CloudUCM – Endpoint Configuration Guide

CloudUCM is a cloud PBX that integrates audio and video calls and collaborative work. Users can register extensions on terminals for communication, including using Grandstream IP phones, Wave Desktop applications, Wave Web clients, and Wave mobile applications for calls/meetings, chats, remotely managing and synchronizing extensions, cloud storage, alerts, statistics reports, etc.

This document describes how to configure CloudUCM services on IP phones so that users can register with CloudUCM on external and internal networks.

Configuring SIP Accounts on End Devices

IP phones can be registered to the CloudUCM in external and internal network environment for internal communication and remote communication.

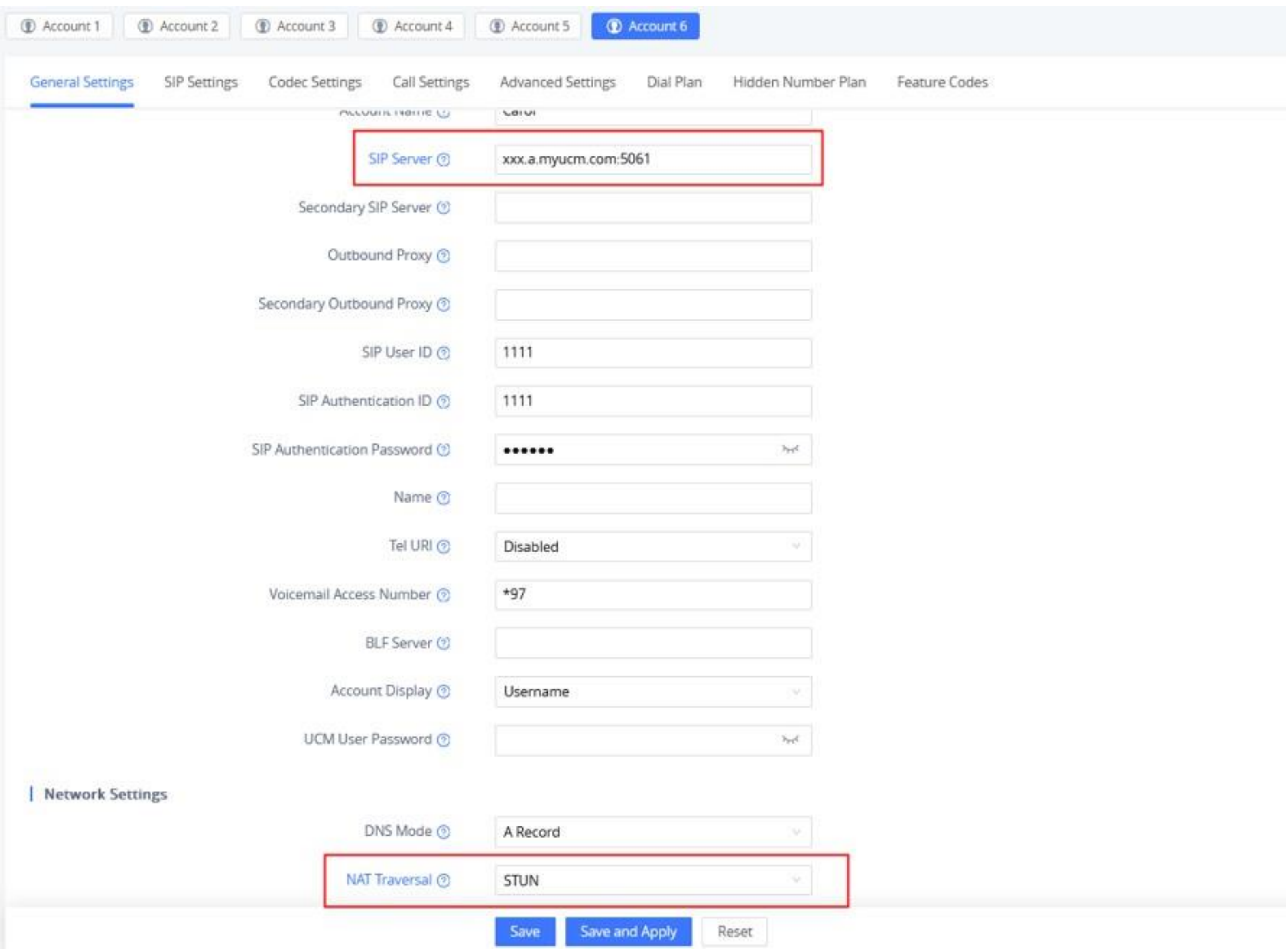
Method 1: Configure SIP Account from End Device Web User Interface

In this method, the user needs to configure the SIP server on the end device web GUI using the CloudUCM SIP Server Address and configure NAT to STUN. For SIP transport protocol, it must be set to TLS using TLS version 1.2 or 1.3.

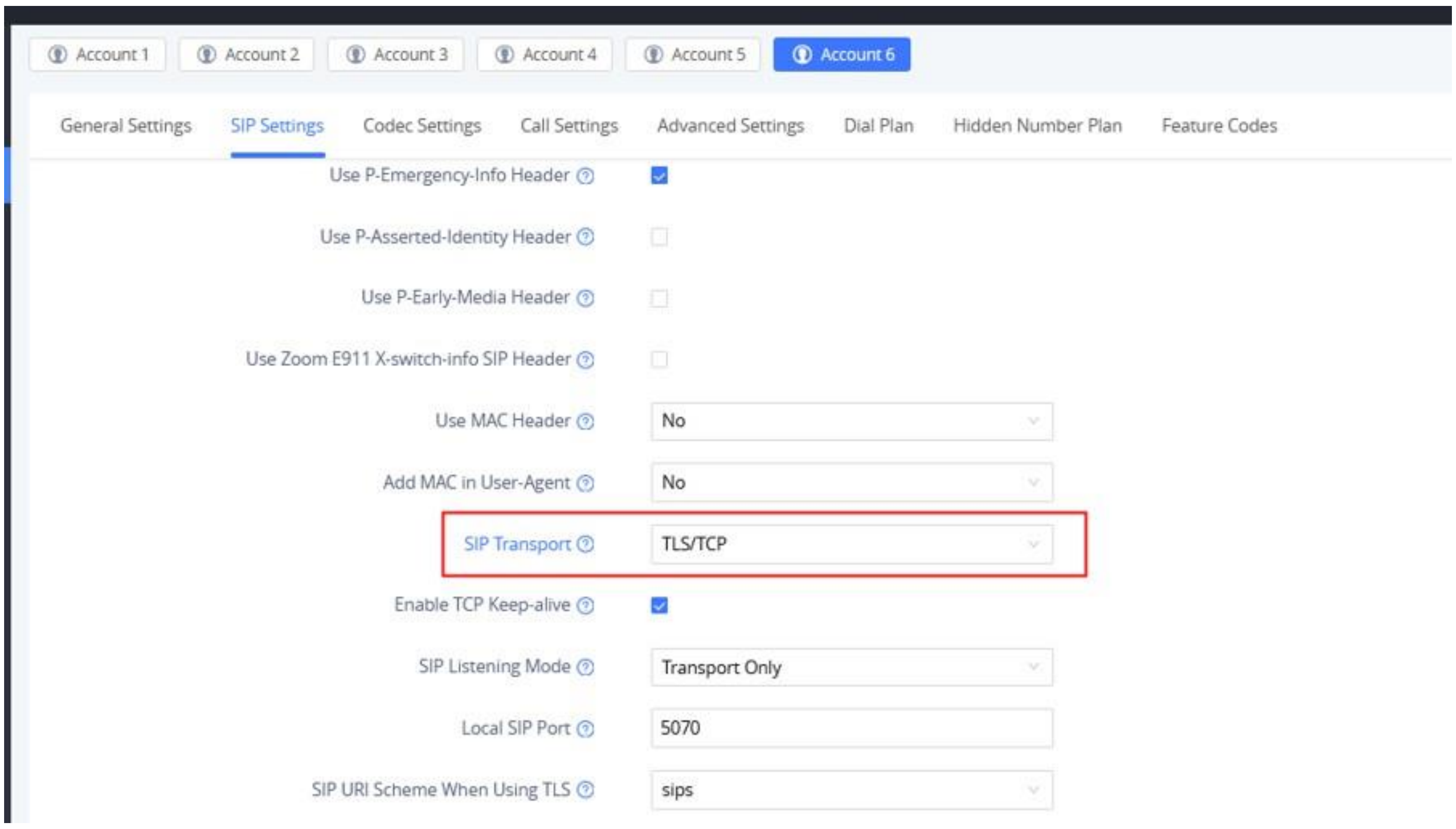
Please refer to the below configuration example on GRP2604P.

Step 1: Log in GRP2604P web UI as admin, navigate to **Account->Basic Settings** page and configure the following:

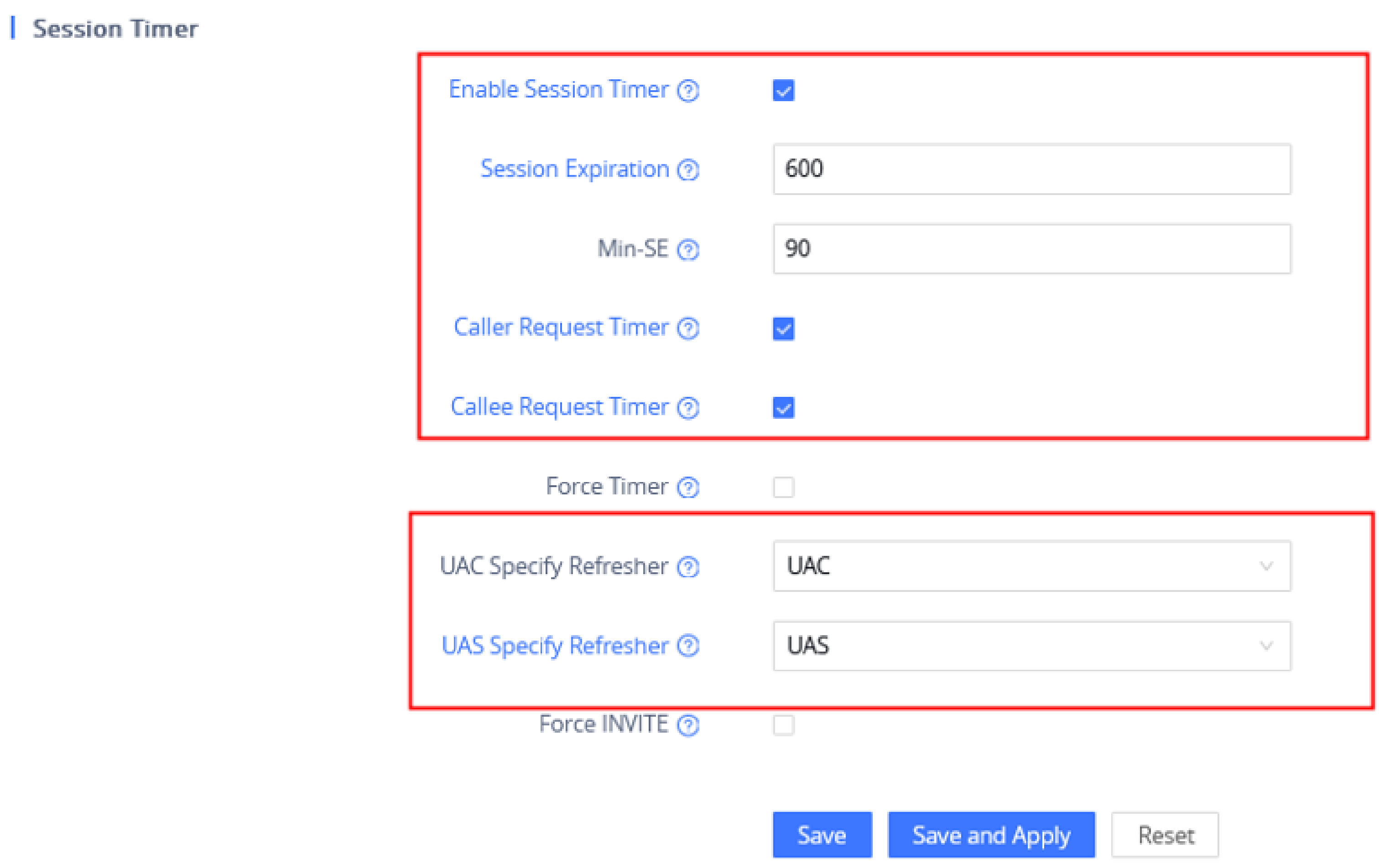
- **SIP Server**: Enter the CloudUCM SIP Server Address. This information can be found under UCM **Web GUI -> CloudUCM Plan** page.
- **NAT Traversal**: STUN



Step 2: Go to **Account → SIP Settings** and configure SIP transport to “TLS” or “TLS/TCP”.

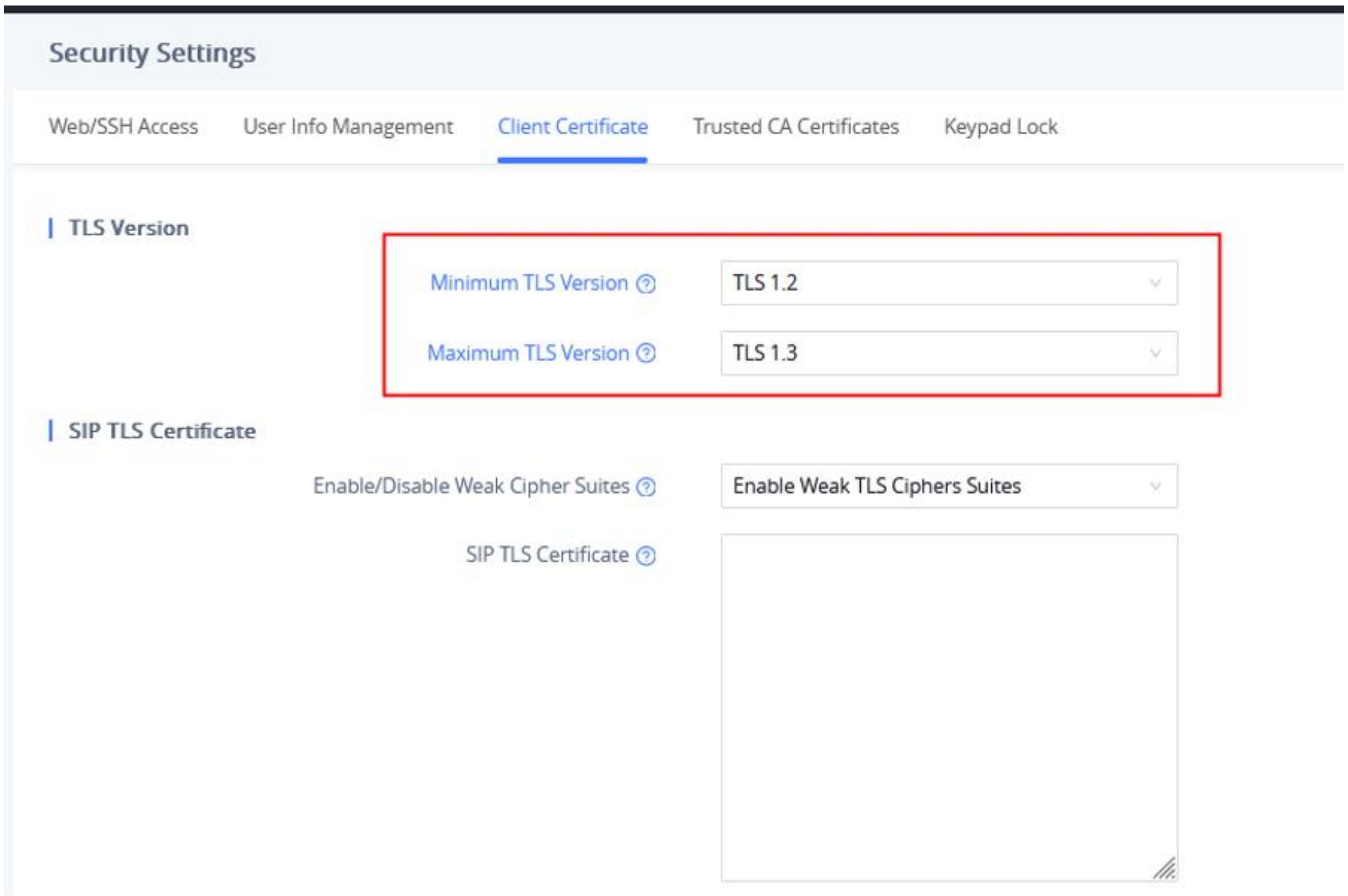


Step 3: Go to **Account -> SIP Settings** page, you can configure REGISTER Expiration, SUBSCRIBE Expiration, and Session Timer following the configurations below. The configurations are not required, but if your network is unstable, it is recommended to configure those settings, so that the CloudUCM services will be more stable.



REGISTER Expiration	3600
SUBSCRIBE Expiration	3600
Enable Session Timer	Yes
Session Expiration	600
Min-SE	90
Caller Request Timer	Yes
Callee Request Timer	Yes
UAC Specify Refresher	UAC
UAS Specify Refresher	UAS

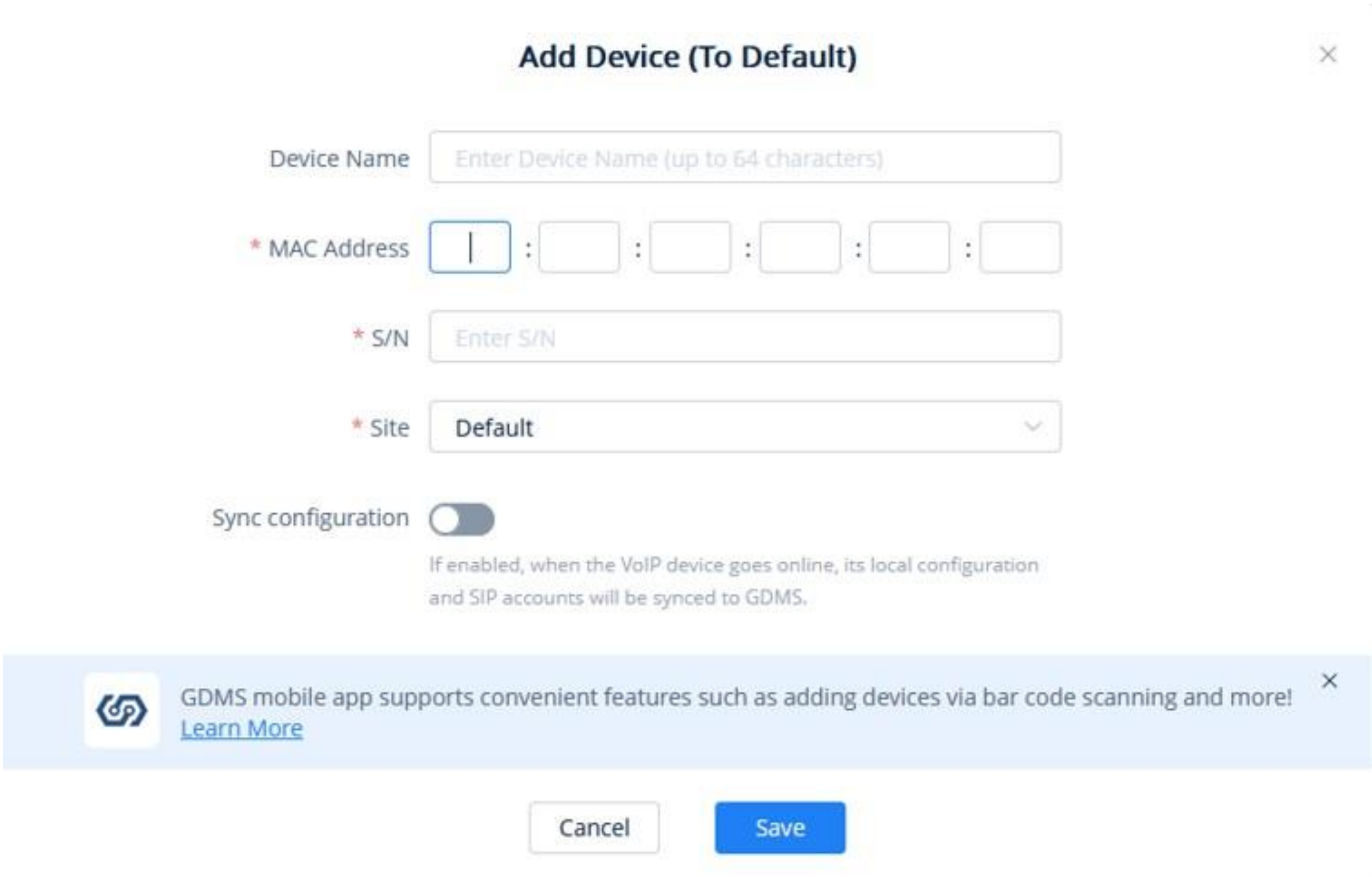
Step 4: Go to the phone's web **UI→System Settings→Security Setting→TLS** page, configure "Minimum TLS Version" and "Maximum TLS Version" to be 1.2 or 1.3



Method 2: Assign SIP Account for End Device from GDMS

In this method, user needs to log in GDMS to assign SIP account to the end device.

Step 1: Add the end device as VOIP Device to GDMS. Click on "Save" to save the configuration.



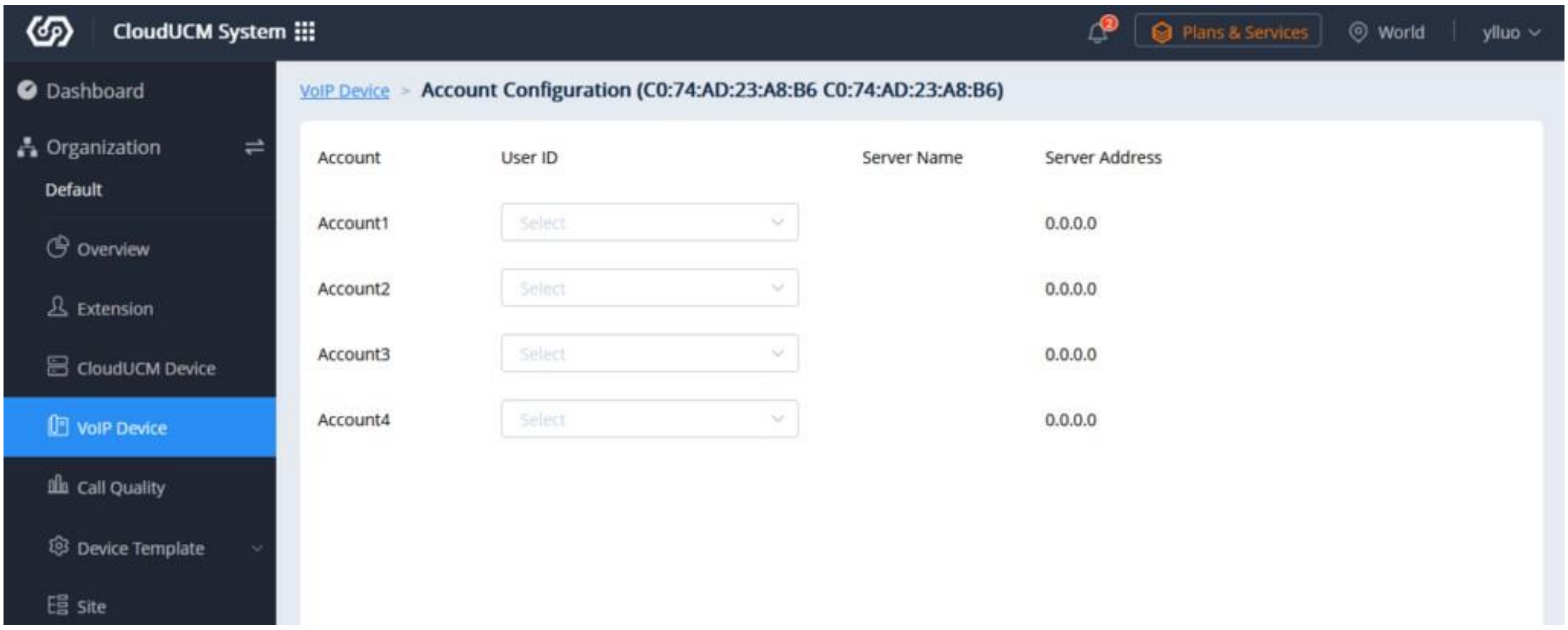
Note

- Each end device can only be added to **ONE** GDMS account at a time.
- Users can use the "Device Name", "MAC Address", or "Site Name" to search for the end device.

Step 2: There are two methods to assign the SIP accounts for the end device on GDMS.

Assign Method (1): Configure SIP Accounts from VoIP Device Page

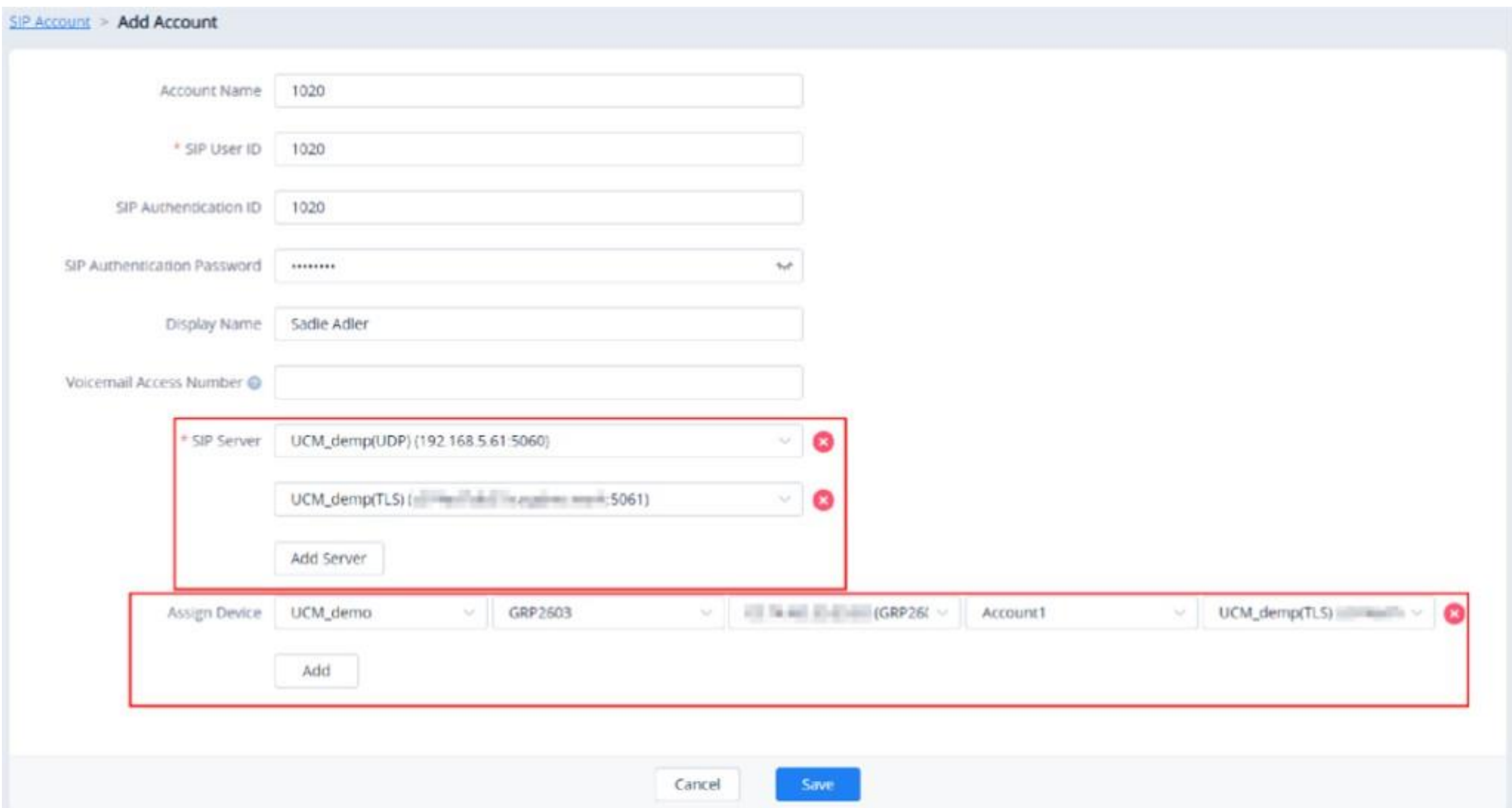
1. In the VoIP Device Page, click on to enter the Account Configuration page.
2. In this page, users can choose to deploy the extensions from the Extension page to the device. They can also change the existing extension to another or remove existing extensions.
3. Save and apply the configuration.



> **Note**
>
> - If the device becomes offiine during the account deployment, GDMS will push the settings once the devices comes back online.
> - Configurations from other methods such as via phone's web UI, Zero Config, etc… will not sync to GDMS.

Assign Method (2): Assign SIP Account from Extension Page

Step 1: Select "CloudUCM System" on the upper left page in GDMS and go to the Extension page, select the extension that needs to be assigned to the phone, and click to assign an account.



Step 2: Click the "Add Server" button and select the server address that the user wants to add for the extension.

Step 3: Assign the server to the phone. Select the site, device, MAC address, account location and the CloudUCM server address to assign.

Step 4: Click the "Save" option to complete assigning the extension to the phone.

Note

- When configuring the UCMRC server address to the phone, in order to make UCMRC work normally, the system will automatically issue the following configuration to the phone:
  1. NAT Traversal: STUN
  2. SIP transport is configured as "TLS".
  3. The Session Timer settings will be modified following the configurations below:
     Enable Session Timer Yes
     Session Expiration 600
     Min-SE 90
     Caller Request Timer Yes
     Callee Request Timer Yes
     UAC Specify Refresher UAC
     UAS Specify Refresher UAS
  4. "Minimum TLS Version" and "Maximum TLS Version" to be 2 or 1.3
- After assigning an account to the phone on the GDMS, if the phone cannot be registered or there is a problem with the call after the registration, please go to the phone to check whether the configuration is correct according to section [CONFIGURING SIP ACCOUNTS ON END DEVICES]
- IP phones which are not supported on GDMS cannot be remotely managed and deployed.
- Clients will not be able to edit SIP UserID, Authentication ID, Authentication Password, Display Name or Voicemail Access Number from this page.
- The available devices for configuration will be the devices listed in the VoIP Device page.

Make calls using IP Phones

After configuring the IP phones with CloudUCM service, users can use the phone to make audio/video calls and join GS Wave audio/video meetings.

Alert

Presentation on end device IP phone is currently not supported.