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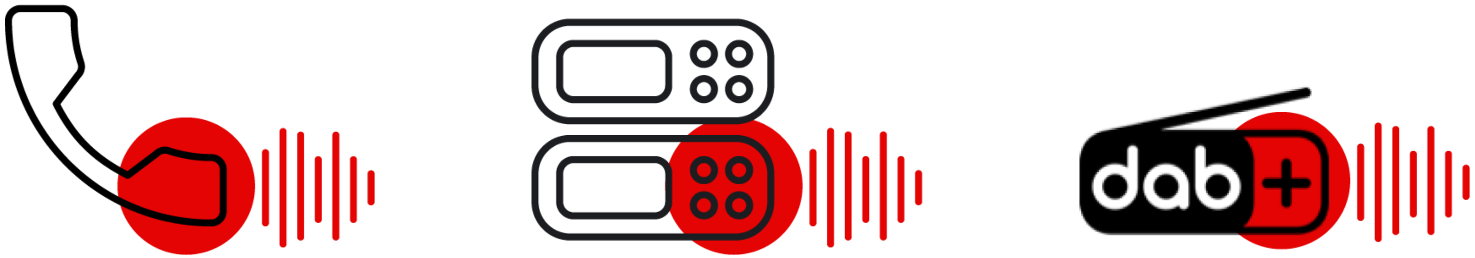

Grandstream HT881 - POTS Gateway

Quick Guide

Version 1.000 (13. February 2025)

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Content

Grandstream HT881

Overview

- Hardware
- Application

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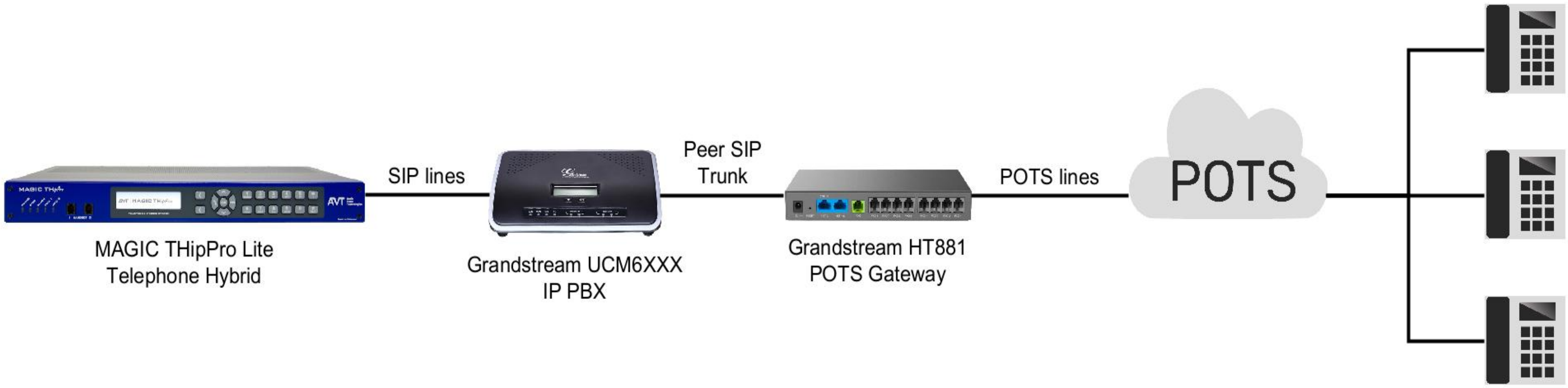
Hardware – Grandstream HT881

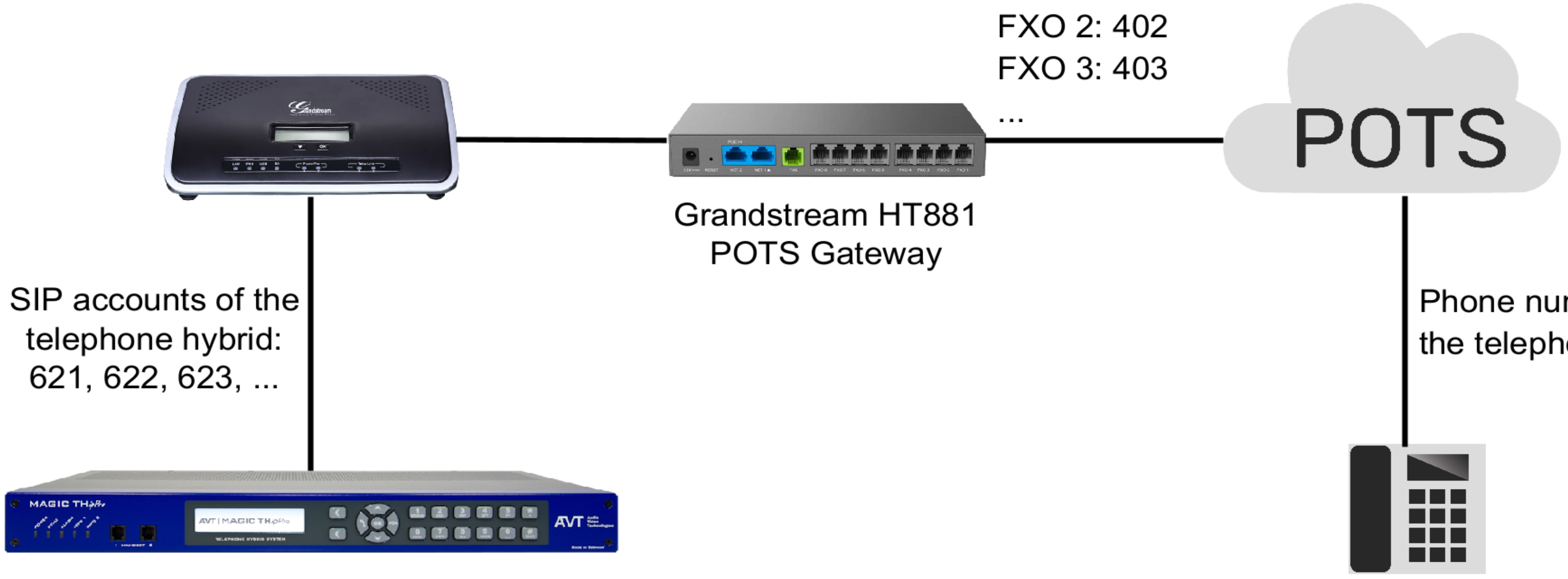
- 8 channel POTS / PSTN gateway
- Does not include a SIP server, a Grandstream UCM6XXX series PBX is required
- Supports only one connection per channel (no call transfer or conferencing)
- A VoIP licence is required on the telephone hybrid (POTS Gateway mode cannot be used)
- A version with 4 channels (HT841) is also available

Application



- The telephone hybrid connects via standard SIP lines to a Grandstream UCM6XXX series PBX
- The IP BPX connects to the Grandstream HT881 POTS gateway via a Peer SIP trunk
- Inbound and outbound routes in the IP PBX interconnect the calls between the telephone hybrid and the POTS gateway
- The Grandstream HT881 POTS Gateway connects via the FXO ports to the POTS network (POTS PBX or provider)

Operation - Example



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Grandstream HT881

Configuration of the Grandstream UCM6XXX PBX

- SIP Settings
- Call Termination
- Staging Method

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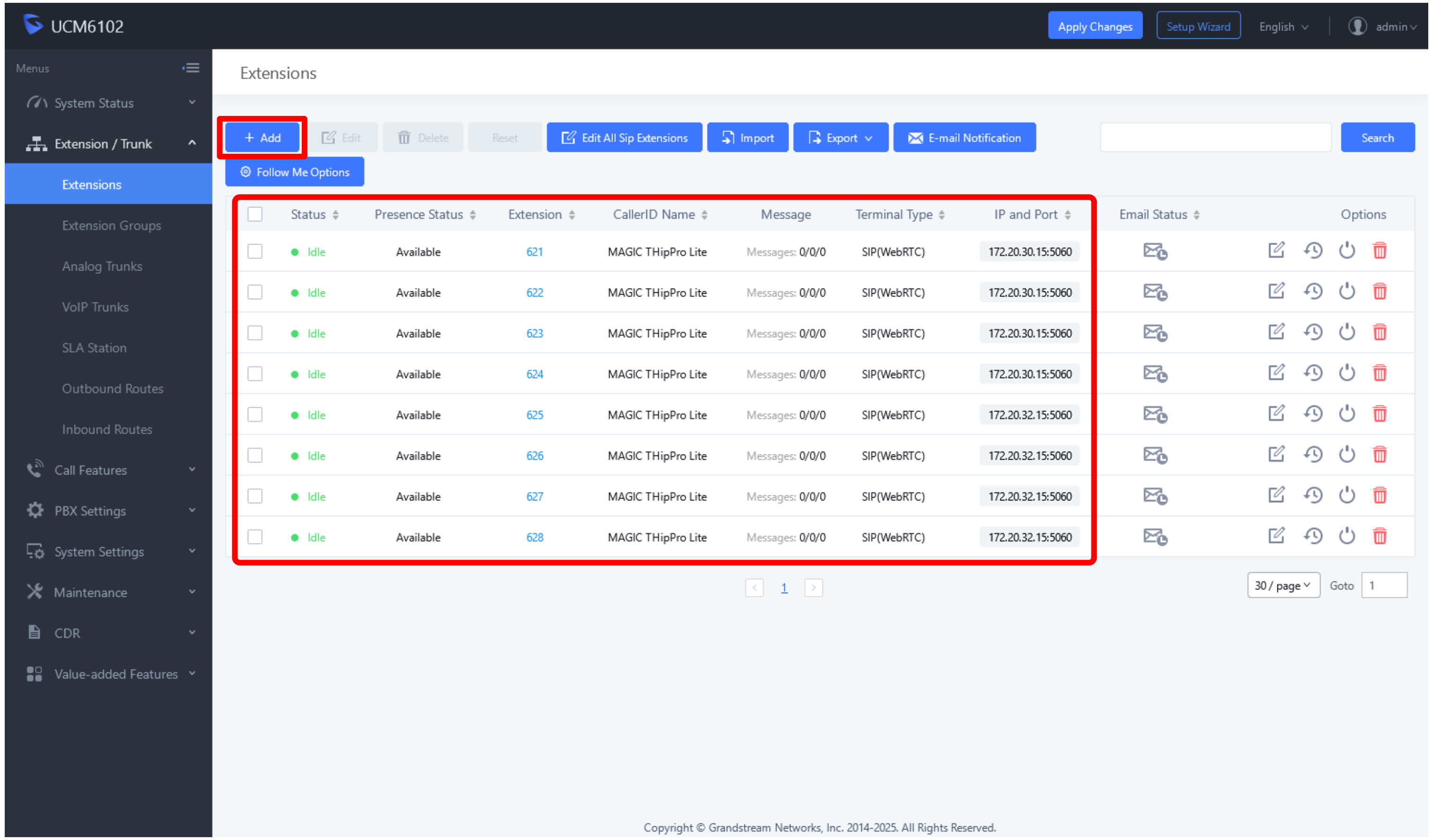
Enter the Web GUI

- Enter that IP address of the UCM6XXX PBX in a web browser to open the Web GUI
- Refer to the User Manual of your specific model for default IP address and login details

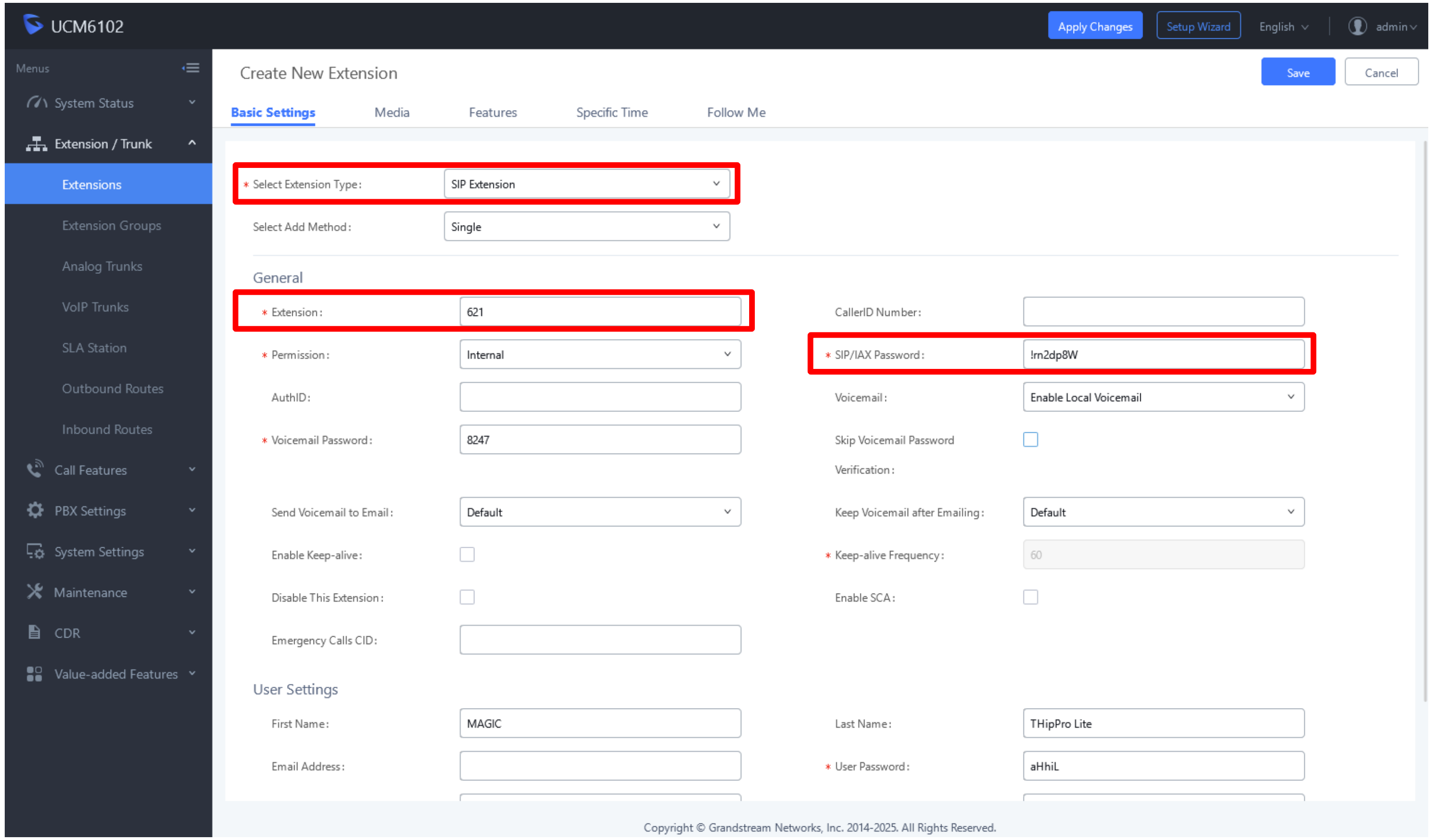
SIP Extensions

- Go to **Extension / Trunk > Extensions**
- **Add** an extension for each VoIP channel of the Telephone Hybrid



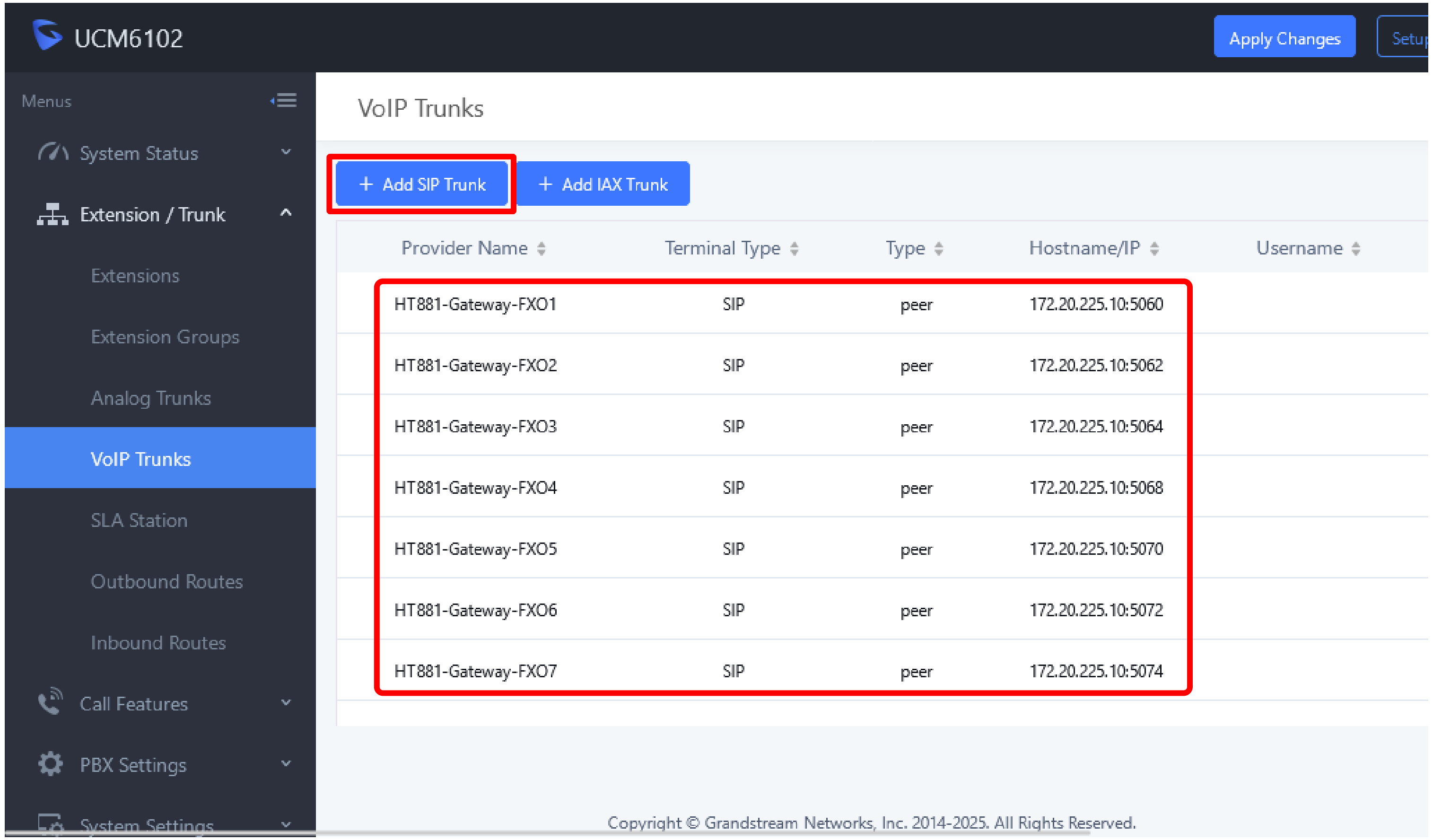
New SIP Extension

- Set **Select Extension Type** to **SIP Extension**
- Enter the phone number of the SIP account under **General > Extension**
- Set a password for the SIP account under **SIP/IAX Password**



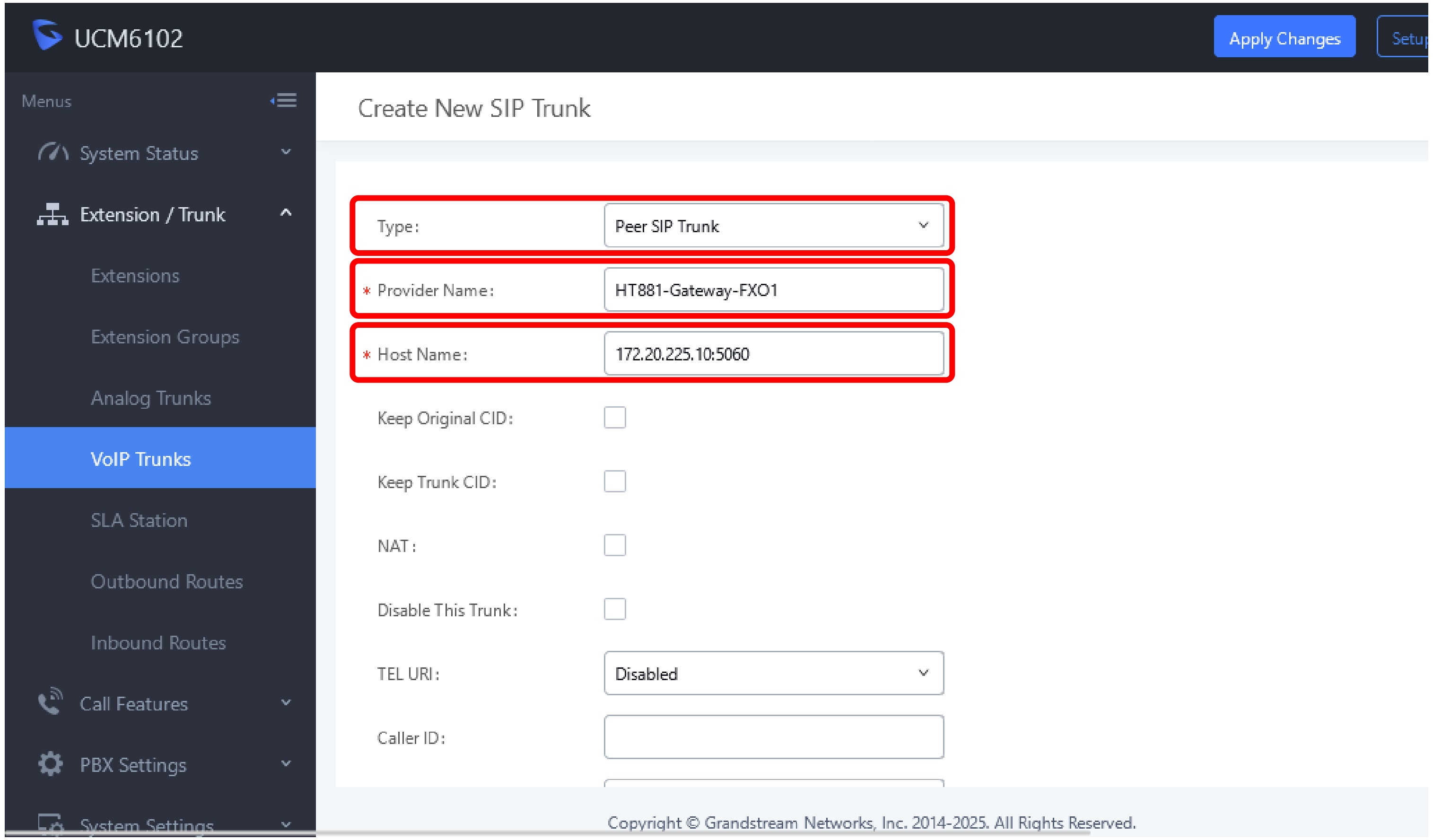
Peer SIP Trunks

- Go to **Extension / Trunk > VoIP Trunks**
- Add a SIP Trunk for each FXO port of the Grandstream HT881 POTS Gateway



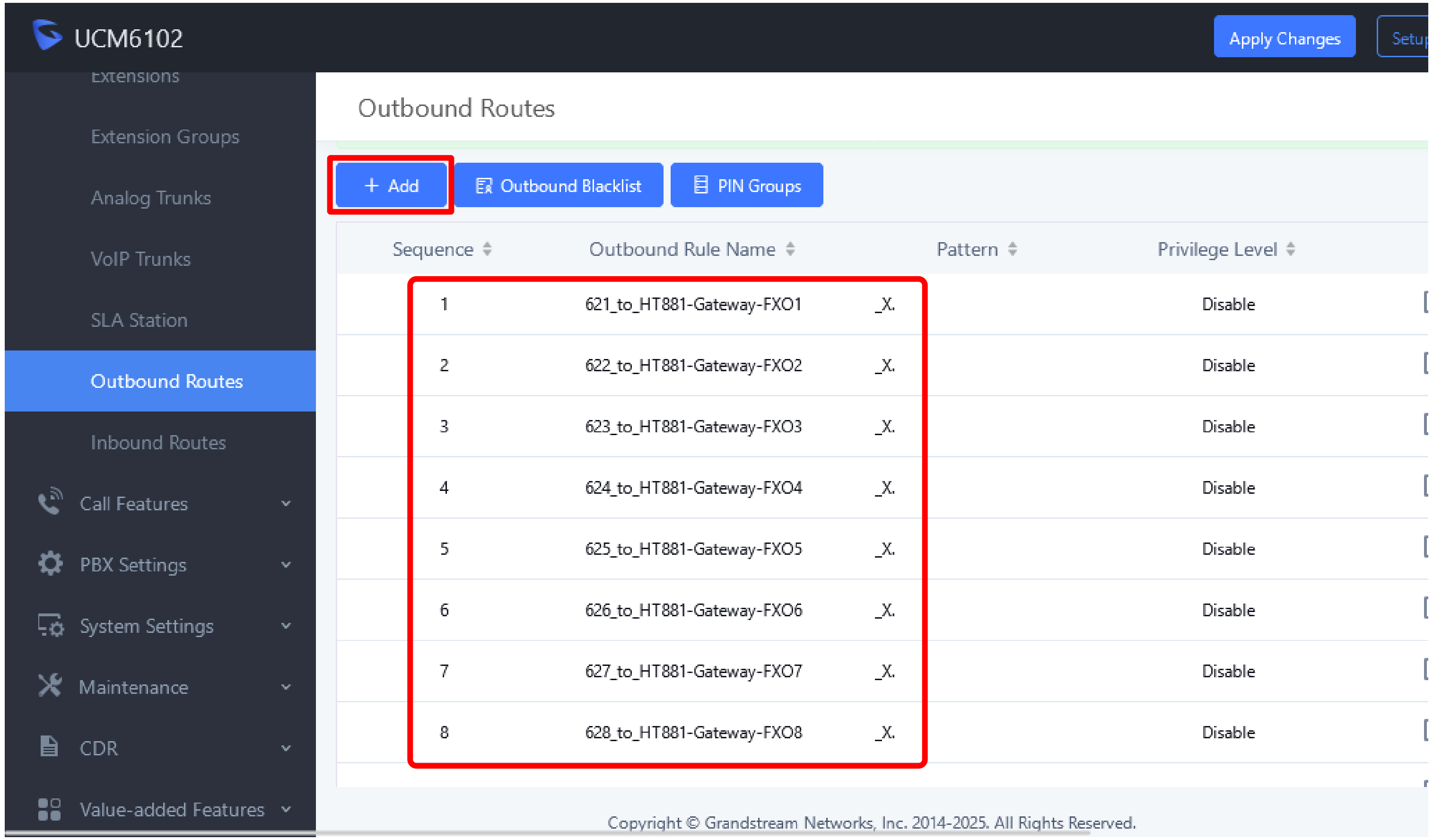
New Peer SIP Trunk

- Set **Type** to **Peer SIP Trunk**
- Set a **Provider Name**
- Set **Host Name** to the **IP address** of the HT881 POTS Gateway and append a specific **port number** for each FXO port using a colon
  - FXO 1: 5060
  - FXO 2: 5062
  - FXO 3: 5064
  - …
  - FXO 8: 5074



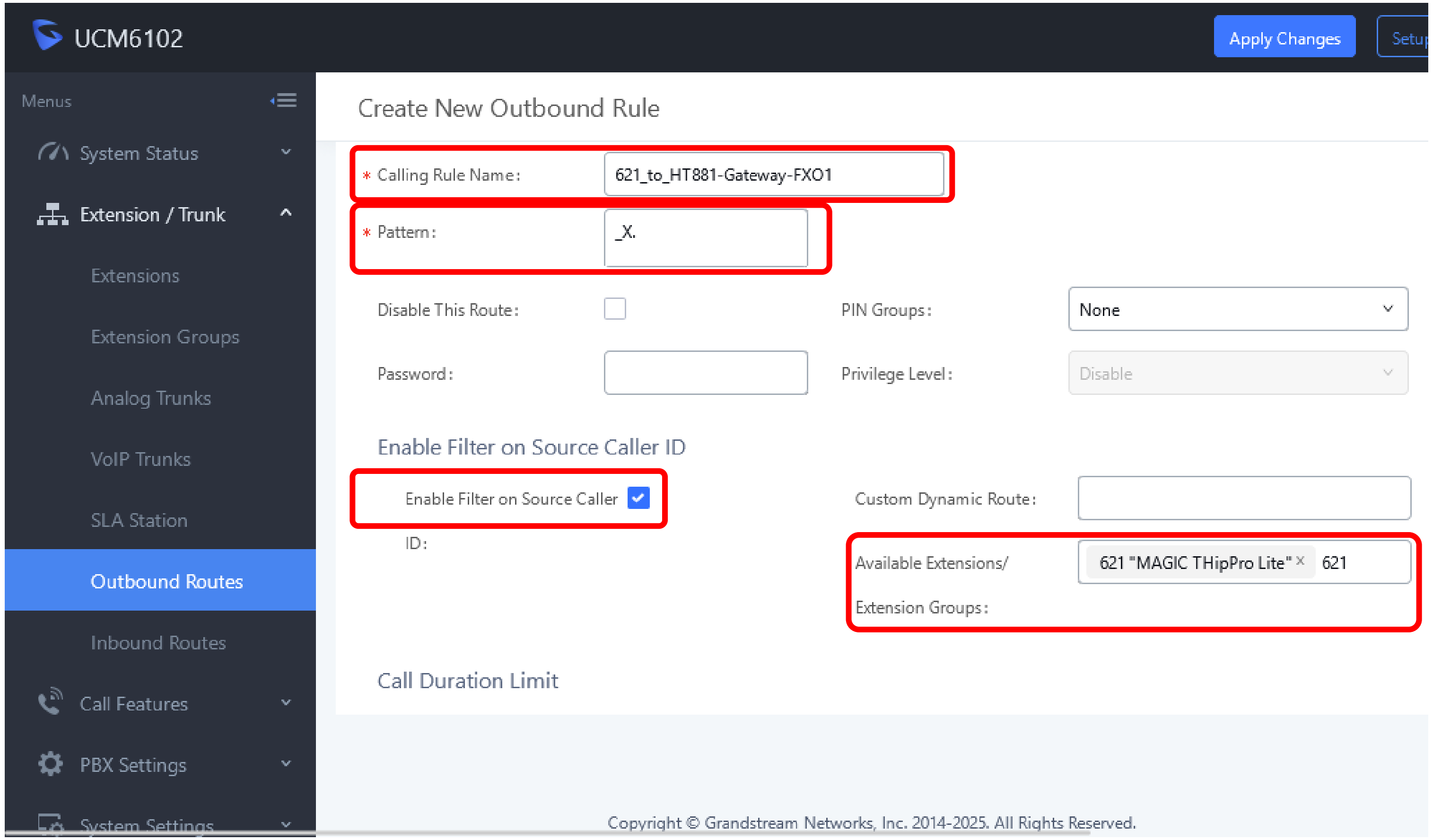
Outbound Routes

- Go to **Extension / Trunk > Outbound Routes**
- Add an Outbound Route for each FXO port of the Grandstream HT881 POTS Gateway



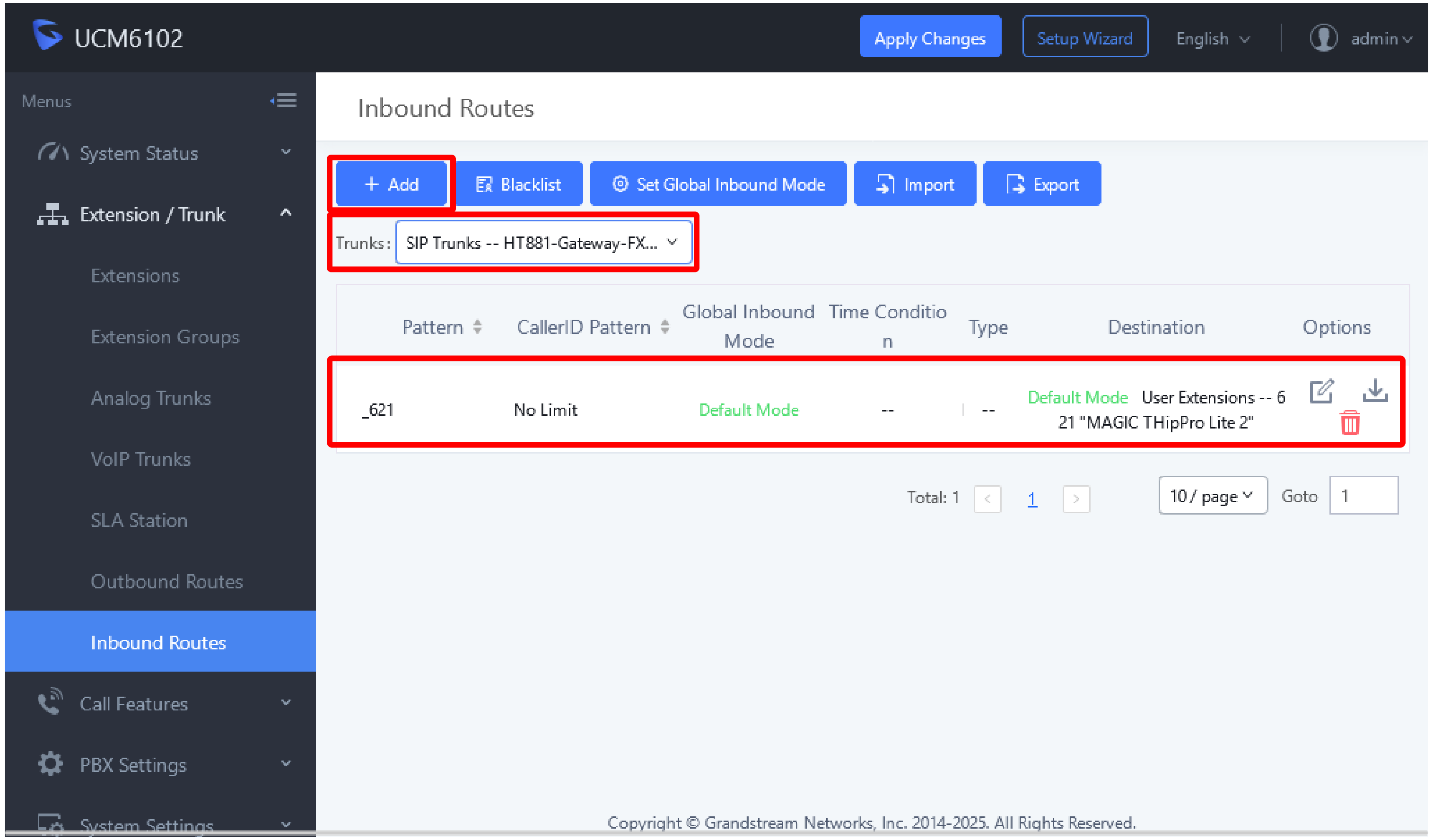
New Outbound Route

- Set a **Calling Rule Name**
- Set **Pattern** to **_X.**
- **Enable Filter on Source Caller**
- Under **Available Extensions / Extension Group**, select the SIP Username of the respective line of the Telephone Hybrid



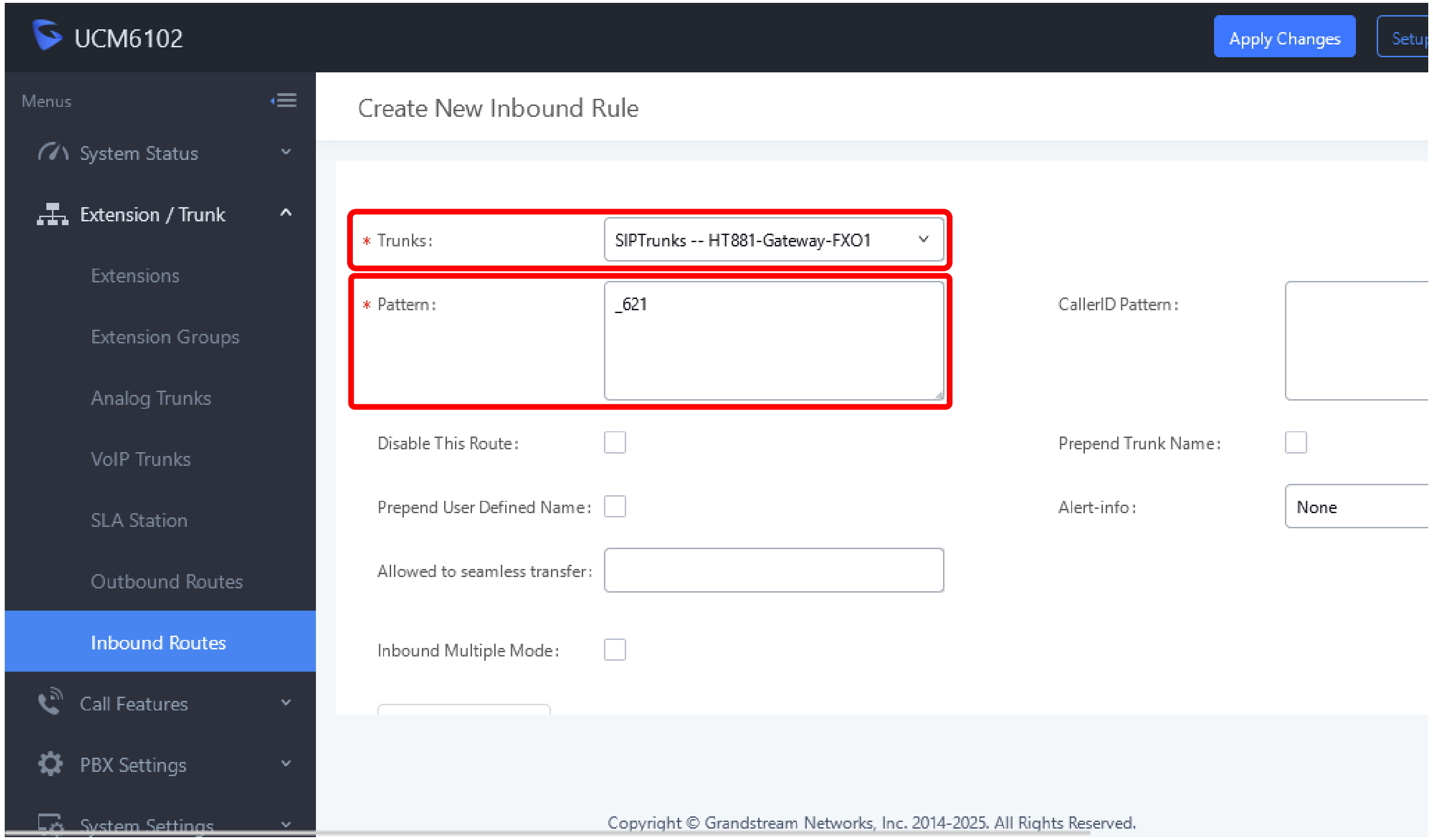
Inbound Routes

- Go to **Extension / Trunk > Inbound Routes**
- Add one Inbound Route for each Peer SIP **Trunk**



New Inbound Route

- The Peer SIP Trunk is already selected.
- Enter the phone number of the respective VoIP line of the THipPro as the **Pattern.** Prefix it with an underscore ("_<number>", in this example: _621).



Grandstream HT881

Configuration of the Grandstream HT881

- SIP Settings
- Call Termination
- Staging Method

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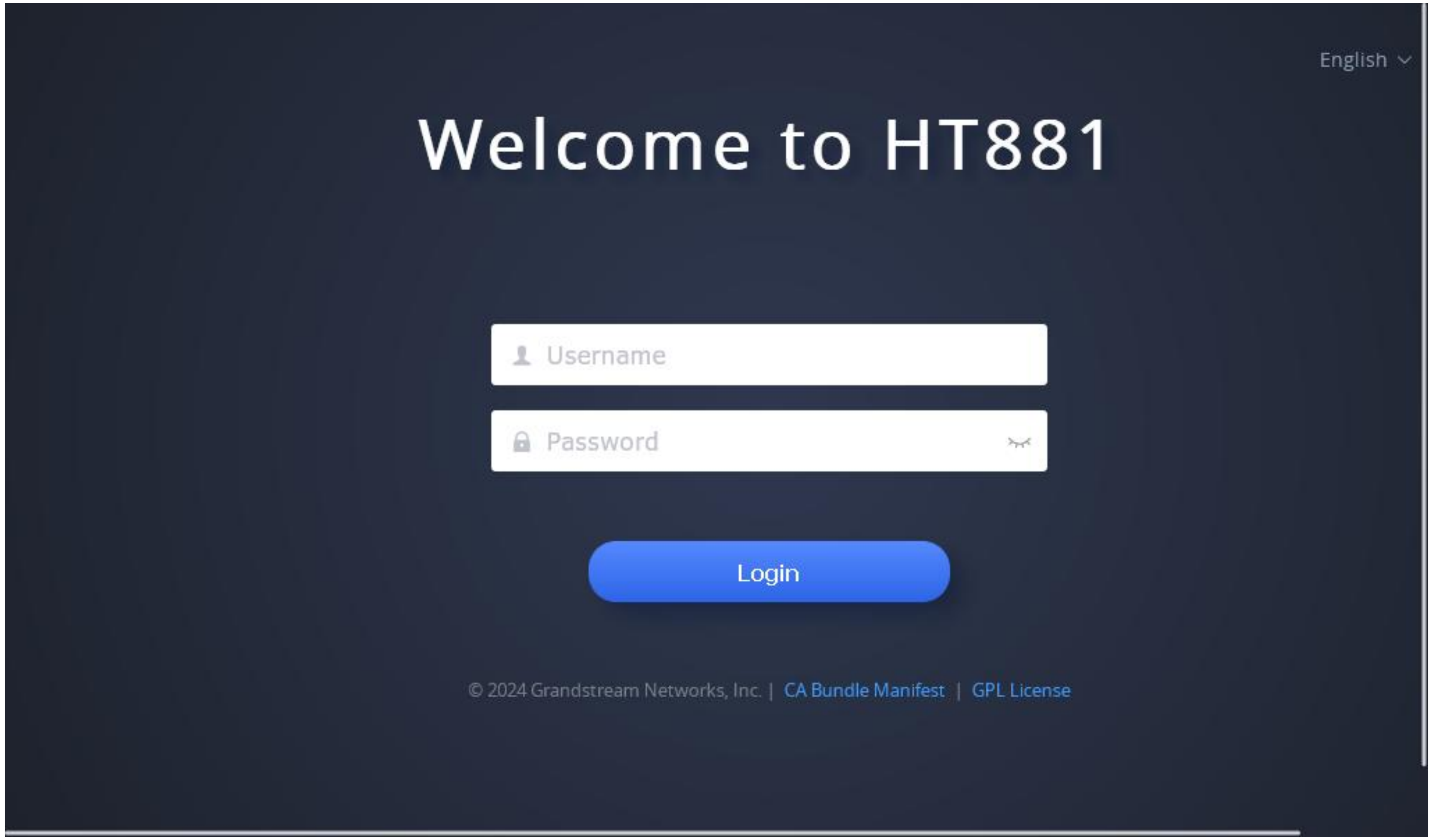
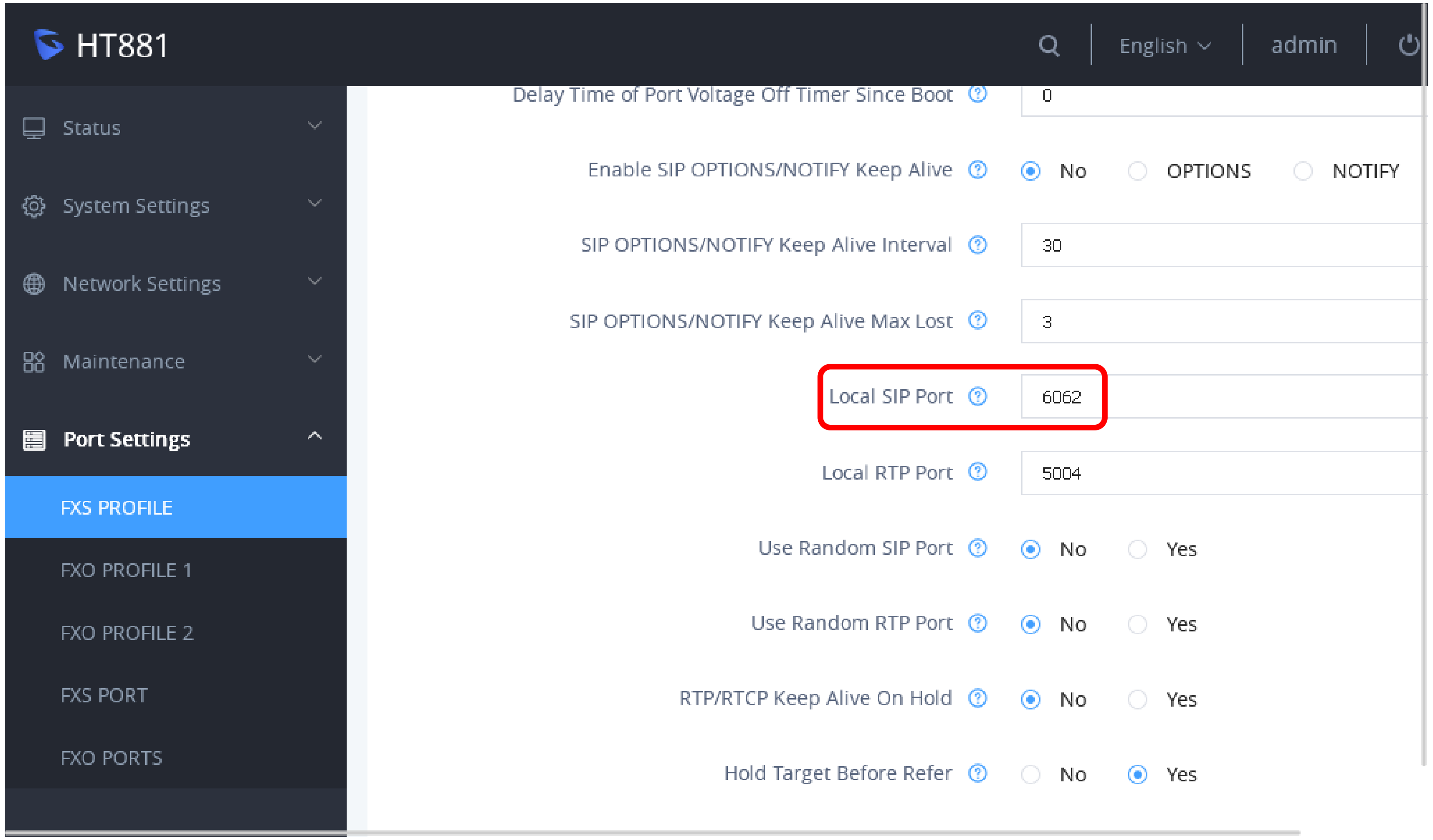
Enter the Web GUI

- By default, the Ethernet interfaces NET 1 and NET 2 operate in bridge mode
  - NET 1 is the WAN port which will request an IP address via DHCP
  - NET 2 is the LAN port with DHCP pass through from WAN port
- Connect NET 1 of the HT881 to the local network and lookup the IP address it received in the router / DHCP server
- Enter that IP address in a web browser to open the Web GUI
- Default login:
  - Username: admin
  - Password: written on the sticker at the bottom of the device



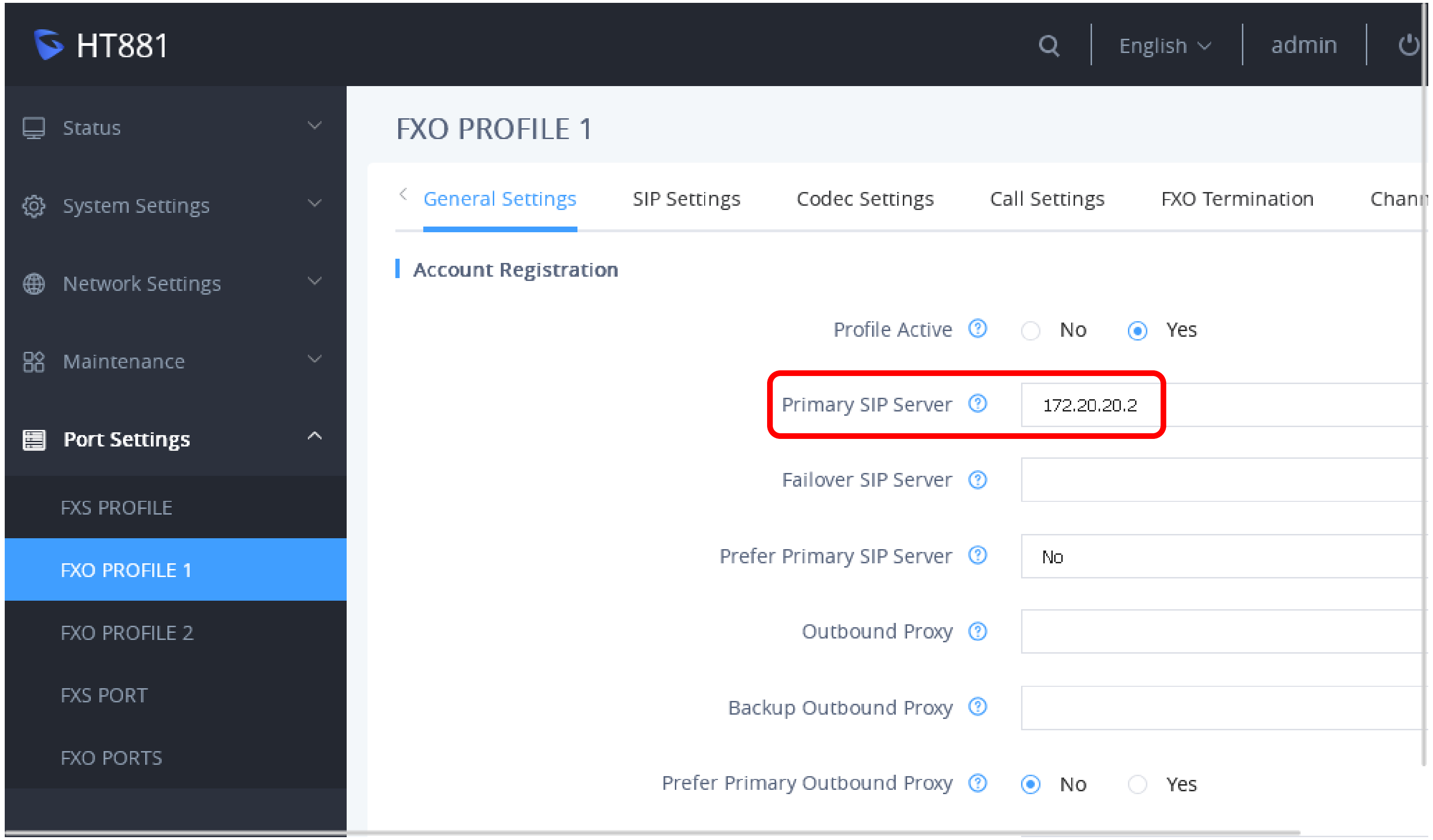
FXS Profile

- Go to **Port Settings > FXS Profile > SIP Settings**
- Change **Local SIP Port** to **6062**



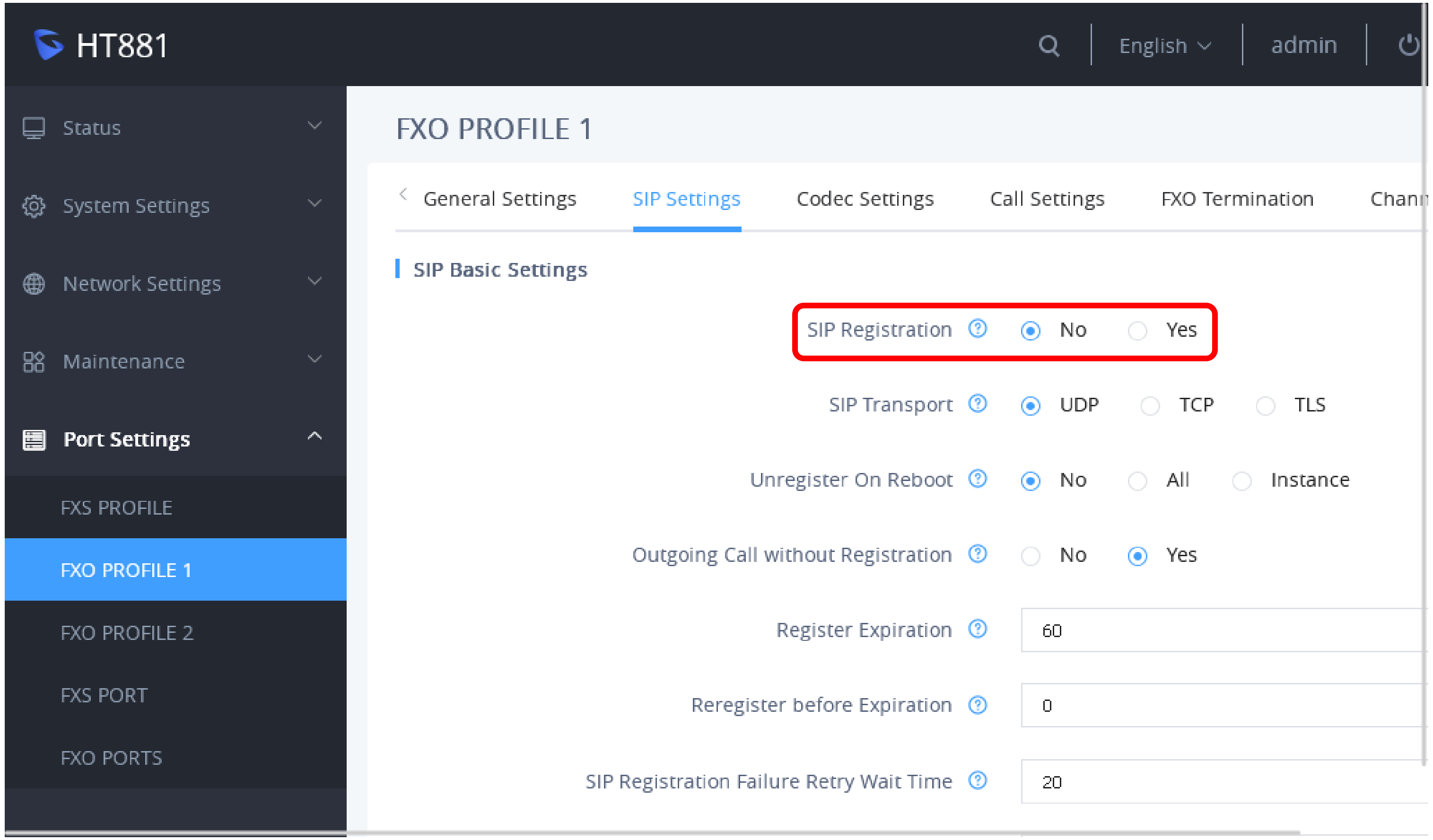
FXO Profile 1 – General Settings

- Go to **Port Settings > FXO Profile 1 > General Settings**
- Enter the IP address of the Grandstream UCM 6XXX PBX as **Primary SIP Server**



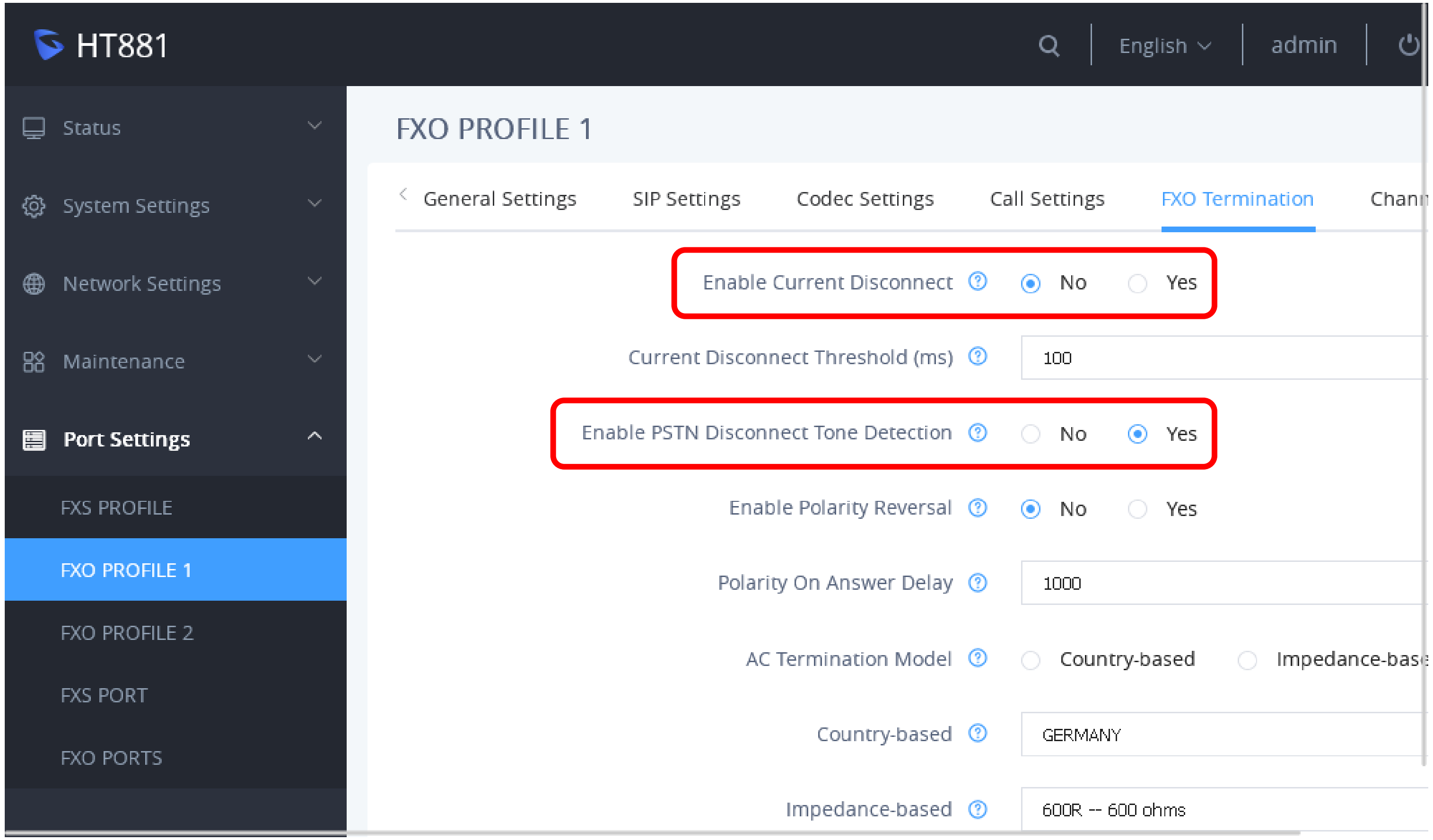
FXO Profile 1 – SIP Settings

- Go to **Port Settings > FXO Profile 1 > SIP Settings**
- Set **SIP Registration** to **No**



FXO Profile 1 – FXO Termination

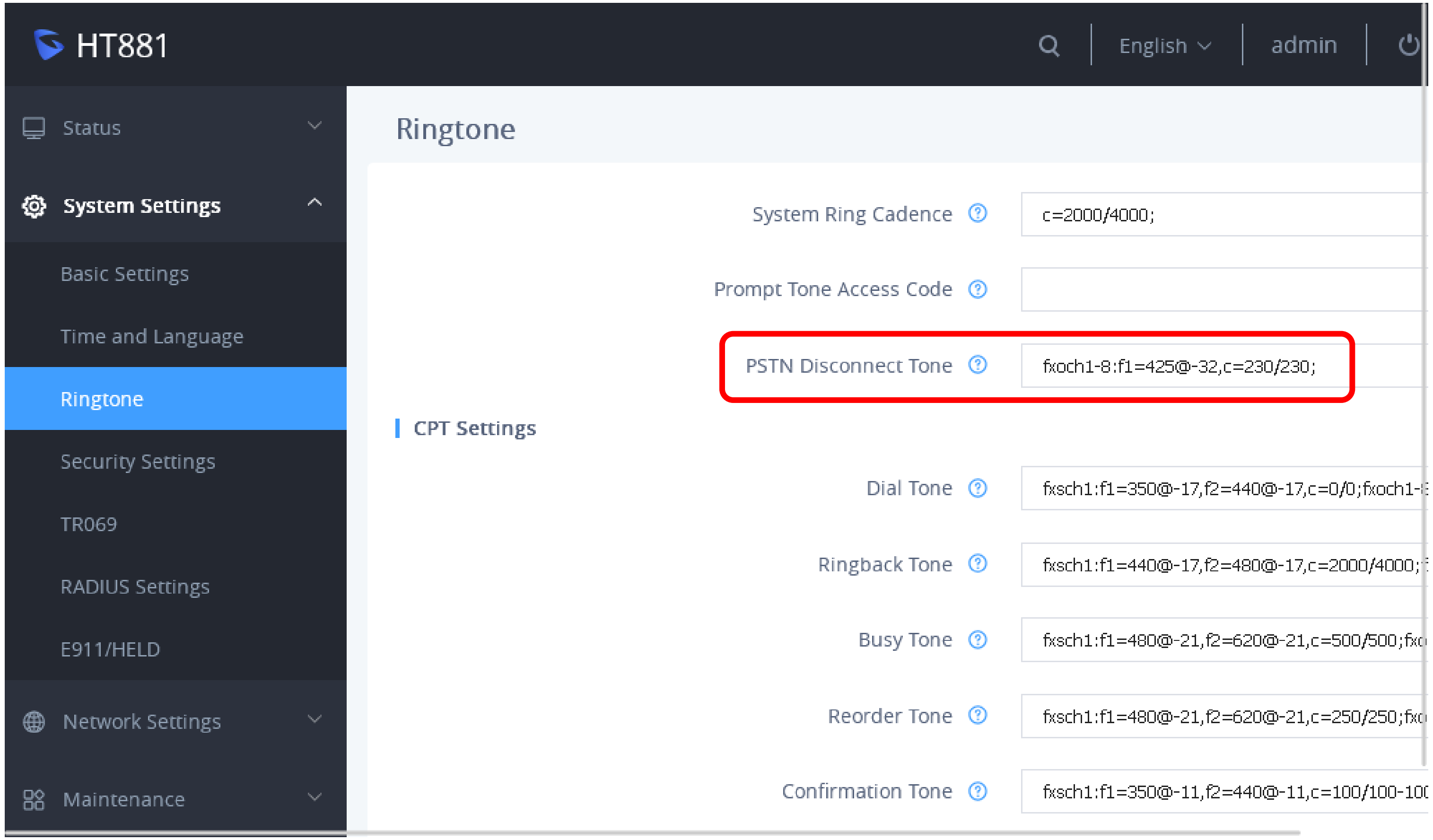
- Go to **Port Settings > FXO Profile 1 > FXO Termination**
- Find out if the local POTS network uses **Current Disconnect** or **Disconnect Tone** to disconnect a call and configure the options accordingly



PSTN Disconnect Tone

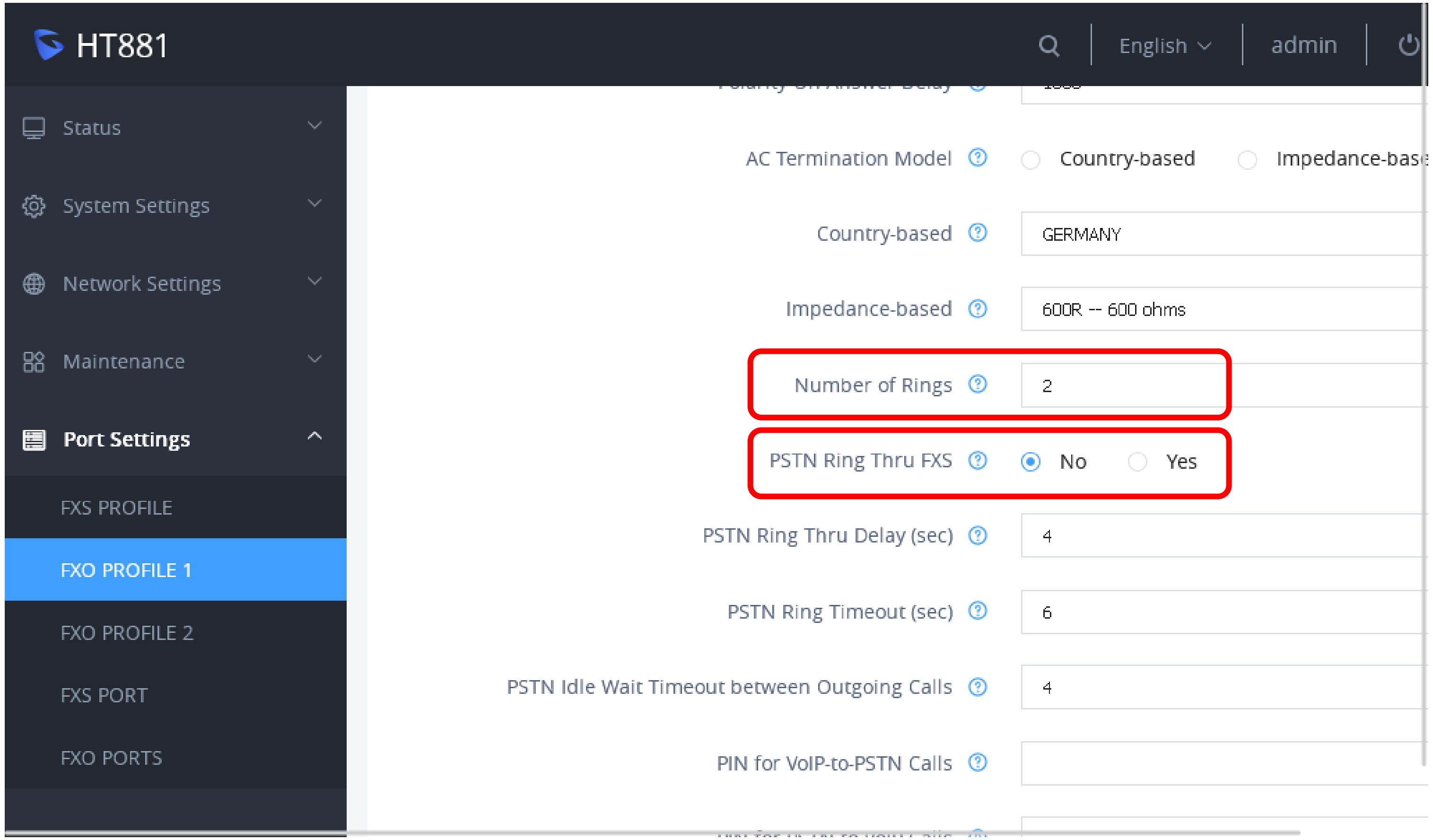
- If Disconnect Tone is used, the tone sequence must be configured under **System Settings > Ringtone > PSTN Disconnect Tone**
- Tone sequences of many countries can be looked up here:

  https://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf



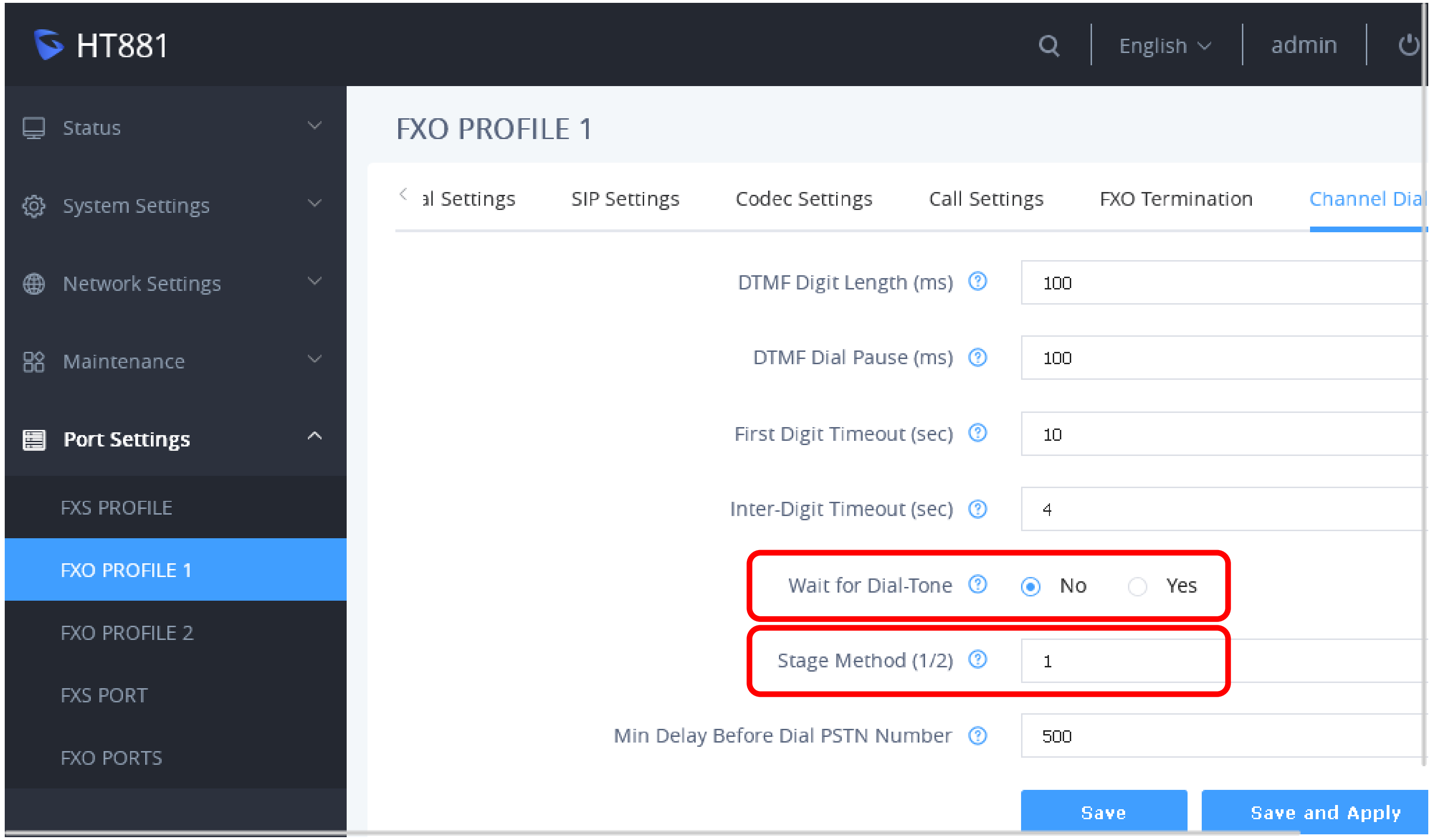
FXO Profile 1 – FXO Termination

- Go to **Port Settings > FXO Profile 1 > FXO Termination**
- Set **Number of Rings** to **2** so, the call will be passed through after the second ringing when the phone number was received.
- Set **PSTN Ring Thru FXS** to **No**.



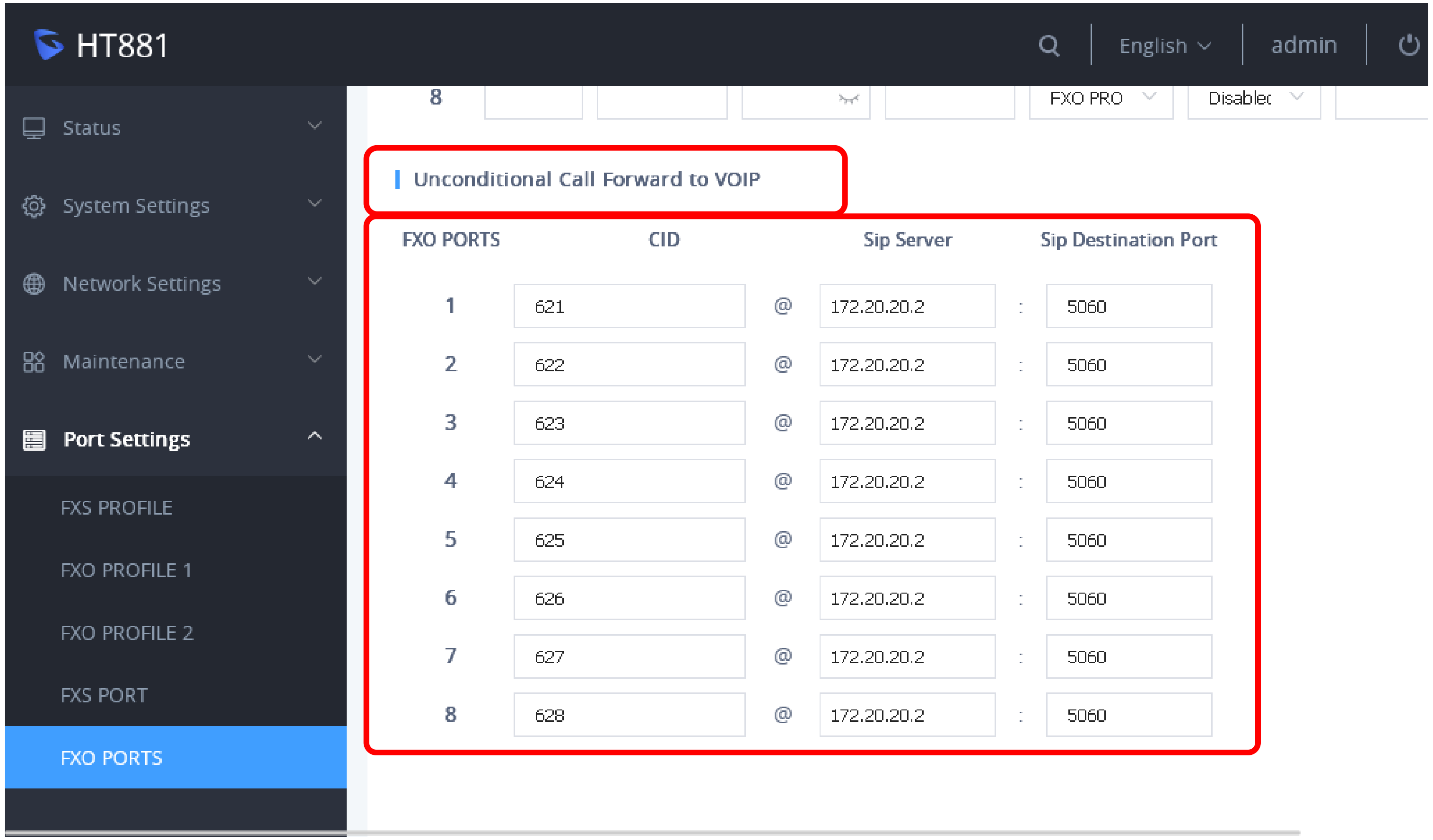
FXO Profile 1 – Channel Dialling

- Go to **Port Settings > FXO Profile 1 > Channel Dialling**
- Set **Wait for Dial-Tone** to **No** so.
- Set **Stage Method (1/2)** to **1**.



FXO Ports

- Go to **Port Settings > FXO Ports**
- Enter the SIP Usernames of the respective VoIP lines of the telephone hybrid as **CIDs**
- Set the IP address of the UCM6XXX PBX as **SIP Server** for each port
- Set the **SIP Destination Port** to 5060 for each port



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Configuration of the Telephone Hybrid

- Line Interface
- VoIP (LAN / SIP)

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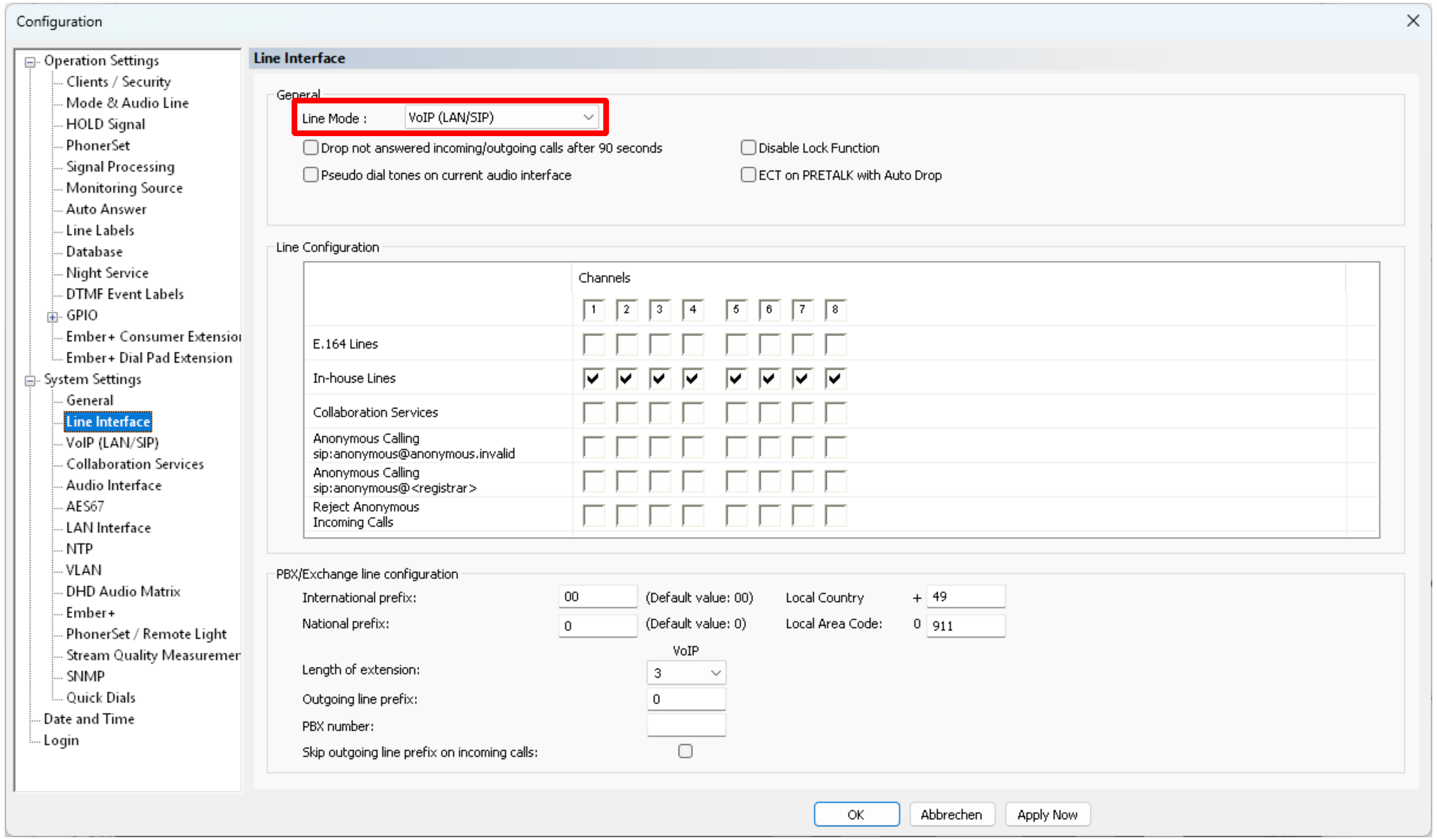
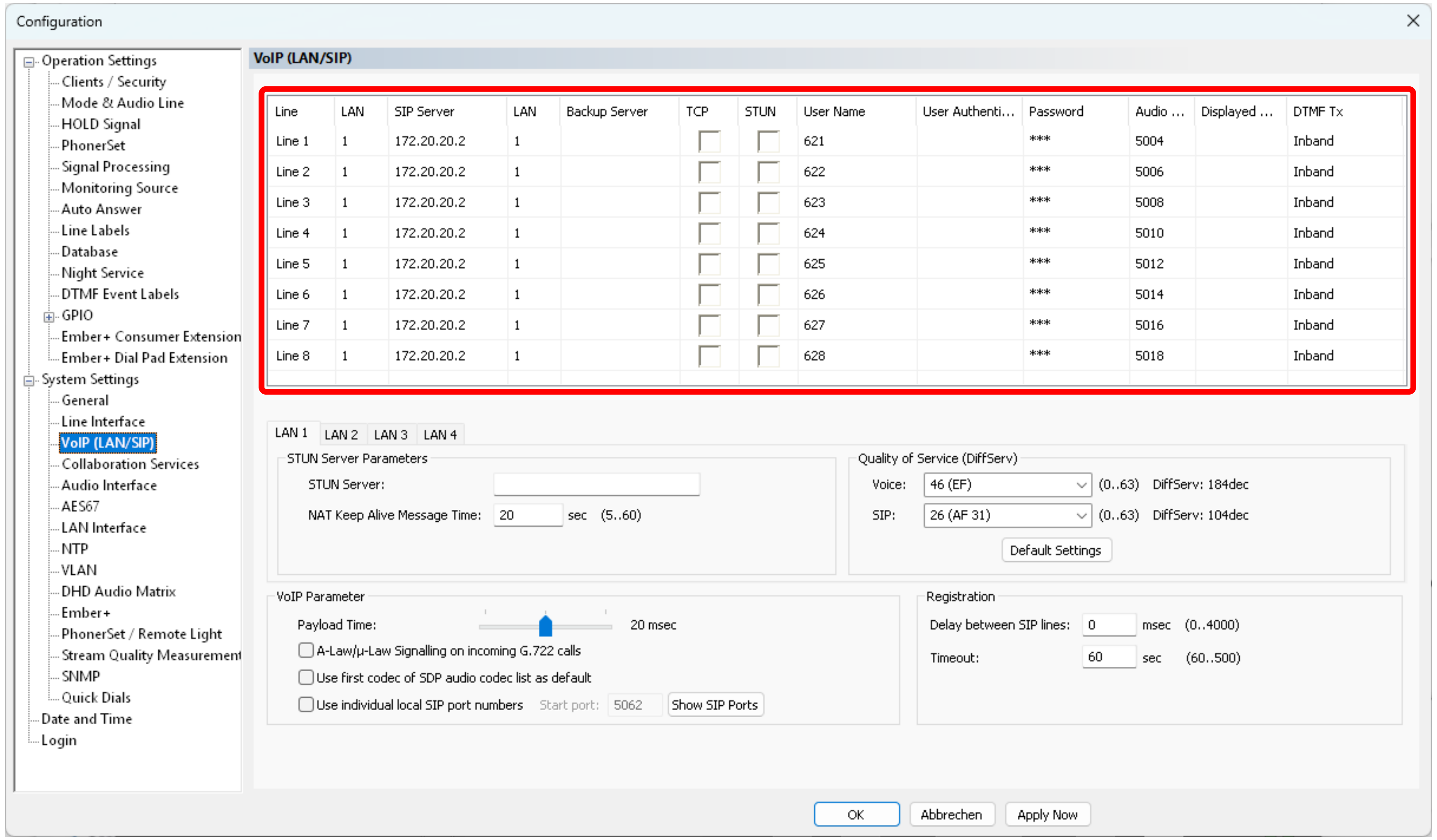
Line Interface

- In the PC-Software, go to **Menu > Configuration > System > System Settings > Line Interface**
- Set **Line Mode** to **VoIP (LAN / SIP)**.
  - Note: POTS Gateway Mode cannot be used with the Grandstream HT881.



VoIP (LAN /SIP)

- In the PC-Software, go to **Menu > Configuration > System > System Settings > VoIP (LAN /SIP)**
- Enter the SIP accounts which were prepared in the Grandstream IP PBX



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