Grandstream Networks, Inc.

Interoperability Tutorial:

Configuring UCM6100 Series with FreePBX ®

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 FreePBX via SIP register trunk. Once properly configured, the extensions on both PBX can securely make calls to each other.

⚠ **Warning:**

- When the UCM6100 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 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 call allowed via "Dial Trunk".
- There are vast deployment possibilities when peering and interconnecting PBX systems. Due to highly customizable nature of both the UCM6100 series and FreePBX, please use this tutorial as a basic sample to get UCM6100 series work with the FreePBX. The actual implementation may be customized and different from this basic configuration.

CONFIGURING SIP TRUNK

Create Extension on UCM6100

On the UCM6100 web GUI, create an extension under **PBX**->**Basic/Call Routes**->**Extensions**. This extension is used for FreePBX to register SIP trunk to the UCM6100.

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

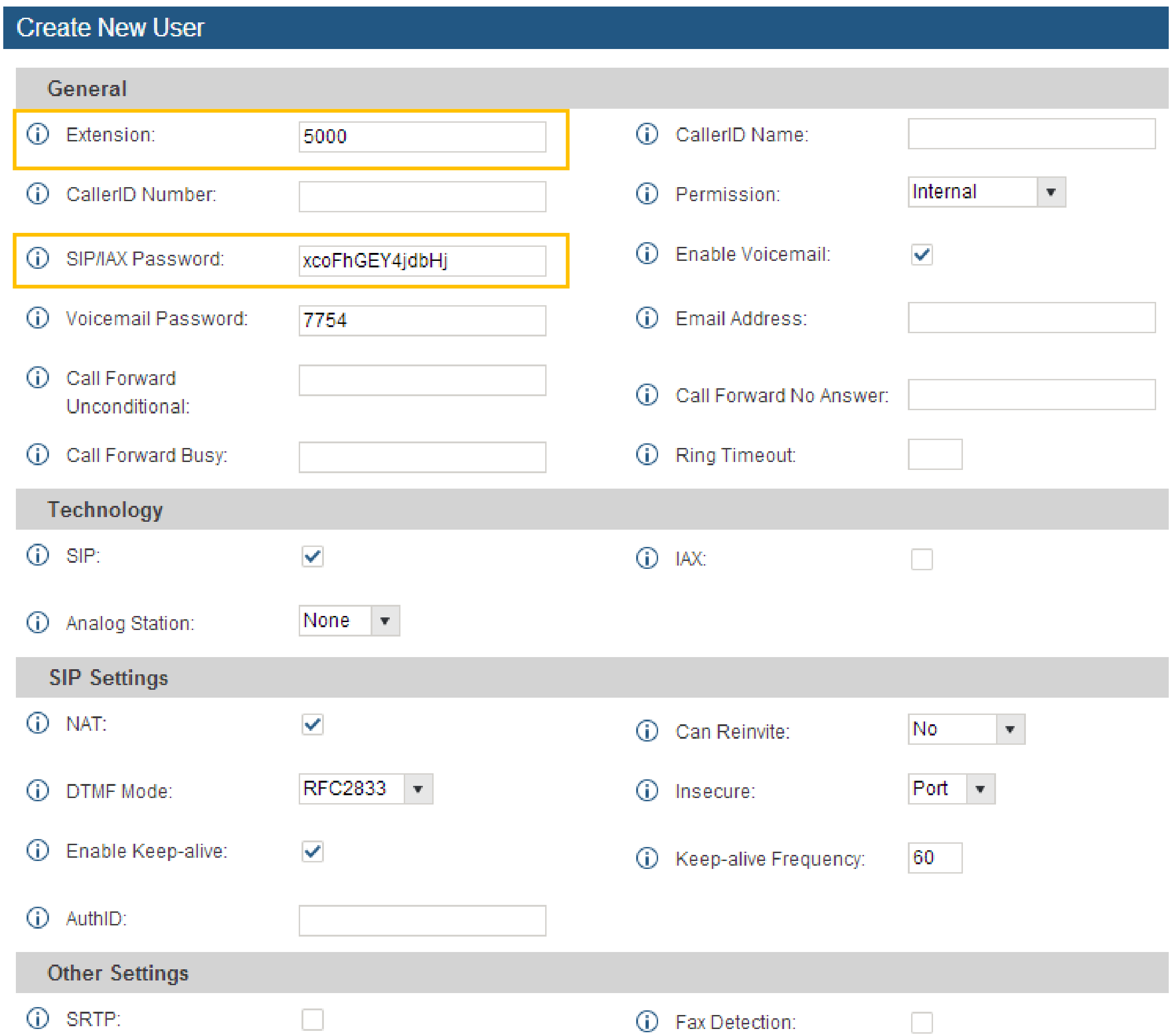

Figure 1: Create an Extension on UCM6100

Configure SIP Trunk on FreePBX

1. On the FreePBX web GUI, go to trunk setting page to create a SIP trunk. Then configure the following in **Outgoing Settings** section of this trunk.

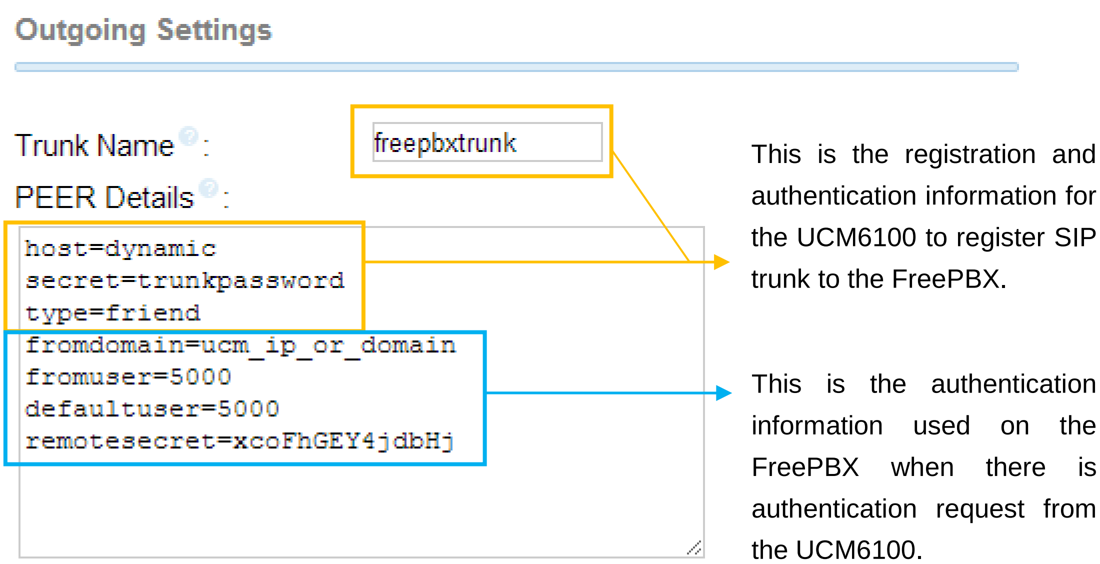

Figure 2: FreePBX Trunk - Outgoing Settings

⚠ **Note:**

- ***secret=trunkpassword***
  Please use a secure password to replace "trunkpassword" as the actual password for the FreePBX to authenticate UCM6100.
- ***fromdomain=ucm_ip_or_domain***
  Please replace ucm_ip_or_domain with the actual IP address of host domain of the UCM6100.
- ***remotesecret=xcoFhGEY4jdbHj***
  This is the password of the extension created on UCM6100 in section *[Create Extension on UCM6100]*.

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2. On the same trunk setting page, configure the registration string so that the FreePBX can register the SIP trunk to the extension created on UCM6100 in section *[Create Extension on UCM6100]*.

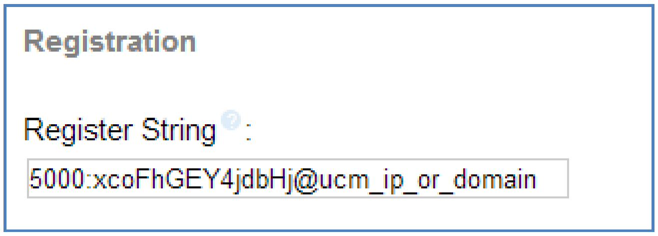

Figure 3: FreePBX Trunk - Registration

The registration string has the following format:
extension:password@ip_or_domain

Please make sure the registration information here matches the extension information (extension, password, IP or domain address) on the UCM6100.

Configure SIP Trunk on UCM6100

1. On the UCM6100 web GUI, go to **PBX**->**Basic/Call Routes**->**VoIP Trunks** to create a new SIP trunk using "Register SIP Trunk" type.

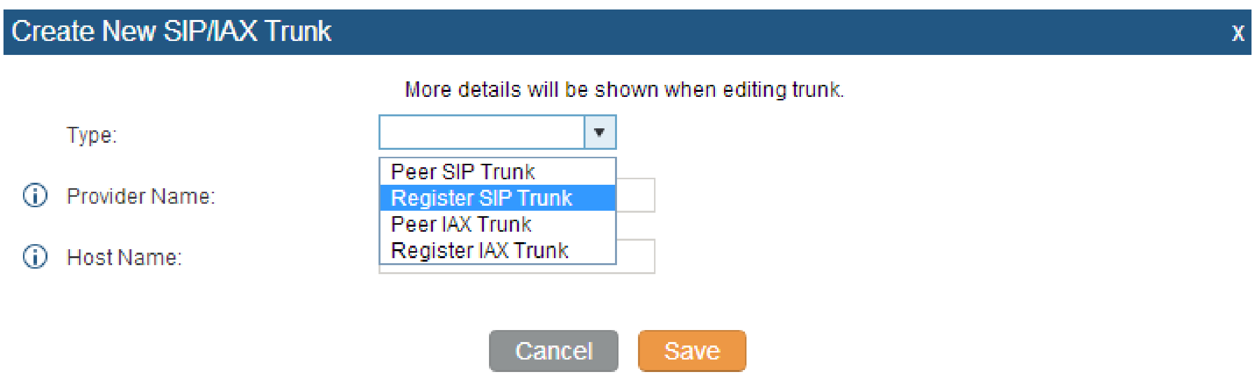

Figure 4: Create Register SIP Trunk on the UCM6100

2. Configure the following information for this trunk so that the UCM6100 can register to the FreePBX trunk we just created.

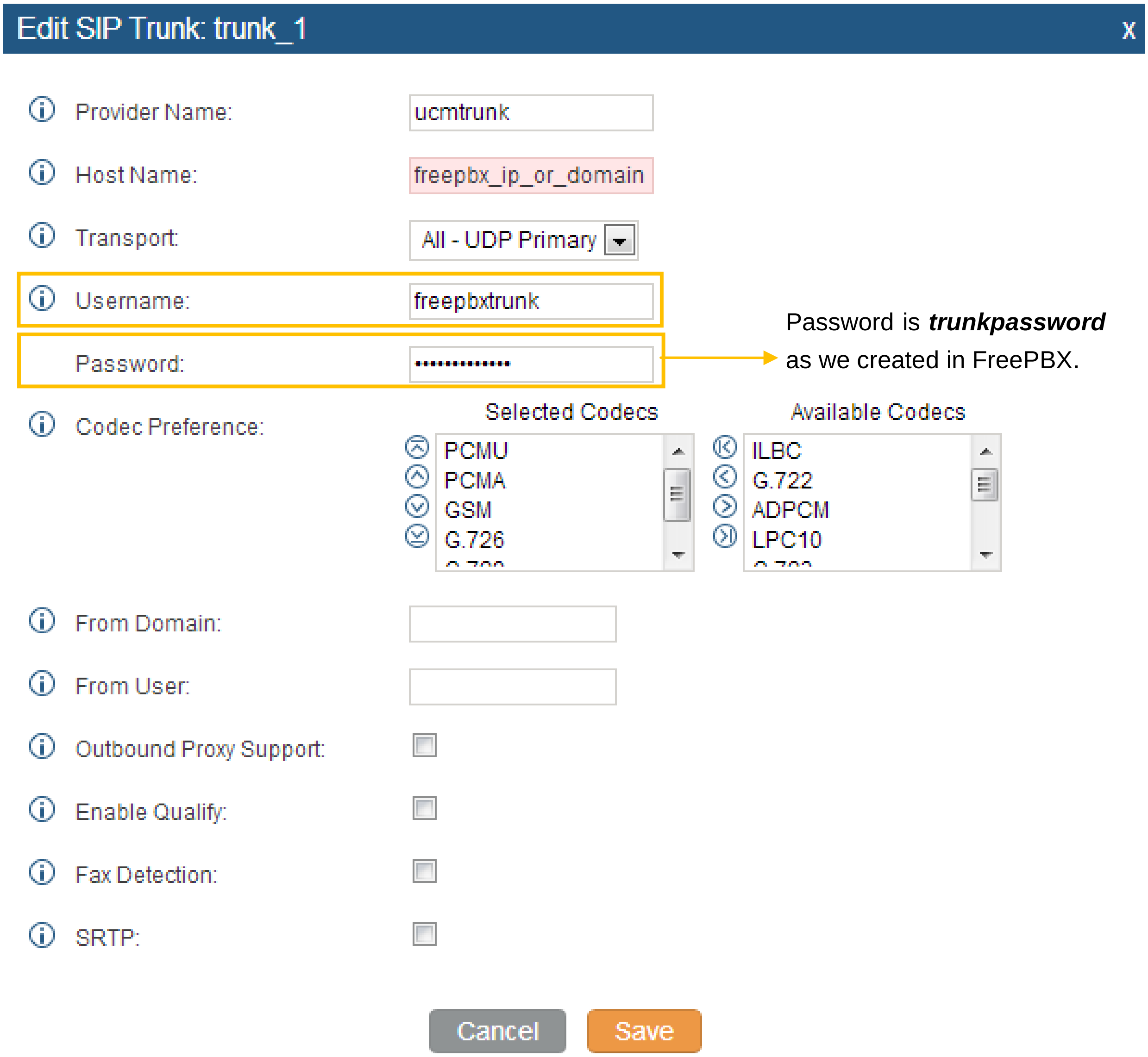

Figure 5: Configure Register SIP Trunk on the UCM6100

The other fields in the above figure are optional.

- ***From Domain***: Configure the FreePBX IP or domain address.
- ***From User***: Same as username, i.e., ***freepbxtrunk***.

Configure Call Routes on FreePBX

1. On the FreePBX web GUI, go to outbound route setting page to create an outbound route for the SIP

trunk.

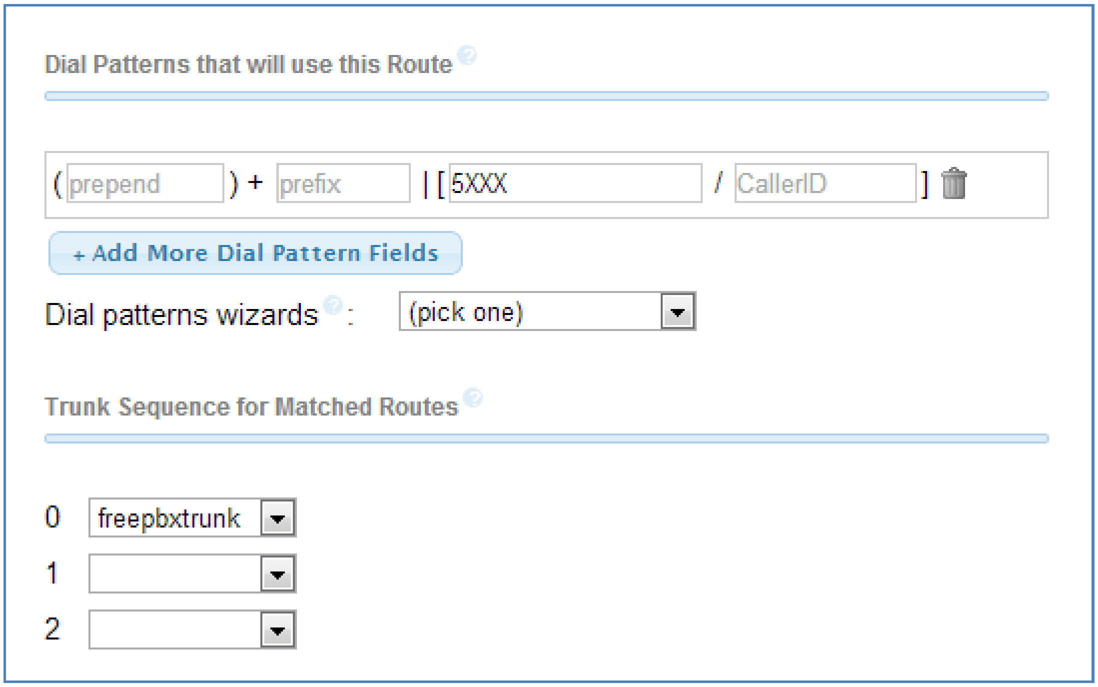

Figure 6: Configure Outbound Route on FreePBX

2. The FreePBX uses DID for inbound route by default. Therefore the extensions on the UCM6100 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 UCM6100

1. 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 extensions (5xx, in this example) on the FreePBX.

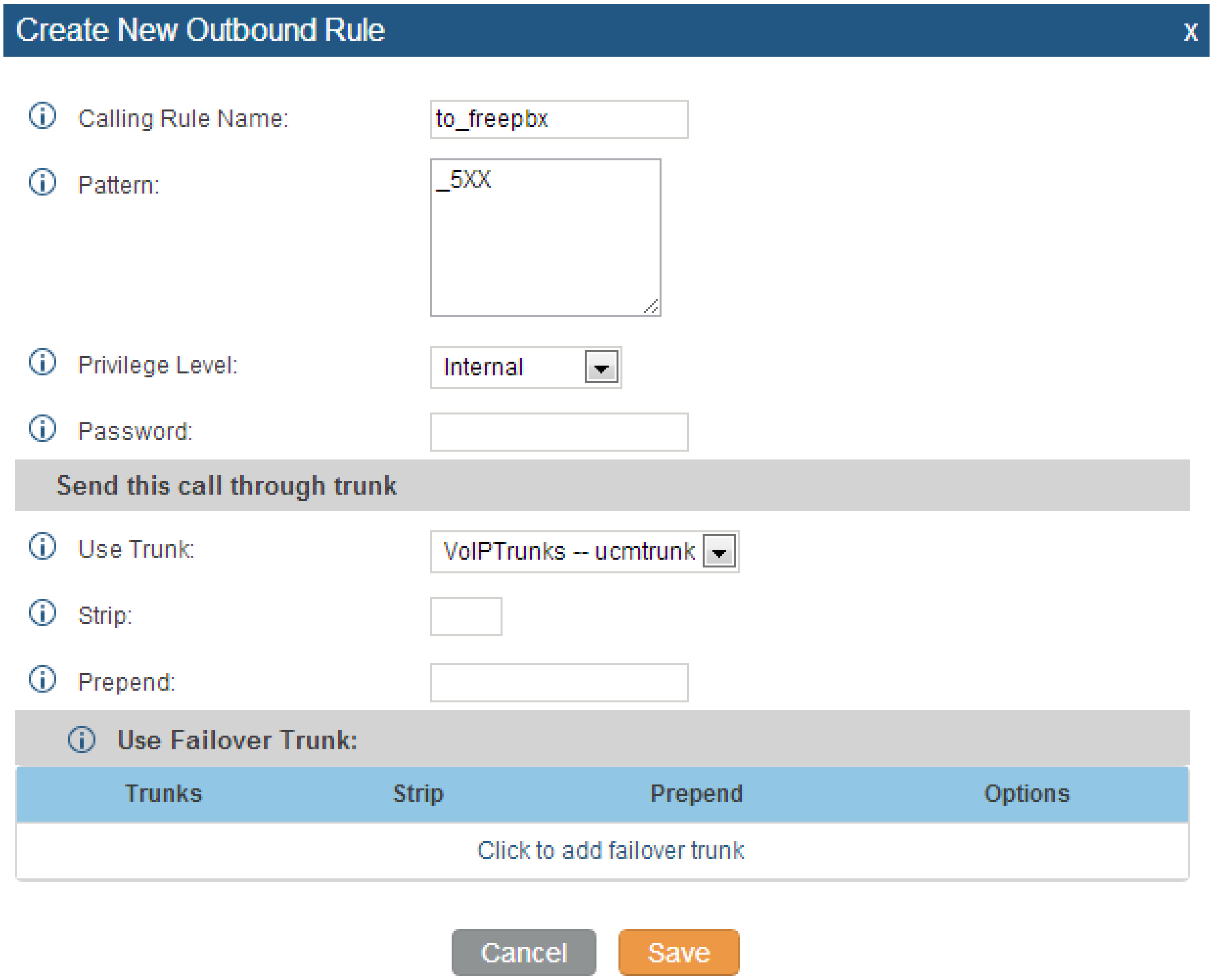

Figure 7: Configure Outbound Route on the UCM6100

2. On the UCM6100 web GUI, go to **PBX**->**Basic/Call Routes**->**Inbound Routes** to create a new inbound rule.

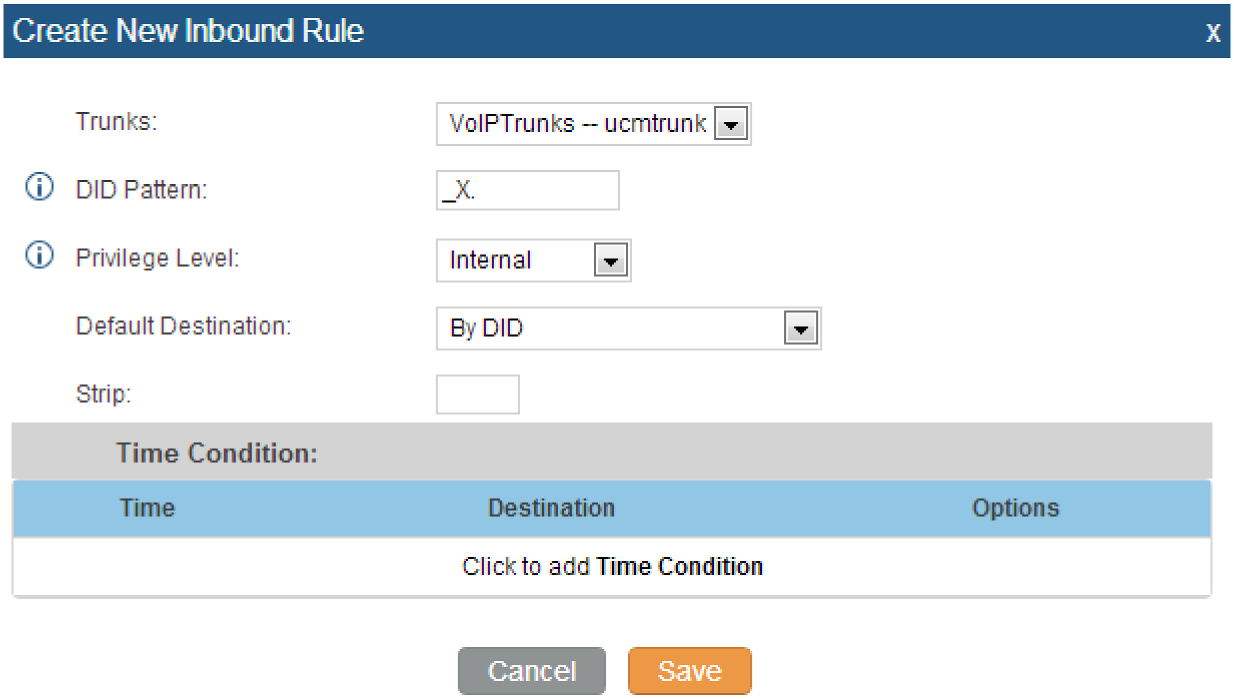

Figure 8: Configure Inbound Route on UCM6100

Now the FreePBX and UCM6100 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 UCM6100.

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