Grandstream Networks, Inc.

Connecting UCM6XXX with FreePBX®

Table of Contents

Table of Figures

Table of Tables

OVERVIEW

This document describes basic configuration to interconnect the UCM6XXX IP-PBX series with FreePBX® via SIP register trunk or SIP peer trunk. Once properly configured, the extensions on both PBXs can securely make calls to each other. Users need to have separate extension ranges on each side to avoid calls failure.

For this guide, we are using FreePBX® ver13 and UCM6202, also we are using extension range 5XXX on the FreePBX® side and extension range 3XXX are on the UCM side.

Caution

This document is subject to change without notice. The latest electronic version of this document is available for download here: http://www.grandstream.com/support

⚠ **Warning:**

- When the UCM6XXX series is interconnected with other PBX, it is NOT recommended to turn on "Allow Guest Calls" under web GUI->**PBX**->**SIP Settings**->**General**. Turning on this option will allow unauthenticated calls coming through the UCM6XXX series. Please be aware of the security concerns when using this option.
- When using the IVR in UCM6XXX 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 UCM6XXX'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 call allowed via "Dial Trunk".
- There are vast deployment possibilities when peering and interconnecting PBX systems. Due to highly customizable nature of both the UCM6XXX series and FreePBX, please use this tutorial as a basic sample to get UCM6XXX series work with the FreePBX. The actual implementation may be customized and different from this basic configuration.

CONNECTING UCM6XXX WITH FREEPBX®

Using SIP Trunk with Registration

Configure SIP Trunk on FreePBX®

First you need to go under FreePBX® web GUI and create the trunk which will be used to connect with the UCM, we need this first step since on FreePBX® you can either choose to send registration (regular ITSP case, or receive registration where in this case the FreePBX® will play the role of provider). In our case, we will choose to receive registration from the UCM where we will create a Register type SIP trunk.

1. On the FreePBX® web GUI, access to trunk setting page "**Connectivity -> Trunks**" to create and configure the SIP trunk as displayed on the following screenshot.

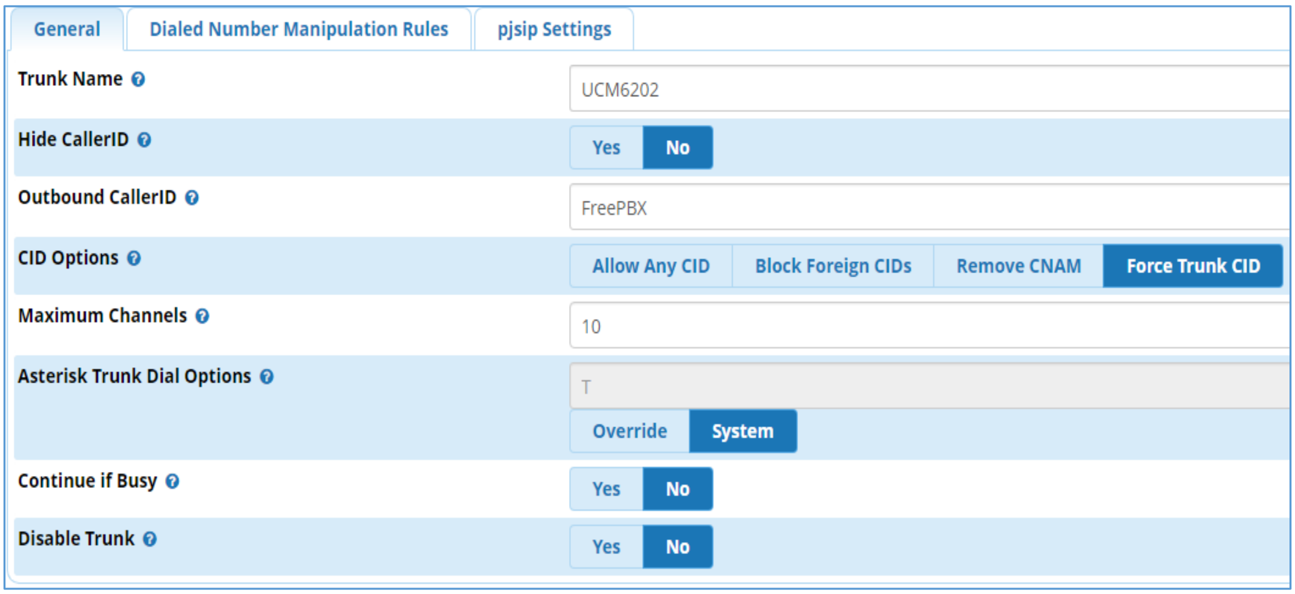

Figure 1: FreePBX® Trunk General Settings

2. After setting the trunk name and outbound caller ID, access PJSIP Settings tab and set the following parameters.

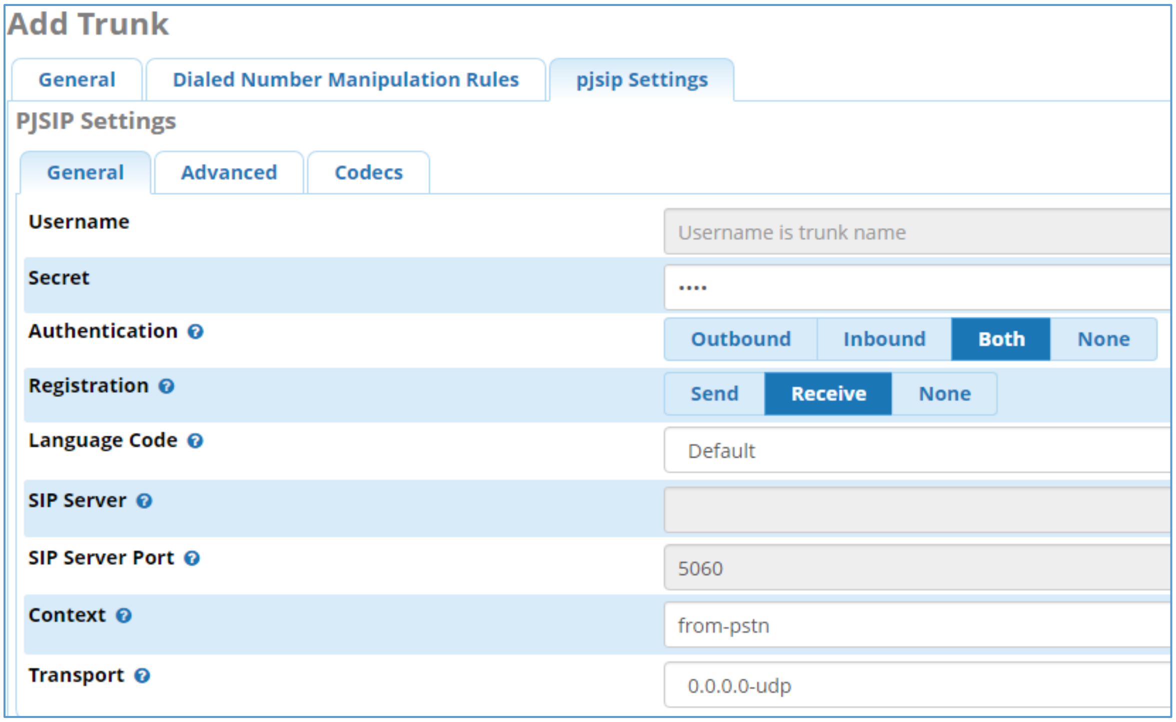

Figure 2: FreePBX® Trunk Config to Receive Registration

Following table summarizes the important options:

Table 1: FreePBX® Trunk PJSIP Settings

Option	Description
Username	This is the trunk's name and it will be used by UCM to send registration to FreePBX®.
Secret	The Trunk's account password
Authentication	Enable authentication for incoming and/or outgoing calls.
Registration	Choose **Receive** registration since the UCM will send register request to FreePBX®.
SIP Server	This is filed is used to set IP or domain name of the SIP server, when the trunk is configured to **SEND Registration**, in our example it's not needed since we receive the registration.
SIP Server Port	The port number to which the registration should be sent.
Context	Asterisk Context used to route calls to/from the configured peer.
Transport	Select transport protocol (UDP, TCP or TLS).

3. Submit and save the settings to apply the new configuration.

Configure SIP Trunk on UCM6XXX

1. On the UCM6XXX web GUI, access to **PBX**->**Basic/Call Routes**->**VoIP Trunks** to create a new SIP trunk using "Register SIP Trunk" type.

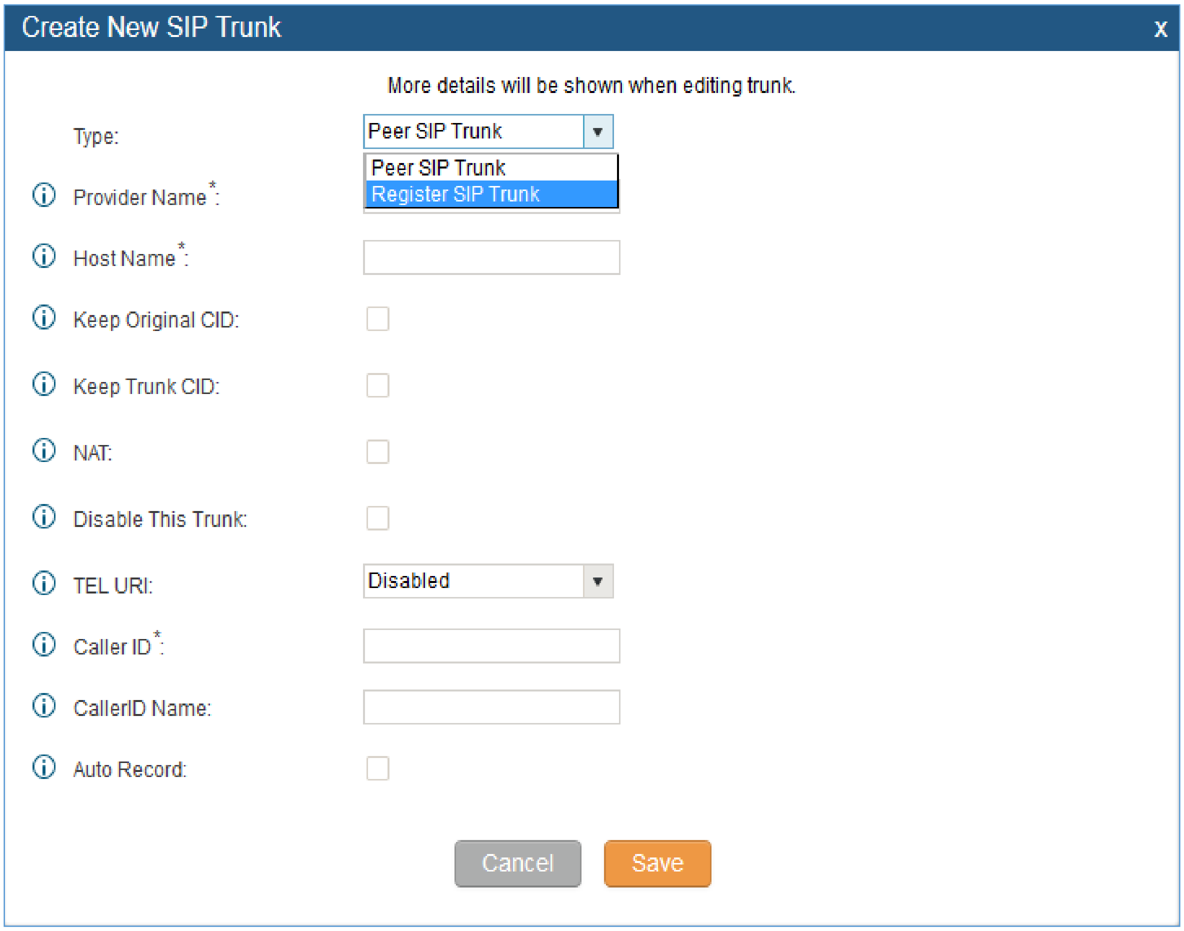

Figure 3: Create Register SIP Trunk on the UCM6XXX

2. Configure the below information for this trunk so that the UCM6XXX can register to the trunk we just created on FreePBX®.

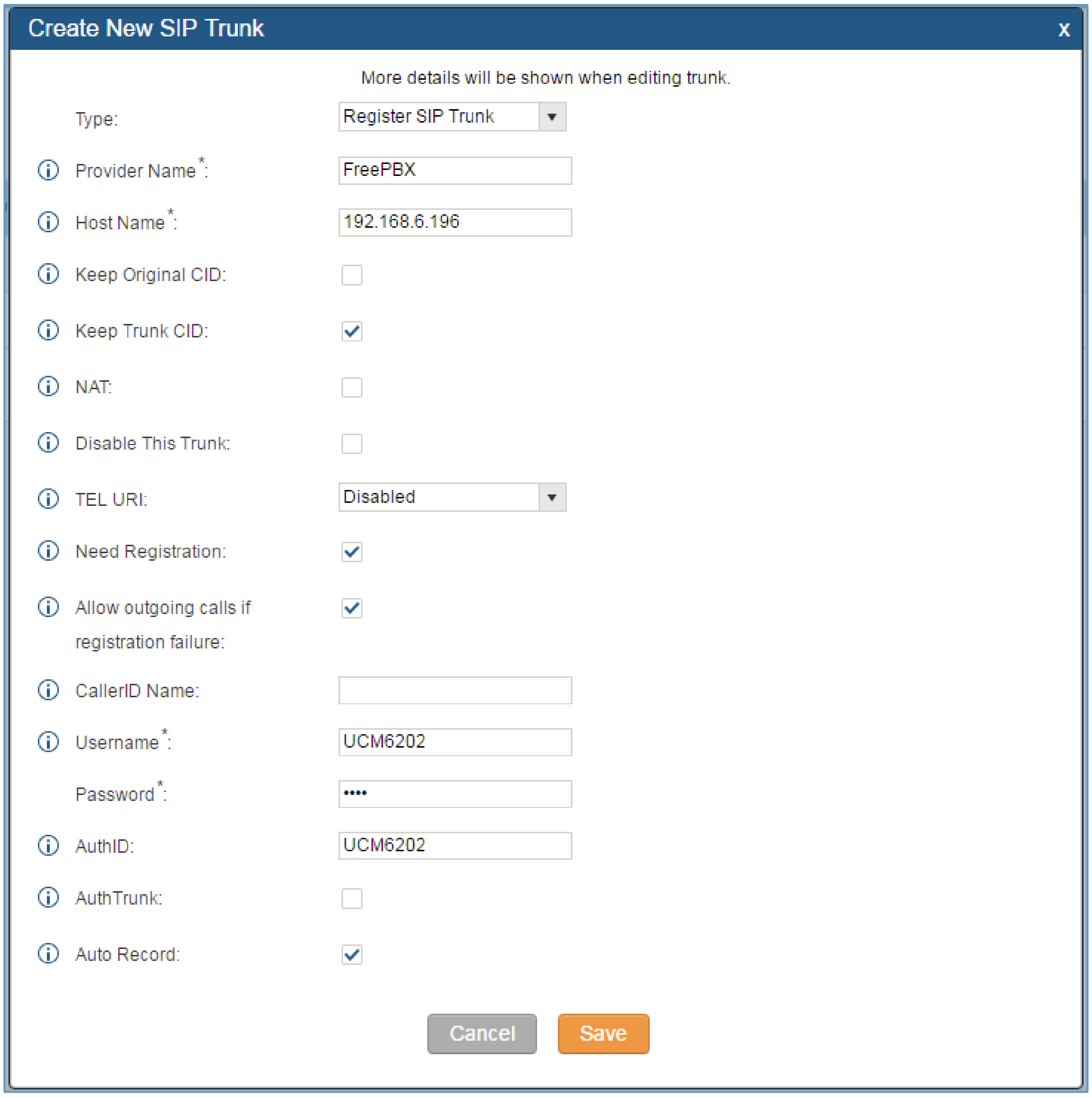

Figure 4: Configure Register SIP Trunk on the UCM6XXX

Following table summarizes the important options:

Table 2: Register trunk UCM settings

Option	Description
Provider Name	Description of the trunk
Hostname	Insert the IP or domain name of the FreePBX® machine.
Username	Username used for the registration. **Should be the same as the trunk's name on FreePBX® settings.**
Authentication	Password used for the registration. **Should be the same as the trunk's password configured on FreePBX®.**

Please refer to the following Guide for more details about SIP trunk parameters: SIP Trunk Guide.

3. After configuring the trunk on the UCM6XXX, save and apply the new settings.

Note: Users can verify the registration's status of the configured trunk under PBX Status page as displayed on the following screenshot:

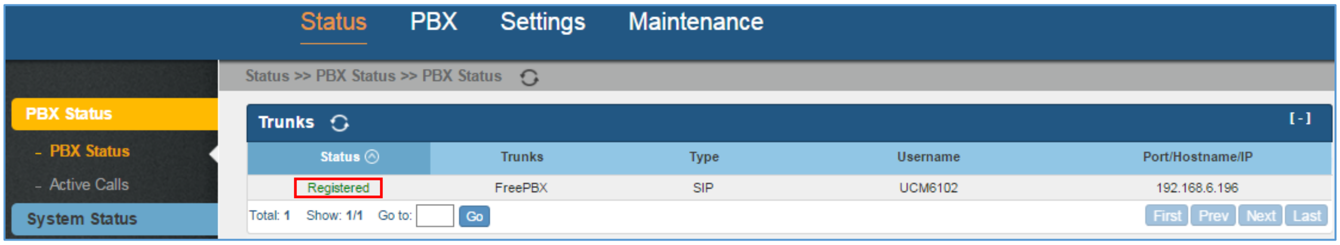

Figure 5: Registered Trunk Status

Using SIP Peer Trunks

Configure SIP Trunk on FreePBX®

1. Access to **Connectivity -> Trunks** Settings page and create new trunk, and set a name and a caller ID name which is optional (in our example, we used "UCM6202"),
2. Navigate to PJSIP Settings tab and set the following parameters as shown below:

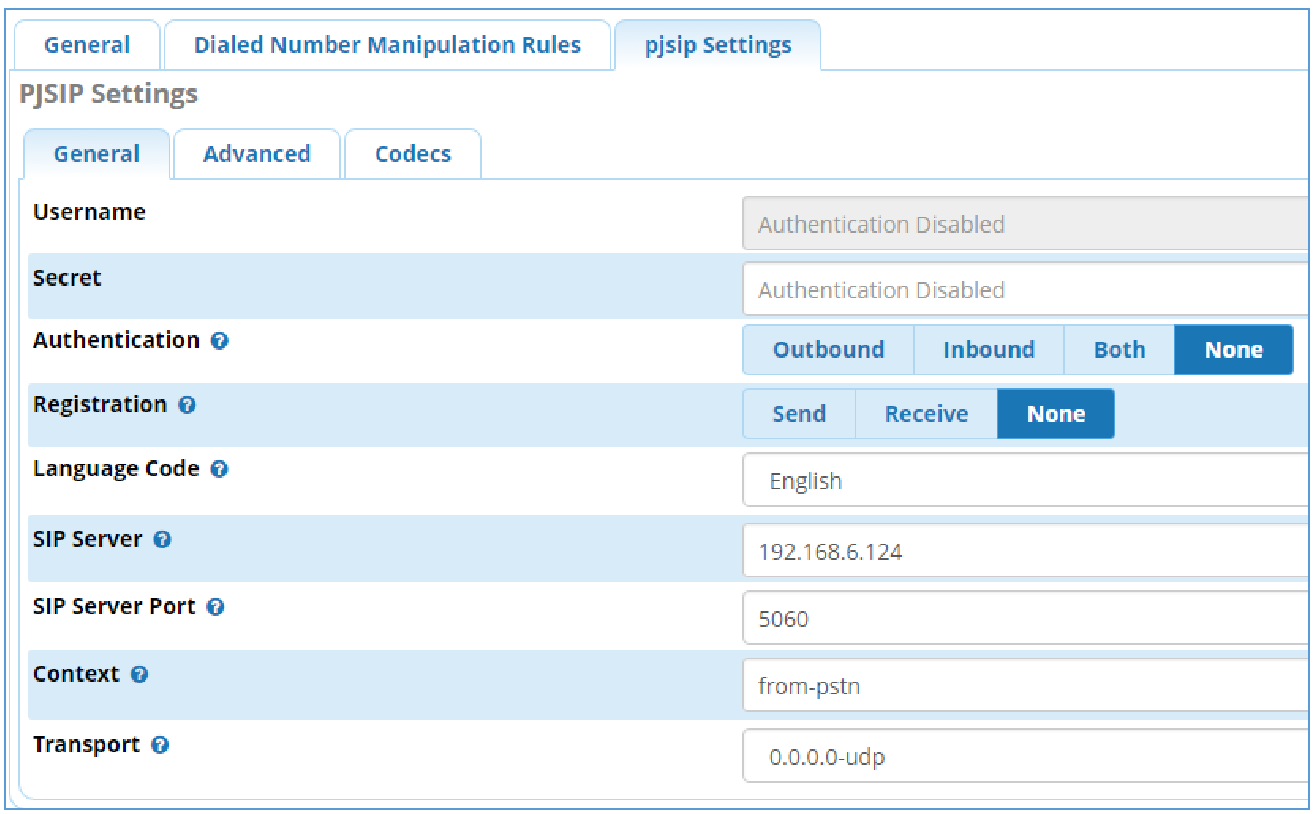

Figure 6: FreePBX® Peer Trunk

As you can see we have disabled the authentication and registration which are now set to None, since the two sides trust each other and we can only specify the IP addresses to have connectivity.

Table 3: FreePBX® Peer Trunk

Option	Description
SIP server	IP address of the UCM6XXX.
SIP Server port	Listening port of the UCM6XXX.

Configure SIP Trunk on UCM6XXX

1. Access to **“PBX -> Basic/Call Routes -> VoIP Trunks -> Create New Trunk**” and create a SIP Peer trunk, then set the name and the IP address of FreePBX® server as shown below:

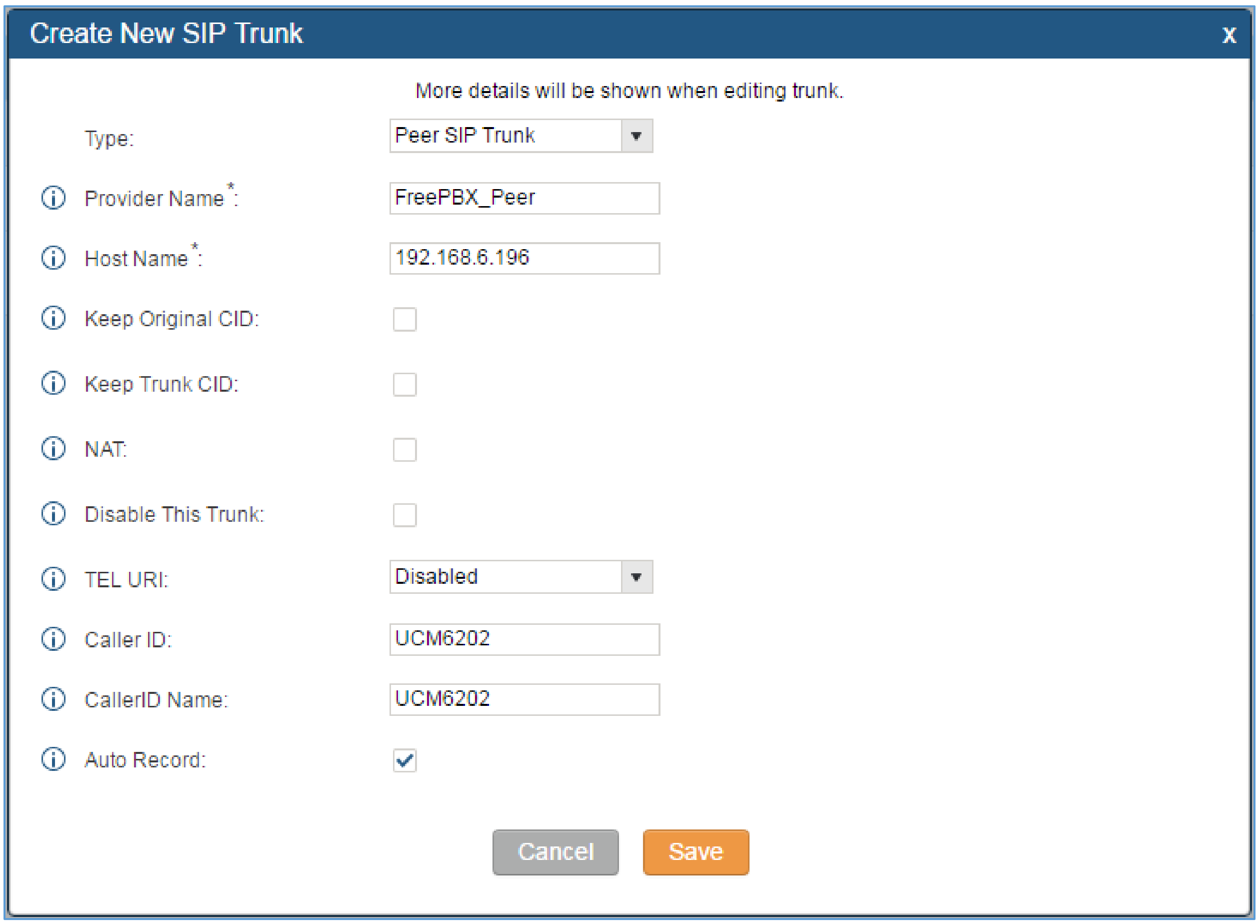

Figure 7: UCM Peer SIP Trunk

2. Navigate to advanced settings tab and enable the option of heartbeat to monitor the trunks status, once enabled the UCM will keep sending periodic keep alive SIP messages to FreePBX®.

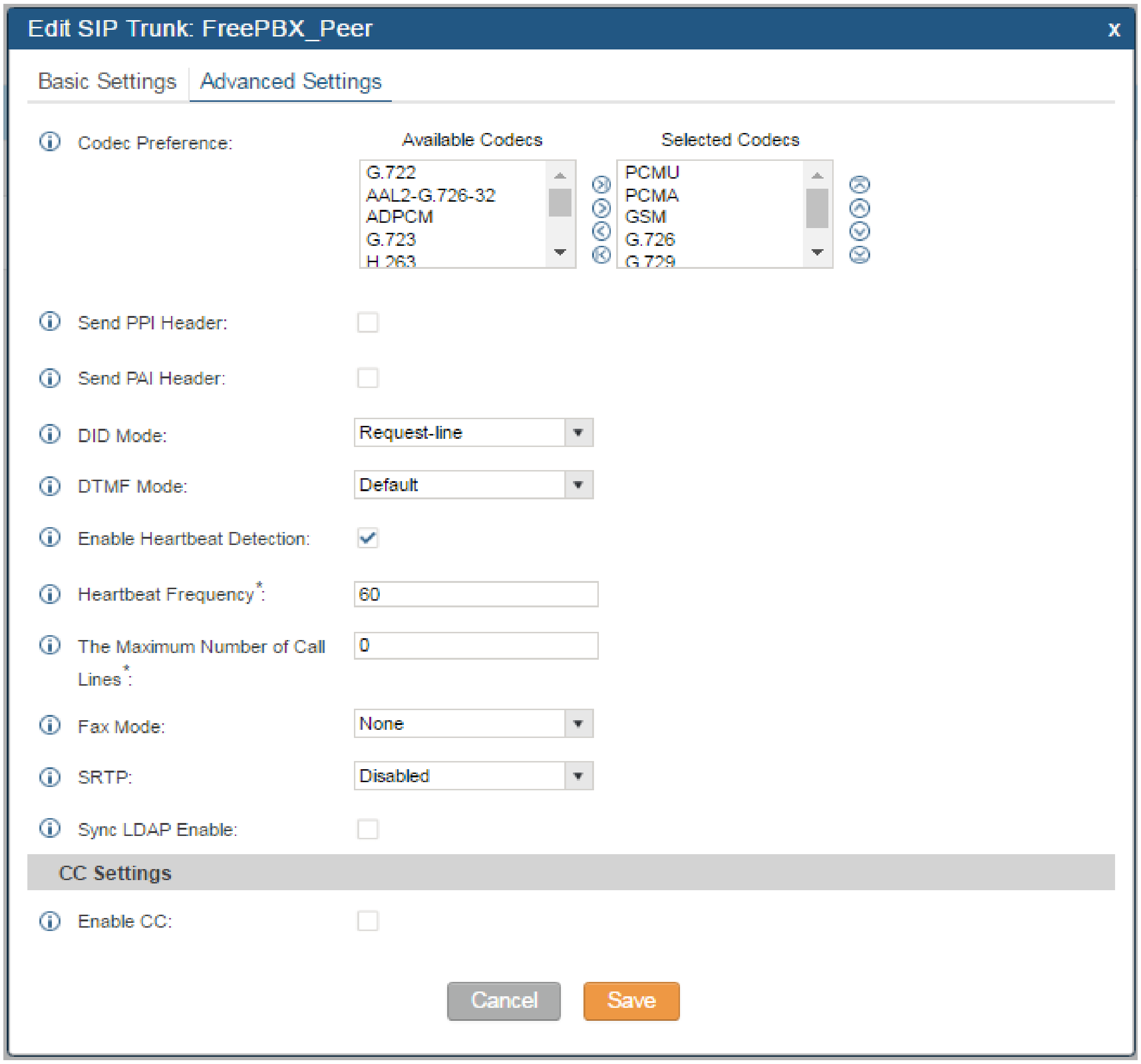

Figure 8: Enable Heartbeat Detection

Note: Users can verify the registration's status of the configured trunk under PBX Status page as displayed on the following screenshot:

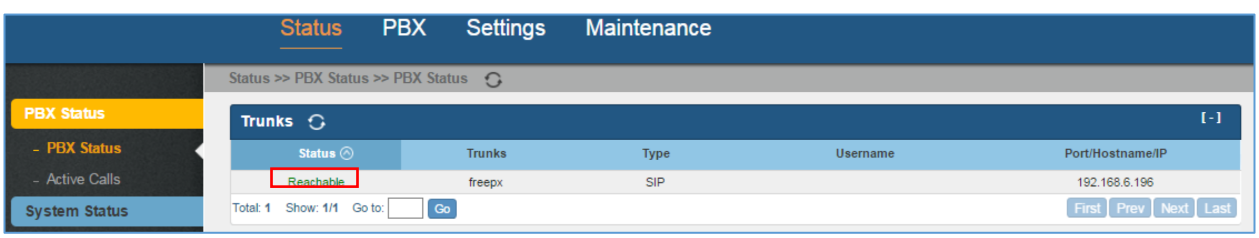

Figure 9: Peer Trunk UCM Status

CALL ROUTING

After creating and configuring SIP trunks on both UCM and FreePBX® (either Peer trunk or with registration), then you need next to configure the call routing for inbound and outbound calls on both sides.

Configure Call Routes on FreePBX®

Outbound Calls Routing

1. On the FreePBX® web GUI, access to outbound route setting page to create an outbound route for the SIP trunk. As displayed on following screenshot, we configured the dial pattern to 3XXX which matches the extension range on our UCM.

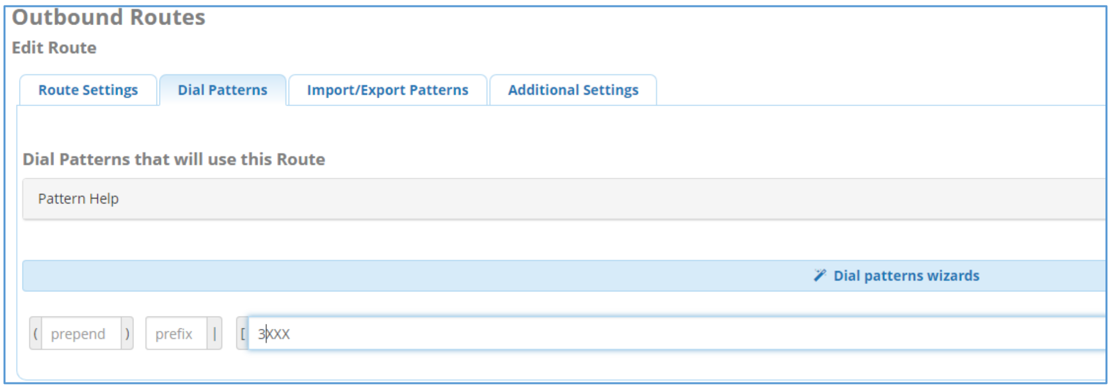

Figure 10: FreePBX® Outbound Routes Pattern

2. On the route settings page select the trunk through which the calls will be routed.

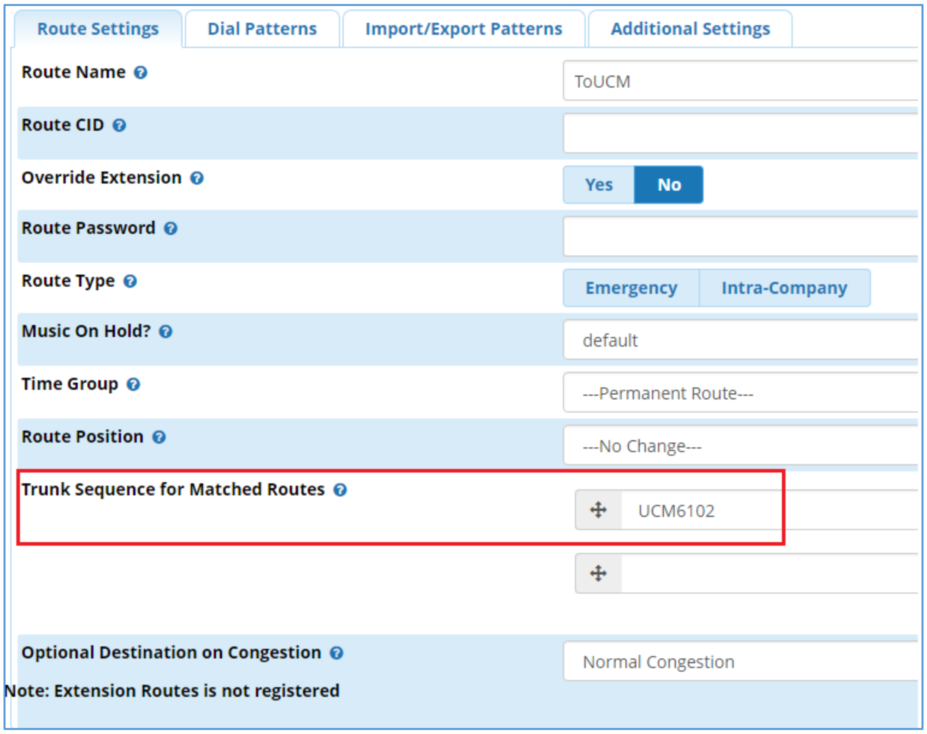

Figure 11: FreePBX® Outbound Routes Trunk Selection

Inbound Calls Routing

The FreePBX® uses DID for inbound route by default. Therefore, the extensions on the UCM6XXX can directly reach the extensions on the FreePBX. There is no additional configuration required for inbound route as a basic configuration sample.

Configure Call Routes on UCM6XXX

Outbound Calls Routing

On the UCM6XXX web GUI, access to **PBX**->**Basic/Call Routes**->**Outbound Routes** to create a new outbound rule. This will allow the registered extension on the UCM6XXX to reach registered extensions (5XXX range, in this example) on the FreePBX.

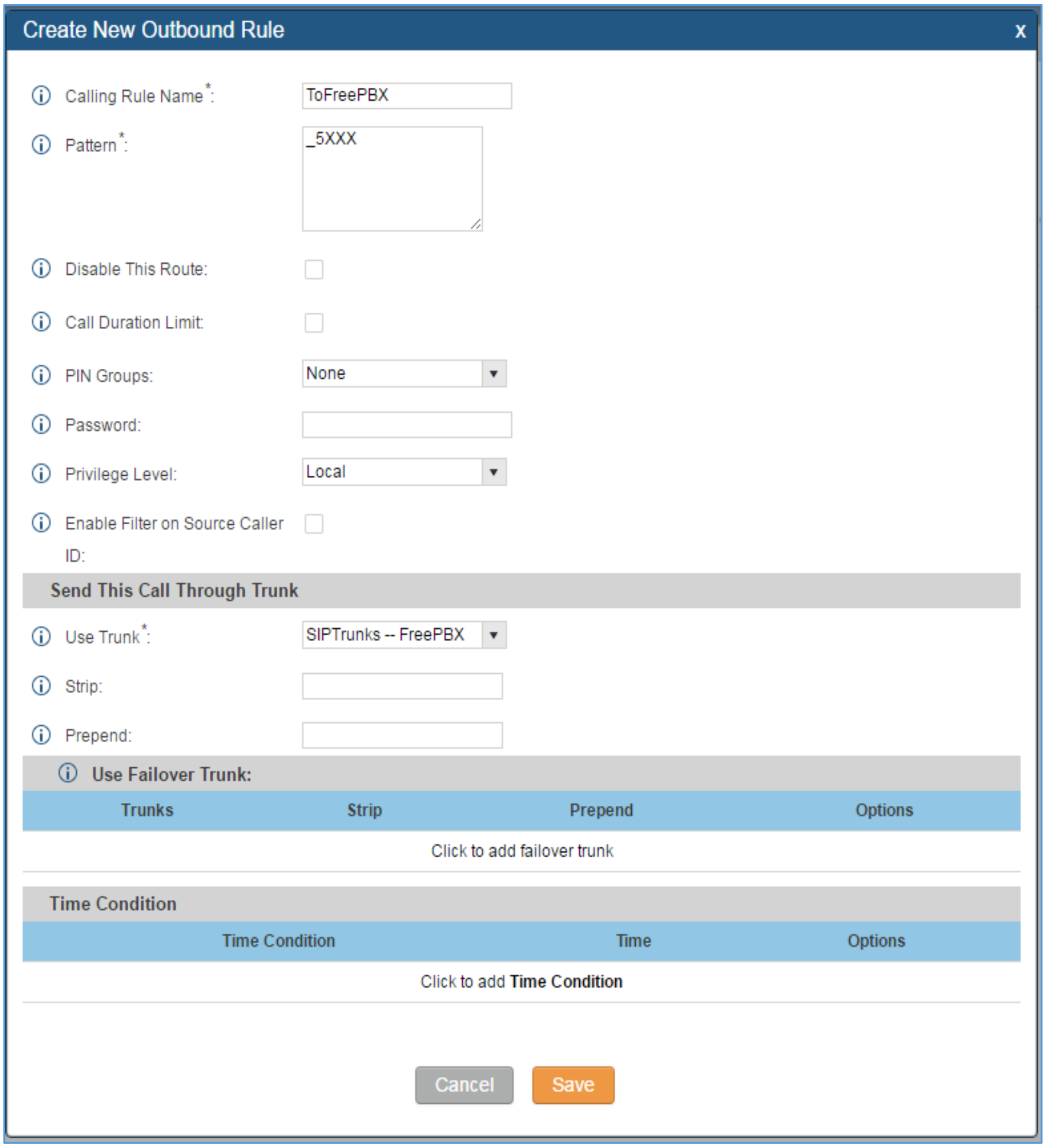

Figure 12: Configure Outbound Route on the UCM6XXX

Note: You need to make sure to give extensions permission level equal or higher than the privilege level configured on the outbound rule.

For more detailed explanation on outbound and inbound rules on the UCM, please refer to this following [How to Guide]

Inbound Calls Routing

On the UCM6XXX web GUI, access to **PBX->Basic/Call Routes->Inbound Routes** to create a new inbound rule.

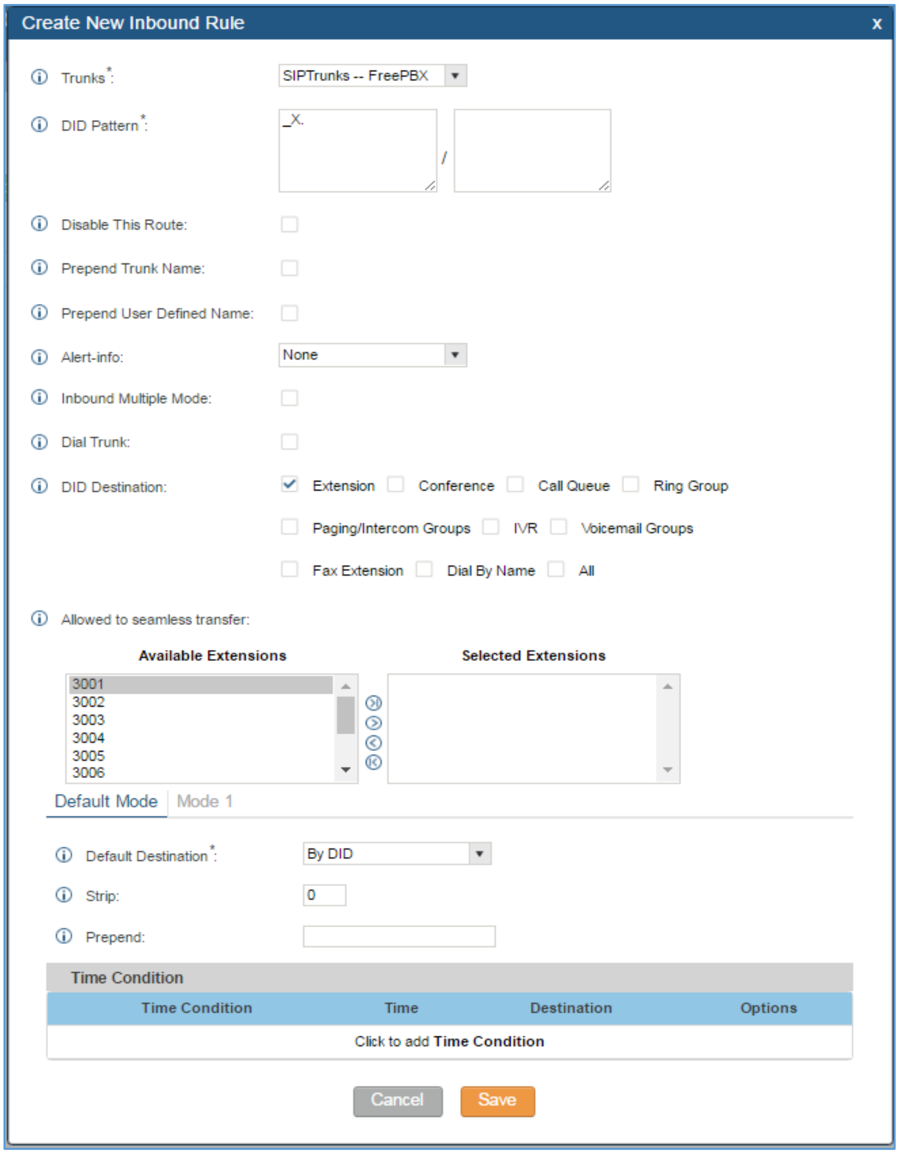

Figure 13: Configure Inbound Route on UCM6XXX

Now the FreePBX® and UCM6XXX are interconnected and configured to make calls to extensions both ways. You can further configure the inbound rule, outbound rule, IVR and the corresponding permission/privilege levels to control the calls through the UCM6XXX.

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