UCM6102/6104/6108/6116 Configuration (1/24/2017)

This document introduces manual configuration steps performed for interoperability testing between Twilio and Grandstream UCM6102/6104/6108/6116. Configuration parameters not explained in this document were kept at factory default settings. Screen shots were taken with the UCM6102 running version 1.6A.

1. Connect UCM6102/6104/6108/6116 to the network and power it up.

2. From the LCD and navigation keys, press "OK" button and "DOWN" button to view the IP address on the UCM6102/6104/6108/6116.

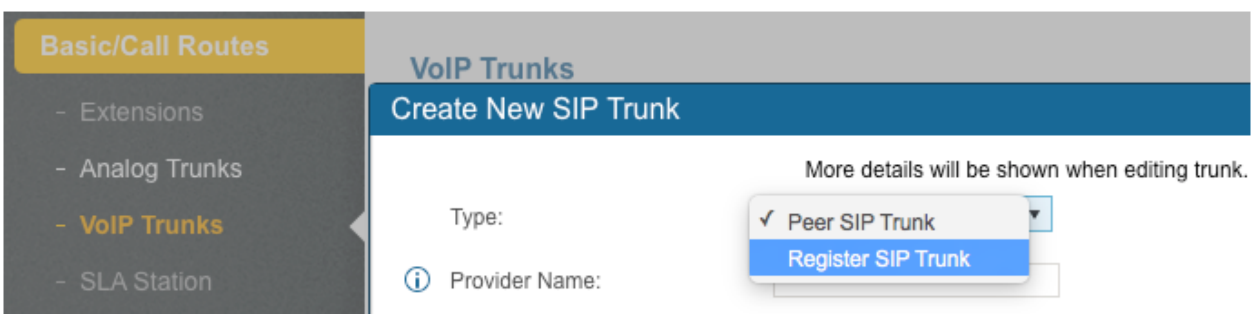
   Note: If there is a separate WAN and LAN port then for initial setup be sure to connect the WAN port to your to network to successfully receive an IP address via DHCP.

3. From a PC connected in the same LAN, log into the device's Web UI by typing the IP address of the UCM6102/6104/6108/6116 on the PC's web browser.

4. Default login name and password for the administrator is **admin/admin** for Grandstream devices.

5. Navigate to Network Settings → Basic Settings → Method – select "Switch" mode. This ensures both the WAN and LAN RJ45 port behave as a layer-2 switch which is preferred if your UCM is already behind a firewall.

6. Select tab "PBX"->"Basic/Call Routes"->"VoIP Trunks" in the web UI and click on "Create New SIP Trunk" and select "**Register SIP Trunk**".

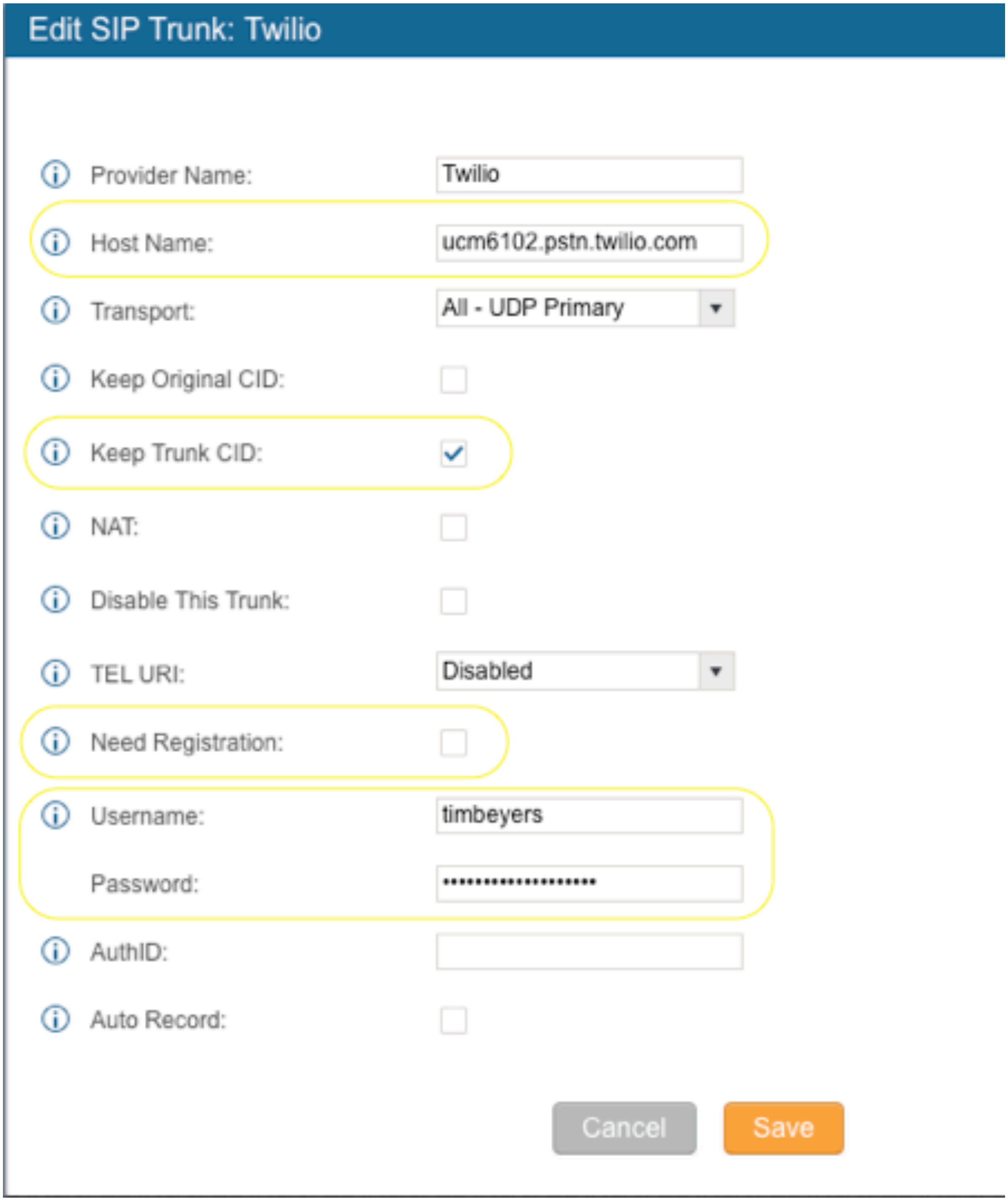


In the following dialog, configure the elements circled in yellow which include - **Host Name, Username, Password**. You can also add a Provider name like Twilio. Please make sure that **Keep Trunk CID** is enabled in order to match authentication credential on Twilio.

Be sure to uncheck "**Need Registration**".

Careful note:

AuthID, when non-null, takes precedence over username for credential authentication. So that means the username field is used for Authentication when no AuthID is present, but that when present AuthID is used instead of username. The following example just leaves AuthID blank.

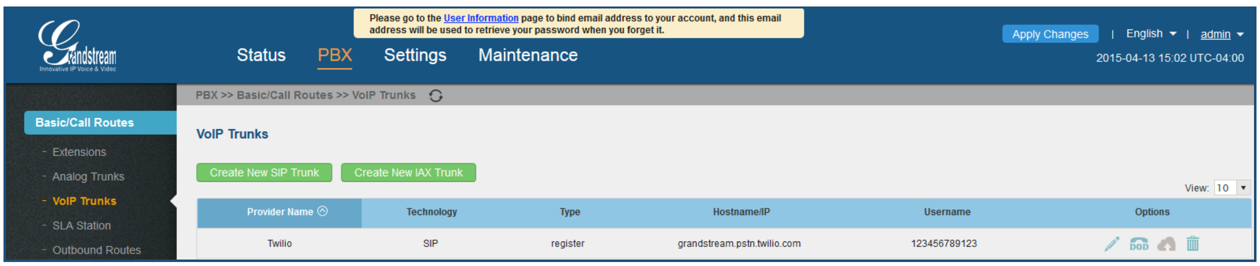


Using the "Register SIP Trunk" with username/password authentication is the preferred way to secure your termination calls. It is recommended that you also whitelist the public IP address in the Twilio console.

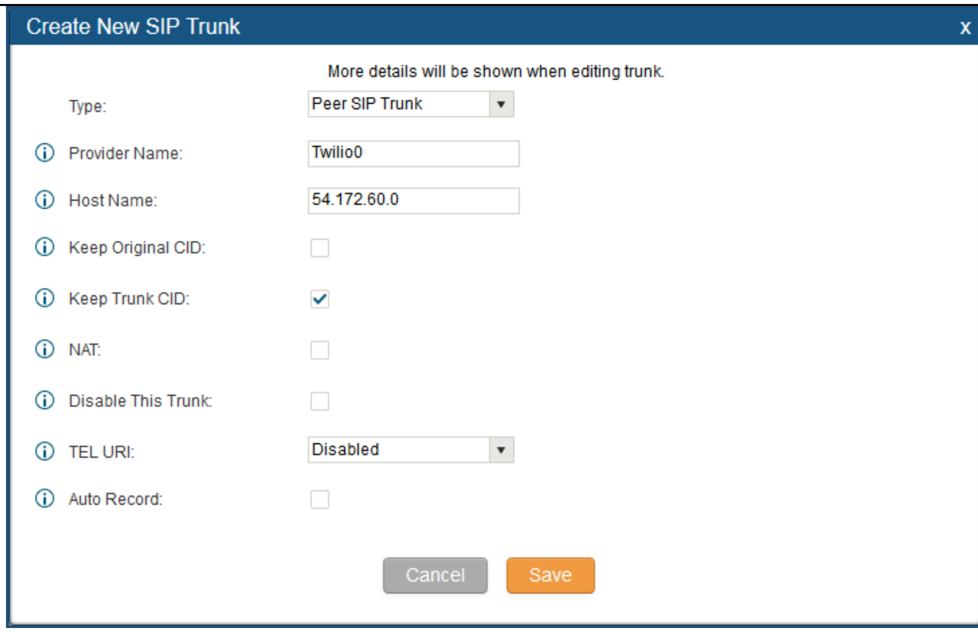
While it is not recommended for security reasons, it is also possible to create a PEER SIP TRUNK for the termination direction. This setting does not allow you to provide credentials so instead you would whitelist the public IP on the Twilio console. This is not considered as secure as the “Register SIP Trunk”.

Click on "Save" to save the configuration.

7. Click on "Apply Changes" on the upper right of the web page to apply the configuration. A VoIP trunk is successfully created.

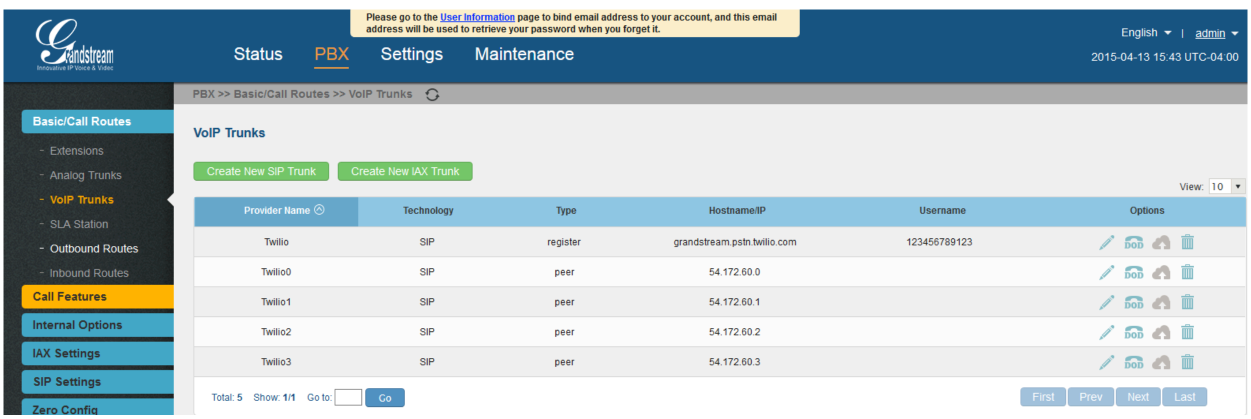


8. Since Twilio domain name will have 4 IP address for incoming call (based on DNS query). UCM 6102/6104/6108/6116 will only use the first DNS query address. Therefore, in order to receive an inbound call, we need to create 4 SIP Peer Trunk to support Twilio server.
   Please note:
   According to the DNS trace file, in this example, Twilio domain has 54.172.60.0, 54.172.60.1, 54.172.60.2, 54.172.60.3 IP address that will communicate with UCM 6102/6104/6108/6116. According to the Twilio documentation, 54.172.60.0/23 network will be used for signaling. Please adjust SIP Peer Trunk accordingly.
9. Select tab "PBX"->"Basic/Call Routes"->"VoIP Trunks", click on "Create New SIP Trunk". In the following dialog, configure Type, Provider Name and Host Name. Please make sure Keep Trunk CID is enabled.

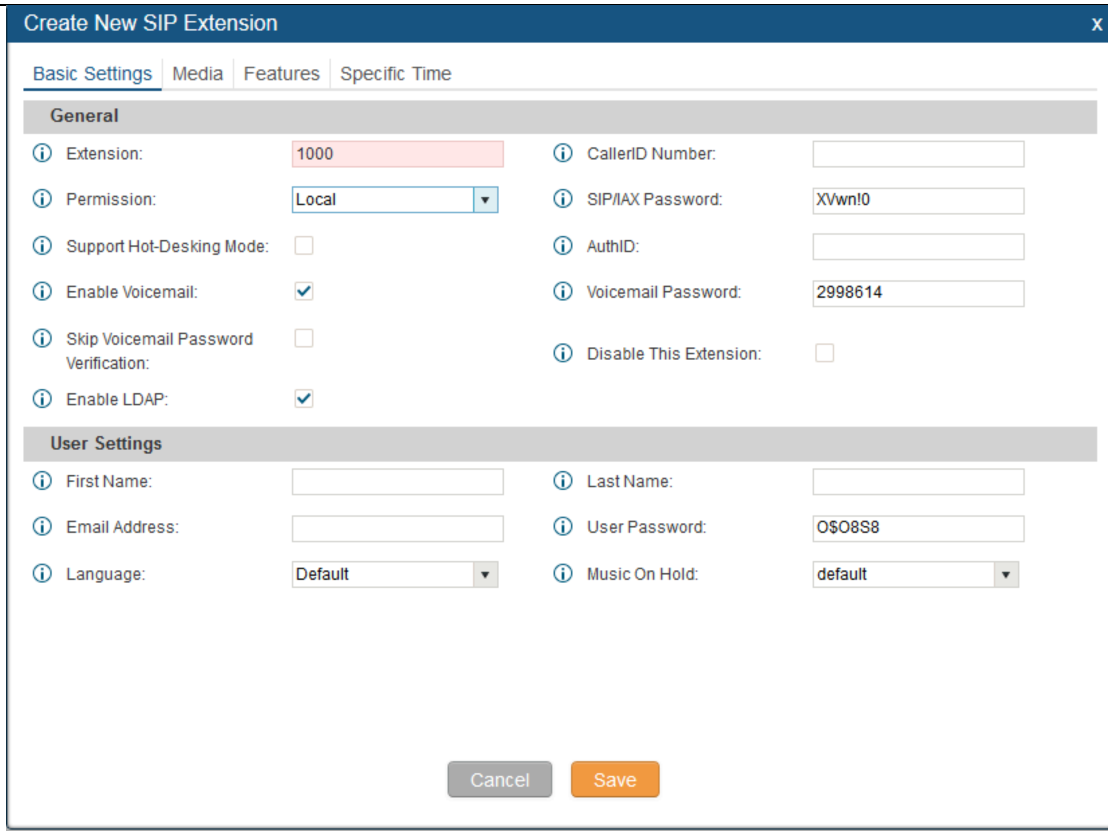


Click on "Save" to save the configuration.

10. Repeat Step 9 to create another 3 SIP Peer Trunks. One SIP Register Trunk should be created for outbound call and four SIP Peer Trunks should be created to support inbound call.



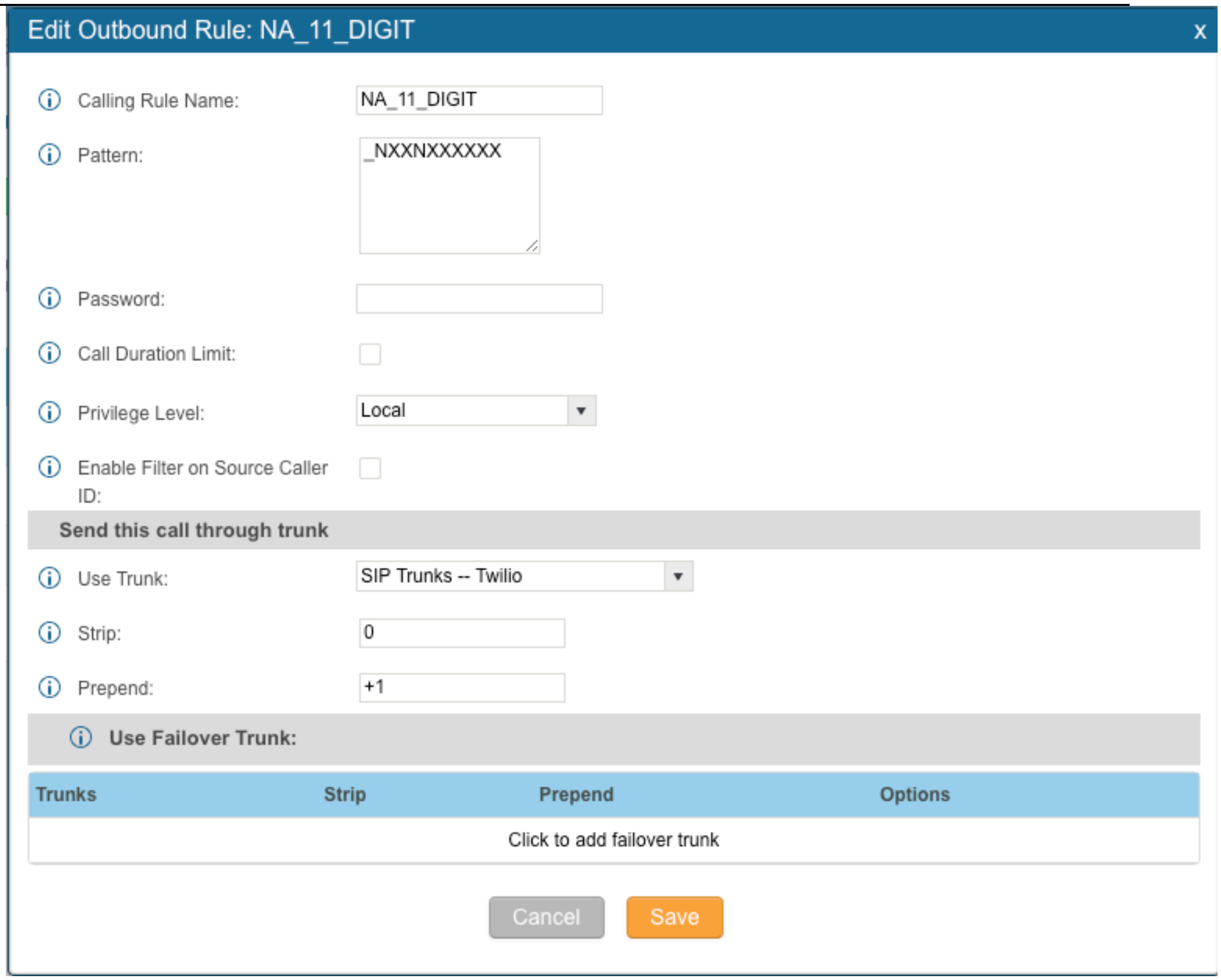
11. Select tab "PBX"->"Basic/Call Routes"->"Extensions" in the web UI, click on "Create New SIP Extension". In the following dialog, configure Permission (set to "Local" in this example). Enter First Name, Last Name and other information as needed.



Click on "Save" to save the configuration.

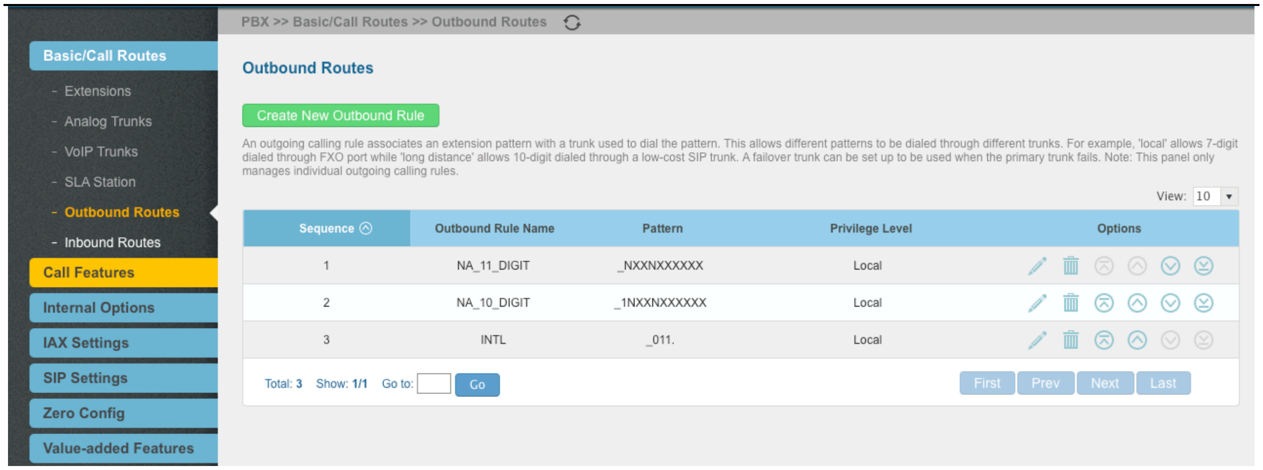
12. Repeat this step to create more extensions as needed. These extensions can be used to make outbound calls or answer inbound calls. Once done, click on "Apply Changes" on the upper right of the web page to apply the configuration. The created extensions in step 11 will be successfully applied on the PBX.

13. Select tab "PBX"->"Basic/Call Routes"->"Outbound Routes" in the web UI, click on "Create New Outbound Rule". In the following dialog, configure Caller Rule Name, Pattern, select Privilege Level, Use Trunk and Prepend value.
The example route is for calls to & from US/CAN following the NPA-NXX format.



Please note: Twilio only supports E.164-formated. Therefore, please prepend "+1" to the calls that have been placed on current trunk. SIP Register Trunk is used to support outbound calls, therefore, outbound rules are not required for rest of the 4 SIP Peer Trunks since they are used for taking inbound calls.

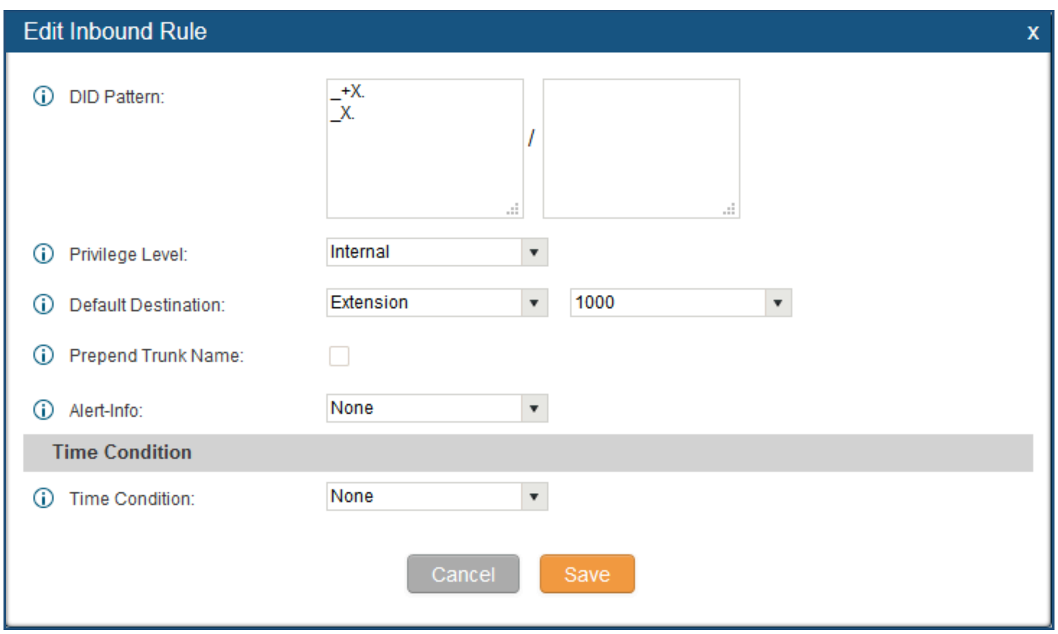
Please add a total of 3 outbound routes as seen summarized in next screen shot. This covers the two common domestic formats and one international format. For international calls you may need to enable countries in [Geo-permissions]() of the Twilio Console.



Please note:

If "Privilege Level" is set to "Local" in outbound rule, the extension that is used to make outbound call needs to have "Permission Level" set to "Local" or higher. The extension settings is under web UI->"PBX"->"Basic/Call Routes"->"Extension" page, as described in step 11.

14. Select tab "PBX"->"Basic/Call Routes"->"Inbound Routes" in the web UI, click on "Create New Inbound Rule" to create Inbound Rule for each Trunks. In the following dialog, configure Trunks, DID Pattern and Default Destination.



Click on "Save" to save the configuration.

Please note:

In this example, SIP Register Trunk is used only to support outbound calls, we don't need to create an inbound rule for it. Total of 4 inbound rules are required for each of the SIP Peer Trunks, as they are supporting Twilio incoming calls.

15. Repeat Step 14 to create rest of the inbound rules. A total of 4 inbound rules should be created to support incoming calls.
16. Click on "Apply Changes" on the upper right of the web page to apply the configuration. The created inbound rule in step 14 will be successfully applied to the PBX. In this example, the inbound call to the UCM6102/6104/6108/6116 will be routed Extension 1000.

The following table shows the configuration information for SIP trunk.

Create New SIP Trunk	
Type	Select the VoIP trunk type. • Peer SIP Trunk • Register SIP Trunk
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
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".
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.
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.
From Domain	Configure the actual domain name where the extension comes from. This can be used to override the From Header. For example, "trunk.UCM6100.provider.com" is the From Domain in From Header: sip:1234567@trunk.UCM6100.provider.com.
From User	Configure the actual user name of the extension. This can be used to override the From Header. There are cases where there is a single ID for registration (single trunk) with multiple DIDs. For example, "1234567" is the From User in From Header: sip:1234567@trunk.UCM6100.provider.com.

The following table shows the configuration information for extensions.

General	
Extension	The extension number associated with the user.
CallerID Number	Configure the CallerID Number that would be applied for outbound calls from this user.

	Note: The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
Permission	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". **Note:** Users need to have the same level as or higher level than an outbound rule's privilege in order to make outbound calls using this rule.
SIP/IAX Password	Configure the password for the user. A random secure password will be automatically generated. It is recommended to use this password for security purpose.
Enable Voicemail	Enable voicemail for the user. The default setting is "Yes".
Voicemail Password	Configure voicemail password (digits only) for the user to access the voicemail box. A random numeric password is automatically generated. It is recommended to use the random generated password for security purpose.
Call Forward Unconditional	Configure the Call Forward Unconditional target number. If not configured, the Call Forward Unconditional feature is deactivated. The default setting is deactivated.
CFU Time Condition	Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time condition are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”. **Note:** • “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. • Office Time and Holiday could be configured on page **Settings->Time Settings->Office Time/Holiday** page.
Call Forward No Answer	Configure the Call Forward No Answer target number. If not configured, the Call Forward No Answer feature is deactivated. The default setting is deactivated.
CFN Time Condition	Select time condition for Call Forward No Answer. The available time condition are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”. **Note:** • “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. • Office Time and Holiday could be configured on page **Settings->Time Settings->Office Time/Holiday** page.
Call Forward Busy	Configure the Call Forward Busy target number. If not configured, the Call Forward Busy feature is deactivated. The default setting is deactivated.

CFB Time Condition	Select time condition for Call Forward Busy. The available time condition are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”. **Note:** • “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. • Office Time and Holiday could be configured on page **Settings->Time Settings->Office Time/Holiday** page.
Ring Timeout	Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the UCM6100, which can be configured in the global ring timeout setting under web GUI->**Internal Options**->**IVR Prompt**: General Preference. The valid range is between 5 seconds and 600 seconds. **Note:** If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.
Auto Record	Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under web GUI->**CDR**->**Recording Files**.
Skip Voicemail Password Verification	When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default this option is disabled.
Support Hot-Desking Mode	If enabled, SIP Password will accept only alphabet characters and digits; AuthID will be changed to the same as Extension.
Disable This Extension	If selected, this extension will be disabled on the UCM6100. Note: The disabled extension still exists on the PBX but can’t be used on the end device.
Music On Hold	Select which Music On Hold class to suggest to extension when putting it on hold.
User Settings	
First Name	Configure the first name of the user. The first name can contain characters, letters, digits and _.
Last Name	Configure the last name of the user. The last name can contain characters, letters, digits and _.
Email Address	Fill in the Email address for the user. Voicemail will be sent to this Email address.
Language	Select the voice prompt language to be used for this extension. The default setting is "Default" which is the selected voice prompt language under web GUI->**PBX**->**Internal Options**->**Language**. The dropdown list shows all the current available voice prompt languages on the UCM6100. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under web GUI->**PBX->Internal Options**->**Language**.
SIP Settings	

NAT	Use NAT when the UCM6100 is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is one-way audio issue, usually it's related to NAT configuration or Firewall's support of SIP and RTP ports. The default setting is enabled.
Can Reinvite	By default, the UCM6100 will route the media steams from SIP endpoints through itself. If enabled, the PBX will attempt to negotiate with the endpoints to route the media stream directly. It is not always possible for the UCM6100 to negotiate endpoint-to-endpoint media routing. The default setting is "No".
DTMF Mode	Select DTMF mode for the user to send DTMF. The default setting is "RFC2833". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit PCMU and PCMA are required. When "Auto" is selected, RFC2833 will be used if offered, otherwise "Inband" will be used.
Insecure	• Port: Allow peers matching by IP address without matching port number. • Very: Allow peers matching by IP address without matching port number. Also, authentication of incoming INVITE messages is not required. • No: Normal IP-based peers matching and authentication of incoming INVITE. The default setting is "Port".
Enable Keep-alive	If enabled, empty SDP packet will be sent to the SIP server periodically to keep the NAT port open. The default setting is "Yes".
Keep-alive Frequency	Configure the Keep-alive interval (in seconds) to check if the host is up. The default setting is 60 seconds.
Auth ID	Configure the authentication ID for the user. If not configured, the extension number will be used for authentication.
TEL URI	If the end device/phone 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.
Other Settings	
SRTP	Enable SRTP for the call. The default setting is disabled.
Fax Detection	Enable to detect Fax signal from the user/trunk during the call and send the received Fax to the Email address configured for this extension. If no Email address can be found for the user, send the received Fax to the default Email address in Fax setting page under web GUI->PBX->Internal Options->Fax/T.38. **Note:** If enabled, Fax Pass-through cannot be used.
Skip Trunk Auth	If enabled, users will not need enter the "PIN Set" required by the outbound rule to make outbound calls. The default setting is "No".
Dial Trunk Password	Configure personal password when making outbound calls via trunk.
Strategy	This option controls how the extension can be used on devices within different types of network. • Allow All Device in any network can register this extension. • Local Subnet Only

	Only the user in specific subnet can register this extension. Up to three subnet addresses can be specified. • A Specific IP Address Only the device on the specific IP address can register this extension. The default setting is "Allow All".
Codec Preference	Select audio and video codec for the extension. 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 and H.263p.
Specific Time	
Time Condition	Click on add Time Condition to configure specific time for this extension.

The following table shows the configuration information for outbound route.

Calling Rule Name	Configure the name of the calling rule (e.g., local, long_distance, and etc). Letters, digits, _ and - are allowed.
Pattern	• All patterns are prefixed with the "_". • Special characters: X: Any Digit from 0-9. Z: Any Digit from 1-9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately. Example: [12345-9] - Any digit from 1 to 9.
Password	Configure the password for users to use this rule when making outbound calls.
Call Duration Limit	Enable to configure the maximum duration for the call using this outbound route.
Maximum Call Duration	Configure the maximum duration of the call (in seconds). The default setting is 0, which means no limite.
Warning Time	Configure the warning time for the call using this outbound route. If set to x seconds, the warning tone will be played to the caller when x seconds are left to end the call.
Warning Repeat Interval	Configure the warning repeat interval for the call using this outbound route. If set to x seconds, the warning tone will be played every x seconds after the first warning.
Privilege Level	Select privilege level for the outbound rule. • Internal: The lowest level required. All users can use this rule. • Local: Users with Local, National, or International level are allowed to use this rule. • National: Users with National or International level are allowed to use this rule. • International: The highest level required. Only users with international level can use this rule. The default setting is "Disable". Please be aware of the potential security risks when using "Internal" level, which means all users can use this outbound rule to dial out from the trunk.

Enable Filter on Source Caller ID	When enabled, users could specify extensions allowed to use this outbound route. "Privilege Level" is automatically disabled if using "Enable Filter on Source Caller ID". The following two methods can be used at the same time to define the extensions as the source caller ID. 1. Select available extensions/extension groups from the left to the right. This allows users to specify arbitrary single extensions available in the PBX. 2. Custom Dynamic Route: define the pattern for the source caller ID. This allows users to define extension range instead of selecting them one by one. • All patterns are prefixed with the "_". • Special characters: X: Any Digit from 0-9. Z: Any Digit from 1-9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately. Example: [12345-9] - Any digit from 1 to 9.
Send This Call Through Trunk	
Use Trunk	Select the trunk for this outbound rule.
Strip	Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk. Example: The users will dial 9 as the first digit of a long distance calls. However, 9 should not be sent out via analog lines and the PSTN line. In this case, 1 digit should be stripped before the call is placed.
Prepend	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.

The following table shows the configuration information for inbound route.

Trunks	Select the trunk to configure the inbound rule.
DID Pattern	• All patterns are prefixed with the "_". • Special characters: X: Any Digit from 0-9. Z: Any Digit from 1-9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately. Example: [12345-9] - Any digit from 1 to 9. • The pattern can be composed of two parts, divided by a '/' character. The first part is used to specify the dialed number the second part is used to specify the caller ID and it is optional, if set it means only the extension with the specific caller ID is allowed to call in or call out. For example, patter '_2XXX/1234' means the only extension with the caller ID '1234' is allowed to use this rule.

Privilege Level	Configure the privilege level for this inbound route.
Default Destination	Select the default destination for the inbound call. • Extension • Voicemail • Conference Room • Queue • Ring Group • Paging/Intercom • Voicemail Group • Fax • DISA • IVR • Dial By Name • External Number • By DID When "By DID" is used, the UCM6100 will look for the destination based on the number dialed, which could be local extensions, conference, call queue, ring group, paging/intercom group, IVR, voicemail groups and Fax extension as configured in "DID destination". If the dialed number matches the DID pattern, the call will be allowed to go through.
Strip	Configure the number of digits to be stripped from the beginning of the DID. This option shows up only when "By DID" is selected.
Prepend Trunk Name	This option shows up only when “By DID” is selected. If enabled, the trunk name will be prepended to the display name.
Dial Trunk	This option shows up only when "By DID" is selected. If enabled, the external users dialing in to the trunk via this inbound route can dial outbound call using the UCM6100’s trunk.
DID Destination	This option shows up only when "By DID" is selected. This controls the destination that can be reached by the external caller via the inbound route. The DID destination are: • Extension • Conference • Call Queue • Ring Group • Paging/Intercom Groups • IVR • Voicemail Groups • Fax Extension • Dial By Name
Alert-Info	When present in an INVITE request, the Alert-Info header field specifies an alternative ring tone to the UAS.

Note:

For more configuration information, please refer to UCM6100 user manual on our web site:

http://www.grandstream.com/products/ucm_series/ucm61xx/documents/ucm61xx_usermanual_english.pdf