



## Grandstream Networks, Inc.

Interoperability Tutorial:

Configuring UCM6100 Series with FreePBX®

Grandstream Networks, Inc.

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## 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.

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 **Warning:**

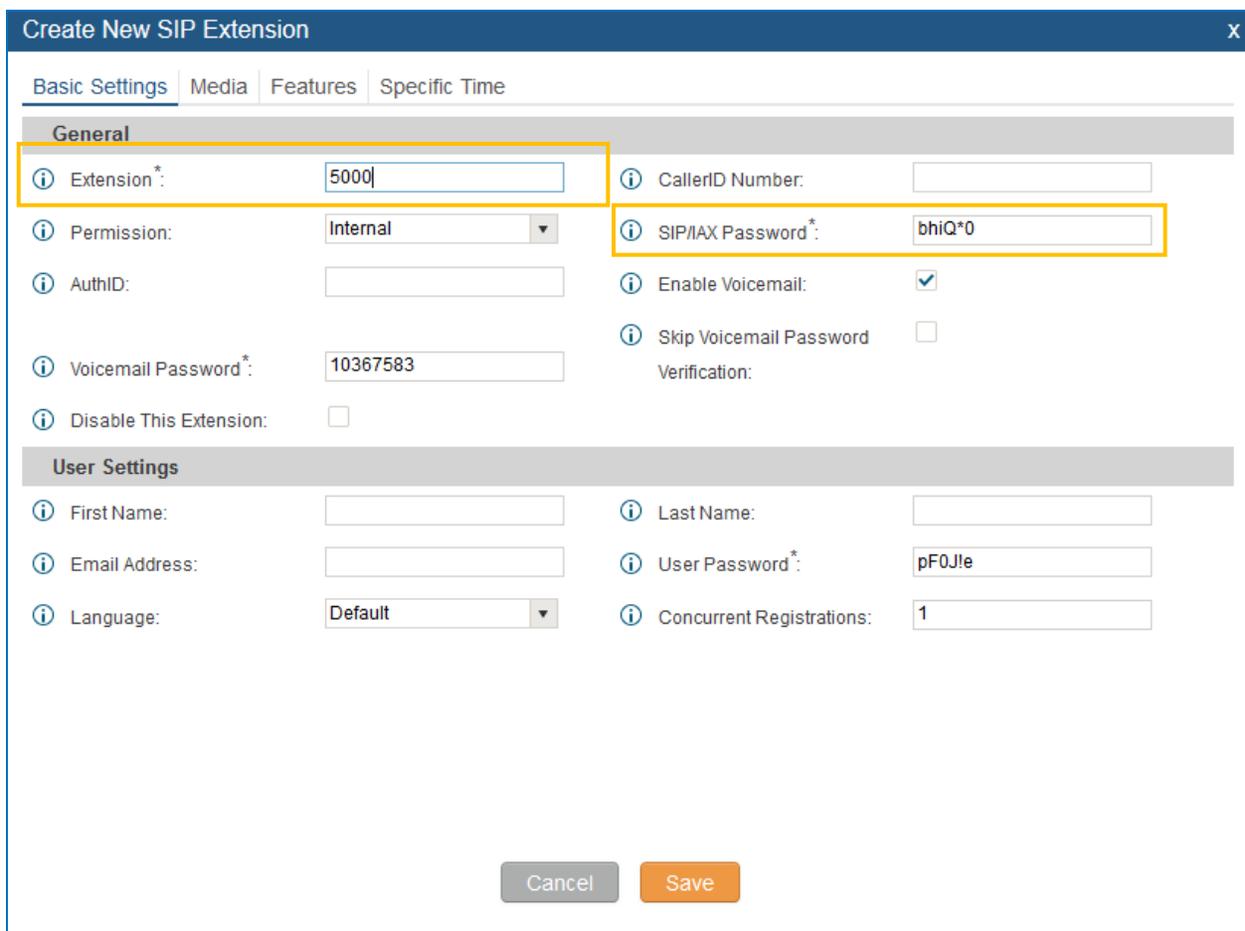
- When the UCM6100 series is interconnected with other PBX, it is NOT recommended to turn on "Allow Guest Calls" under web GUI->**PBX->SIP Settings->General**. Turning on this option will allow unauthenticated calls coming through the UCM6100 series. Please be aware of the security concerns when using 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.



General	
Extension *	5000
Permission:	Internal
AuthID:	
Voicemail Password *	10367583
Disable This Extension:	<input type="checkbox"/>
CallerID Number:	
SIP/IAX Password *	bhiQ*0
Enable Voicemail:	<input checked="" type="checkbox"/>
Skip Voicemail Password Verification:	<input type="checkbox"/>
User Settings	
First Name:	
Email Address:	
Language:	Default
Last Name:	
User Password *	pF0J!e
Concurrent Registrations:	1

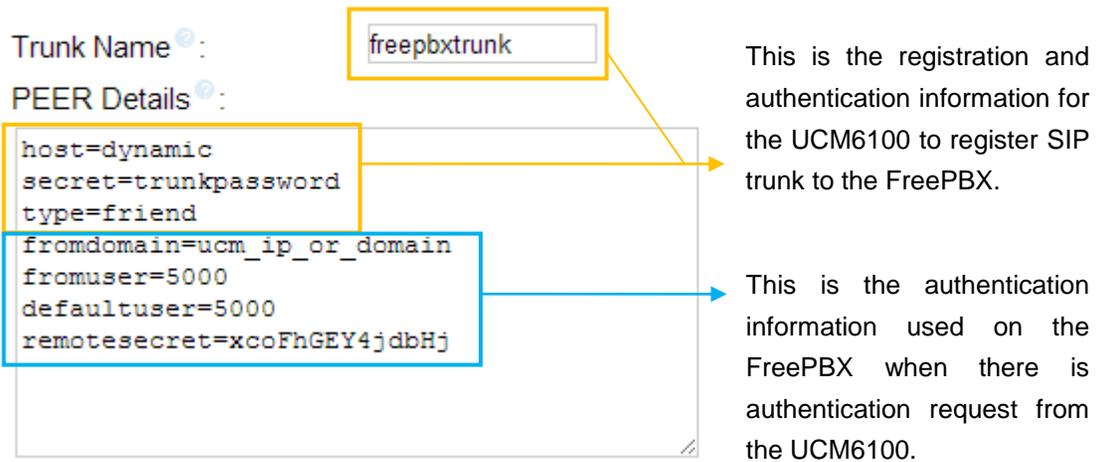
Figure 1: Create an Extension on UCM6100

### 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.

### Outgoing Settings



Trunk Name: **freepbxtrunk**

PEER Details:

```

host=dynamic
secret=trunkpassword
type=friend
fromdomain=ucm_ip_or_domain
fromuser=5000
defaultuser=5000
remotesecond=xcoFhGEY4jdbHj

```

This is the registration and authentication information for the UCM6100 to register SIP trunk to the FreePBX.

This is the authentication information used on the FreePBX when there is authentication request from the UCM6100.

**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.
- **remotesecond=xcoFhGEY4jdbHj**  
This is the password of the extension created on UCM6100 in section *[Create Extension on UCM6100]*.

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]*.

**Registration**

Register String <sup>?</sup>:

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.

Create New SIP Trunk
X

More details will be shown when editing trunk.

Type:	<span style="border: 1px solid #ccc; padding: 2px;">Peer SIP Trunk</span> ▼
ⓘ Provider Name *:	<span style="border: 1px solid #ccc; padding: 2px;">Peer SIP Trunk</span> <span style="border: 1px solid #ccc; padding: 2px; background-color: #007bff; color: white;">Register SIP Trunk</span>
ⓘ Host Name *:	<input style="width: 100%;" type="text"/>
ⓘ Keep Original CID:	<input type="checkbox"/>
ⓘ Keep Trunk CID:	<input type="checkbox"/>
ⓘ NAT:	<input type="checkbox"/>
ⓘ Disable This Trunk:	<input type="checkbox"/>
ⓘ TEL URI:	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span> ▼
ⓘ Caller ID *:	<input style="width: 100%;" type="text"/>
ⓘ CallerID Name:	<input style="width: 100%;" type="text"/>
ⓘ Auto Record:	<input type="checkbox"/>

Cancel
Save

**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.

Create New SIP Trunk
X

More details will be shown when editing trunk.

Type:	Register SIP Trunk ▼
<i>i</i> Provider Name *:	ucmtrunk
<i>i</i> Host Name *:	freepbx_ip_or_domain
<i>i</i> Keep Original CID:	<input type="checkbox"/>
<i>i</i> Keep Trunk CID:	<input checked="" type="checkbox"/>
<i>i</i> NAT:	<input type="checkbox"/>
<i>i</i> Disable This Trunk:	<input type="checkbox"/>
<i>i</i> TEL URI:	Disabled ▼
<i>i</i> Need Registration:	<input checked="" type="checkbox"/>
<i>i</i> Username *:	freepbxtrunk
Password *:	●●●●●●
<i>i</i> AuthID:	
<i>i</i> AuthTrunk:	<input type="checkbox"/>
<i>i</i> Auto Record:	<input type="checkbox"/>

Cancel
Save

→ Password is ***trunkpassword*** as we created in FreePBX.

**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.

The screenshot shows a configuration page for an outbound route. At the top, there is a section titled "Dial Patterns that will use this Route" with a blue header bar. Below this, there is a form with several input fields: a "prepend" field, a "+" sign, a "prefix" field, a vertical bar "|", a "5XXX" field, a "/" sign, and a "CallerID" field. To the right of the "CallerID" field is a trash icon. Below the form is a blue button labeled "+ Add More Dial Pattern Fields". Underneath the button is a label "Dial patterns wizards" followed by a dropdown menu currently showing "(pick one)". Below this is another section titled "Trunk Sequence for Matched Routes" with a blue header bar. Underneath, there are three rows, each with a number (0, 1, 2) and a dropdown menu. The first row (0) has the value "freepbxtrunk" selected. The second and third rows (1 and 2) are currently empty.

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.

Create New Outbound Rule
X

i Calling Rule Name\*:

i Pattern\*:

i Password:

i Call Duration Limit:

i Privilege Level:  Warning: Setting privilege level at 'Internal' has potential security risks.

i Enable Filter on Source Caller ID:

Send this call through trunk

i Use Trunk\*:

i Strip:

i Prepend:

Use Failover Trunk:

Trunks	Strip	Prepend	Options
Click to add failover trunk			

Figure 7: Configure Outbound Route on the UCM6100

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

Create New Inbound Rule
X

**Trunks \***: SIPTrunks -- ucctrunk ▼

**DID Pattern \***: \_X. /

**Prepend Trunk 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

---

**Default Mode** | Mode 1

**Default Destination \*:** By DID ▼

**Strip:** 0

**Prepend:**

**Time Condition**

Time Condition	Time	Destination	Options
Click to add Time Condition			

Cancel
Save

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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