



Grandstream Networks, Inc.

Connecting UCM6XXX with FreePBX®



Table of Contents

OVERVIEW	4
CONNECTING UCM6XXX WITH FREEPBX®	5
Using SIP Trunk with Registration	5
<i>Configure SIP Trunk on FreePBX®</i>	5
<i>Configure SIP Trunk on UCM6XXX</i>	7
Using SIP Peer Trunks	9
<i>Configure SIP Trunk on FreePBX®</i>	9
<i>Configure SIP Trunk on UCM6XXX</i>	10
CALL ROUTING	12
Configure Call Routes on FreePBX®	12
<i>Outbound Calls Routing</i>	12
<i>Inbound Calls Routing</i>	13
Configure Call Routes on UCM6XXX.....	13
<i>Outbound Calls Routing</i>	13
<i>Inbound Calls Routing</i>	14

Table of Figures

Figure 1: FreePBX® Trunk General Settings	5
Figure 2: FreePBX® Trunk Config to Receive Registration.....	6
Figure 3: Create Register SIP Trunk on the UCM6XXX	7
Figure 4: Configure Register SIP Trunk on the UCM6XXX	8
Figure 5: Registered Trunk Status	9
Figure 6: FreePBX® Peer Trunk.....	9
Figure 7: UCM Peer SIP Trunk	10
Figure 8: Enable Heartbeat Detection.....	11
Figure 9: Peer Trunk UCM Status.....	11
Figure 10: FreePBX® Outbound Routes Pattern.....	12
Figure 11: FreePBX® Outbound Routes Trunk Selection	13
Figure 12: Configure Outbound Route on the UCM6XXX	14
Figure 13: Configure Inbound Route on UCM6XXX	15

Table of Tables

Table 1: FreePBX® Trunk PJSIP Settings	6
Table 2: Register trunk UCM settings	8
Table 3: FreePBX® Peer Trunk.....	10



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>

Reproduction or transmittal of the entire or any part, in any form or by any means, electronic or print, for any purpose without the express written permission of Grandstream Networks, Inc. is not permitted.

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.
-

[FreePBX®](#) is a Registered Trademark of [Schmooze Com, Inc.](#)



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.

General	Dialed Number Manipulation Rules	pjsip Settings
Trunk Name	<input type="text" value="UCM6202"/>	
Hide CallerID	<input type="button" value="Yes"/> <input checked="" type="button" value="No"/>	
Outbound CallerID	<input type="text" value="FreePBX"/>	
CID Options	<input type="button" value="Allow Any CID"/> <input type="button" value="Block Foreign CIDs"/> <input type="button" value="Remove CNAM"/> <input checked="" type="button" value="Force Trunk CID"/>	
Maximum Channels	<input type="text" value="10"/>	
Asterisk Trunk Dial Options	<input type="text" value="T"/>	
	<input type="button" value="Override"/> <input checked="" type="button" value="System"/>	
Continue if Busy	<input type="button" value="Yes"/> <input checked="" type="button" value="No"/>	
Disable Trunk	<input type="button" value="Yes"/> <input checked="" type="button" value="No"/>	

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.



Add Trunk

General | Dialed Number Manipulation Rules | **pjsip Settings**

PJSIP Settings

General | Advanced | Codecs

Username

Secret

Authentication Outbound Inbound Both None

Registration Send Receive None

Language Code

SIP Server

SIP Server Port

Context

Transport

Figure 2: FreePBX® Trunk Config to Receive Registration

Following table summarizes the important options:

Table 1: FreePBX® Trunk PJSIP Settings

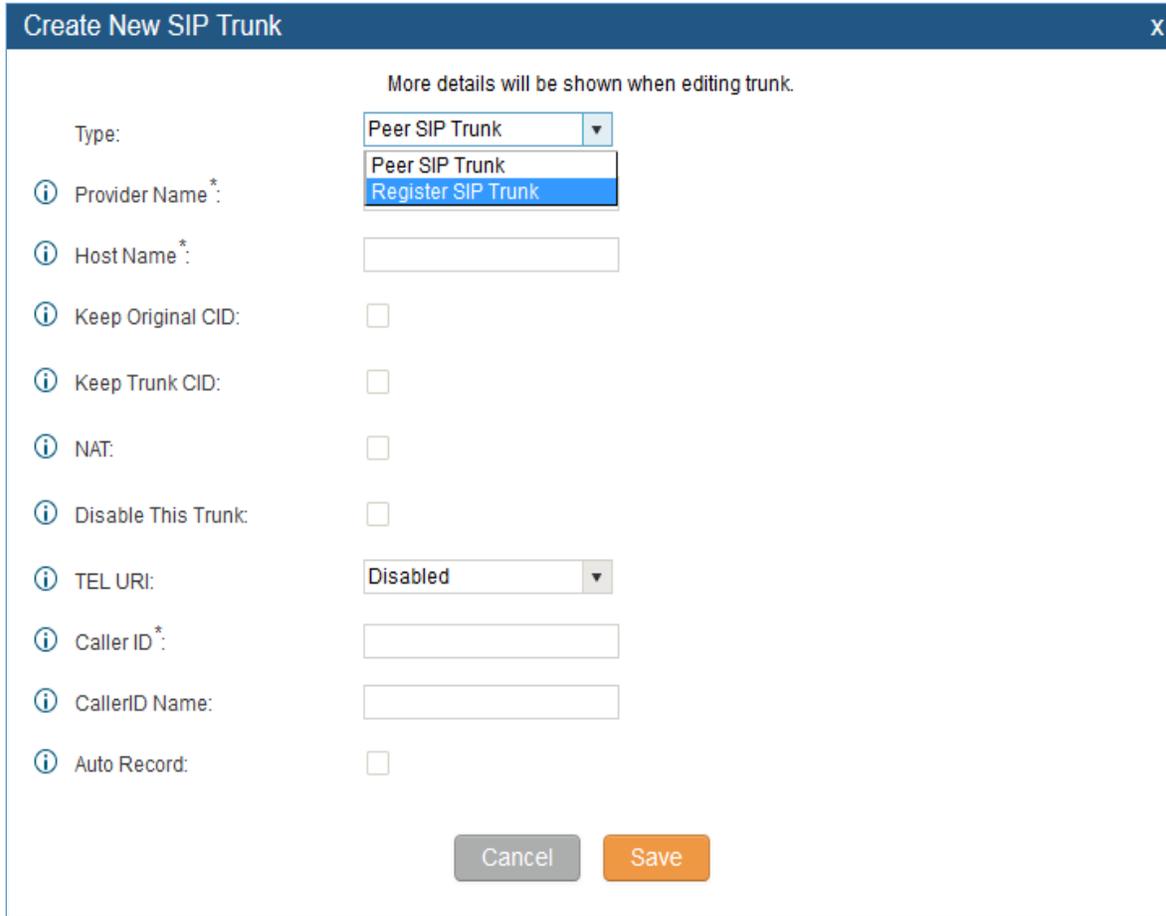
Option	Description
Username	This is the trunk's name and it will be used by UCM to send registration to FreePBX®.
Secret	The Trunk's account password
Authentication	Enable authentication for incoming and/or outgoing calls.
Registration	Choose Receive registration since the UCM will send register request 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.



More details will be shown when editing trunk.

Type: Peer SIP Trunk

Peer SIP Trunk
Register SIP Trunk

Provider Name *:

Host Name *:

Keep Original CID:

Keep Trunk CID:

NAT:

Disable This Trunk:

TEL URI: Disabled

Caller ID *:

CallerID Name:

Auto Record:

Cancel Save

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®.



Create New SIP Trunk
X

More details will be shown when editing trunk.

Type:	<input type="text" value="Register SIP Trunk"/>
(i) Provider Name*:	<input type="text" value="FreePBX"/>
(i) Host Name*:	<input type="text" value="192.168.6.196"/>
(i) Keep Original CID:	<input type="checkbox"/>
(i) Keep Trunk CID:	<input checked="" type="checkbox"/>
(i) NAT:	<input type="checkbox"/>
(i) Disable This Trunk:	<input type="checkbox"/>
(i) TEL URI:	<input type="text" value="Disabled"/>
(i) Need Registration:	<input checked="" type="checkbox"/>
(i) Allow outgoing calls if registration failure:	<input checked="" type="checkbox"/>
(i) CallerID Name:	<input type="text"/>
(i) Username*:	<input type="text" value="UCM6202"/>
Password*:	<input type="password" value="...."/>
(i) AuthID:	<input type="text" value="UCM6202"/>
(i) AuthTrunk:	<input type="checkbox"/>
(i) Auto Record:	<input checked="" type="checkbox"/>

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:



Status	Trunks	Type	Username	Port/Hostname/IP
Registered	FreePBX	SIP	UCM6102	192.168.6.196

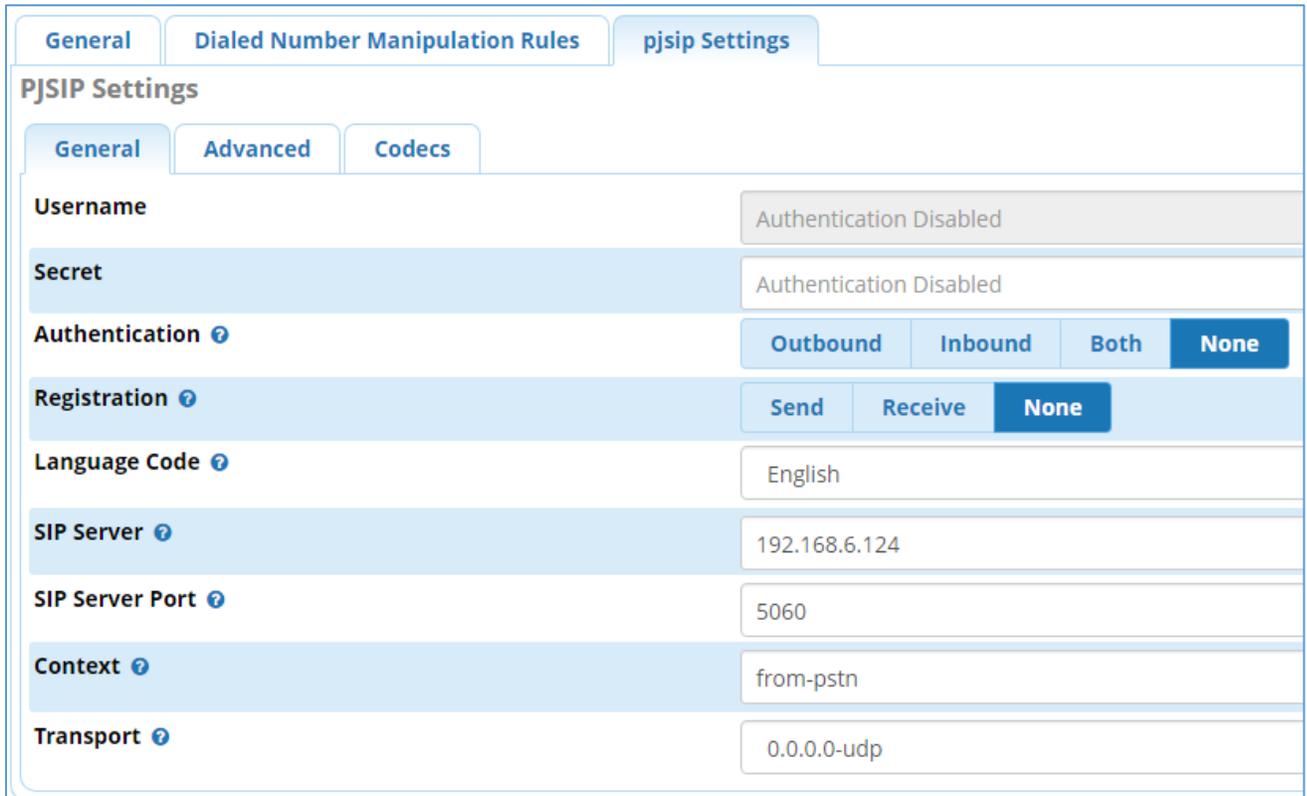
Total: 1 Show: 1/1 Go to: Go First Prev Next Last

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:



General | Dialed Number Manipulation Rules | **pjsip Settings**

PJSIP Settings

General | Advanced | Codecs

Username: Authentication Disabled

Secret: Authentication Disabled

Authentication: Outbound | Inbound | Both | **None**

Registration: Send | Receive | **None**

Language Code: English

SIP Server: 192.168.6.124

SIP Server Port: 5060

Context: from-pstn

Transport: 0.0.0.0-udp

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:

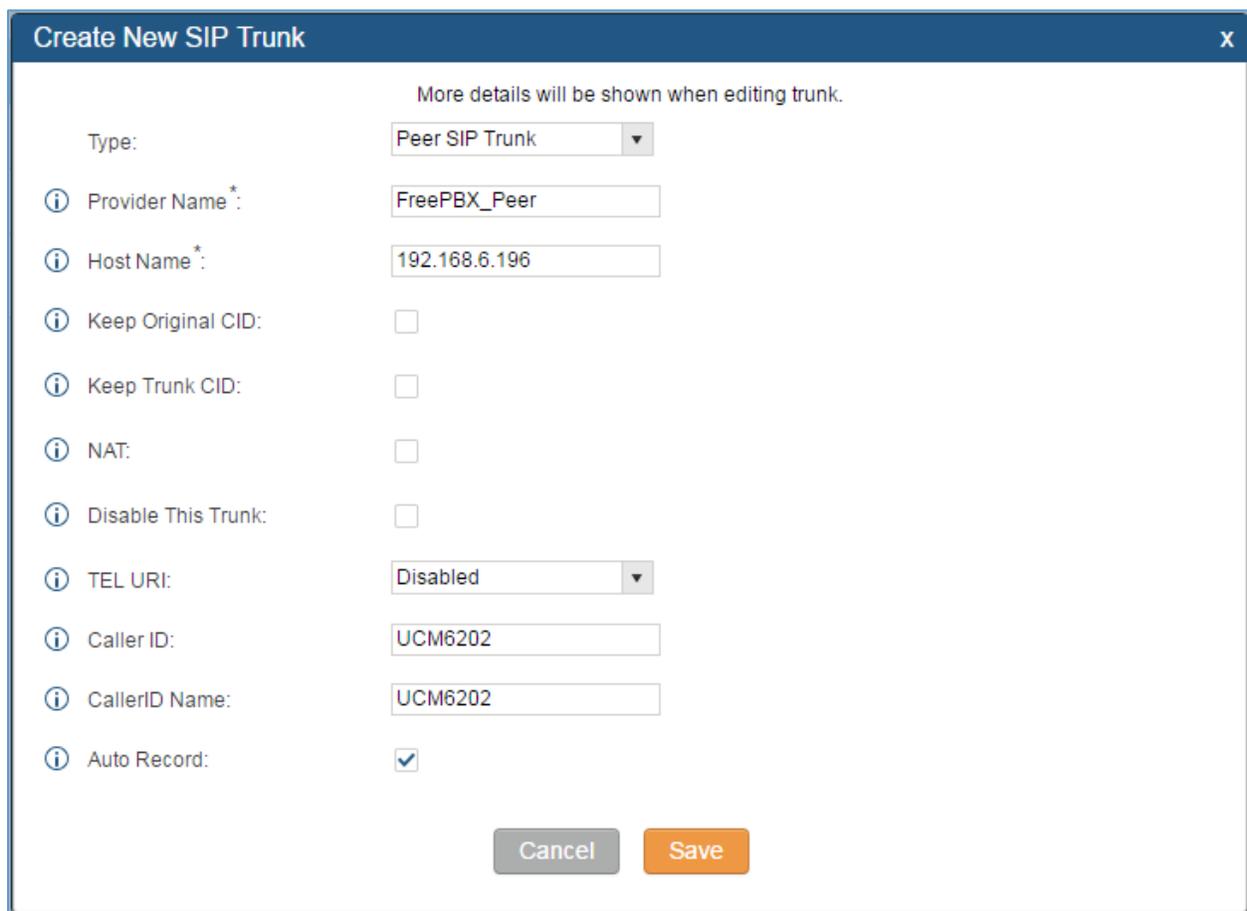


Figure 7: UCM Peer SIP Trunk

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®.



Edit SIP Trunk: FreePBX_Peer
X

Basic Settings
Advanced Settings

Codec Preference:

Available Codecs
 G.722
 AAL2-G.726-32
 ADPCM
 G.723
 H.263

⊗
⊗
⊗
⊗

Selected Codecs
 PCMU
 PCMA
 GSM
 G.726
 G.729

⊗
⊗
⊗
⊗

Send PPI Header:

Send PAI Header:

DID Mode:

DTMF Mode:

Enable Heartbeat Detection:

Heartbeat Frequency*:

The Maximum Number of Call Lines*:

Fax Mode:

SRTP:

Sync LDAP Enable:

CC Settings

Enable CC:

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:

Status
PBX
Settings
Maintenance

Status >> PBX Status >> PBX Status

PBX Status
- PBX Status
- Active Calls

System Status

Trunks
[-]

Status	Trunks	Type	Username	Port/Hostname/IP
Reachable	freepx	SIP		192.168.6.196

Total: 1 Show: 1/1 Go to:

Figure 9: Peer Trunk UCM Status



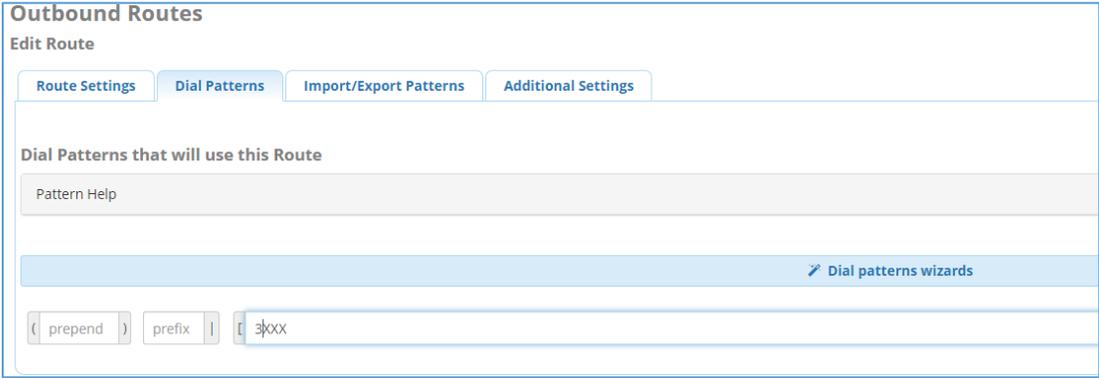
CALL ROUTING

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.



The screenshot shows the 'Outbound Routes' configuration page in FreePBX. The page title is 'Outbound Routes' and it is in 'Edit Route' mode. There are four tabs: 'Route Settings', 'Dial Patterns', 'Import/Export Patterns', and 'Additional Settings'. The 'Dial Patterns' tab is active. Below the tabs, there is a section titled 'Dial Patterns that will use this Route'. Under this section, there is a 'Pattern Help' link. A blue bar with a pencil icon and the text 'Dial patterns wizards' is visible. At the bottom, there is a text input field with the value '3XXX'. To the left of the input field are three buttons: 'prepend', 'prefix', and 'suffix'.

Figure 10: FreePBX® Outbound Routes Pattern

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



Route Settings	Dial Patterns	Import/Export Patterns	Additional Settings
Route Name ?	ToUCM		
Route CID ?			
Override Extension ?	<input type="radio"/> Yes <input checked="" type="radio"/> No		
Route Password ?			
Route Type ?	<input type="radio"/> Emergency <input checked="" type="radio"/> Intra-Company		
Music On Hold? ?	default		
Time Group ?	---Permanent Route---		
Route Position ?	---No Change---		
Trunk Sequence for Matched Routes ?	<input type="button" value="+"/> UCM6102 <input type="button" value="+"/>		
Optional Destination on Congestion ?	Normal Congestion		
Note: Extension Routes is not registered			

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.



Create New Outbound Rule x

Calling Rule Name*

Pattern*

Disable This Route:

Call Duration Limit:

PIN Groups:

Password:

Privilege Level:

Enable Filter on Source Caller ID:

Send This Call Through Trunk

Use Trunk*

Strip:

Prepend:

Use Failover Trunk:

Trunks	Strip	Prepend	Options
Click to add failover trunk			

Time Condition

Time Condition	Time	Options
Click to add Time Condition		

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.



Create New Inbound Rule
X

Trunks*: SIPTunks -- FreePBX

DID Pattern*: _X /

Disable This Route:

Prepend Trunk Name:

Prepend User Defined Name:

Alert-info: None

Inbound Multiple Mode:

Dial Trunk:

DID Destination: Extension Conference Call Queue Ring Group
 Paging/Intercom Groups IVR Voicemail Groups
 Fax Extension Dial By Name All

Allowed to seamless transfer:

Available Extensions

3001
3002
3003
3004
3005
3006

Selected Extensions

Default Mode | **Mode 1**

Default Destination*: By DID

Strip: 0

Prepend:

Time Condition	Time	Destination	Options
Click to add Time Condition			

Cancel
Save

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.

** Asterisk is a Registered Trademark of Digium, Inc.*

