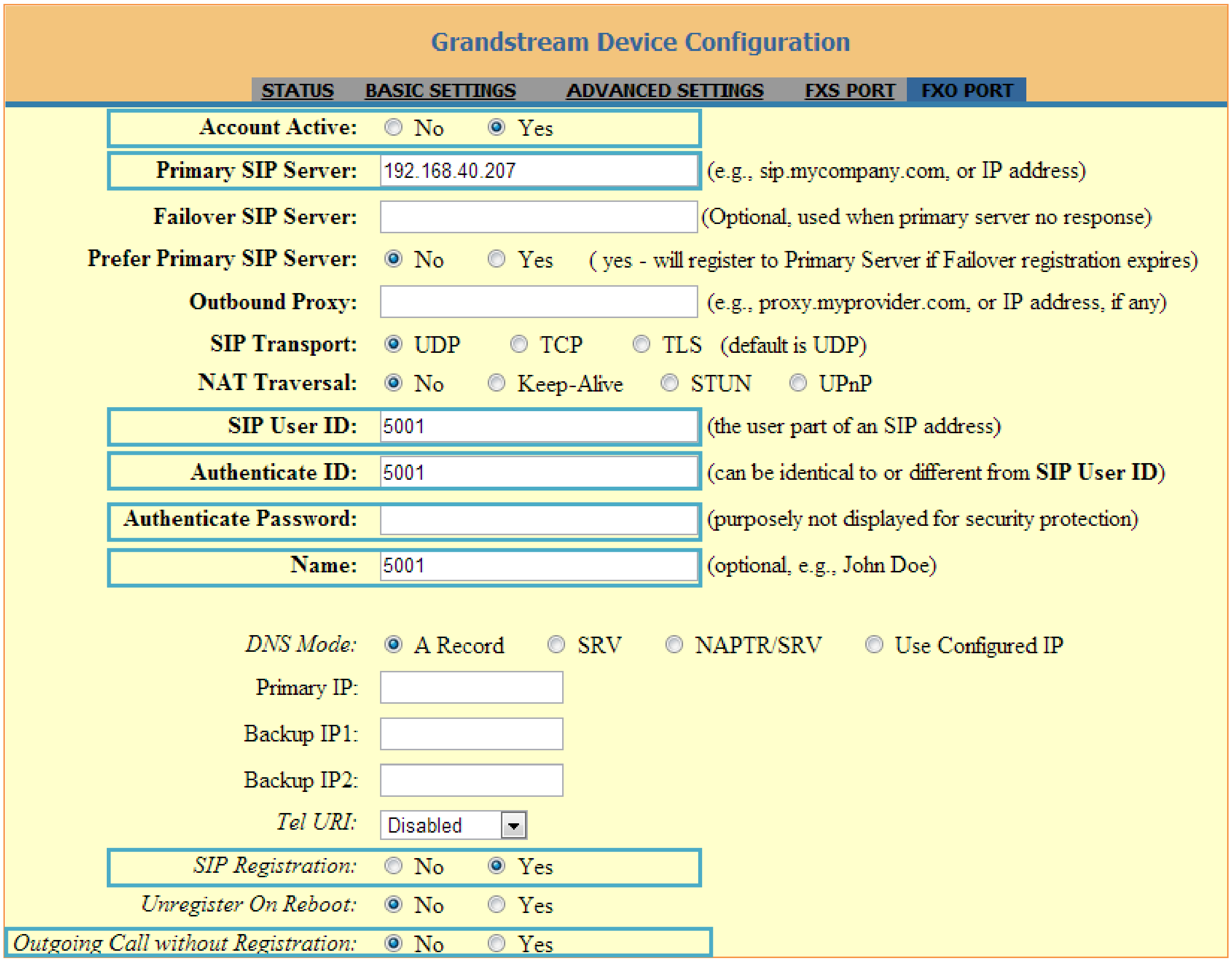

Figure 5: Method 1 - Configure FXO Port on the HT503: Registration

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 HT503 FXO port DTMF settings as below as an initial setup.

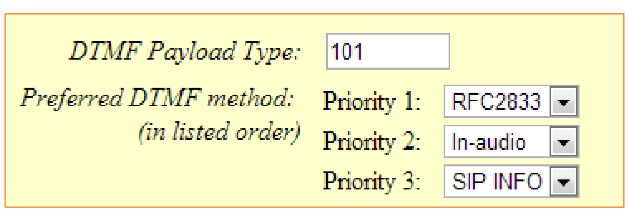

Figure 6: Method 1 - Configure FXO Port on the HT503: DTMF Settings

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

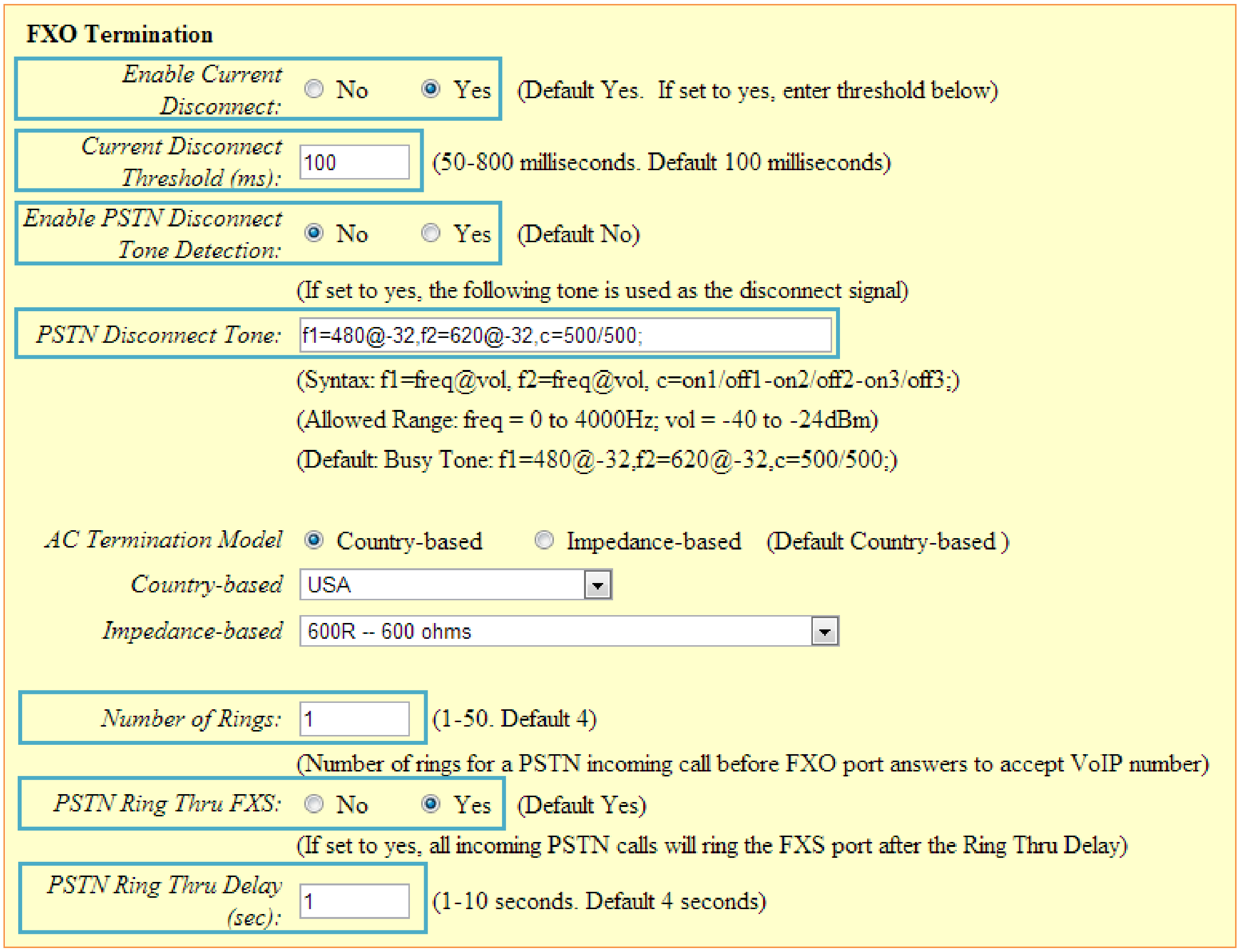

Figure 7: Method 1 - Configure FXO Port on the HT503: FXO Termination

- 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.
  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.

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

- Set "PSTN Ring Thru FXS" to "No" if you prefer not to ring the FXS port 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 to 2.

- Set the "Stage Method (1/2)" to 2 for 2 stage dialing.

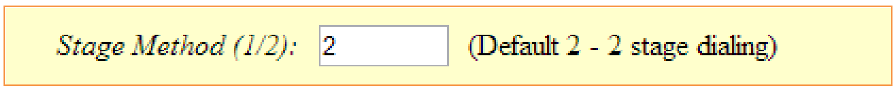

Figure 8: Method 1 - Configure FXO Port on the HT503: Channel Dialing

Configure Unconditional Call Forward on HT503

On the HT503 web GUI, go to Basic setting page, configure "Unconditional Call Forward to VOIP" to the IVR extension on the UCM6100. In this example, the UCM6100 IP address is 192.168.40.207.

Figure 9: Method 1 - HT503 Basic Settings

How to Dial

Once the HT503 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 the HT503's FXO extension number (5001 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 HT503). 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.

METHOD 2: Connect UCM6100 to HT503 Using Peer SIP Trunk

Create Extension on UCM6100

On the UCM6100 web GUI, create one extension under **PBX**->**Basic/Call Routes**->**Extensions**. This extension is used for HT503 FXO registration.

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

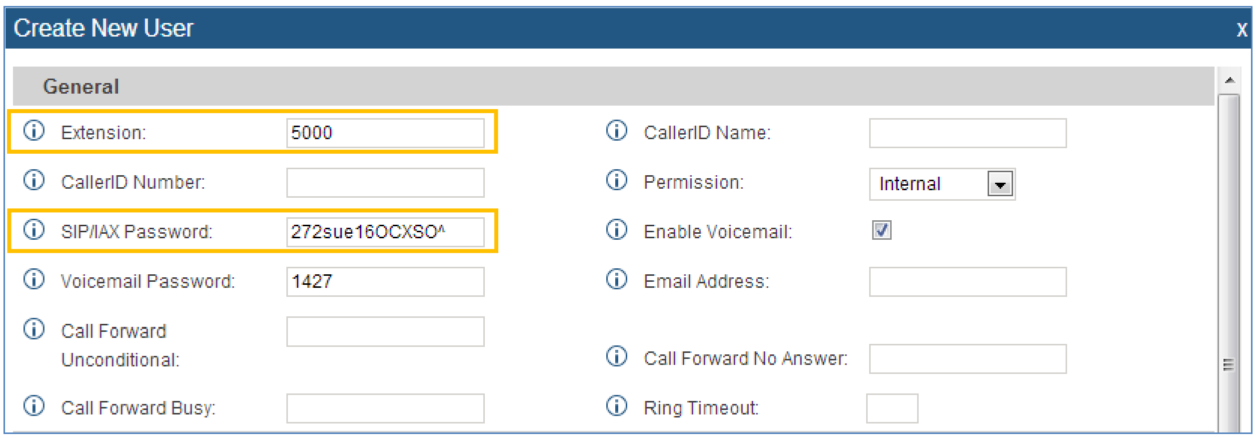

Figure 10: Method 2 - Create Extension 5000 on the UCM6100

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.

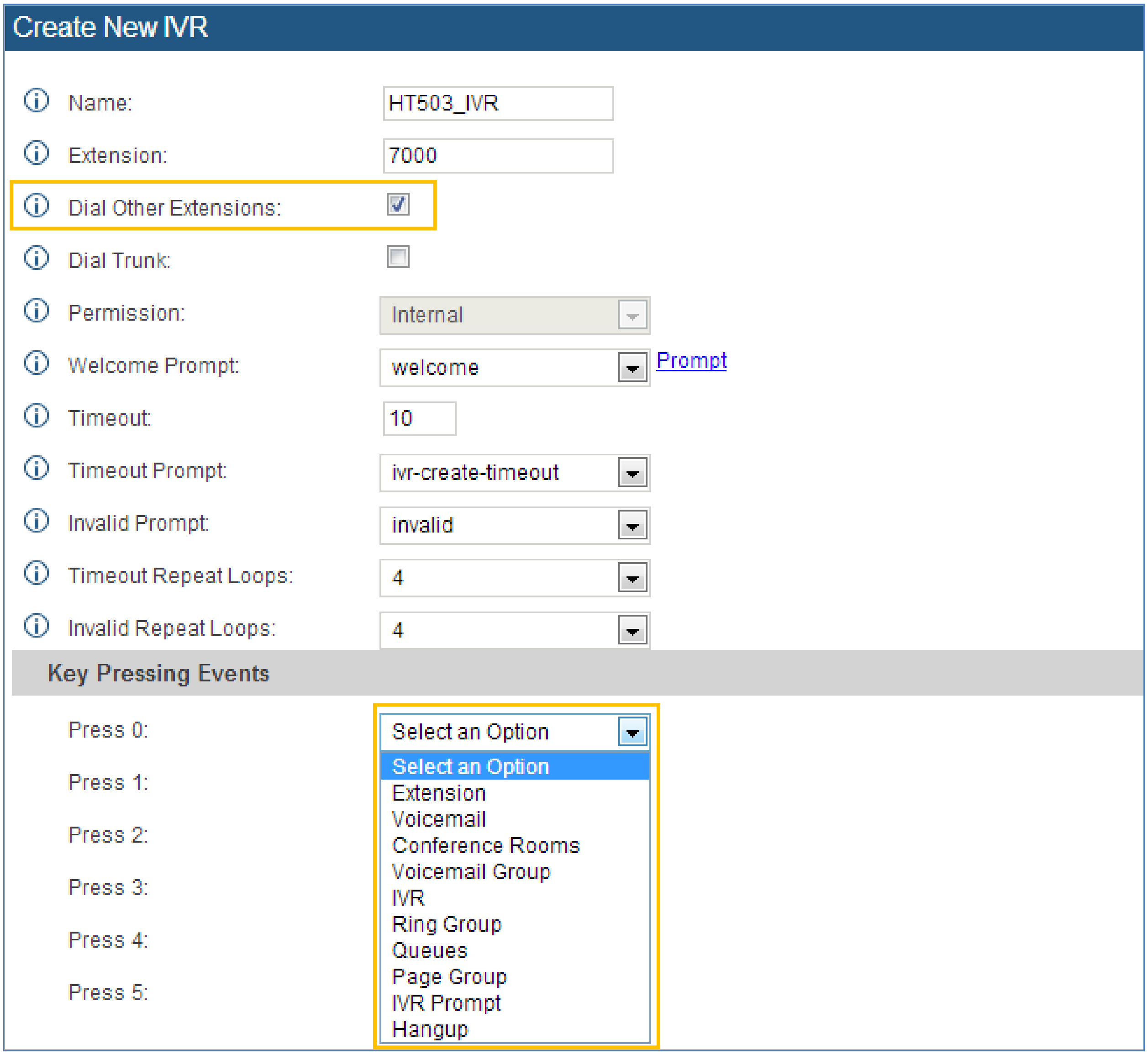

Figure 11: Method 2 - Create IVR 7000 on the UCM6100

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 HT503 IP address is 192.168.40.137.

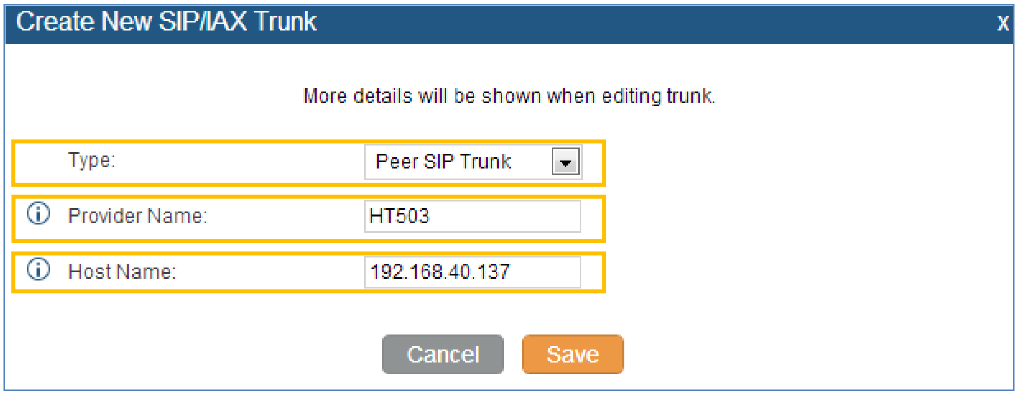

Figure 12: Method 2 - Create Peer SIP Trunk on the UCM6100

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.

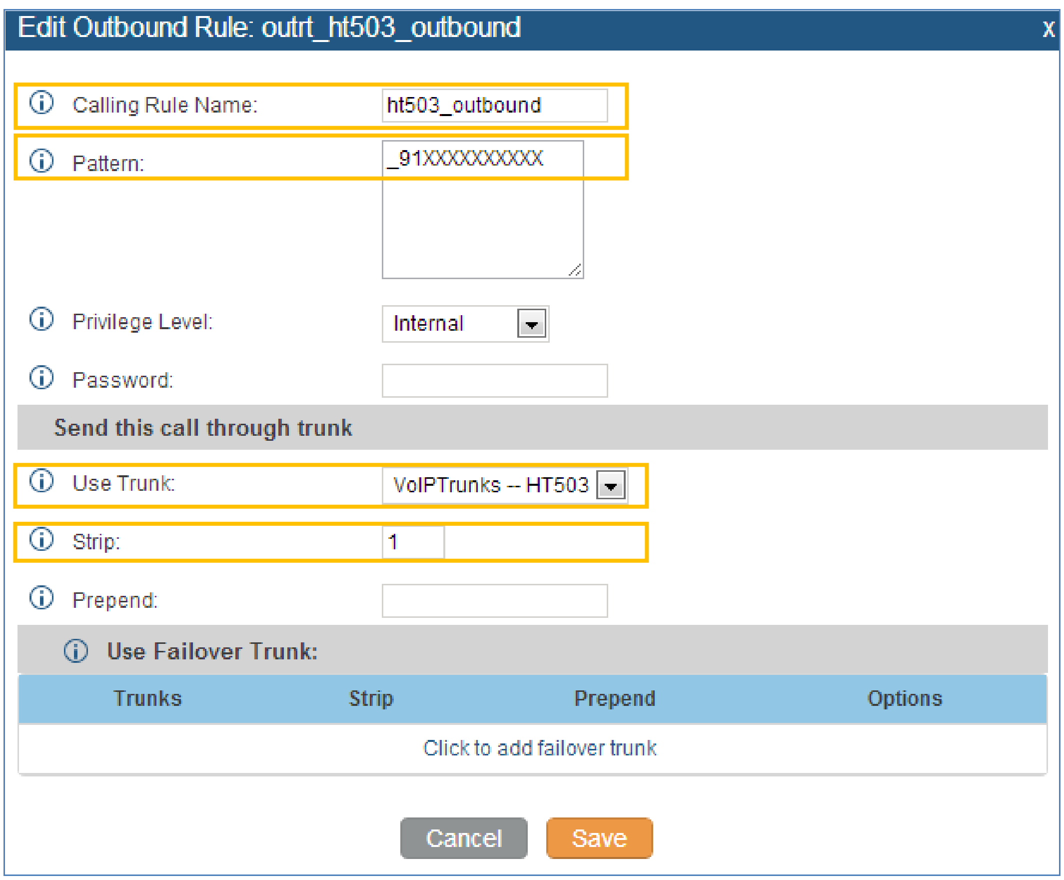

Figure 13: Method 2 - Configure Outbound Rule on the UCM6100

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 HT503 call forward setting.

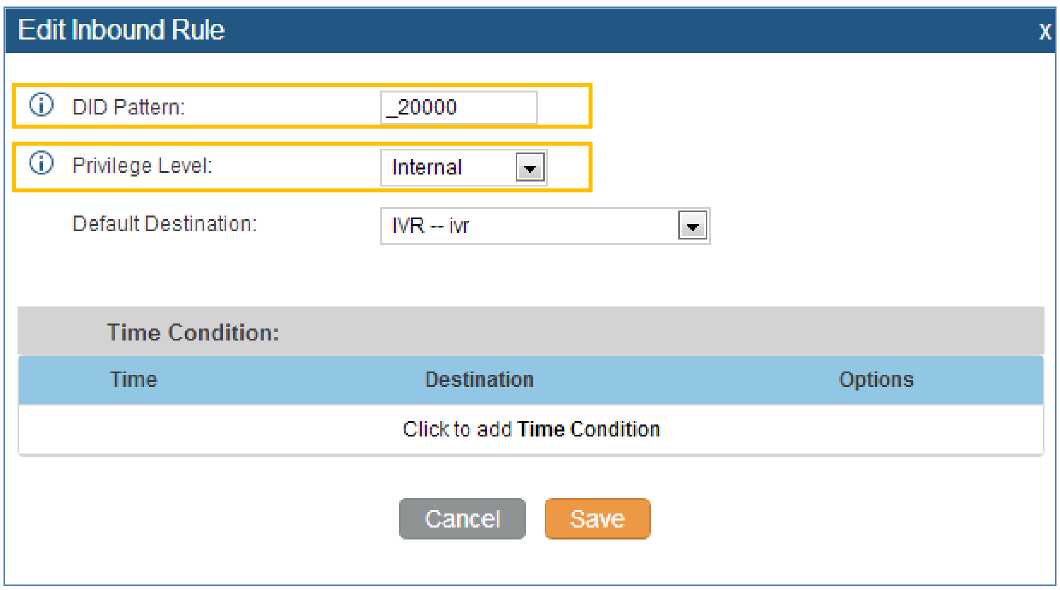

Figure 14: Method 2 - Configure Inbound Rule on the UCM6100

The default destination is configured to IVR.

Configure FXO Port on HT503

1. Connect the PSTN line to the HT503 FXO port.

2. On the HT503 web GUI, go to FXO Port setting page, configure to register the FXO port to the UCM6100 extension 5000. Please refer to the highlighted settings and other necessary settings in the following figures.

   In this example, the UCM6100 IP address is 192.168.40.207.

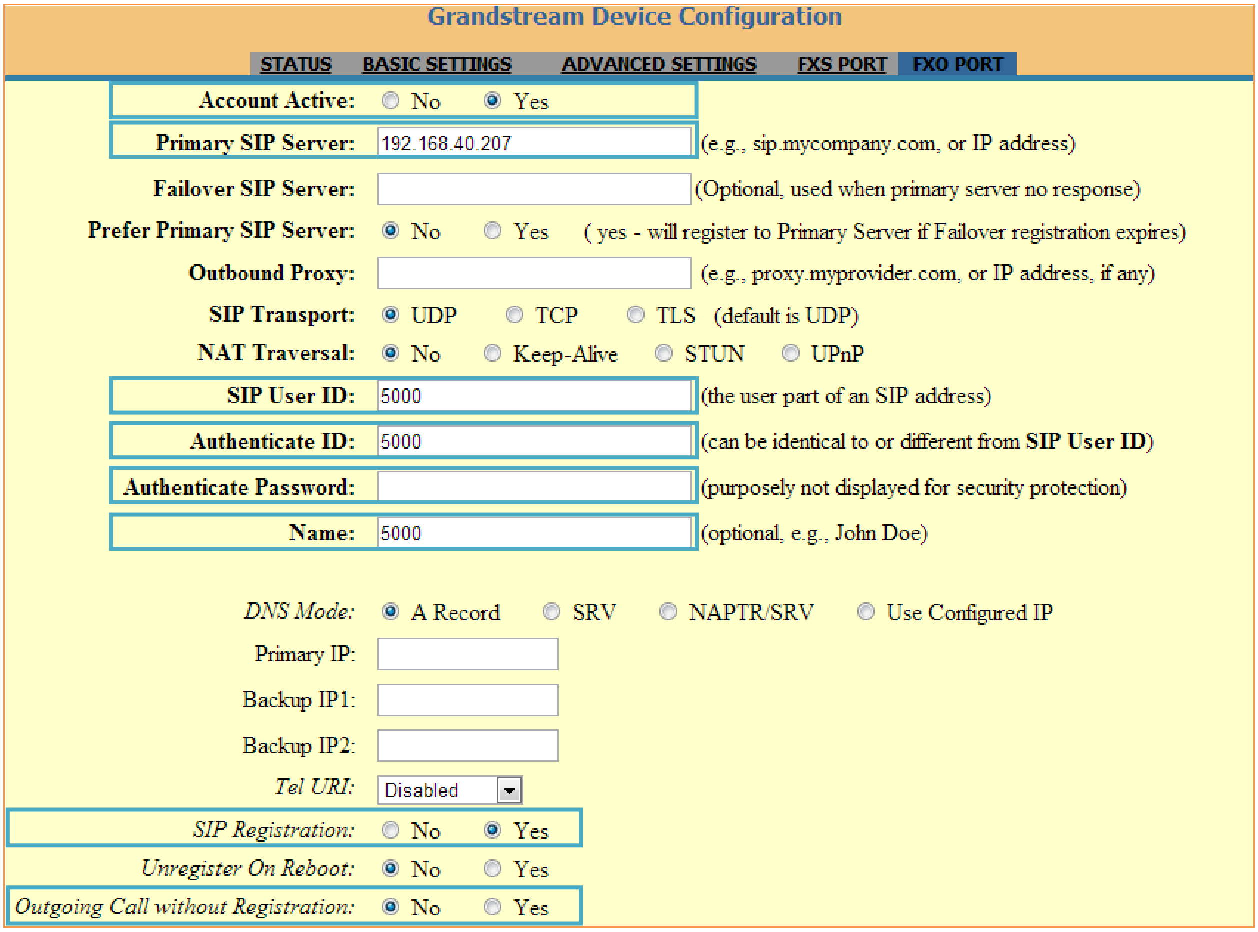

Figure 15: Method 2 - Configure FXO Port on the HT503: Registration

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 HT503 FXO port DTMF settings as below for a initial setup.

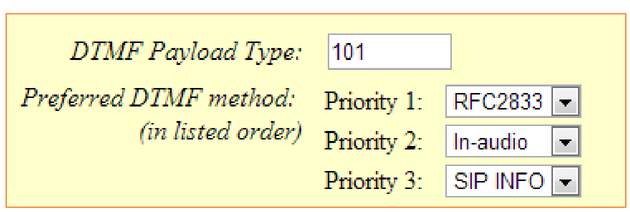

Figure 16: Method 2 - Configure FXO Port on the HT503: DTMF Settings

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

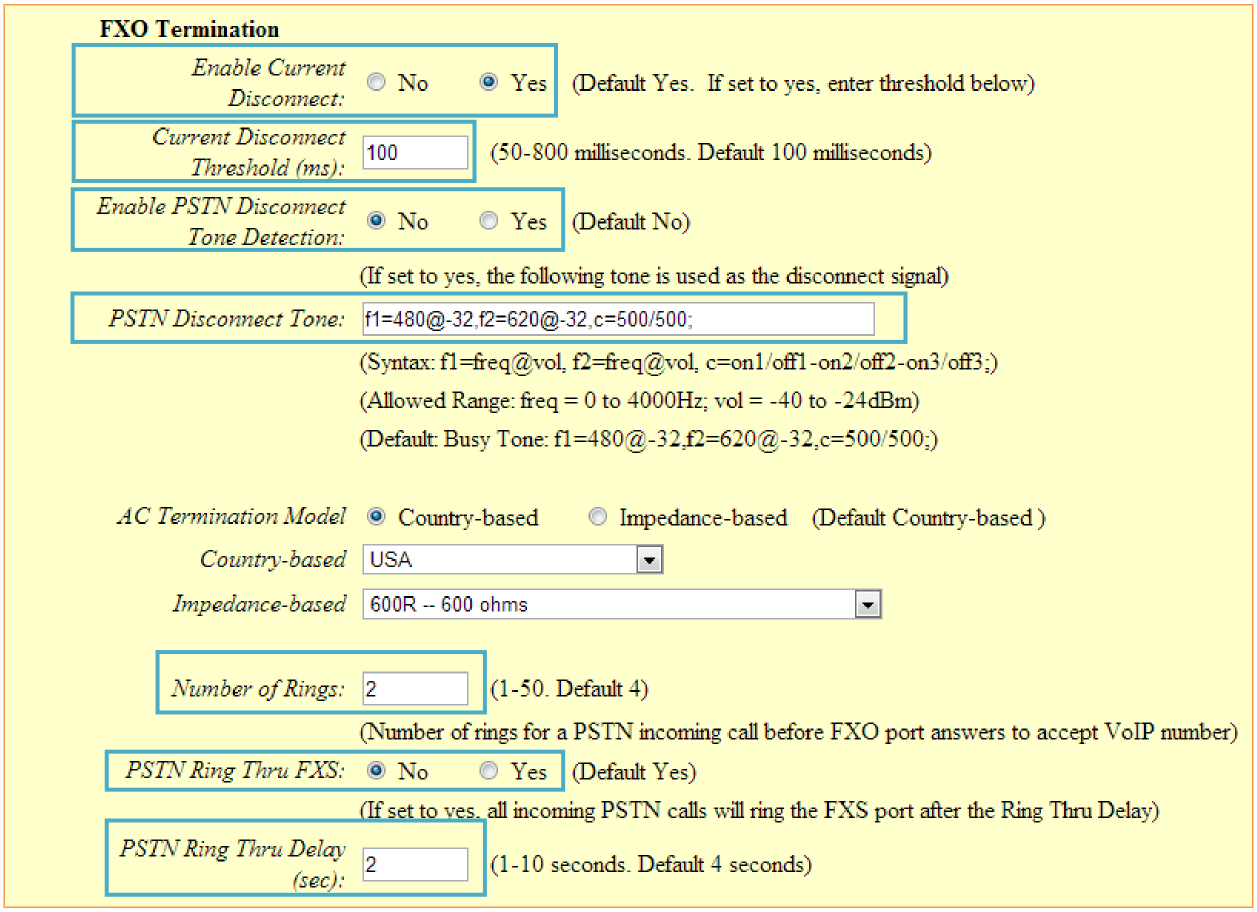

Figure 17: Method 2 - Configure FXO Port on the HT503: FXO Termination

- 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.
  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.

- 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.

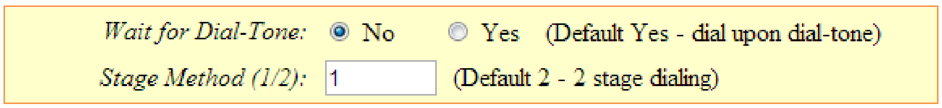

Figure 18: Method 2 - Configure FXO Port on the HT503: Channel Dialing

Exchange SIP Port Settings for FXS and FXO on HT503

- On the HT503 web GUI, go to FXO setting page, configure the "Local SIP Port" to be 5060. (The default setting is 5062.)

- On the HT503 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 HT503

On the HT503 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 UCM6100 inbound route dial pattern. In this example, the UCM6100 IP address is 192.168.40.207.

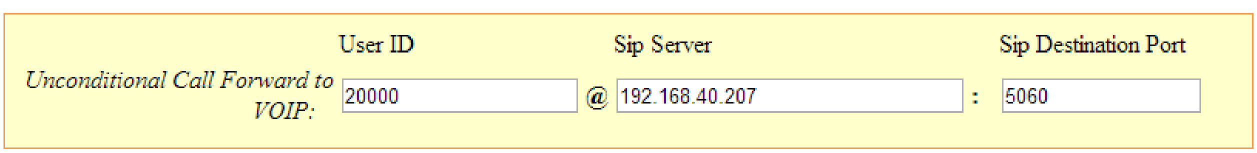

Figure 19: Method 2 - HT503 Basic Settings

How to Dial

Once the HT503 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 HT503). 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.