Grandstream Networks, Inc.

UCM6xxx SIP Trunks Guide

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INTRODUCTION

SIP trunks are a VoIP service that can be provided from an ITSP (Internet Telephony Service Provider) to extend telephony features beyond IPPBX local area. SIP trunks can carry voice calls, video calls, instant messages, multimedia conferences, and other SIP-based, real-time communications services.

Using SIP trunks helps to reduce call rates especially when making long distance calls, since VoIP providers can offer better calling rates compared to local ISP using analog lines.

UCM6xxx series support two types of SIP trunks: “Register SIP trunks”, mainly used to connect with provider’s trunk and “Peer trunks”, that can be used to interconnect multiple IP-PBXs. UCM6xxx series support up to 50 SIP trunks.

This guide describes needed configuration to set up register trunk (with provider) and peer trunk (between two UCM6xxx).

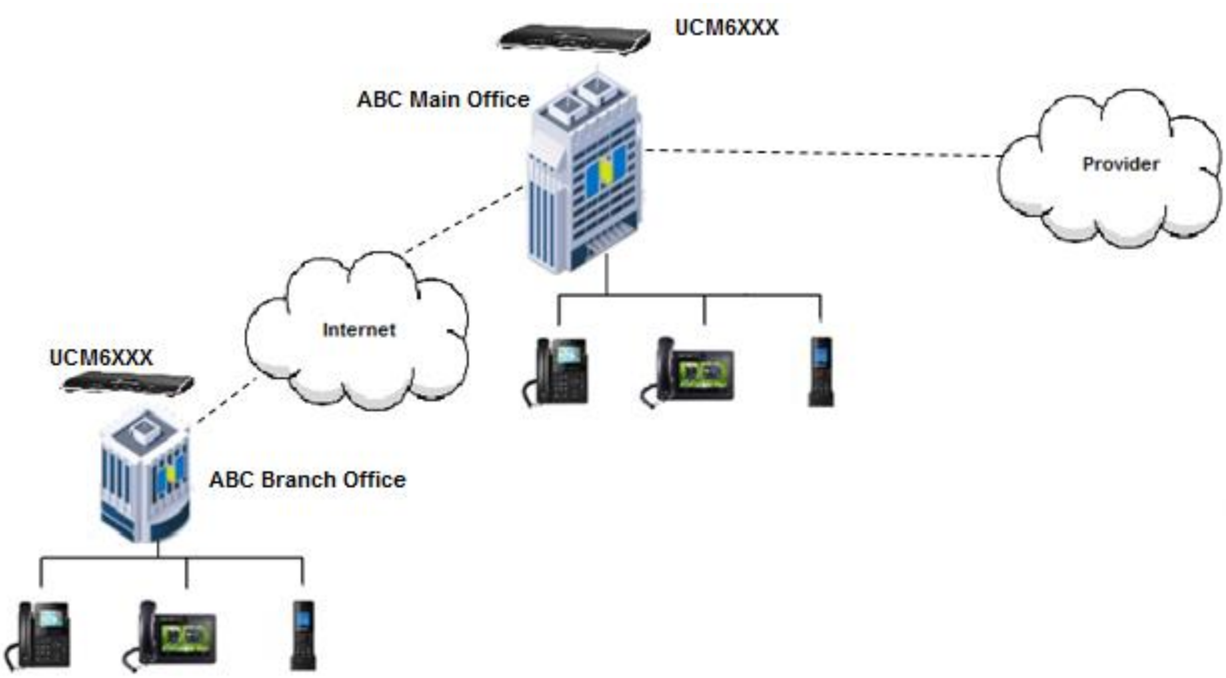

Figure 1: UCM6xxx Typical Scenario

The figure above shows a typical scenario using two UCM6xxx between main and branch offices (connected via peer trunk), and main office UCM6xxx connected to provider via register SIP trunk.

REGISTER SIP TRUNKS

In this guide, we will take company ABC as example. The company has purchased two SIP trunks, the first one will be used for International calls, while the second will be used of national ones.

The ITSP provider gave following trunk information to the company ABC in order to register their trunks.

SIP trunk information	Value
SIP Trunk 1	
Provider address	sip.provider.com
Username	0655441000
Authenticate ID	0655441000
Password	admin123
Main number	0655441000
Provided DIDs	0655441001 / 0655441002 / 0655441003 / 0655441004 / 0655441005 / 0655441006 / 0655441007 / 0655441008 / 0655441009 / 0655441010
SIP Trunk 2	
Provider address	sip.provider.com
Username	0655441100
Authenticate ID	0655441100
Password	admin123
Main number	0655441100
Provided DIDs	0655441101 / 0655441102 / 0655441103 / 0655441104 / 0655441105 / 0655441106 / 0655441107 / 0655441108 / 0655441109 / 0655441110

Setup considerations:

- Extensions range on UCM6xxx in main office is 1000 – 1999.
- Dialing to international numbers should be prefixed with 99 with no size limits.
- National numbers start with 06 with 10-digit length.

Configuration

Below steps to configure SIP Trunk 1 on Main office UCM6xxx. Same steps apply to configure Trunk 2.

1. Access UCM6xxx web GUI → **Extension / Trunk → VoIP Trunks.**
2. Click on + Create New SIP Trunk, in the following screenshot type in Trunk 1 credentials:

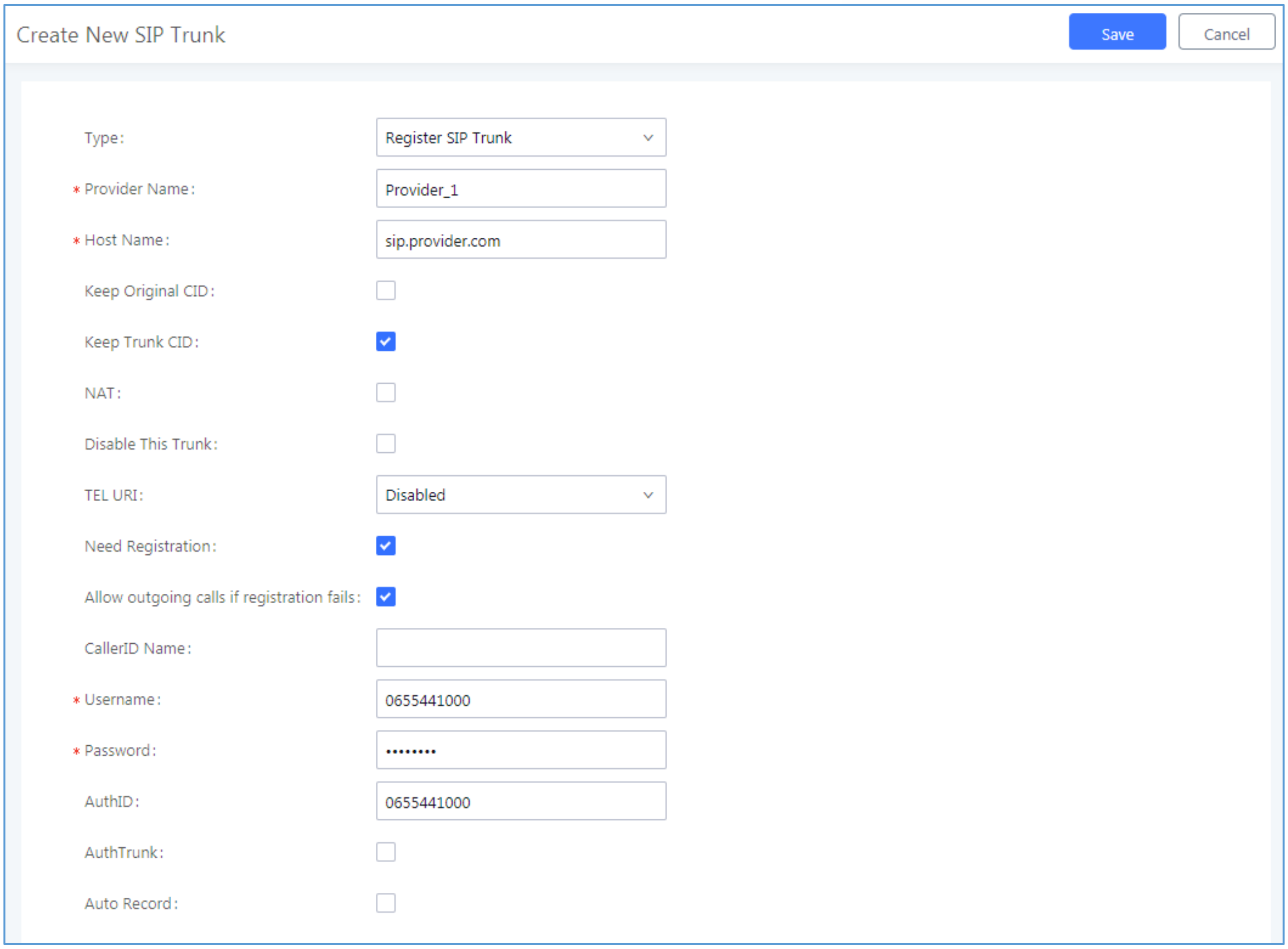

Figure 2: Create New Register Trunk

- Select "**Register SIP Trunk**" as Type.
- Type in a reference name for **Provider Name**. For example: "Provider_1".
- Enter provider's FQDN/IP address in **Host Name** field. In this example: "sip.provider.com".
- Enter **Username** to register to the provider. In this example: "0655441000".
- Enter **Authenticate ID** to register to the provider. In this example: "0655441000".
- Enter **Password** associated with username. In this example: "admin123".

3. Click Save and Apply Changes to store and apply the configuration.

Above steps describe basic configuration needed to register a SIP trunk. Depending on providers, users may need to adjust their settings to successfully register a SIP trunk.

The table below describes main parameters available for Register SIP Trunks:

Table 1: Register Trunk Parameters

Fields	Description
Keep Original CID	Keep the CID from the inbound call when dialing out. This setting will override “Keep Trunk CID” option. Please make sure that the peer PBX at the other side supports to match user entry using “username” field from authentication line.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension’s CID when the extension has CID configured. The default setting is “No”. If enabled and **Keep Original CID** is disabled, the callee will see the ID set on Username field of UCM6xxx, in this case “0655441000“.
NAT	Turn on this option when the PBX is using public IP and communicating with devices behind NAT. If there is one-way audio issue, usually it’s related to NAT configuration or SIP/RTP port configuration on the firewall.
Disable This Trunk	If selected, the trunk will be disabled. **Note**: If a current SIP trunk is disabled, UCM6xxx will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
TEL URI	If the trunk has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled. • If set to **Disabled**, the **TO** header in this example will be “*To: <sip:0655441000@sip.provider.com>*”. • If set to **User=Phone**, the **TO** header in this example will be “*To: <sip: 0655441000@ sip.provider.com;user=phone>*”. • If set to **Enabled,** the **TO** header in this example will be “*To: <tel: 0655441000>*”.
Need Registration	Select whether the trunk needs to register on the external server or not when "Register SIP Trunk" type is selected.
Allow outgoing calls if registration failure	If enabled outgoing calls even if the registration to this trunk fail will still be able to go through. Note that if we uncheck “Need Registration” option, this option will be ignored.
CallerID Name	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored. When making outgoing calls, the following rules are used to determine which CallerID will be used if they exist: • The CallerID configured for the extension will be looked up first. • If no CallerID is configured for the extension, the CallerID configured for the trunk will be used. • If the above two are missing, the "Global Outbound CID" defined in Web GUI- >PBX->Internal Options->General will be used. <u>Example:</u> Caller ID set to “GSTest”, the “From” header will be: *From: "**GSTest**" <sip:0655441000@ucm1.abc.com>;tag=f268*
From Domain	Configure the actual domain where the extension comes from. <u>Example:</u> If set to “gs.test.com”, the “From” header will be as shown below: *From: <sip:0655441000@**gs.test.com**>;tag=f268*

From User	Configure the actual user name of the extension. Example: If set to "0655447777", it will override username in "From" header as shown below: *From: <sip: **0655447777**@gs.test.com>;tag=f268*
Username	Enter the username to register to the trunk from the provider when "Register SIP Trunk" type is selected.
Password	Enter the password to register to the trunk from the provider when "Register SIP Trunk" is selected.
Auth ID	Enter the authentication ID for "Register SIP Trunk" type. **Note:** This is the SIP service subscriber's ID used for authentication. If not configured the Extension Number will be used for authentication.
Auth Trunk	If enabled, the UCM will send 401 response to the incoming call to authenticate the trunk.
Auto Record	Enable automatic recording for the calls using this trunk (for SIP trunk only). The default setting is disabled. The recording files can be accessed under web GUI->CDR->Recording Files.

After saving the new VoIP trunk, it will be displayed under **Web UI → Extension / Trunk → VoIP Trunks** as shown below. Press to edit trunk settings if needed.

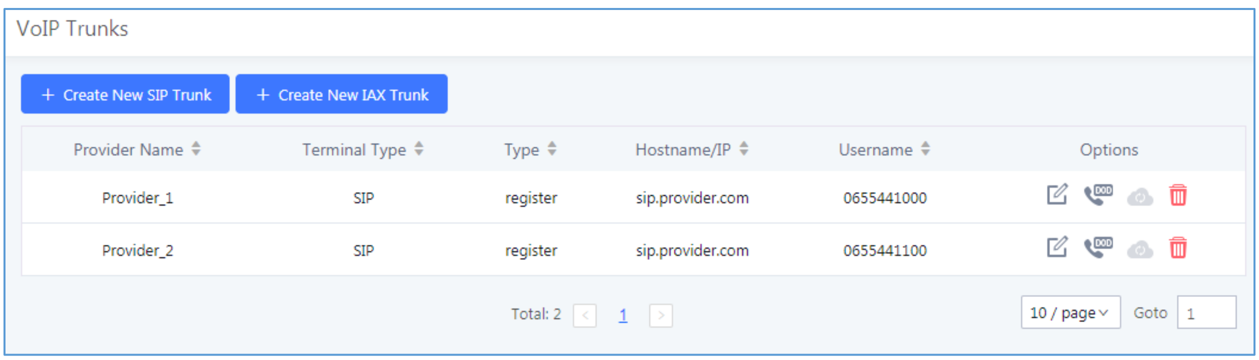

Figure 3: Register Trunk

Once the trunk has been created, users can check its registration status under **System Status → Dashboard** page.

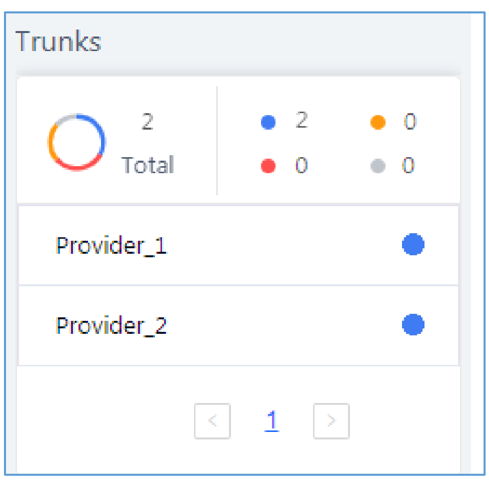

Figure 4: Register Trunk Status

Note: If status shows “Rejected”, it means that UCM6xxx didn’t get registered with the provider. Verify that provider’s server is reachable from UCM6xxx and double confirm trunk credentials.

DID / DOD Configuration

Direct Inward Dialing (DID)

DID (Direct Inward Dialing) is a service provided by the ITSP to subscribers using IP-PBX, adding possibility to route incoming calls to a specific DID to a specific extension. Using DIDs, it will allow extensions to be reachable from outside directly without going through main office number.

In this example, following DIDs have been provided with first trunk “Provider_1”: 0655441000 / 0655441001 / 0655441002 / 0655441003 / 0655441004 / 0655441005 / 0655441006 / 0655441007 / 0655441008 / 0655441009 / 0655441010.

Company wants to redirect incoming calls to specific extensions using provided DIDs.
UCM6xxx in main office is using extensions range: 1000 – 1999.

Note: Make sure **DID Mode** is set correctly under trunk advanced settings. Provider may include DID number in “Request-line” or “To-header”.

Following configuration can be done to achieve this:

1. Access UCM6xxx Web UI → **Extension/Trunk** → **Inbound Routes**. Select “Trunk 1” and press “**Add**” button.
2. In **DID Pattern** field, enter “_0655441XXX”.
3. Configure **ByDID** as **Default Destination** and check “Extension” in “DID Destination” options.
4. Configure **Strip** field. In this example: Strip=6 to remove first 6 digits from incoming number and keep last 4 digits. If call is coming to 0655441007, UCM6xxx will strip 065544 and keep 1007.
5. Click Save and Apply Changes to apply configuration.

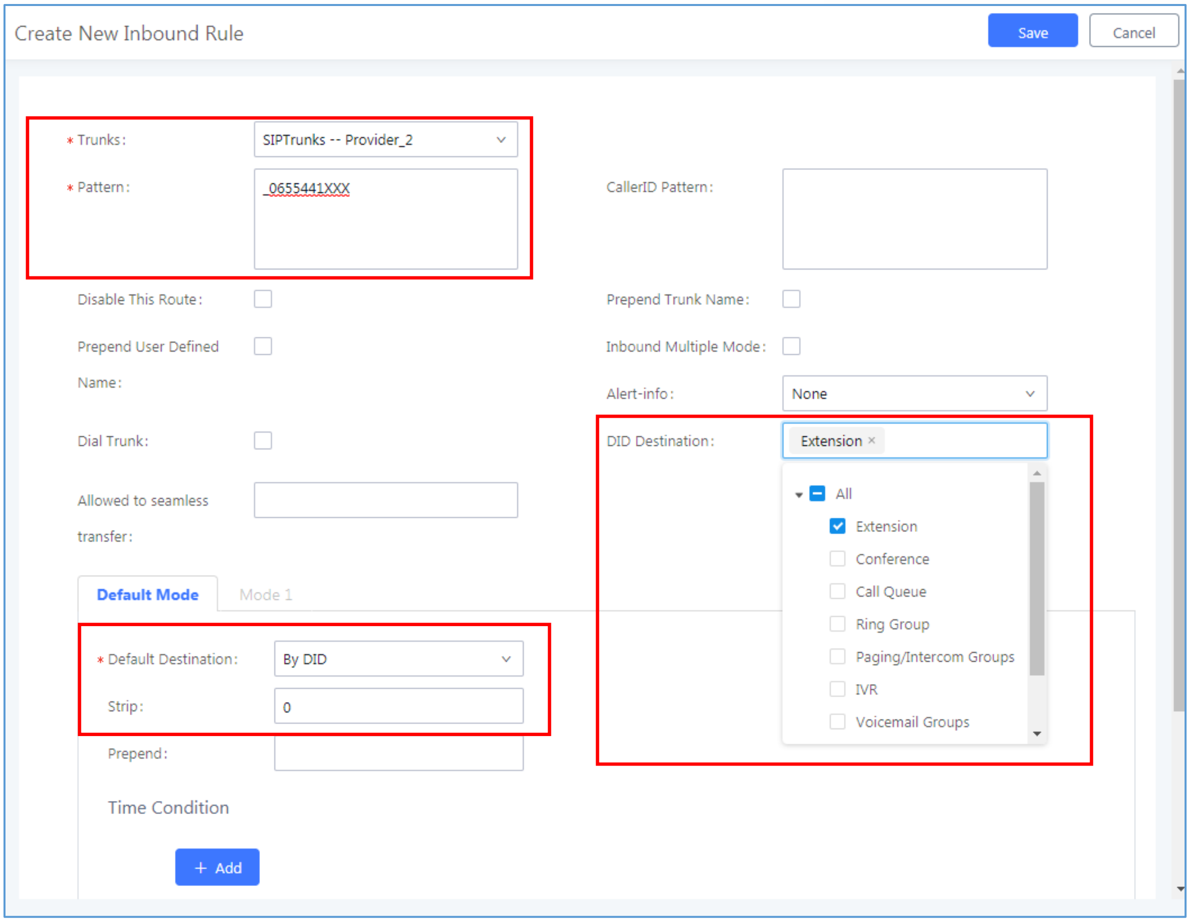

Figure 5: Inbound Rule - ByDID

Direct Outward Dialing (DOD)

The UCM6xxx provides Direct Outward Dialing (DOD) which is a service of a local phone that allows extensions within a company ABC's UCM to connect to outside lines directly.

We will use Company's ABC trunk 0655441000 with 11 DIDs associated to it. At the moment when a user makes an outbound call their caller ID shows up as the main office number which is 0655441000. This create a problem as the CEO would like that his calls to comes from his direct line. This can be accomplished by configuring DOD for the CEO's extension. Other group member also can benefit from DOD to have their own line showed when making calls.

Steps on how to configure DOD on the UCM6xxx:

1. To setup DOD go to UCM6xxx Web UI → **Extension/Trunk → VoIP Trunks** page.
2. Click  to access the DOD options for the selected SIP Trunk.
3. Click "**Create a new DOD**" to begin your DOD setup.
4. For "**DOD Number**", enter one of the numbers (DIDs) from your SIP trunk provider. In this example, we will enter in the number for the CEO's direct line (0655441000).

5. If extension number need to be appended to the DID number click on “Add Extension”.
6. Select an extension from the "Available Extensions" list. Users have the option of selecting more than one extension. In this case, we would select the CEO's extension. After making the selection, click on the > button to move the extension(s) to the "Selected Extensions" list.
7. Click Save and Apply Changes to apply configuration.

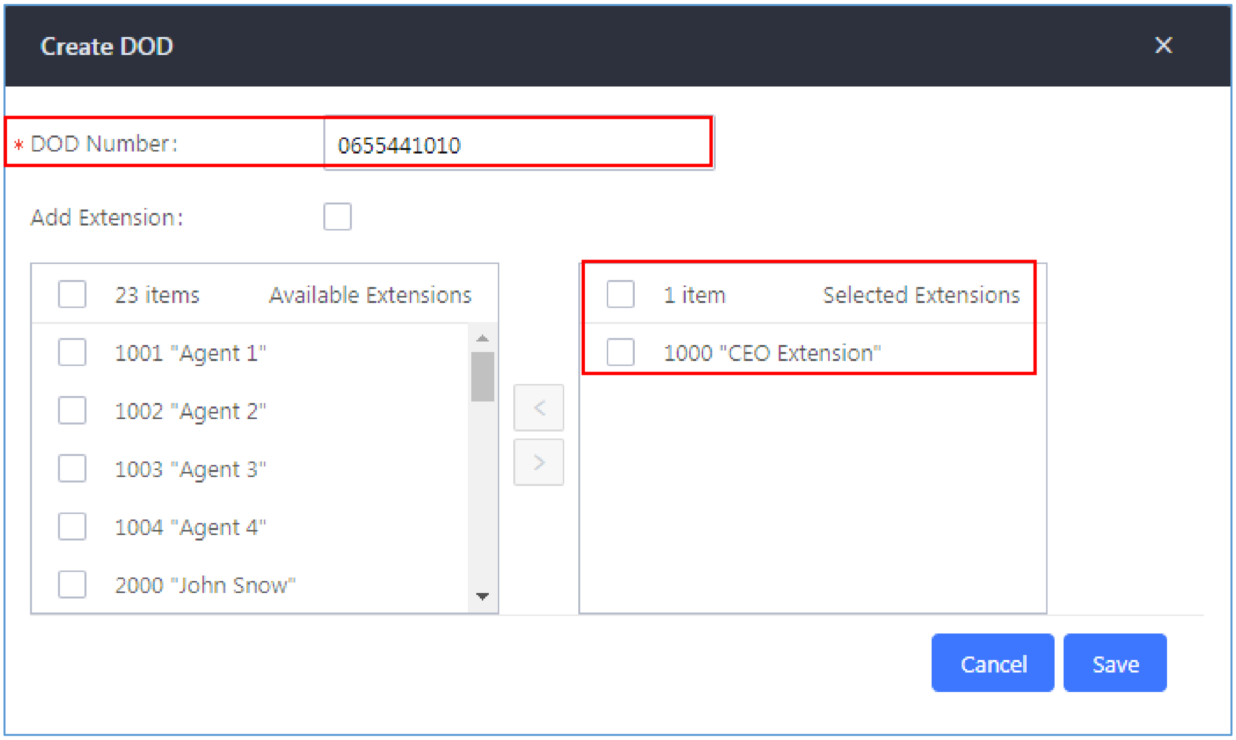

Figure 6: Create New DOD

Users can press “**Edit DOD**” to check/add/delete extensions that are associated to a particular DOD.

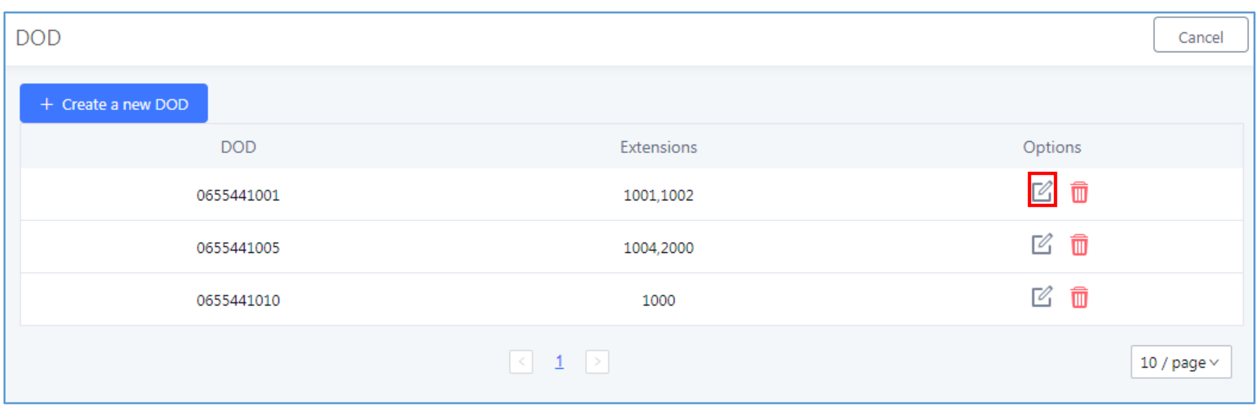

Figure 7: DOD List

Create Outbound/Inbound Routes

In this section we will give an example of creating Outbound and Inbound Rules for trunk 1 “Provider_1” to allow receiving incoming calls and to be redirected to IVR and to allow outgoing calls to be made.

Create Inbound Routes

1. Access UCM6xxx Web UI → **Extension/Trunk** → **Inbound Routes**. Select “Trunk 1” and press “**Create New Inbound Rule**”.
2. In **DID Pattern** field, enter “**_X.**”. This allows all incoming calls to be received.
3. Set **IVR** as **Default Destination**. (Assuming IVR is pre-configured).
4. Click Save and Apply Changes to apply configuration.

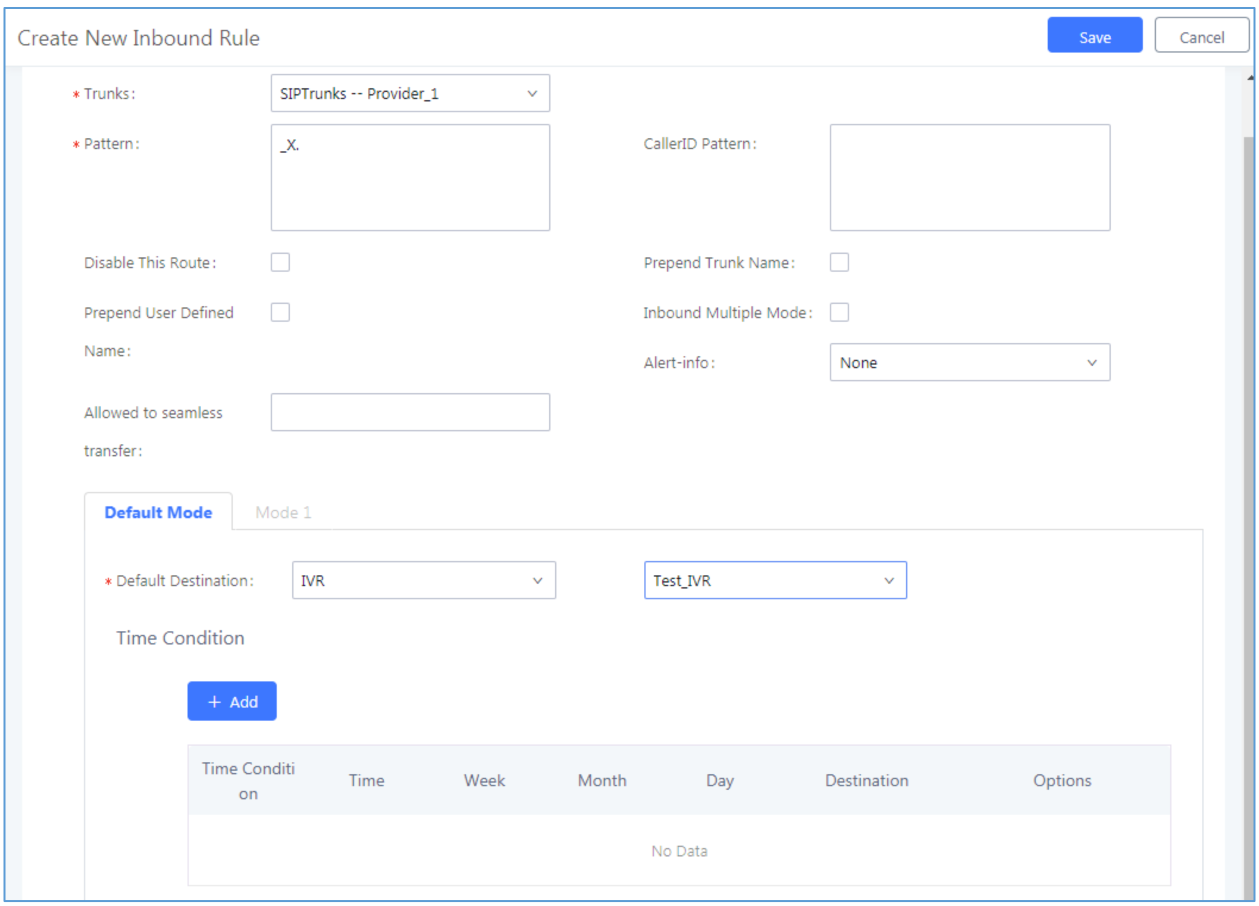

Figure 8: Register Inbound Route

For time condition based Inbound route, please refer to **TIME CONDITION**.

Create Outbound Routes

1. Go to **Web UI → Extension/Trunk → Outbound Routes.**
2. Click on "**Add**" button to create a new outbound route.
3. Enter the **Calling Rule Name**, **Pattern** and choose "Provider_1" in **Use Trunk.**

In below figure, we set "Pattern" to "_06XXXXXXXX" to allow only dialed numbers with leading 06 and 10-digit length to be accepted. Extensions with privilege "Internal" or higher can use this route/trunk.

Note: Users can add comments to a dial plan by typing "/*" and "*/" before and after each comment respectively, in our example we can set the pattern to: "**_06XXXXXXXX /*route1*/** "

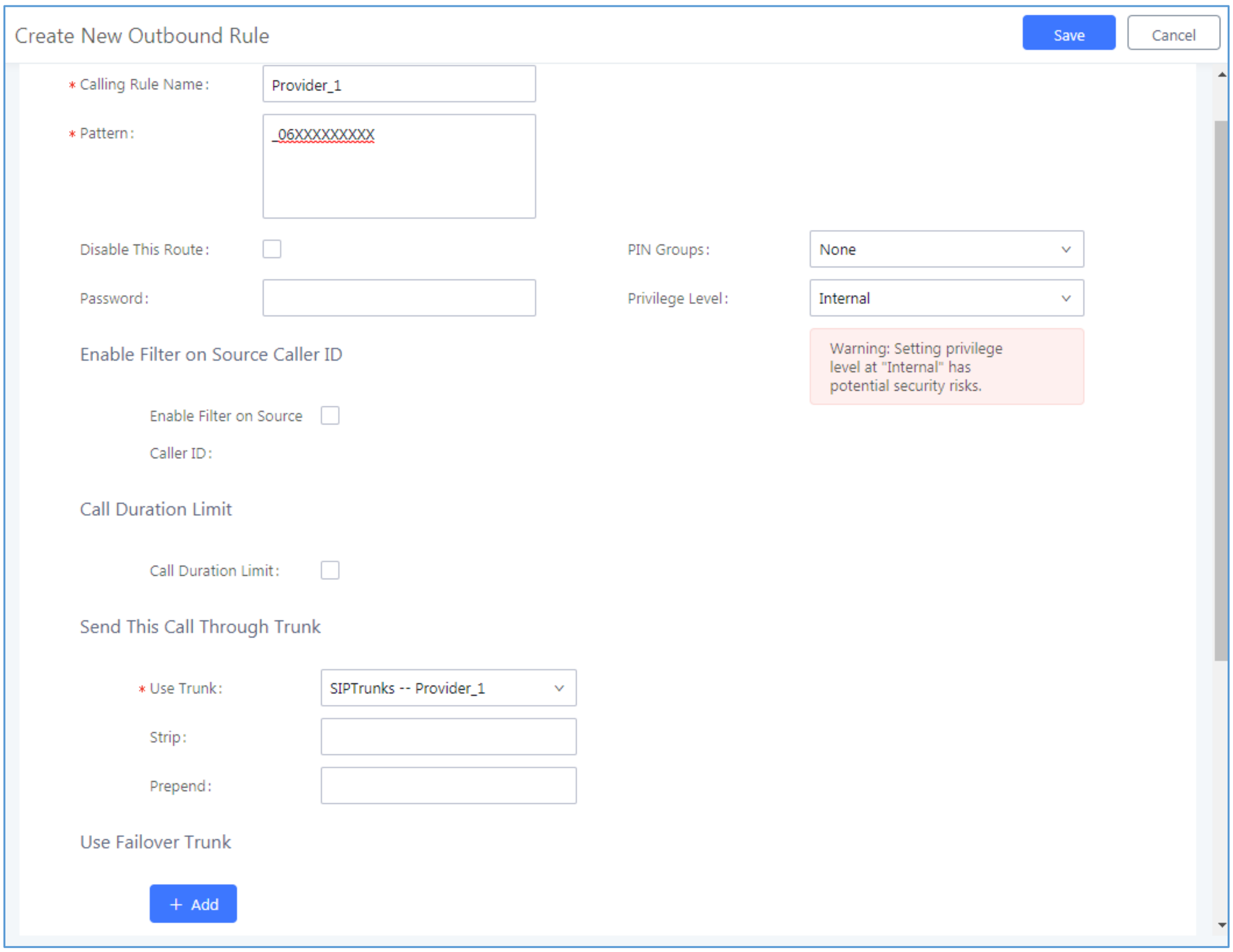

Figure 9: Outbound Route Example

Another outbound route should be set using "Provider_2" with following parameters:

- **Pattern**: "_99X." (to allow only dialed numbers with leading 99 to be authorized).
- **Privilege:** International for instance.
- **Strip:** "2" (to remove prefix 99 before calling out).

To dial international numbers such as 0016175669300, users need to have privilege "International" in used extension and dial 990016175669300.

Note: For time condition based Outbound Routes, please refer to **TIME CONDITION.**

PEER TRUNK

In this chapter, we will consider that company ABC has a branch office equipped with UCM6xxx, and they want to be able to communicate between the two offices.

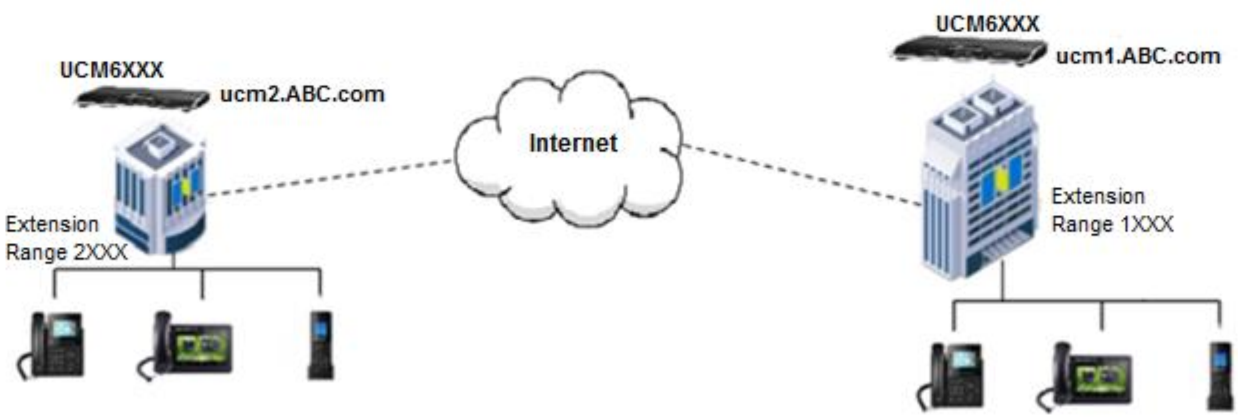

Figure 10: SIP Peer Trunk

Assuming following:

- Both UCM6xxx have public IP addresses;
- Or both UCM6xxx are on the same LAN/VPN using private or public IP addresses;
- Or both UCM6xxx can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).
- UCM6xxx in main office is using extensions range 1XXX, while UCM6xxx in branch office is using extensions range 2XXX.

Configuration

Following steps need to be done on both UCM6xxx in both locations:

1. Access UCM6xxx's web GUI → **Extension / Trunk** → **VoIP Trunks**.
2. Click on + Create New SIP Trunk, and enter following parameters:
   - Select "Peer SIP Trunk" for Type.
   - Enter a reference name. In this example: **toBranch**.
   - In **Host Name** field, enter the IP address/domain of the other UCM6xxx. In this case "ucm2.abd.com" is the domain name of the branch office.
3. Click Save and Apply Changes to apply configuration.

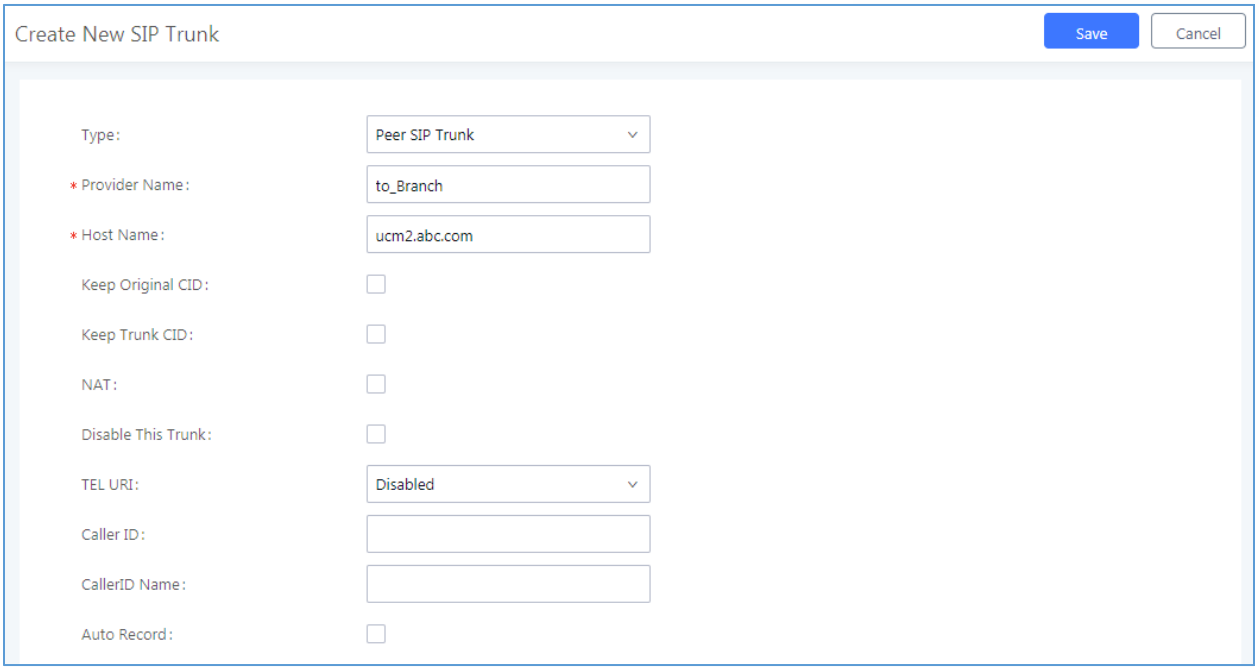

Figure 11: Create New Peer Trunk

The tables below describe basic and advanced parameters available for PEER trunks:

Table 2: Extension/Trunk → VoIP Trunks

Field	Description
Keep Original CID	Keep the CID from the inbound call when dialing out. This setting will override “Keep Trunk CID” option. Please make sure that the peer PBX at the other side supports to match user entry using “username” field from authentication line. • If enabled, user at extension 1000 will see 2000 as UserID.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension’s CID when the extension has CID configured. The default setting is “No”. • If enabled and **Keep Original CID** is disabled, user at extension 1000 will see the ID set on **Caller ID** field.
NAT	Turn on this setting when the PBX is using public IP and communicating with devices behind NAT. If there is one-way audio issue, usually it is related to NAT configuration or SIP/RTP port support on the firewall.
Disable This Trunk	If checked, the trunk will be disabled. Note: If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
TEL URI	If the trunk has an assigned PSTN telephone number, this field should be set to “User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled. • If **TEL URI** is set to **Disabled**, the **TO** header will be “To: <sip:1000@ucm1.abc.com>”. • If **TEL URI** is set to **User=Phone**, the **TO** header will be “To: <sip:1000@ ucm1.abc.com;user=phone>”. • If TEL URI is set to **Enabled,** the **TO** header will be “To: <tel:1000>”.

Caller ID	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored. When making outgoing calls, the following rules are used to determine which CallerID will be used if they exist: • The CallerID configured for the extension will be looked up first. • If no CallerID is configured for the extension, the CallerID configured for the trunk will be used. • If the above two are missing, the "Global Outbound CID" defined in Web GUI->PBX->Internal Options->General will be used.
Caller ID Name	Configure the name of the caller to be displayed when the extension has no CallerID Name configured. **Note:** This Option needs **Keep Trunk CID** Feature enabled • If the Caller ID Name is set on the trunk, the “From” header will include this name as shown below: From: "GSTest" <sip:2000@ ucm2.abc.com >;tag=f268
Auto Record	Enable automatic recording for the calls using this trunk (for SIP trunk only). The default setting is disabled. The recording files can be accessed under web GUI->CDR->Recording Files

Table 3: Extension/Trunk → VoIP Trunks → Advanced Settings

Fields	Description
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p. • The selected codecs will be sent on the SDP field from top to bottom to the other UCM in order to negotiate the codec to use during the call.
Send PPI Header	If enabled, the SIP INVITE message sent to the trunk will contain PPI (P-Preferred-Identity) header. The default setting is “No”. **Note:** “Send PPI Header” and “Send PAI Header” cannot be enabled at the same time. Only one of the two headers is allowed to be contained in the SIP INVITE message. If enabled the SIP header will contain the following: P-Preferred-Identity: sip:2000@ ucm2.abc.com This feature is needed by the provider “some providers use PPI others use PAI”
Send PAI Header	If enabled, the SIP INVITE message sent to the trunk will contain PAI (PAsserted-Identity) header. The default setting is “No”. Note: “Send PPI Header” and “Send PAI Header” cannot be enabled at the same time. Only one of the two headers is allowed to be contained in the SIP INVITE message. If enabled the SIP header will contain the following: P-Asserted-Identity: sip:2000@ucm2.abc.com This feature is needed by the provider “some providers use PPI others use PAI”
DID Mode	Configure where to get the destination ID of an incoming SIP call, from SIP Request-line or To-header. The default is set to "Request-line". • If set to Request-line, the UCM will extract the ID from the Request-Line of the incoming INVITE and set it on the “To header” for the outgoing one.

	• If set to To-Header, the UCM will extract the ID from the To-Header of the incoming INVITE and set it on the "To header" for the outgoing one.
DTMF Mode	Configure the default DTMF mode when sending DTMF on this trunk. • **Default:** The global setting of DTMF mode will be used. The global setting for DTMF Mode setting is under web UI->PBX->SIP Settings- >ToS. • **RFC2833:** Send DTMF using RFC2833. • **Info:** Send DTMF using SIP INFO message. • **Inband:** Send DTMF using inband audio. This requires 64-bit codec, i.e., PCMU and PCMA. • **Auto:** Send DTMF using RFC2833 if offered. Otherwise, inband will be used.
Enable Heartbeat Detection	If enabled the UCM will send regularly SIP OPTIONS to check if the device is online.
Heartbeat Frequency	When "Enable Heartbeat Detection" option is set to "Yes", configure the interval (in seconds) of the SIP OPTIONS message sent to the device to check if the device is still online. The default setting is 60 seconds.
Maximum Number of Call Lines	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limit.
Fax Mode	Select Fax mode. The default setting is "None". • None: Disable Fax. • Fax Detect: Fax signal from the user/trunk during the call can be detected, and the received Fax will be sent to the Email address configured for this extension, or to the and FXS extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under web UI->PBX->Internal Options->Fax/T.38.
SRTP	Enable SRTP for the VoIP trunk. The default setting is "No". • If enabled it will provide encryption, message authentication and integrity for the Audio stream.
Sync LDAP Enable	If enabled, the local UCM6xxx will automatically provide and update the local LDAP contacts to the remote UCM6xxx SIP peer trunk. In order to ensure successful synchronization, the remote UCM6xxx peer also needs to enable this option on the SIP peer trunk. The default setting is "No". • Password and Ports need to be the same on both UCM's in order to synchronize LDAP data. Please refer to UCM6xxx User Manual for more detail about this feature.
Enable CC	If enabled, the system will automatically alert the user when a called party is available, given that a previous call to that party failed for some reason. • For more information about this feature please refer to the Busy Camp On Guide.

After clicking Save the new VoIP trunk will be displayed under **"Web UI → Extension / Trunk > VoIP Trunks"** as shown below.

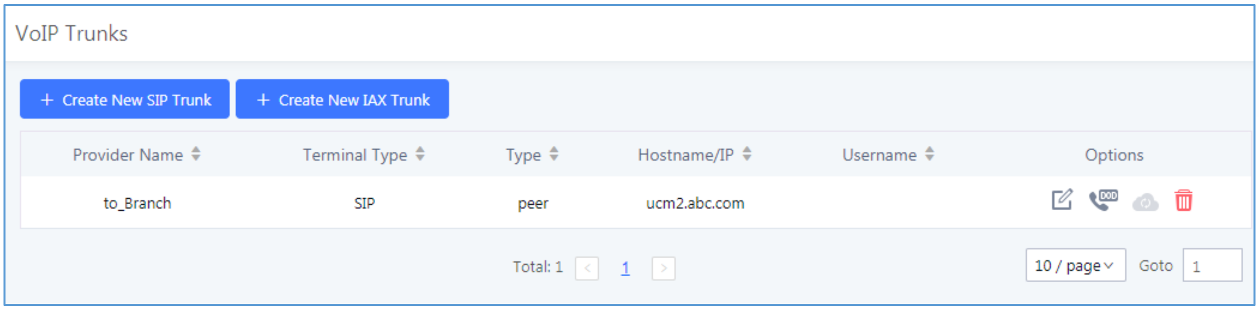

Figure 12: Edit Peer Trunk

Press and go to "Advanced Settings". Check "**Enable Heartbeat Detection**" option to allow UCM6xxx to monitor the status of each other sending regularly SIP OPTIONS to check if it's still online.

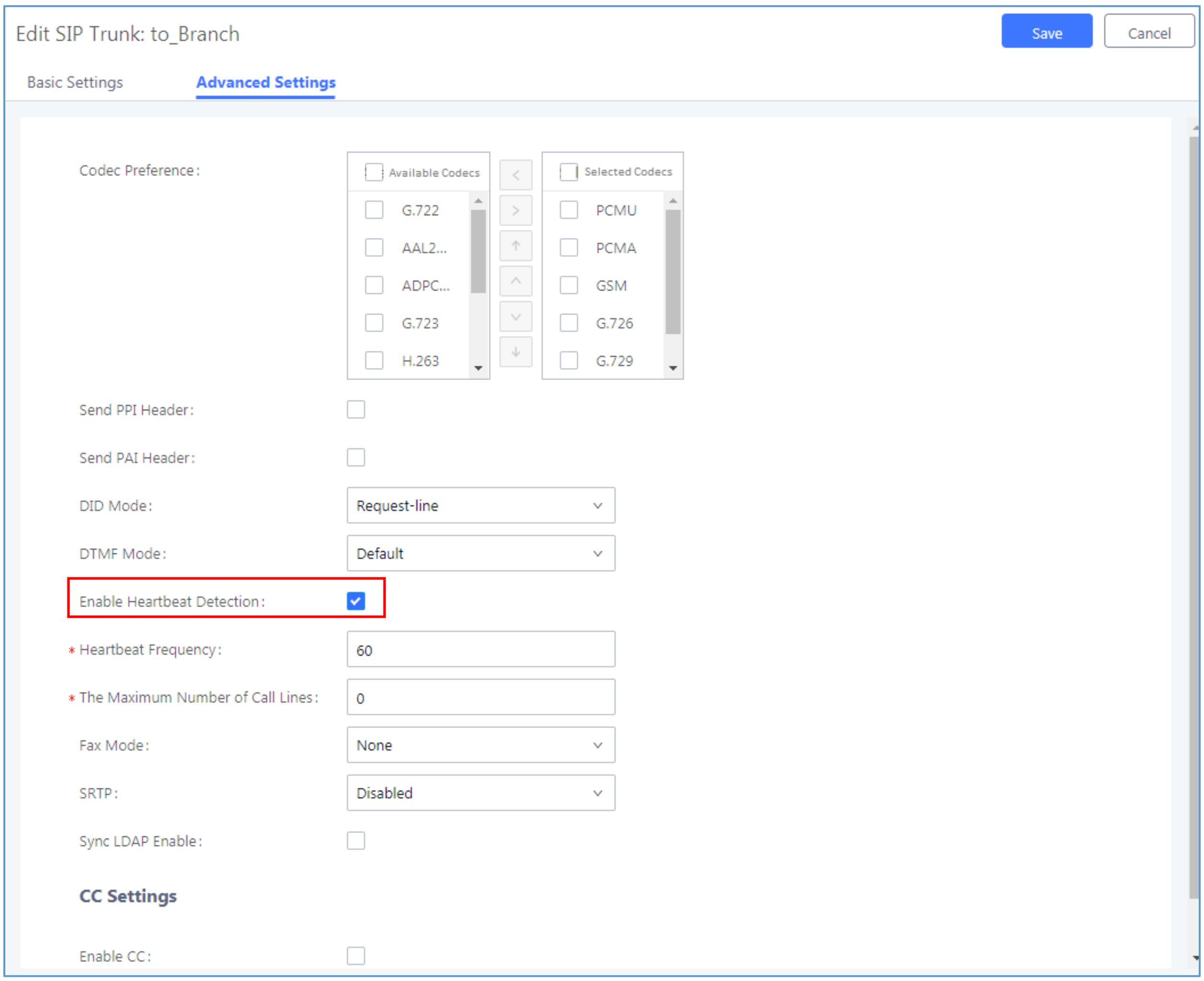

Figure 13: Enable Qualify

Click 

and 

to apply configuration.

Once the trunk has been created and Enable Qualify is set, users can view the status of the peered trunk by navigating to the **Status** page.

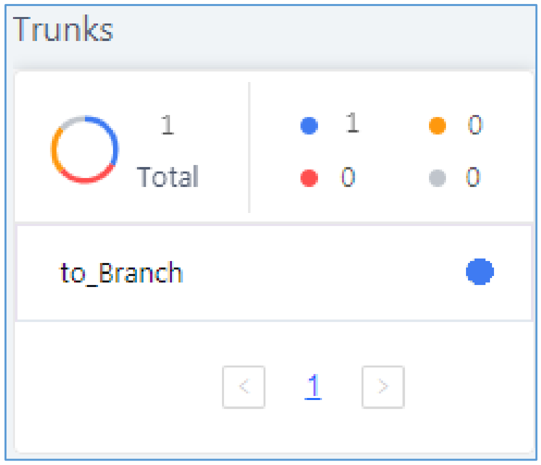

Figure 14: Peer Trunk Status

Outbound Routes Configuration

On the main office UCM6xxx web GUI under "**Web UI → Extension/Trunk → Outbound Routes**", click on + Add. This would allow the extensions on main office to reach extensions on Branch Office.

Configure using following parameters:

1. **Calling Rule Name**: This is for reference purposes so we choose to use "toBranch".
2. **Pattern**: The pattern used in this example is _2XXX since the Branch Office is using extension range 2XXX.
3. **Privilege Level**: Configured as "Internal". User can change it to another privilege depending on the use case.
4. **User Trunk**: Select the SIP Trunk toBranch.
5. Click on Save then Apply Changes.

In figure 15, the pattern "_2XXX" means that whenever a user on the main office call an extension that starts with 2 and have 4 digits, this trunk will be used for the call.

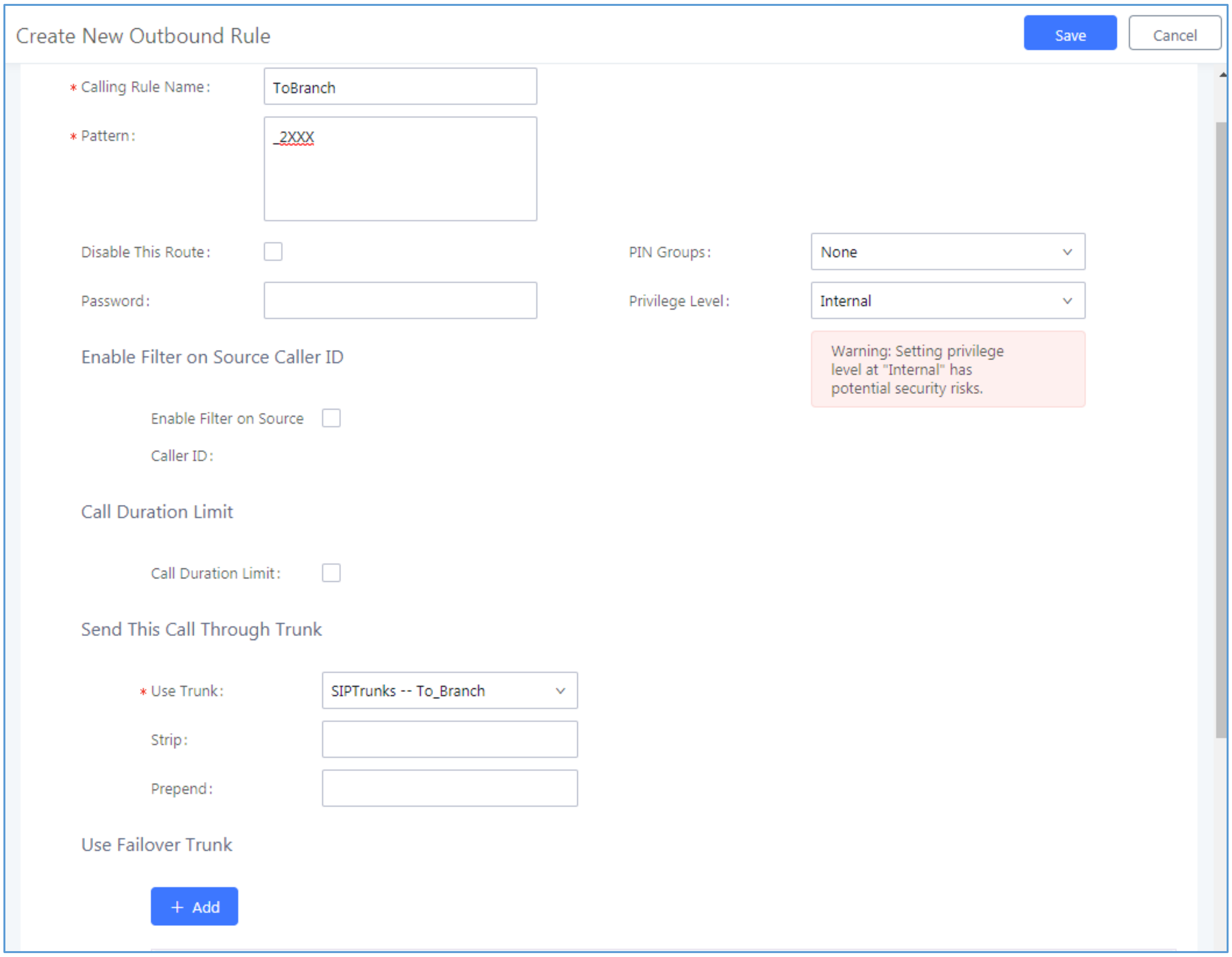

Figure 15: PEER Trunk - Create New Outbound Rule

Notes:

- These steps will also apply when configuring the outbound route from Branch Office UCM6xxx. The pattern used will be _1XXX since the extension range on the Main Office is 1XXX.
- Users also could create Time based Condition to use Outbound Routes on a specific time, please refer to TIME CONDITION for more details.

Inbound Routes Configuration

To create an inbound rule, on the UCM webGUI, under "**Web UI → Extension/trunk → Inbound Routes**" click on + Add button.

For this example, Main Office UCM6xxx inbound rule needs be configured so that when an extension on the Branch Office dials, it will be routed to the specified user. Configure using following parameters:

1. **Trunks:** Select the **toBranch** trunk.
2. **DID Pattern:** Enter in the pattern "_1XXX" | "_2XXX" as shown in below figure.
3. **DID Destination:** By DID.
4. Click on Save then Apply Changes.

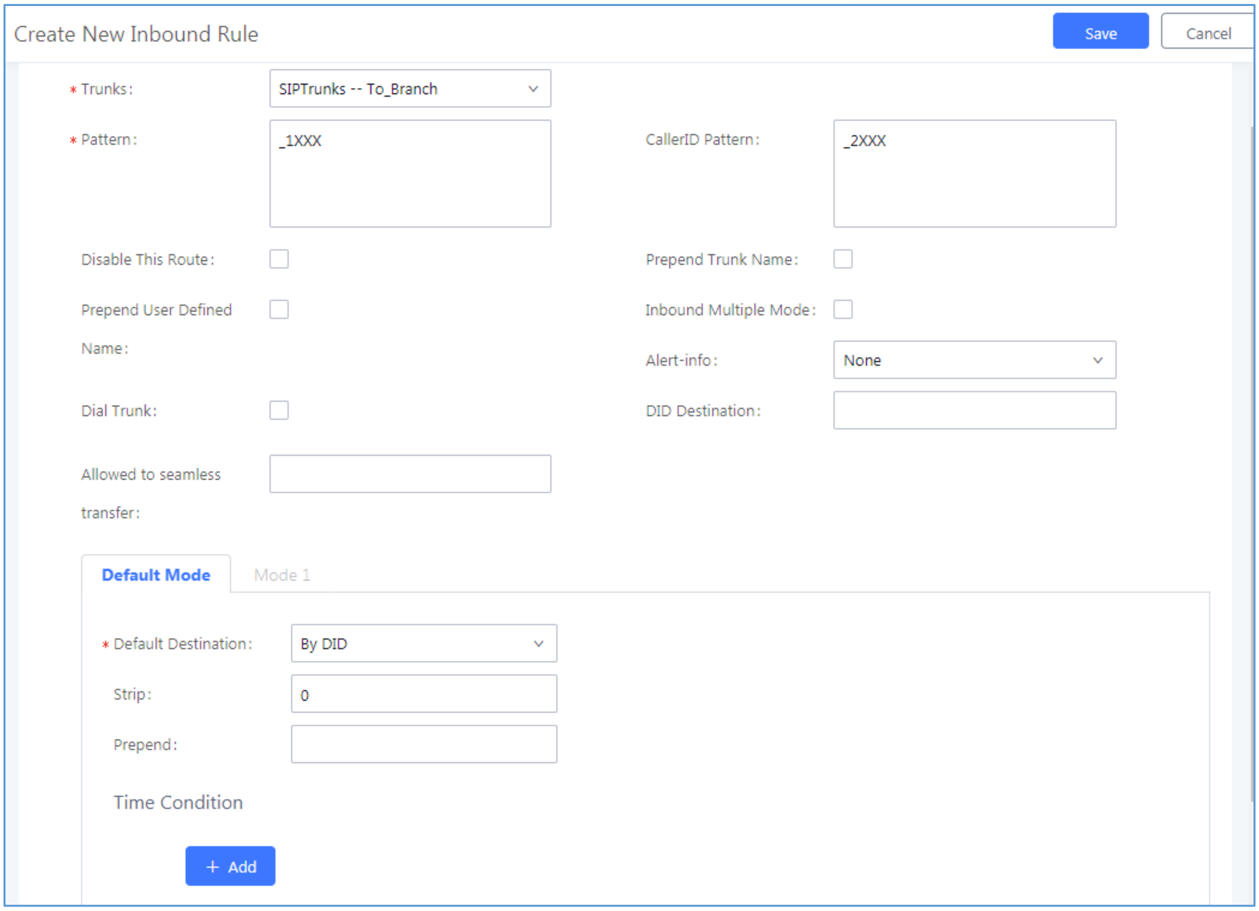

Figure 16: PEER Trunk - Create New Inbound Rule

With this inbound rule configured, if the Main Office UCM6xxx receives a call from any extension with a leading 2 and containing 4 digits to any extension with a leading 1 and contains 4 digits it will be redirected by DID to corresponding extension.

Notes:

- Those steps will also need to be applied to the Branch Office.
- Users also could create Time based Conditions to use Inbound Routes on specific times, please refer to TIME CONDITION for more details.

TIME CONDITION

Users can set time condition on Inbound/Outbound rules in order to use each route on a specific time.

Supposing that Time Conditions are already set, if more details are needed, please refer to the following guide: UCM6XXX Time Condition Guide

Users can setup time conditions for trunks by following below steps:

1. Access the **UCM6xxx Web UI → Extension / Trunk → Outbound Routes**.
2. Navigate to time condition section.
3. Select the time condition when to use this trunk. (Office Time in this example)
4. Click on "Add" to add the time condition.

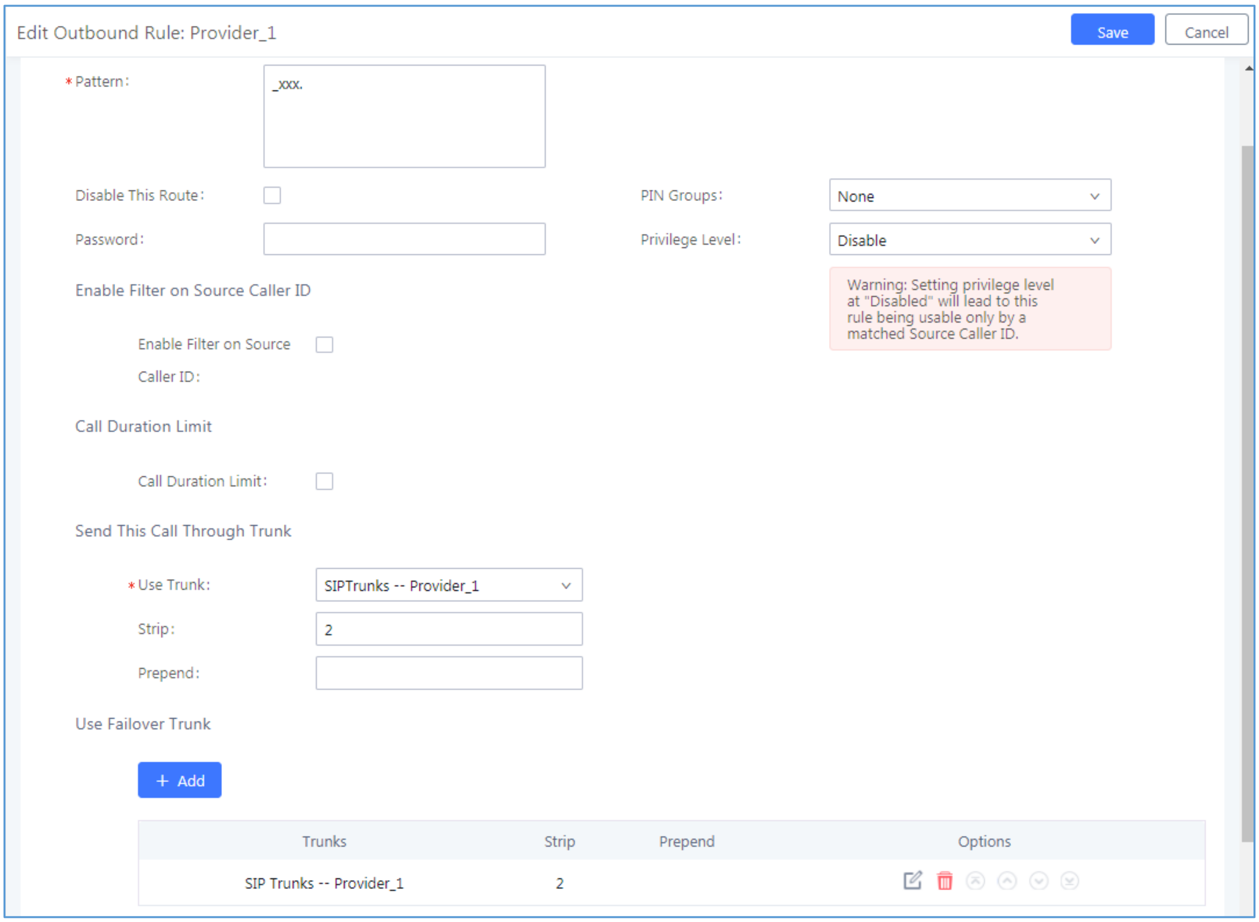

Figure 17: Outbound Time Office

The example above shows that on the Office Time, calls will be made using Provider_1 trunk.

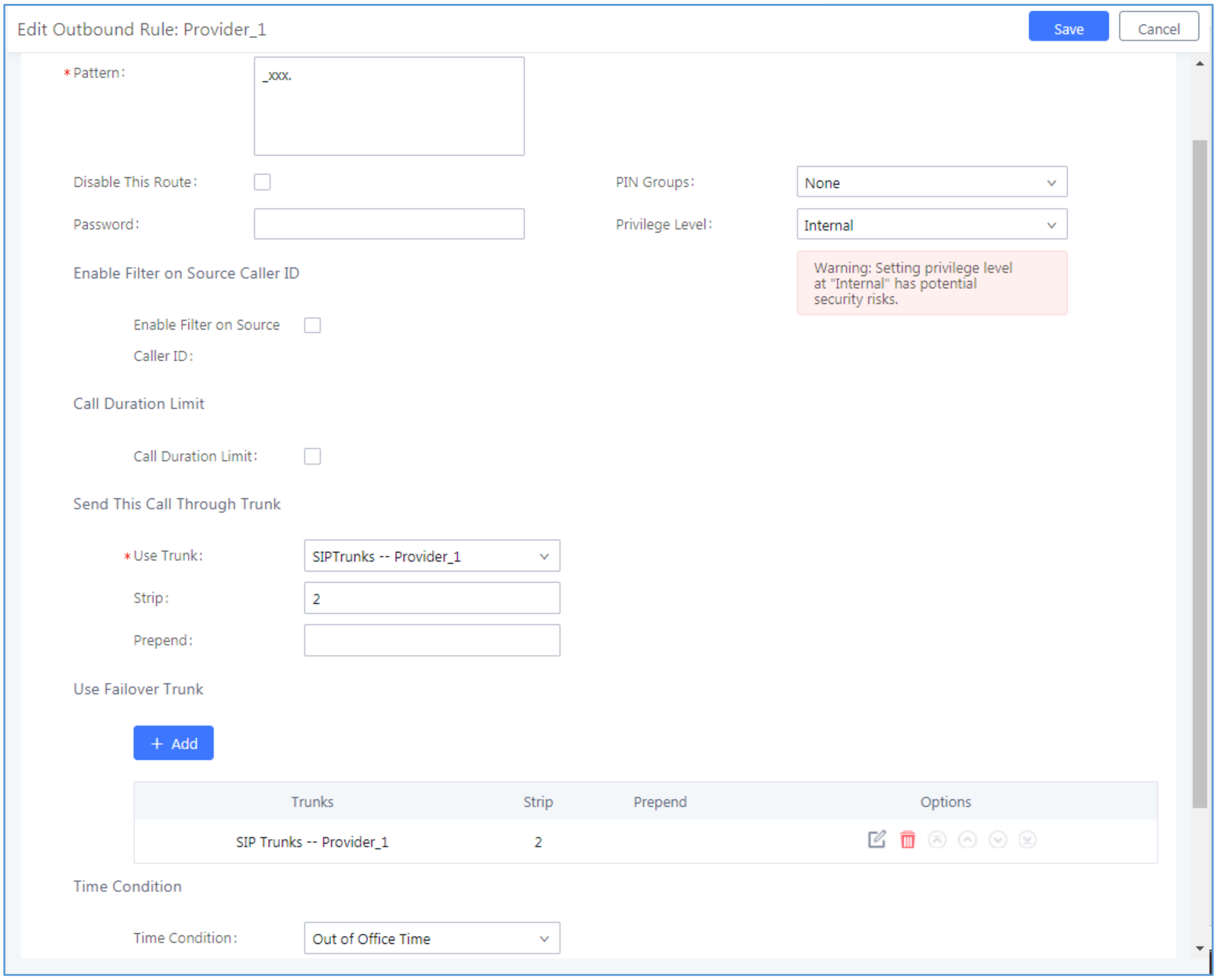

Figure 18: Out Of Office Time

The example above shows that on Out of Office Time, calls will be made using Provider_2 trunk.

FAILOVER TRUNK

Failover trunks can be used to make sure that a call goes through an alternate route, when the primary trunk is busy or down.

UCM6xxx will use Failover trunks in following scenarios:

- No response from first trunk after 32 seconds.
- If UCM6xxx receives 403/407/408/503/603 SIP responses from primary trunk.
- If primary trunk is disabled.
- If primary trunk is an analog trunk and it's busy or not connected.

Users can setup failover for peer and register trunks by following below steps:

1. Go to **PBX->Basic/Call Routes->Outbound Routes**.
2. Click on "**Click to add failover trunk**".
3. Select then the trunk to be set as failover as shown in the below figure.
4. Strip and prepend digits if needed.

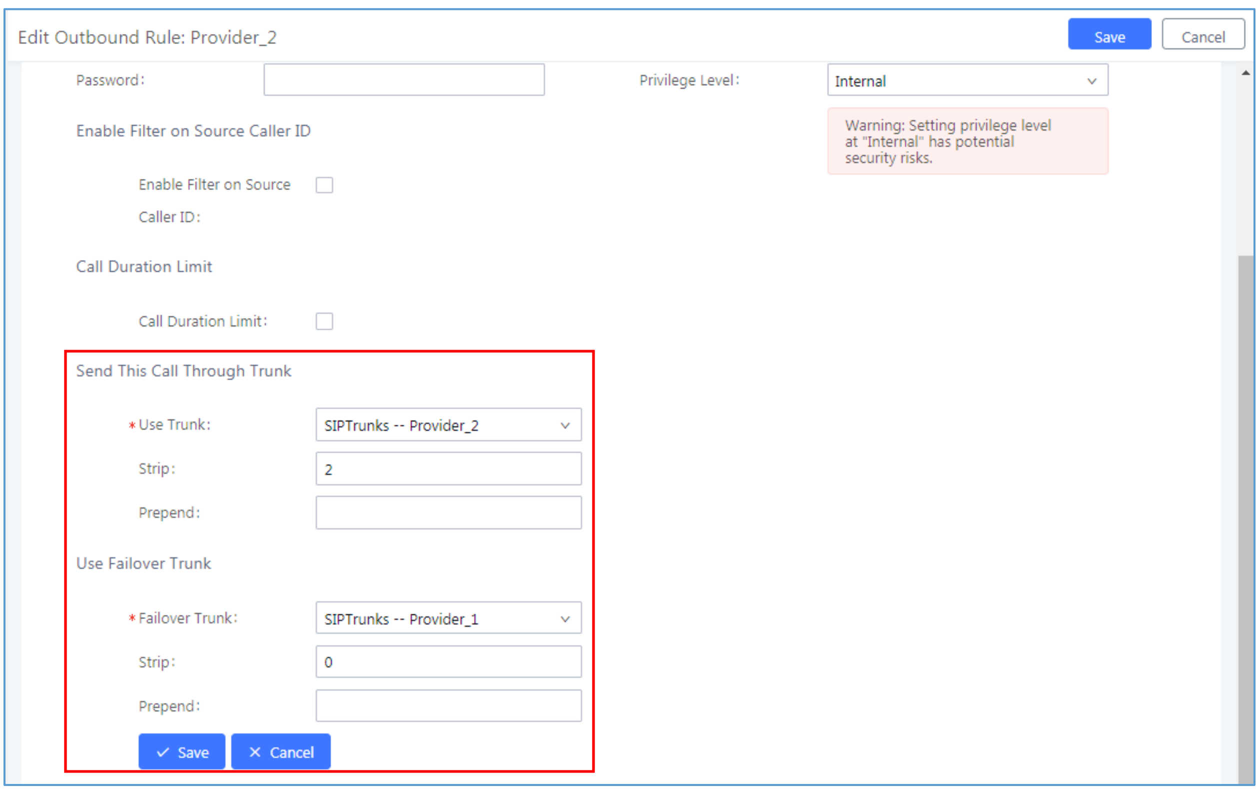

Figure 19: Adding Failover Trunk

Click on + Add to add another failover trunk.

Click on to arrange the order. If there are multiple trunks set as failovers.

Click on to edit, or to delete the Trunk from Failover list.