

## Outbound rule configuration for FXO port between UCM6xxx and GXW410x (Case of UCM6xxx and GXW4104)

It goes to the UCM6xxx and the following changes are made: create a VoIP Trunk in peer mode for each FXO port of the GXW4104 as follows:

Web UI e UCM6XXX → Extension/Trunk → VoIP Trunk → Create New SIP Trunk → Peer SIP Trunk

**Provider Name:** Name of the SIP trunk

**Host Name:** we put the IP of the GXW4104 followed by port 5060 for port FXO1 and for the following ports increase them by 2 (example: Port FXO2 would be port 5062 and so on).

We press the SAVE button and then press Apply Changes

UCM6510

Apply Changes Setup Wizard English admin

Menus

- System Status
- Extension / Trunk
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  - Digital Trunks
  - Data Trunk
  - VoIP Trunks
  - SLA Station
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- System Settings

Edit SIP Trunk: GXW4104\_01

Save Cancel

Basic Settings Advanced Settings

\* Provider Name: GXW4104\_01 \* Host Name: 192.168.78.19:5060

Auto Record:  Keep Original CID:

Keep Trunk CID:  NAT:

Disable This Trunk:  TEL URI: Disabled

Caller ID:  CallerID Name:

From Domain:

Transport: UDP Direct Callback:

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Web UI e UCM6XXX → Extension/Trunk → VoIP Trunk → Outbound Route → +Add

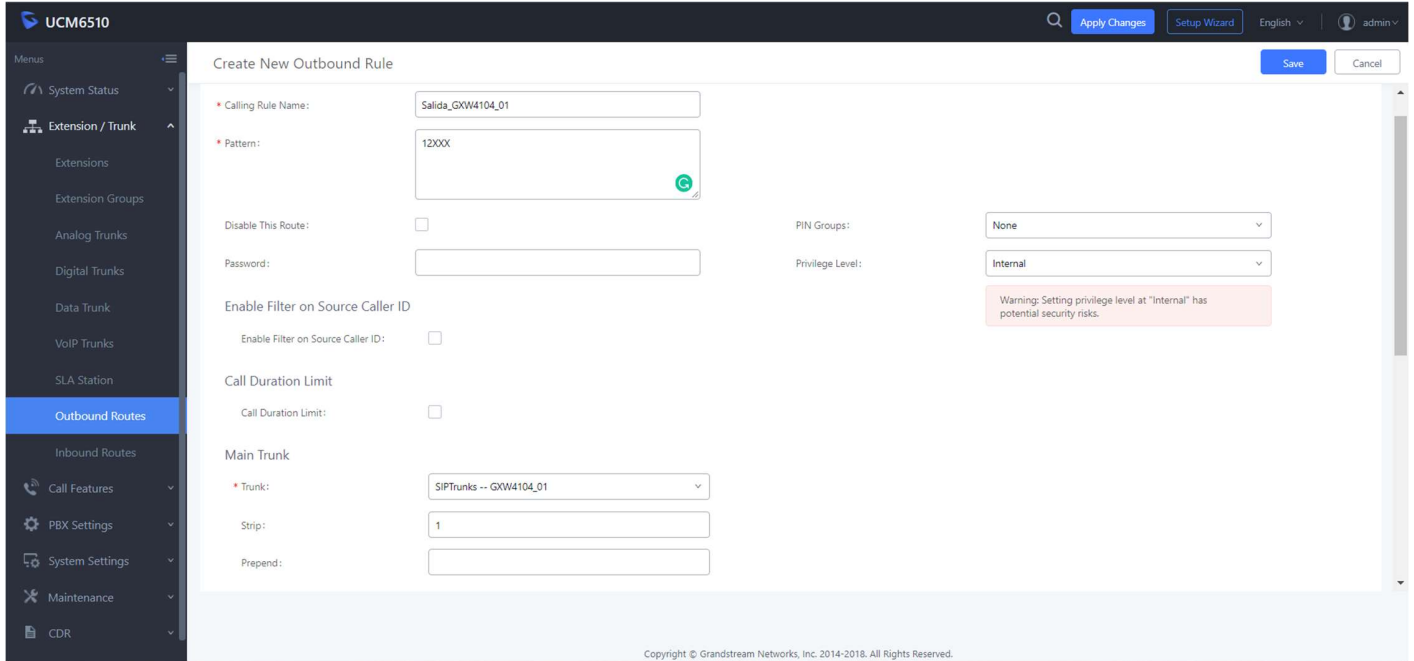
**Calling Route Name:** exit GXW4104 port FXO1

**Pattern:** 1 (to determine that it will be output through the FXO1 port of the GXW4104) pattern of the local numbers

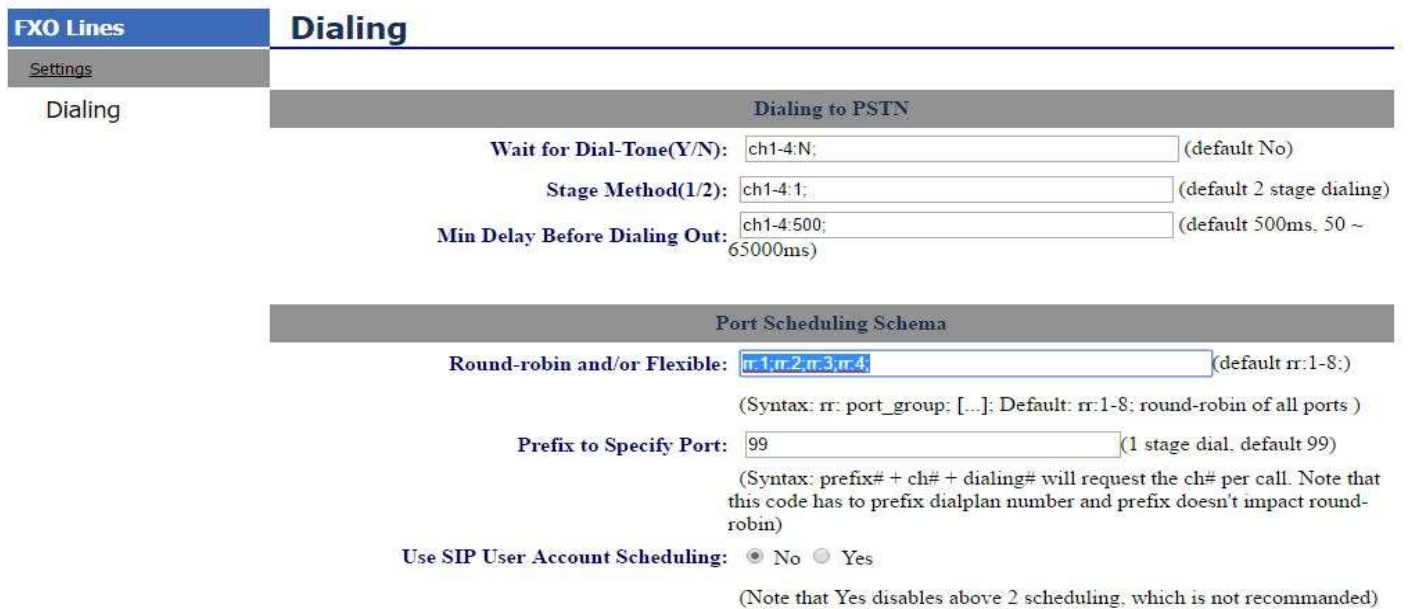
**Privilege Level:** you select the level according to the case

**Use Trunk \*:** select the SIP trunk created for the FXO1 port of the GXW4104

**Strip: 1** to remove the first digit of the established marking pattern. We press the **SAVE** button and then **Apply Changes**



We give Apply changes so that the UCM restarts and takes the configurations made and with this we already configure the output patterns for each FXO port of the GXW4104. Now we access the GXW4104 and make the following configuration changes: FXO Lines >> Dialing >> Round-robin and / or Flexible: rr: 1; rr: 2; rr: 3; rr: 4; we modify in this way



Download the page and click on the Save button

Then we go to Settings>>> Channels Settings

We verify that Local SIP Listen Port: ch1-4: 5060 ++; have this configuration so that for each FXO channel it increases the port by 2 by 2 starting in the 5060

Then we check the User ID: and add a random value in this case 555 as shown in the image

**Grandstream**      **Status**      **Accounts**      **Settings**      **Networks**      **Maintenance**      **FXO Lines**      **Line Analysis**  
Version: 1.4.1.5

**Settings**      **Channels Settings**

General Settings

Call Settings      **SIP Channel Setting**

Channels Settings

**DTMF Methods(1-7):**  (default 1)  
(1:in-audio, 2:RFC2833, 3:1+2, 4:SIP Info, 5:1+4, 6:2+4, 7:1+2+4)

**No Key Entry Timeout(X1s):**  (1-9, default 4)

**Local SIP Listen Port:**  (default ch1-8:5060++;)

**SRTP Mode(1-3):**  (default 1)  
(1:disabled, 2:enabled but not forced, 3:enabled and forced)

**Calling to VoIP**

**Unconditional Call Forward to Following:**

**User ID:**  (i.e ch1-2:223;ch3:224)

**SIP Server:**  (ch1-2:p1;ch3:p2)

**SIP Destination Port:**  (ch1-2:5060;ch2:7080)

Then we click the Save Button to save the changes made and in this way the output conditions of each of the FXO ports in the GXW4104 are configured