Configuration Guide For Grandstream WP810/822/825 series

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Revision History

Version	Date	Change
1.0	07/08/2022	Created document for WP810 Configurations

DUT and Zoom Software Versions

	Equipment	Software Version
Grandstream (Device Under Test)	WP810	1.0.11.16
Zoom	Zoom app Desktop	5.11.1 (6602)
	Zoom app Mobile	5.11.1 (6880)

Features Supported by WP810/822/825 series

- Multiple Line Keys
- Multiple Users per Device
- Custom Time Format and Zone
- Sync time with NTP server
- Enable or disable phone web admin portal
- TLS and SRTP
- Make and Receive Calls
- Inbound and Outbound Call via Opus codec negotiation
- Call Hold and Resume
- Long Duration call
- SIP Session Timer
- Call Waiting
- DND
- Call Forward Always
- Call Forward No Answer
- Call Forward Busy
- Blind/Cold Transfer
- Consultative/Warm Transfer
- 3-party Conference
- VoiceMail
- Call History
- Company Directory
- Speed dial
- Call Monitoring
- Auto Receptionist IVR

- AES-256
- Call Park/Retrieve

1. Overview

This document outlines the configuration best practices for the Grandstream WP810/822/825 as Zoom generic SIP phone.

2. Configuration Steps - Zoom Web Portal

This section provides instructions on how to configure Grandstream WP810 in Zoom Web Portal.

This section is mainly for adding phone devices (WP810) and assign Zoom users to the devices.

Prerequisites:

- Zoom Phone account: a valid Zoom Phone subscription is required in order to assign an WP810 endpoint.
- Zoom approval for provisioning of WP810 as Generic SIP devices. Administrators should contact Zoom Account Executive to start an approval process.

Login to Zoom Web portal at [https://zoom.us/.](https://zoom.us/)

The following Zoom SIP Device configurations are included in this section:

1. **Create Zoom Users**
2. **Add Device**

2.1 Create Zoom Users

Zoom Users are created in order to login to Zoom clients on desktop or mobile, it can also be assigned to SIP Device. The steps for creating a user are as follows:

1. Under Admin, Navigate to **User Management -> Users**. Click **+ Add Users** to create new Zoom users as shown below.

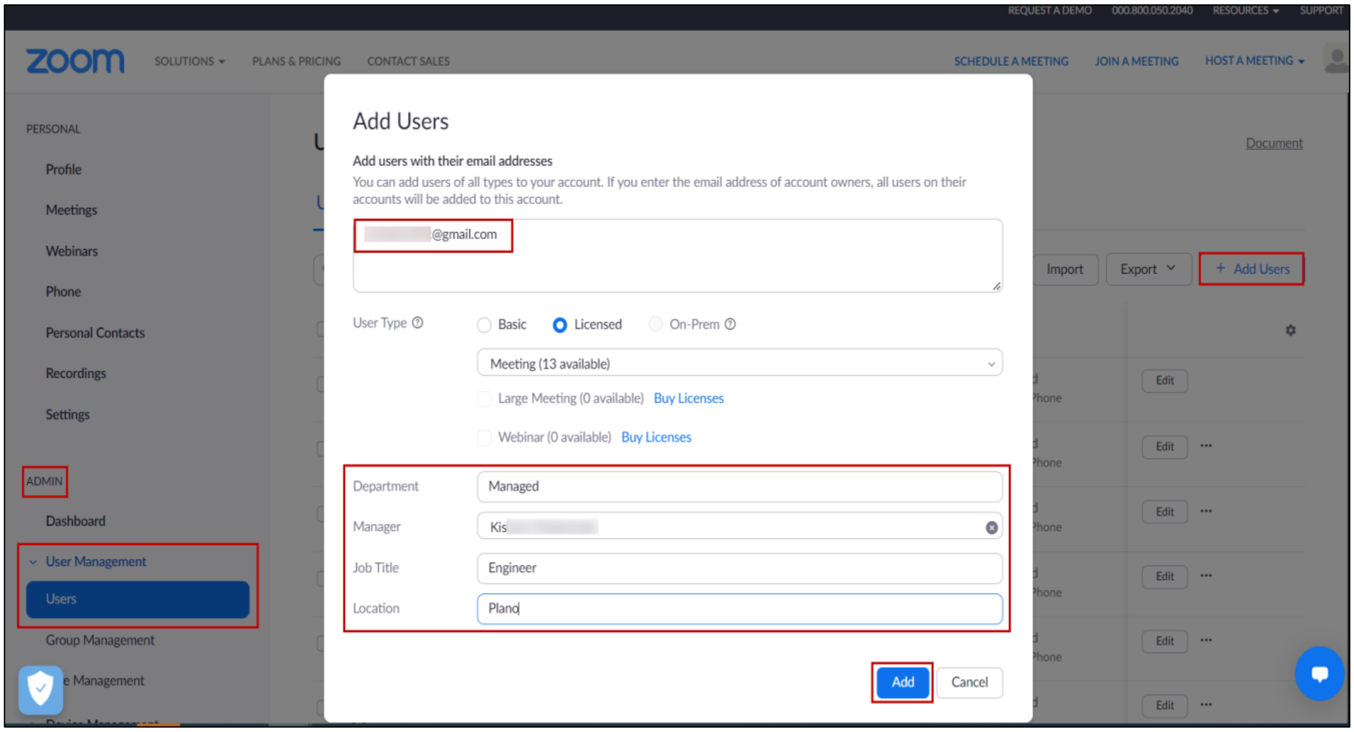

Figure 1 : Add Users

2. A Zoom activation email is sent to the email address while creating the user, follow the instruction to active the zoom account.

3. Navigate to **Phone System Management -> Users & Rooms.** Click **Add.**

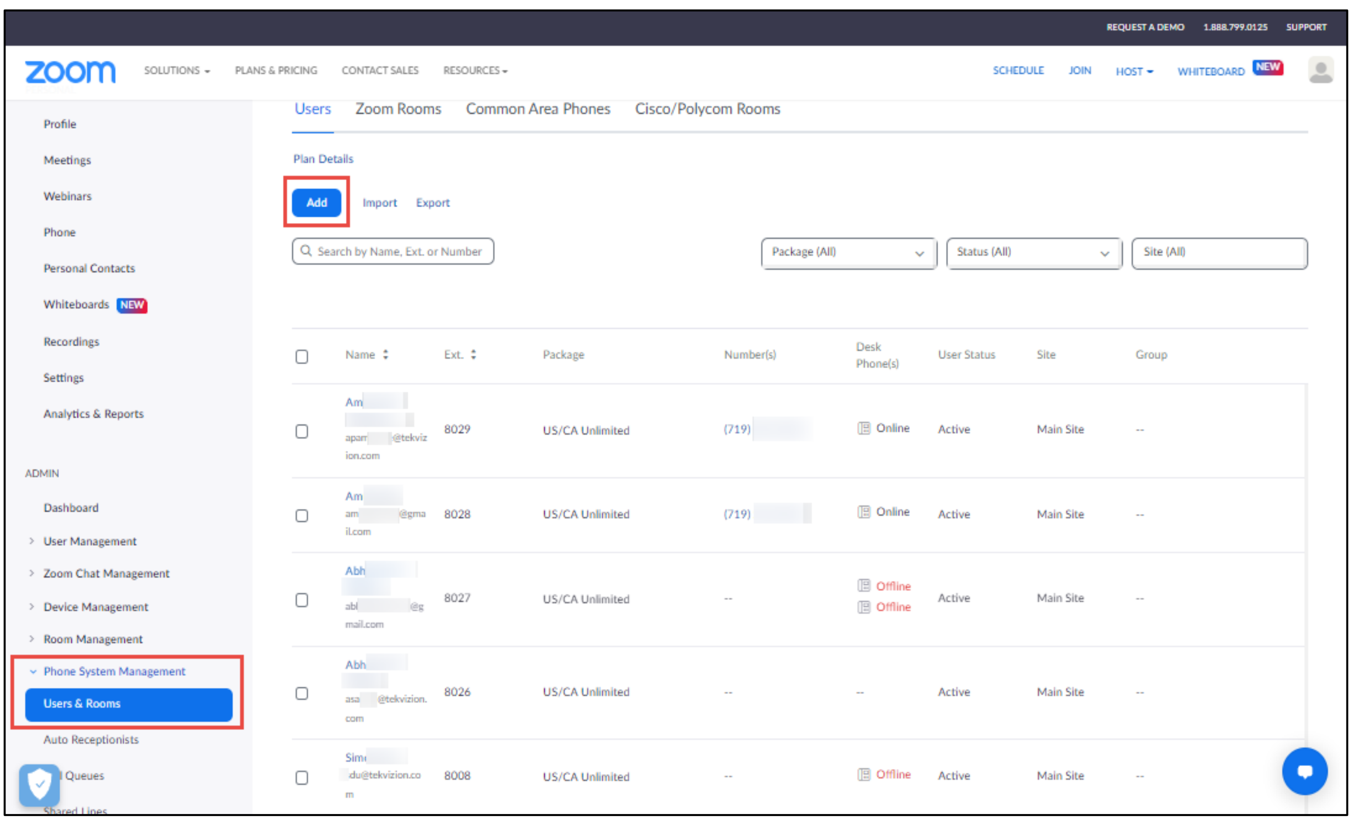

Figure 2 : Add Users and Rooms

4. Click **Choose** beside Users and when the pop-up window opens, select the proper user and **Confirm.**

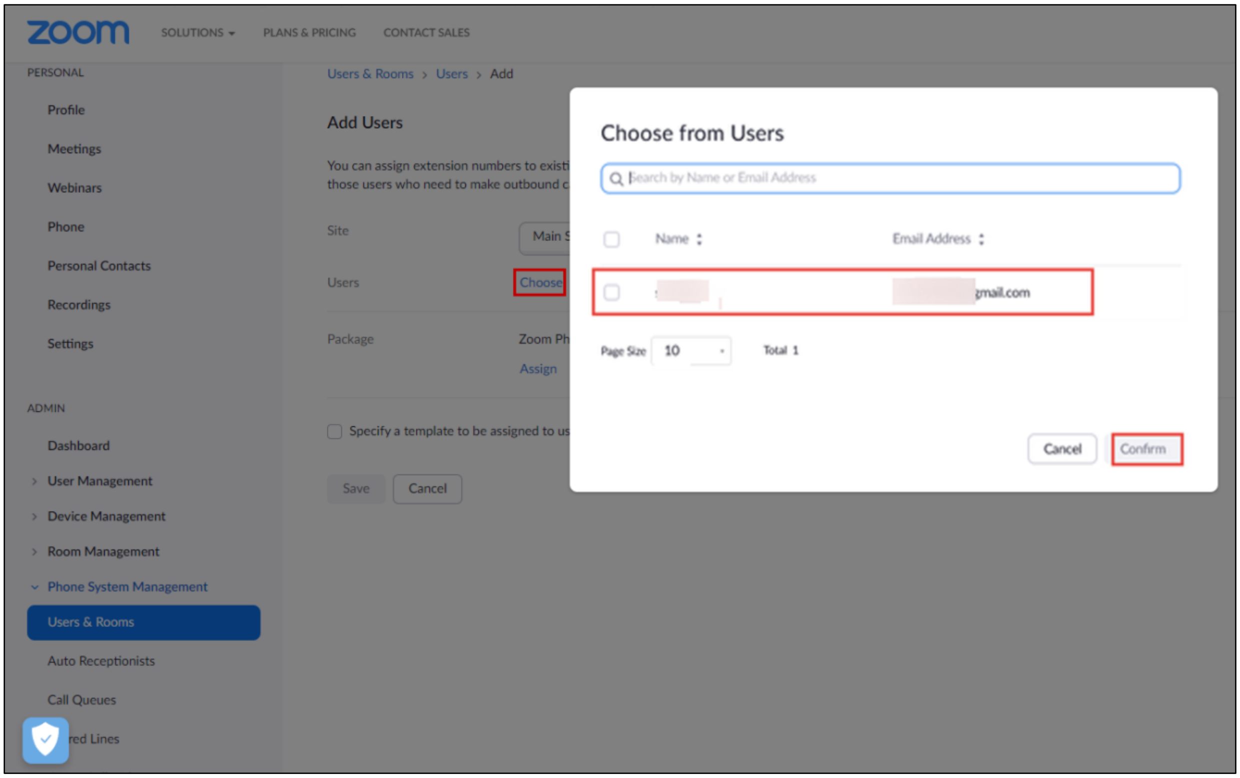

Figure 3 : Choose user

5. Click **Assign** beside Package and at new window, select proper Calling plan, **US/CA Unlimited Calling Plan** was picked up, click **Save** to complete adding users under phone system Management.

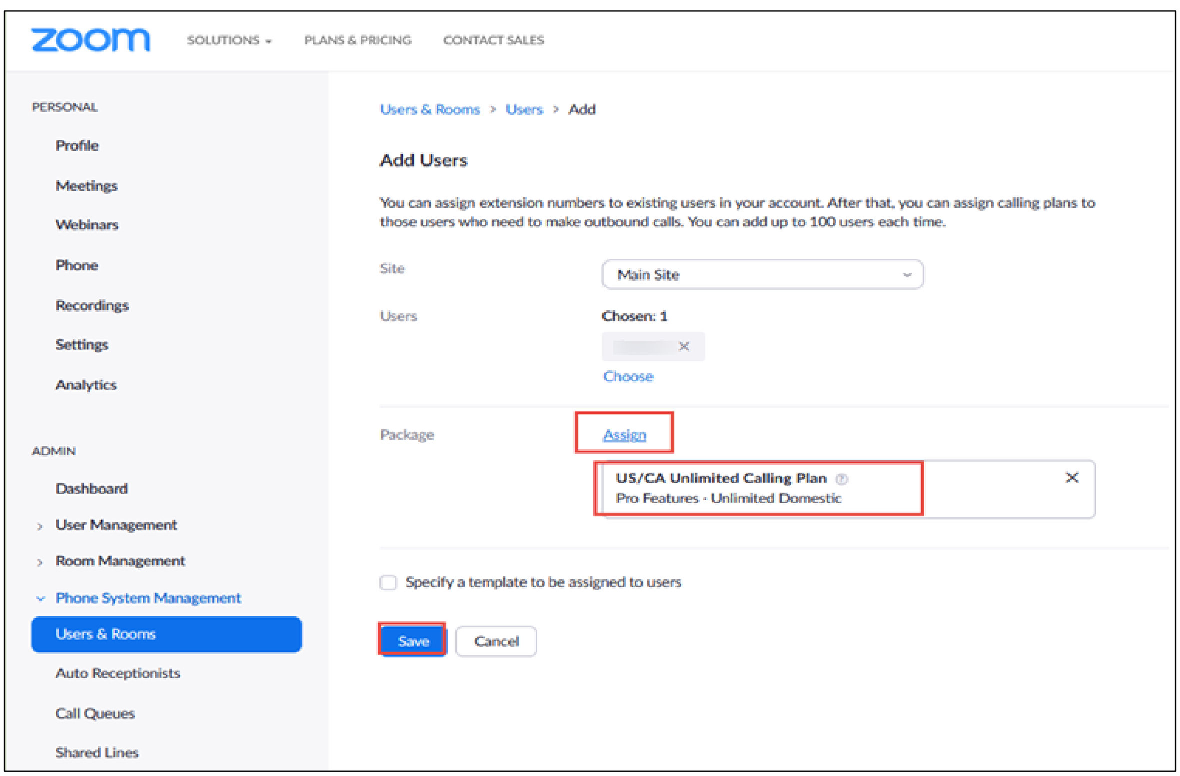

Figure 4 : Assign Calling Plan

6. Select the newly added user, click **Assign** and select **Assign Numbers.**

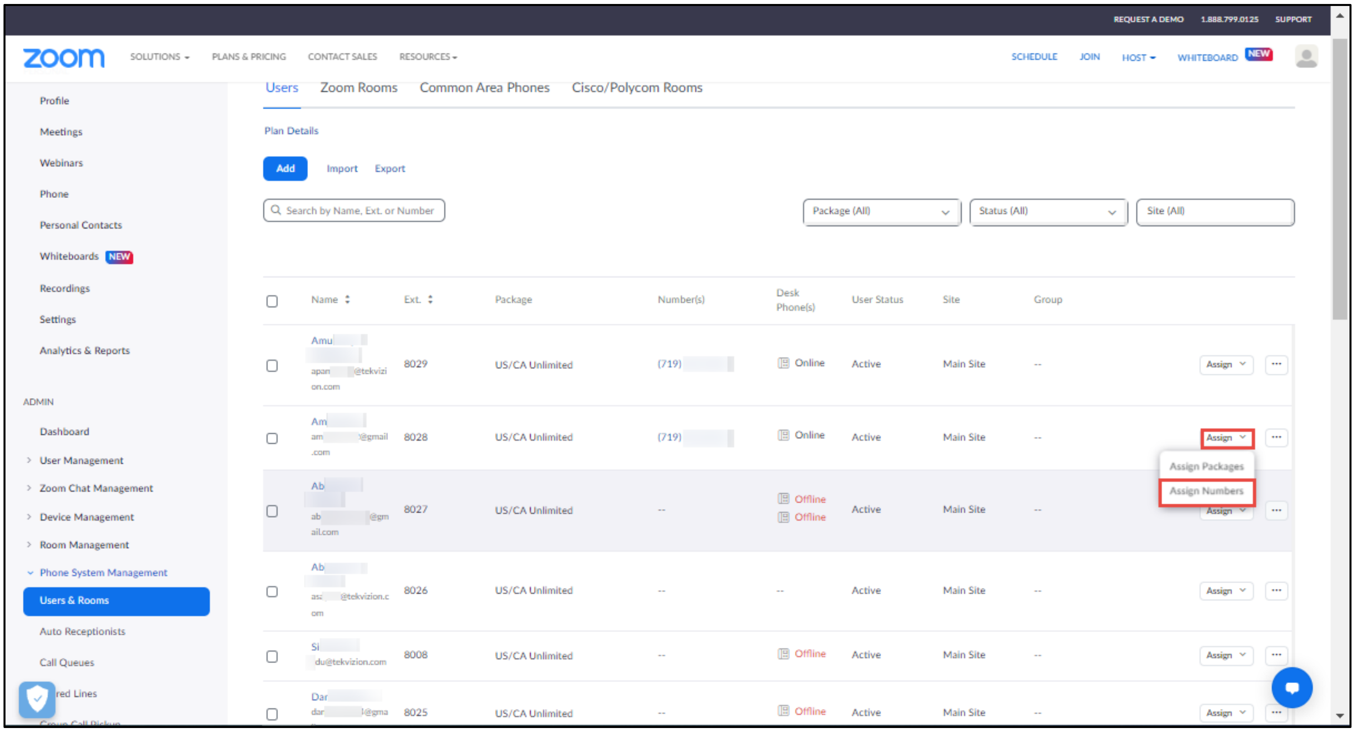

Figure 5 : Assign Number

7. Select the desired DID and click **confirm** to assign the DID to the user.

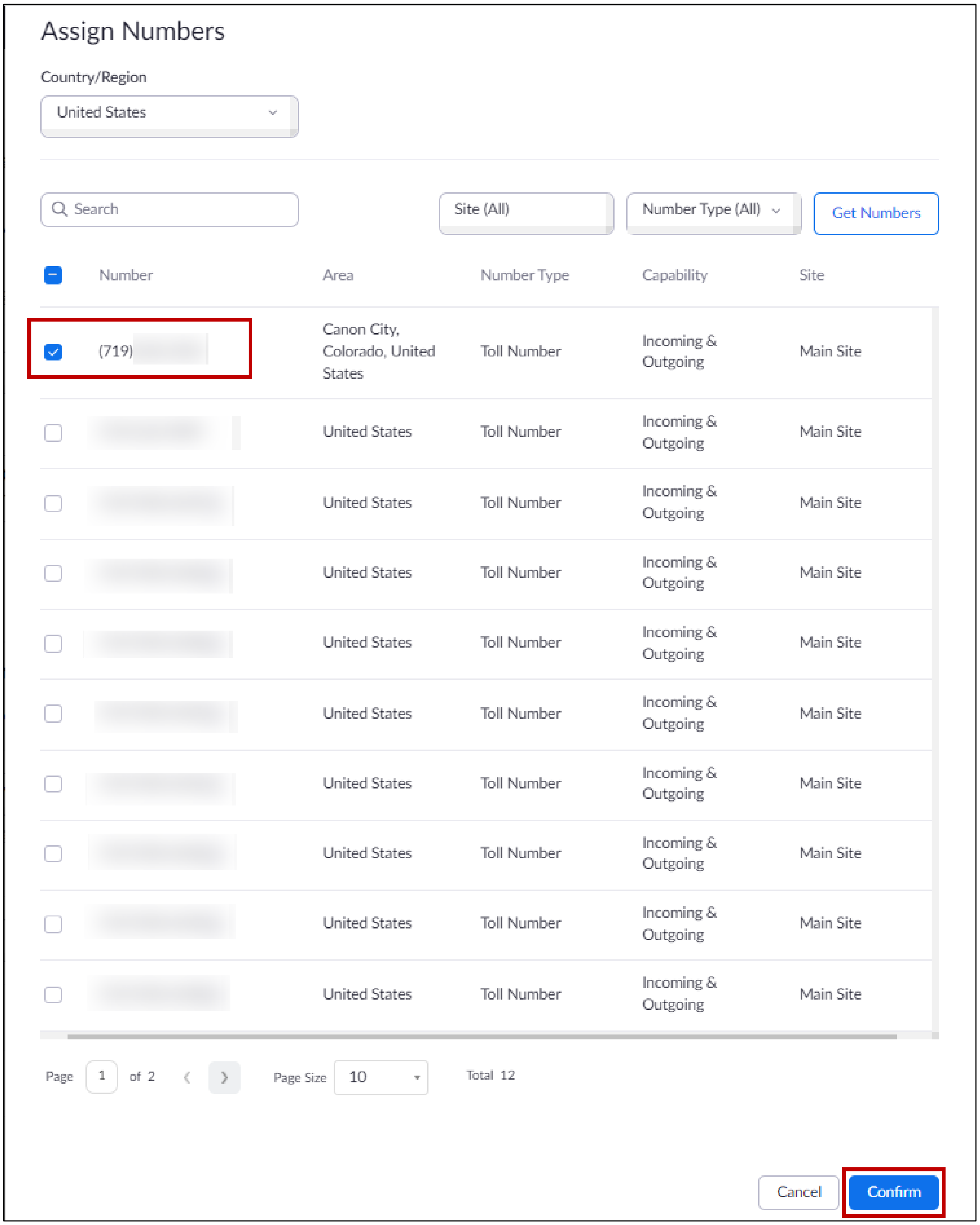

Figure 6 : Select DID Number

2.2 Add SIP Devices

1. Navigate to **ADMIN -> Phone System Management -> Phones & Devices**. Click **Add.**

    - Set **Display Name**: **WP810-2** is set as an example.
    - Set **MAC Address**: add the WP810-2 MAC Address here.
    - Set **Device Type**: select **Other** as the phone type is not certified yet.
    - Click **Assign** under **Assigned To** and select the newly created user in previous steps.
    - Click **Save.**

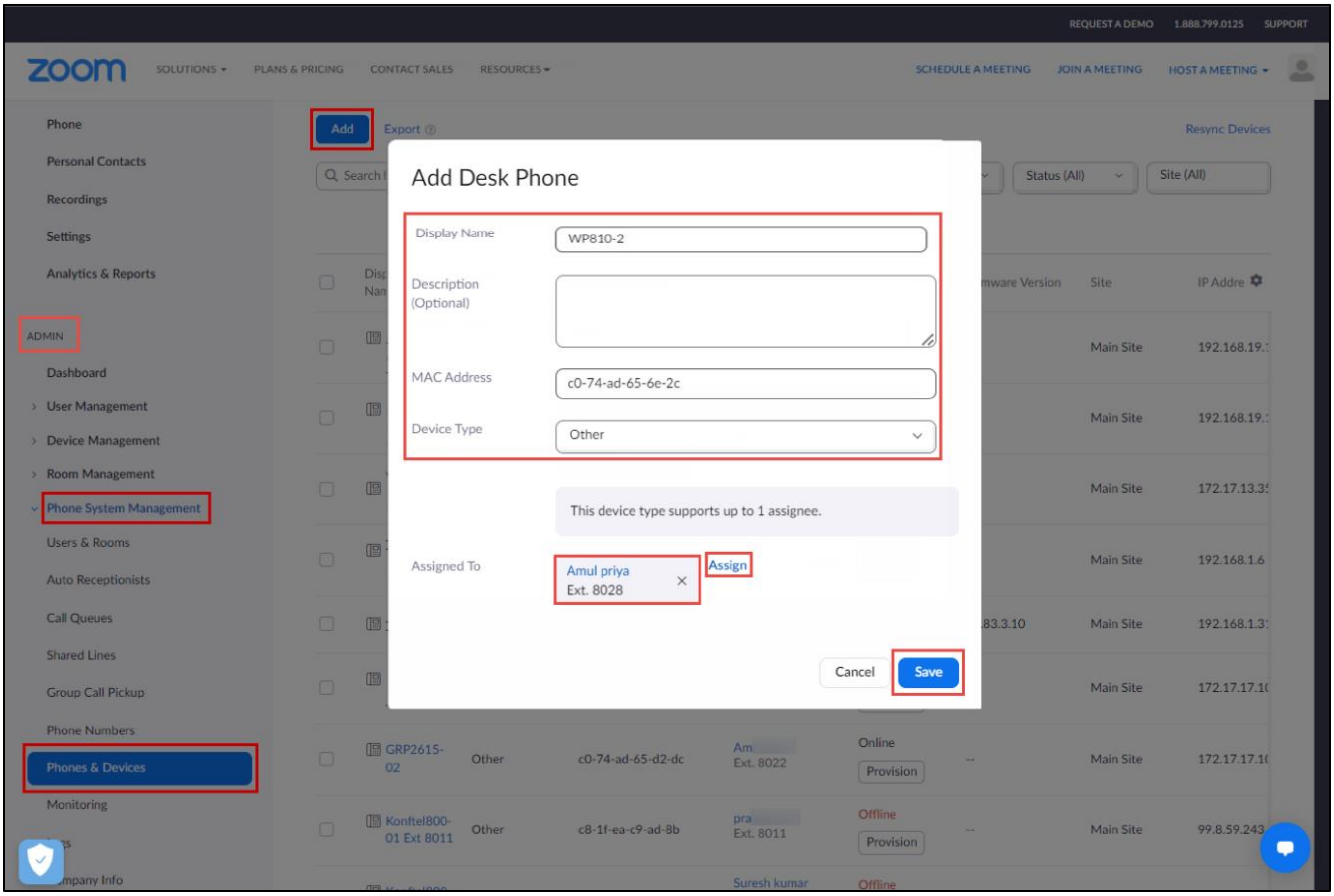

Figure 7 : Add Desk Phone

2. Select the Display Name of the newly created Desk Phone to navigate to its profile and click the **Provision** button. The SIP Account detail is displayed which will be used in the WP810-1 provisioning [(section Provisioning through Phone's Web Interface).](#)

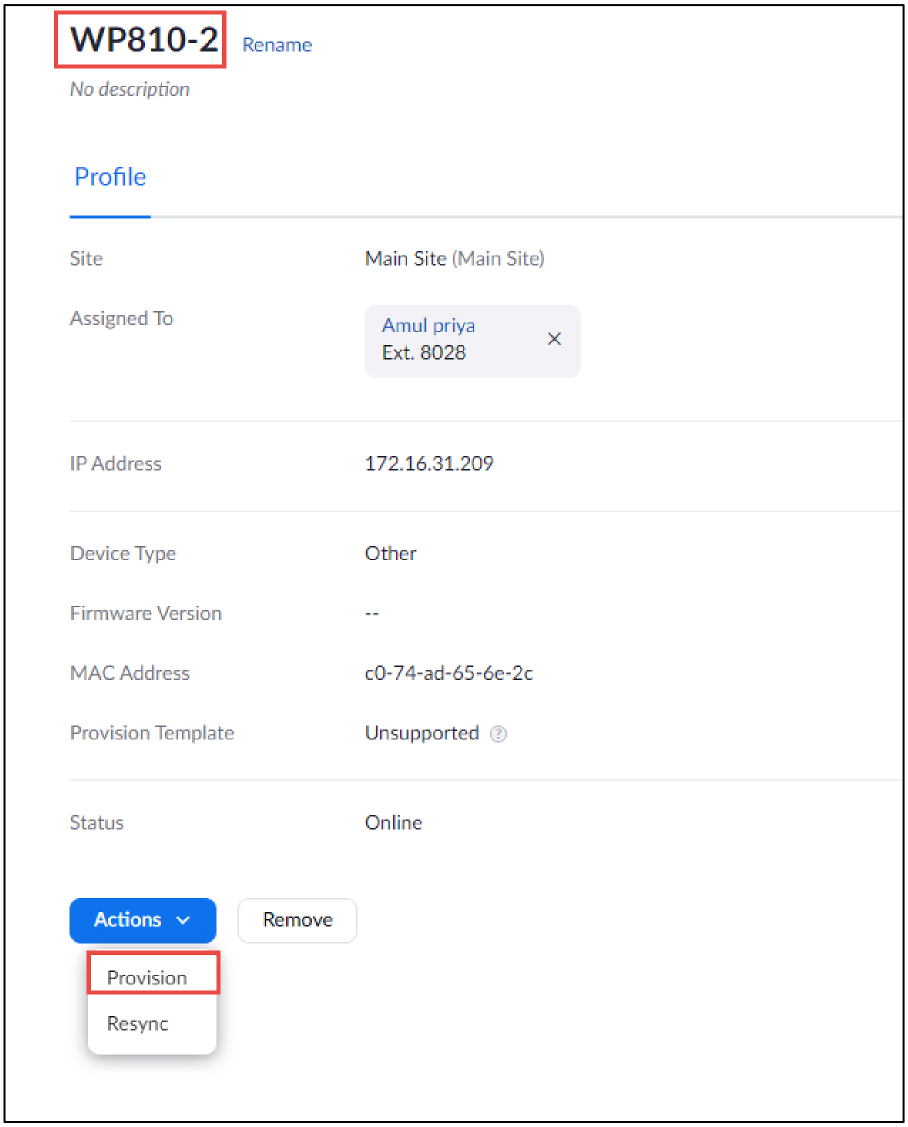

Figure 8 : Provisioning

3. Download the Certificates and import to the device, so that device will be considered in the trust list. **Note:** By default, WP810 natively supports DigiCert CA. So, uploading certificates manually is not required.

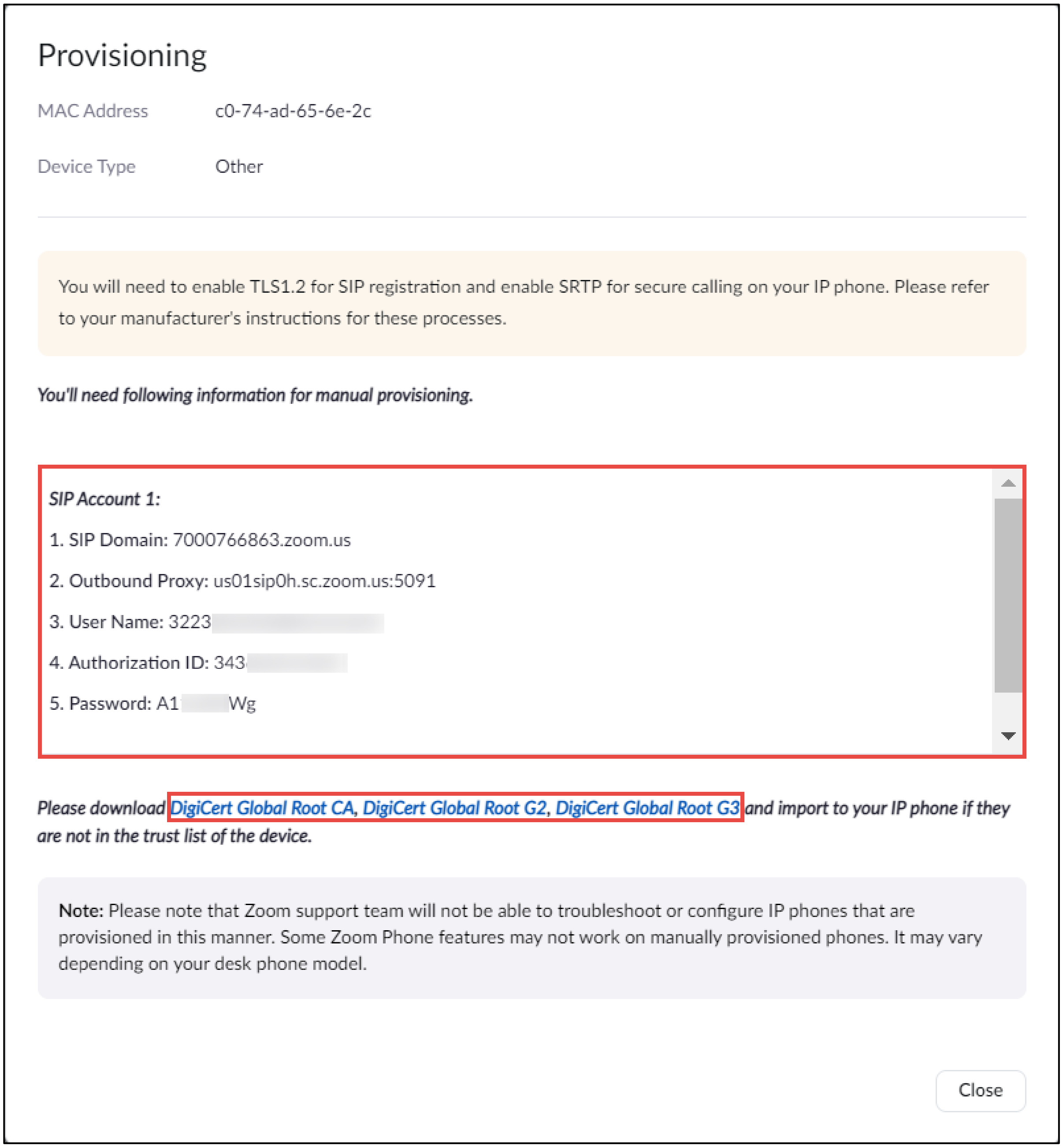

Figure 9 : Provisioning (Cont.,)

3. Grandstream WP810 Provisioning

This section provides instructions on how to configure Grandstream WP810 to register to Zoom Phone Services.

3.1 Deployment Topology Diagram

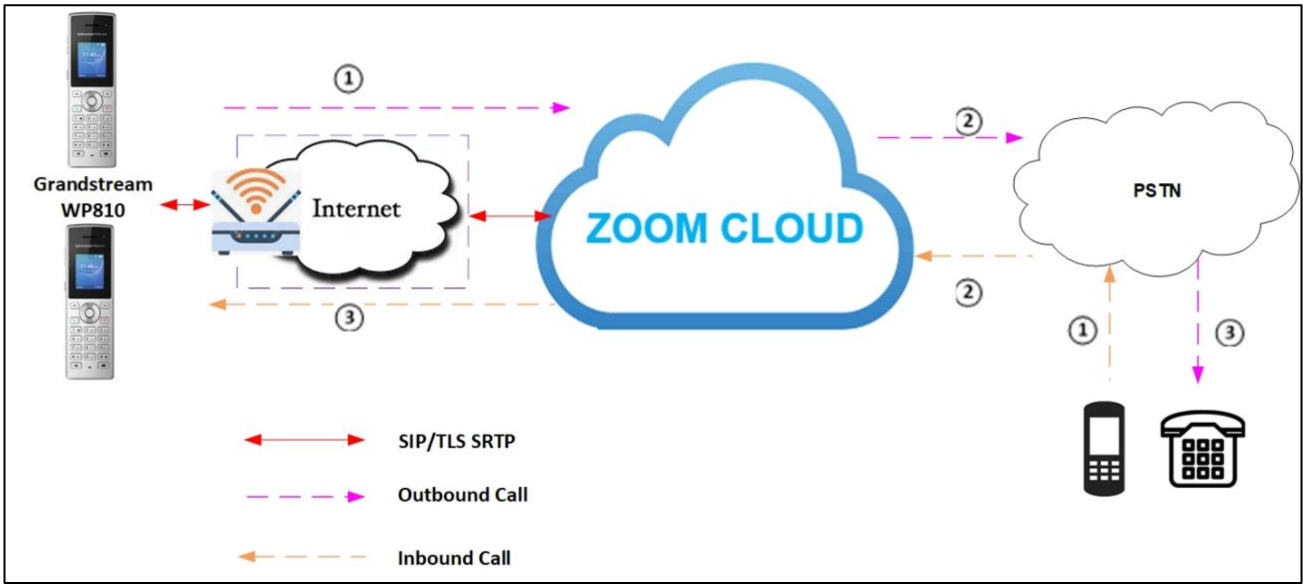

Figure 10 : Network Diagram

3.2 Network

By default, WP810 has DHCP mode enabled, if the router to which phone is connected does not support DHCP, you can configure static IP manually. You can find the IP address of WP810 by pressing the UP-arrow button from WP810.

3.3 Firmware Upgrade

This section ensures the phone is upgraded with the required firmware. The firmware used for this test is **1.0.11.16** From the phone's home page, Navigate to: **Maintenance -> Upgrade and Provisioning**.

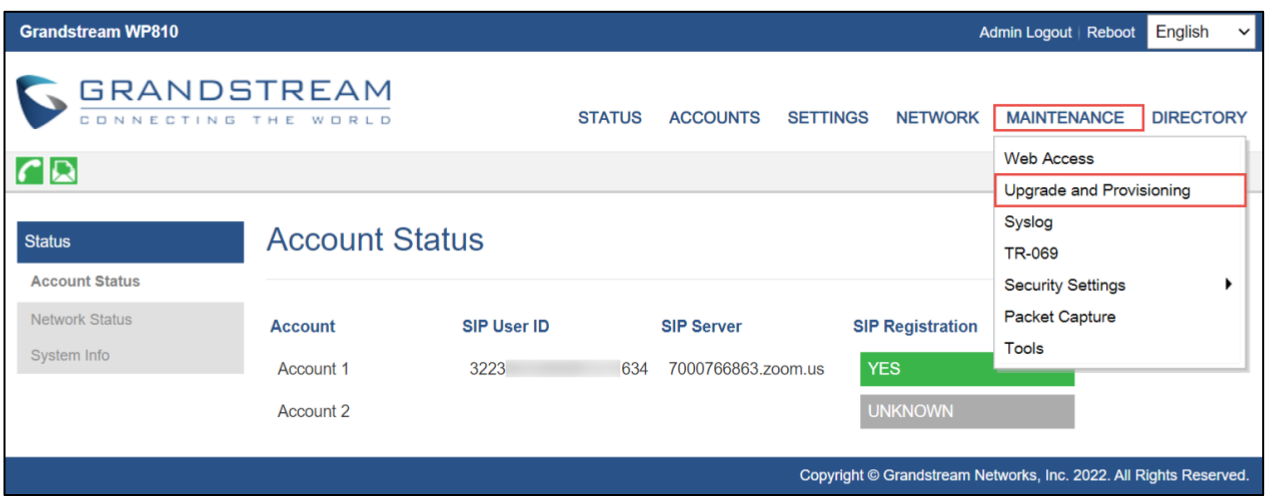

Figure 11 : Firmware Upgrade

- Click **Start** for Upgrade.
- Select and upload the firmware file from the system and then click **Start**.

- WP810 phone displays Upgrade Available. Enter **Yes** to continue (Phone may have to restart to get applied with the changes).

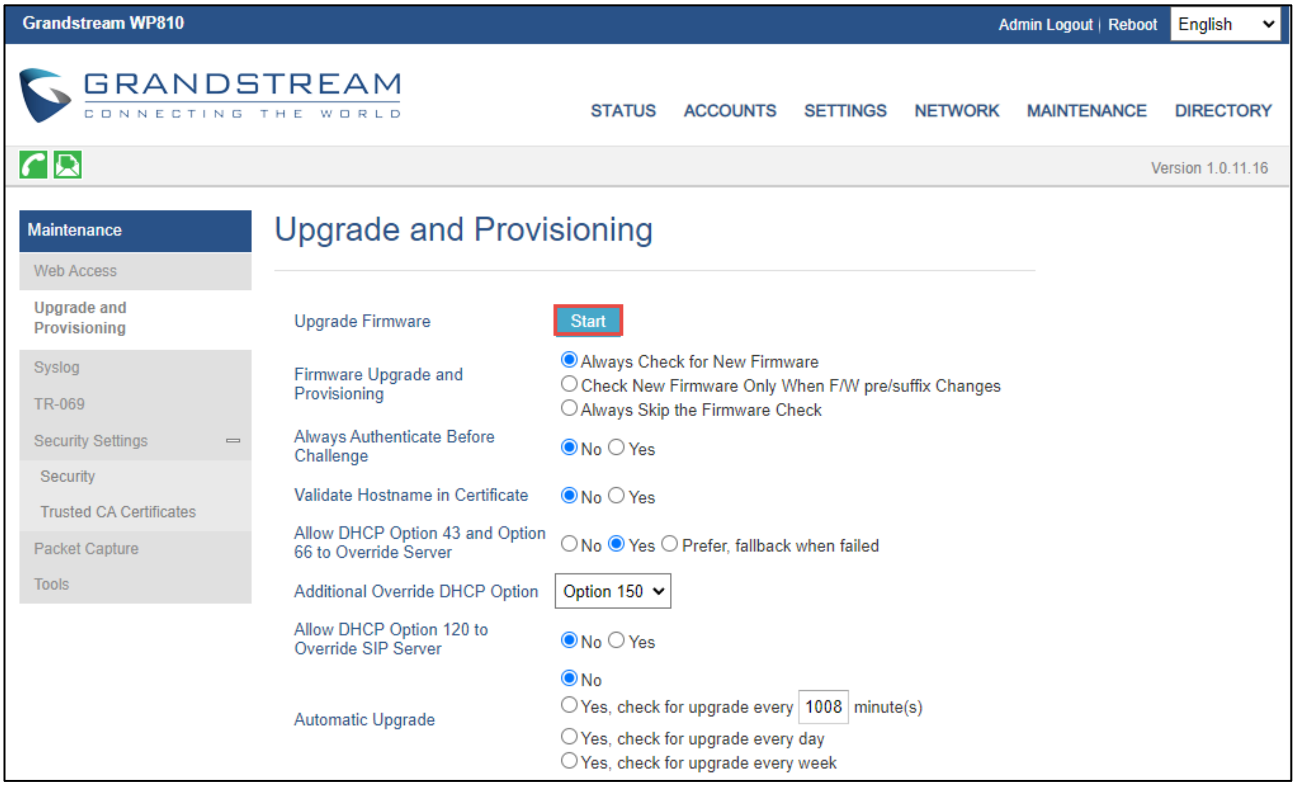

Figure 12 : Firmware Upgrade (Cont.)

3.4 Provisioning

WP810 can be provisioned in two ways:

1. Provisioning through a HTTP Server
2. Provisioning through Web Interface

3.4.1 Provisioning through a HTTP server

1. In a HTTP Server, upload the below file in order for the phone to download the necessary configuration files and get provisioned. They are,

```
<?xml version="1.0" encoding="UTF-8" ?>
<!-- Zoom IOT Provisioning Template - 2 Accounts-->
<gs_provision version="1">
        <config version="1">

<!-- Upgrading and Provisioning Settings -->
                <!-- Firmware Upgrade Protocol. 0 - TFTP, 1 - HTTP, 2 - HTTPS -->
                <P6767>1</P6767>
                <!-- Config Upgrade Protocol. 0 - TFTP, 1 - HTTP, 2 - HTTPS -->
                <P212>1</P212>
                <!-- Firmware Server Path -->
```

```
<P192>http://172.XX.XX.XXX:90/Folder Path</P192>
        <!-- Config Server Path -->
        <P237>http://172. XX.XX.XXX:90/Folder Path</P237>

        <!-- Syslog Server -->
        <P207>172. XX.XX.XXX </P207>
        <!-- Syslog Level. 0 - NONE, 1 - DEBUG, 2 - INFO, 3 - WARNING, 4 - ERROR. Default is 0 -->
        <P208>1</P208>
        <!-- Send SIP Log. 0 - No, 1 - Yes. Default is 0 -->
        <P1387>1</P1387>

        <!-- Use Random Port. 0 - No, 1 - Yes. Default is 1 -->
        <P78>0</P78>

        <!--  Account Settings  -->

        <!-- Account 1 -->
<!-- Account Active (In Use). 0 - No, 1 - Yes -->
<P271>1</P271>
<!-- Account Name. -->
<P270>6E2C</P270>
<!-- SIP User ID -->
<P35>32238XXXXXXXXXXXX634</P35>
<!-- Authenticate ID -->
<P36>3436XXXXX223</P36>
<!-- Authenticate password -->
<P34>A1XXXXXg</P34>
<!-- Display Name (John Doe) -->
<P3>8028</P3>
<!-- SIP Server -->
<P47>7000766863.zoom.us</P47>
<!-- Outbound Proxy -->
<P48>us01sip0h.sc.zoom.us:5091</P48>
<!-- DNS Mode. 0 - A Record, 1 - SRV, 2 - NAPTR/SRV. -->
<P103>1</P103>
    <!-- DNS SRV Failover Mode. 0 - Default, 1 - Saved one until DNS TTL, 2 - Saved one until no
    response, 3 - Failback follows failback expiration timer. -->
    <P26040>2</P26040>
<!-- SIP Registration. 0 - No, 1 - Yes -->
<P31>1</P31>
<!-- Register Expiration (in minutes. default 1 hour, max 45 days) -->
<P32>60</P32>
    <!-- SIP Transport. 0 - UDP, 1 - TCP, 2 - TLS/TCP. Default is 0-->
    <P130>2</P130>
    <!-- SRTP Mode. 0 - Disabled, 1 - Enabled but not forced, 2 - Enabled and forced, 3 - Optional. Default
    is 0 -->
    <P183>2</P183>
    <!-- SRTP Key Length. 0 - AES 128&256 bit, 1 - AES 128 bit, 2 - AES 256 bit. Default is 0 -->
<P2383>0</P2383>
```

```
        <!-- Enable Session Timer. 0 - No, 1 - Yes. Default is 0 -->
    <P2395>1</P2395>
        <!-- Caller ID Display. 0 - Auto, 1 - Disabled, 2 - From Header. Default is 2 -->
    <P2324>0</P2324>
        <!-- NAT Traversal. 0 - No, 1 - STUN, 2 - keep alive, 3 - UPnP, 4 - Auto, 5 - VPN. Default is 4 -->
        <P52>0</P52>
    <!-- Enable Local Call Features. 0 - No, 1 - Yes -->
    <P191>1</P191>
        <!-- Special Feature. 137 - Zoom -->
    <P198>137</P198>


     </config>
</gs_provision>
```

2. From Web UI of phone, Navigate to **Maintenance -> Tools.**

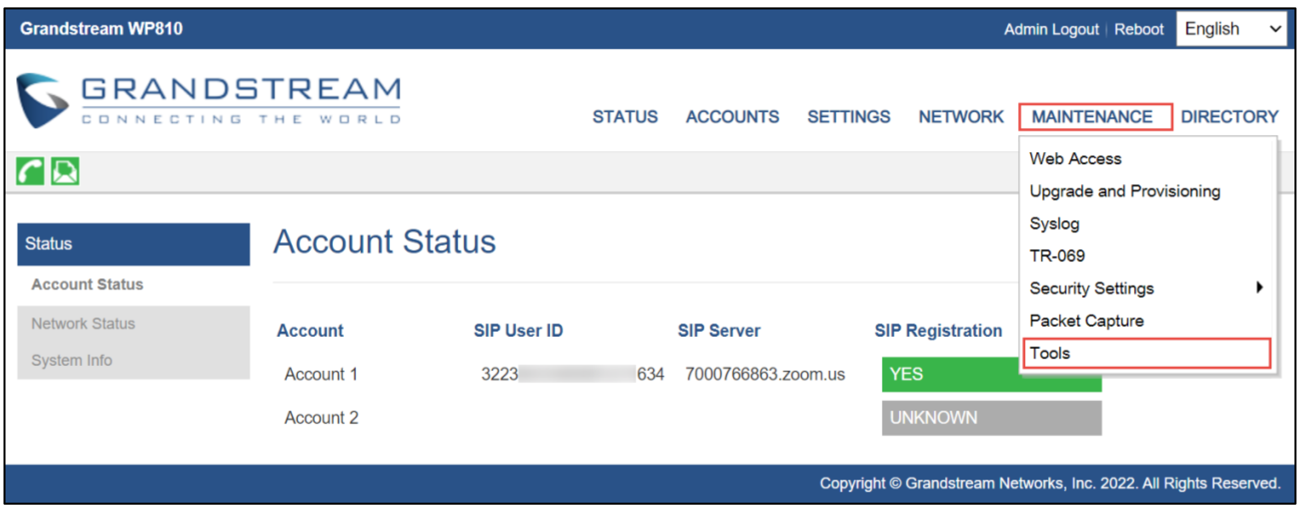

Figure 13 : Factory reset

3. Click **Start** to Factory Reset.

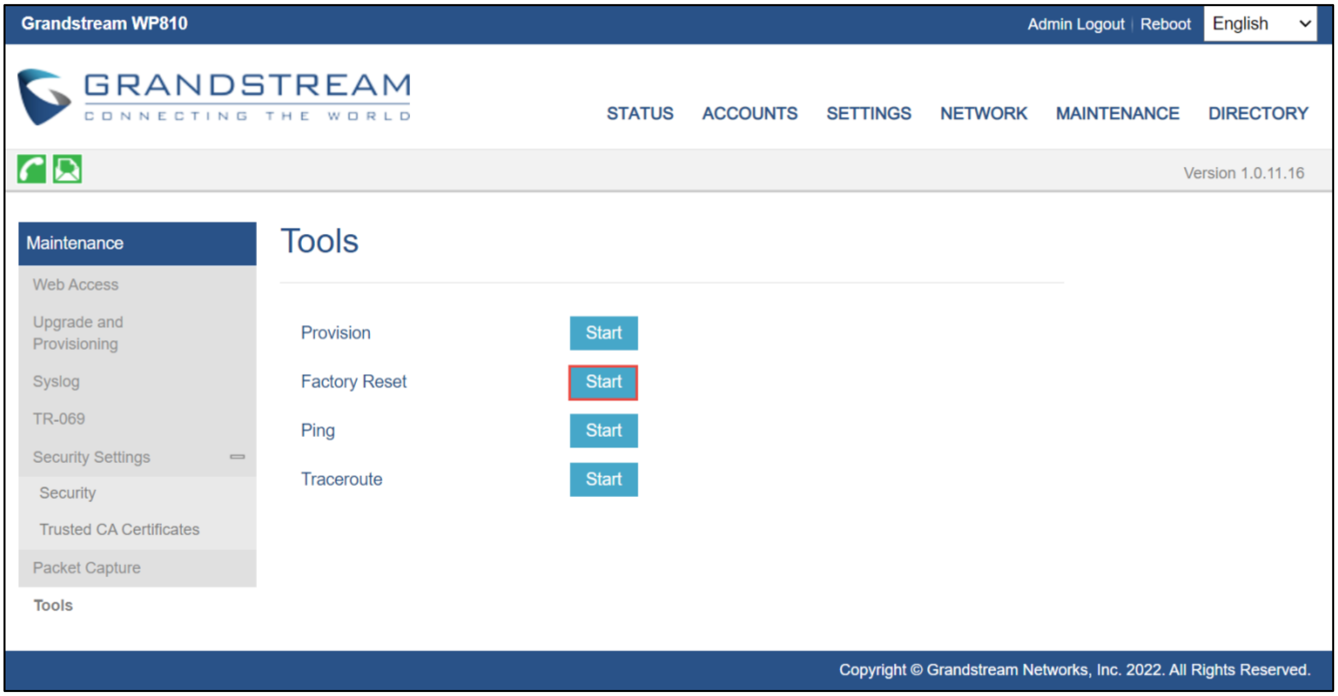

Figure 14 : Factory reset - (Cont.)

4. Give **OK** to confirm to factory reset the phone.

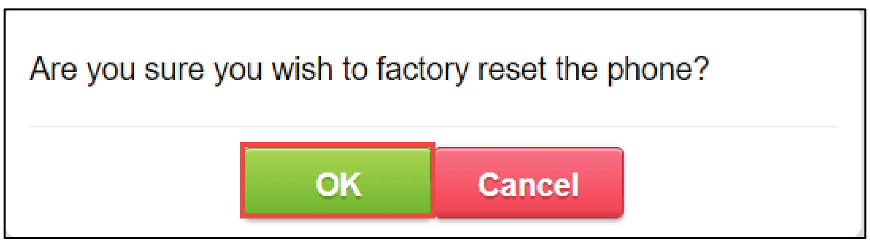

Figure 15 : Factory reset - (Cont.)

5. Once the Phone reboots, enter the phone's IP address in the browser's address bar **(https://ip-address)** and enter. Input phone's **username and password.** The default administrator username is "**admin**", and the random password can be found on the sticker at the back of the unit.

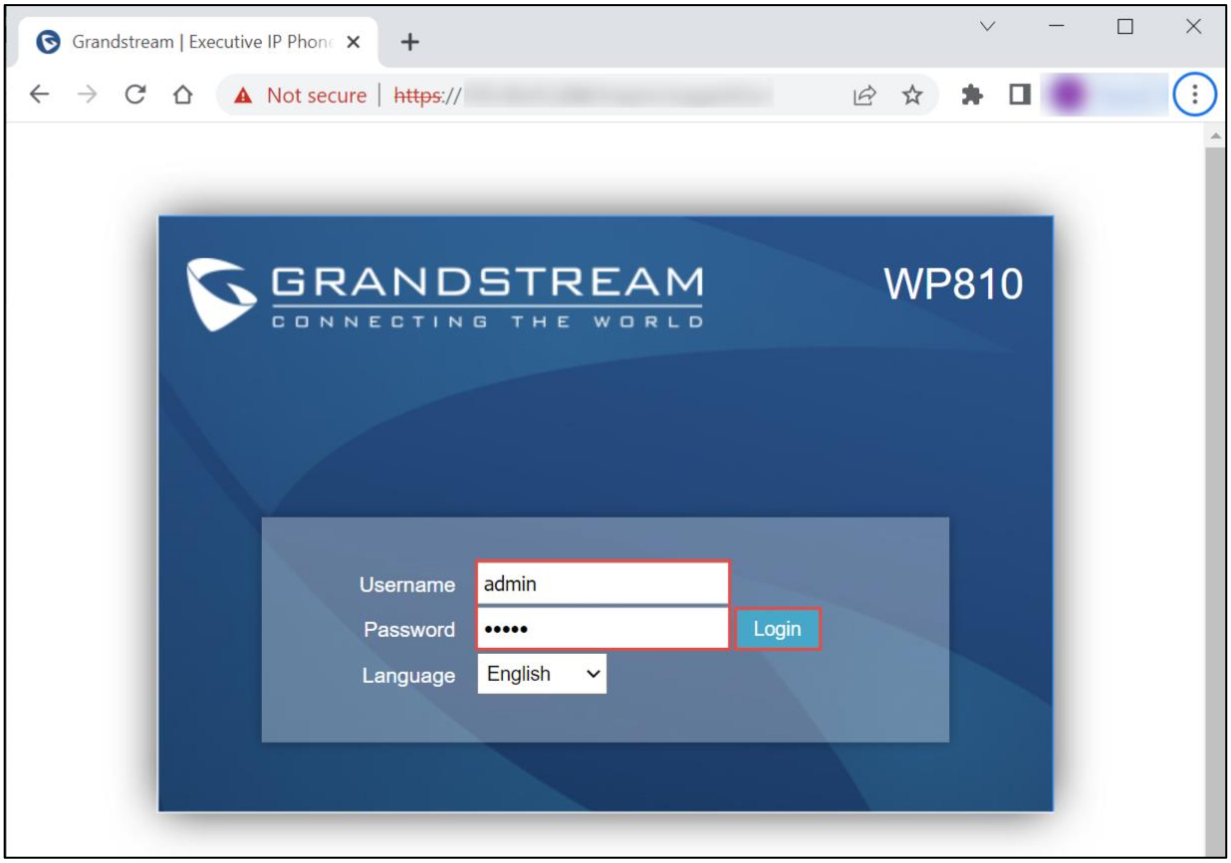

Figure 16 : Login details

6. Once Logged in, Update with the New Password instead of Default password. Navigate to **Maintenance -> Web Access** to change the Password.

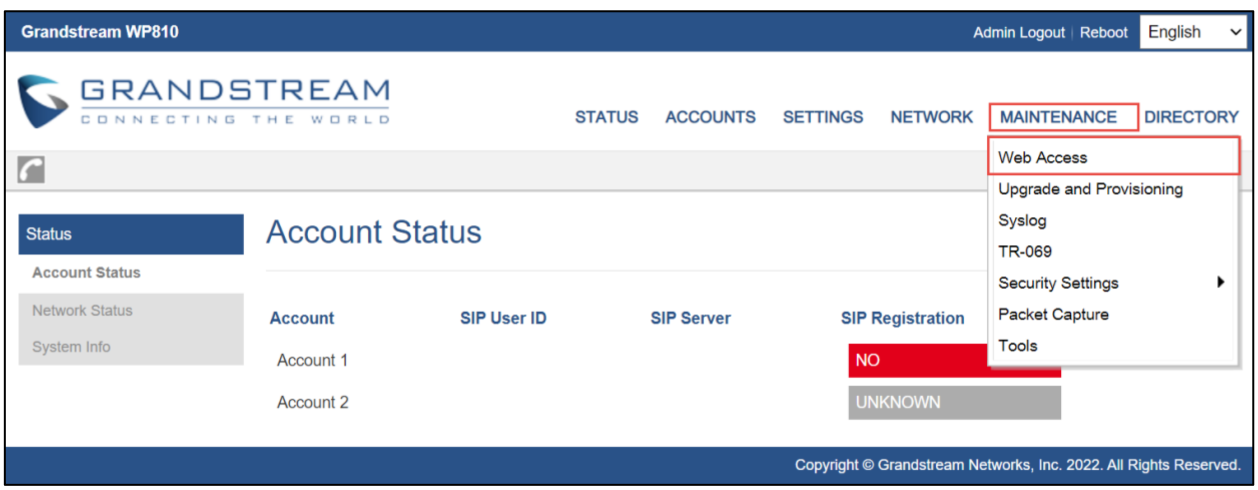

Figure 17 : Login details - (Cont.)

7. Under the **Admin Password,** Enter the **Current Password (i.e., found on the sticker at the back of the unit)**, Type in your new admin password in **New Password** field. Type in again same entered password

in **Confirm Password** field. Click **Save.** On saving, a notification appears saying that Password changed successfully.

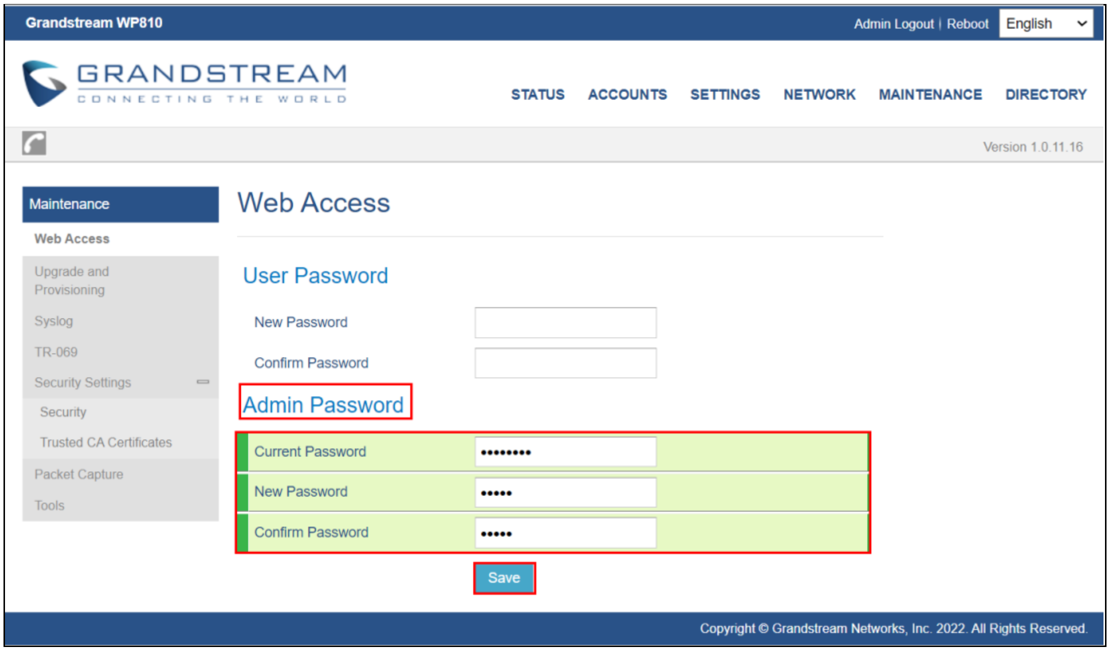

Figure 18 : Login details - (Cont.)

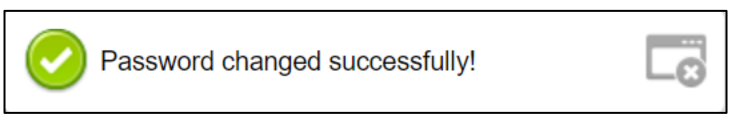

Figure 19 : Password change

8. From the Web UI of phone, Navigate to **Maintenance -> Upgrade and Provisioning**.

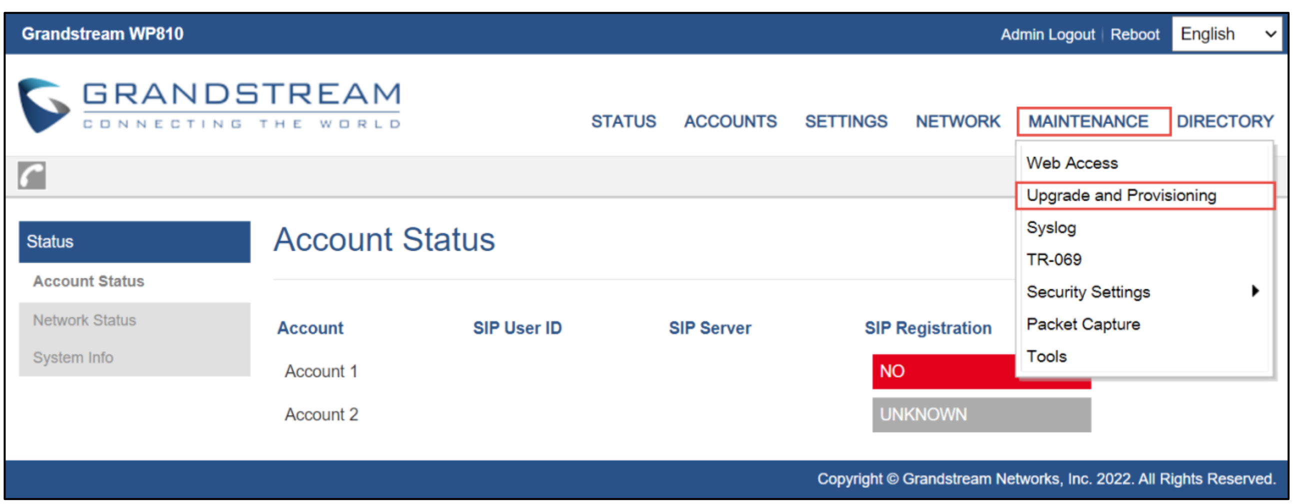

Figure 20 : Provisioning through a HTTP server

9. Scroll down and under **Config** section. Check **HTTP** in Config Upgrade via and enter **(http://ipaddress:90/Folder Path)** in Config Server Path.

10. Click **Save and Apply.**

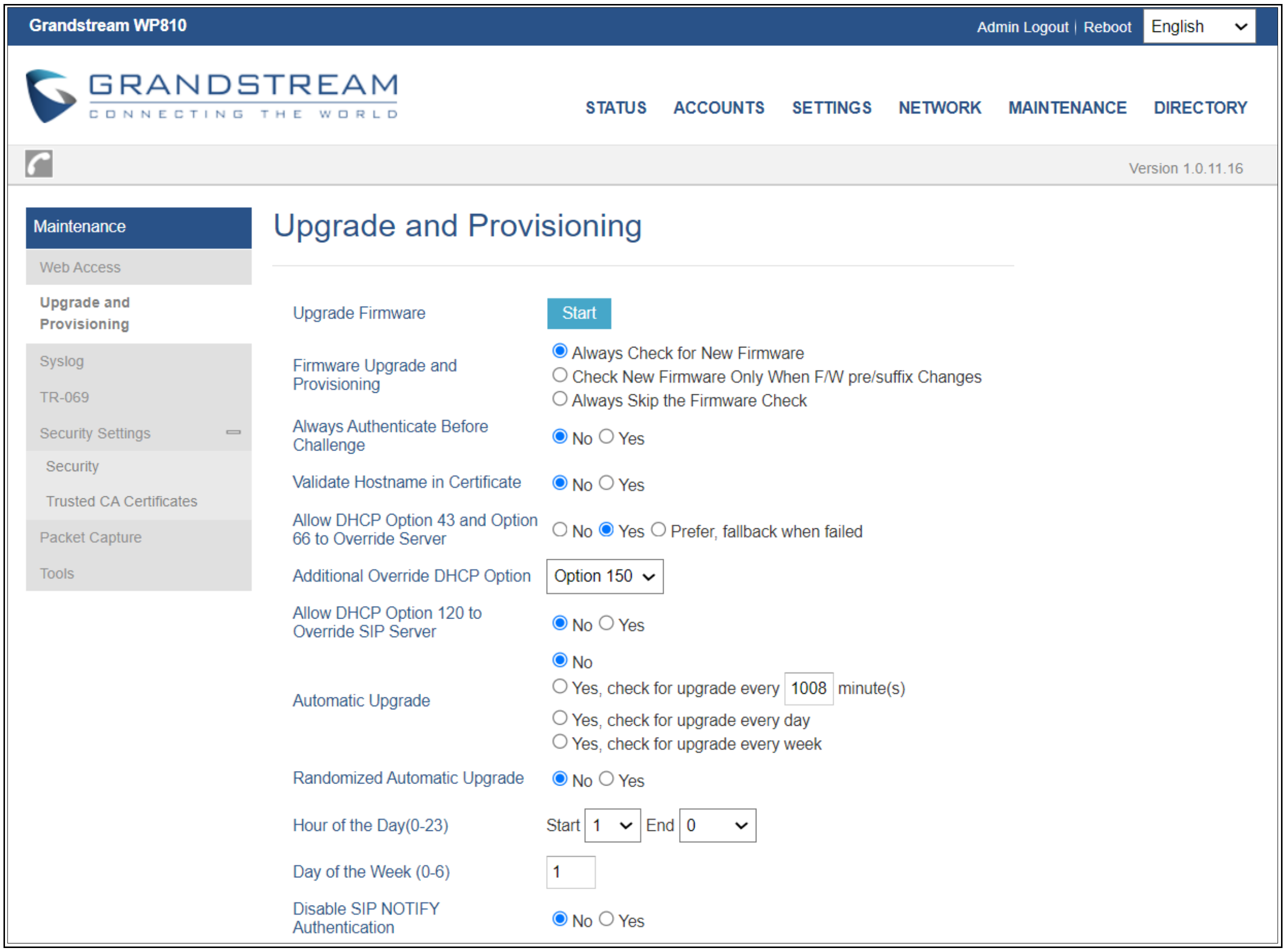

Figure 21 : Provisioning through a HTTP server - (Cont.)

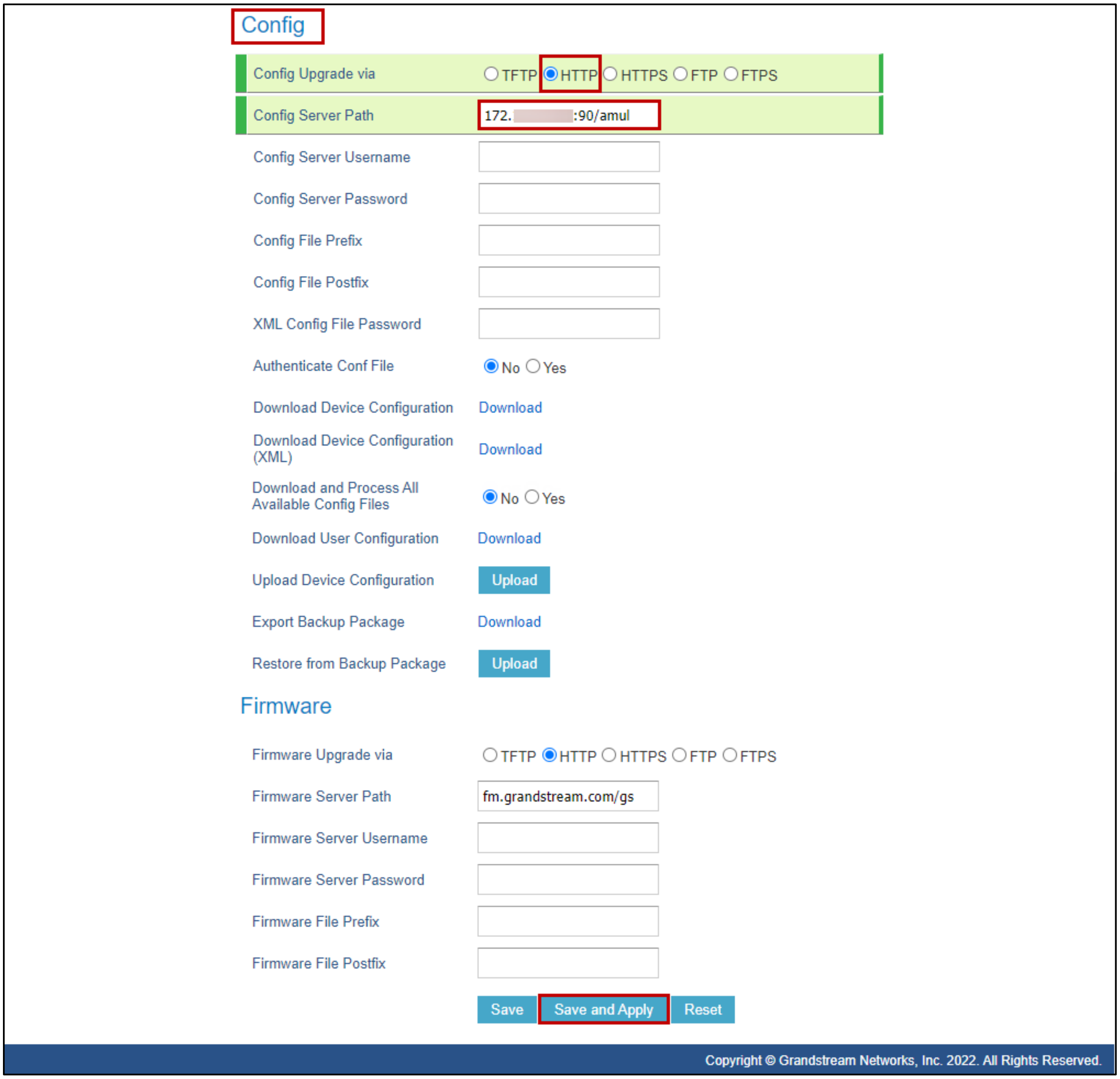

Figure 22 : Provisioning through a HTTP server - (Cont.)

11. After the phone restarts, the phone will start downloading the necessary configuration files from the HTTP server and get provisioned.

12. Navigate to **Status -> Account Status** to verify the provision status.

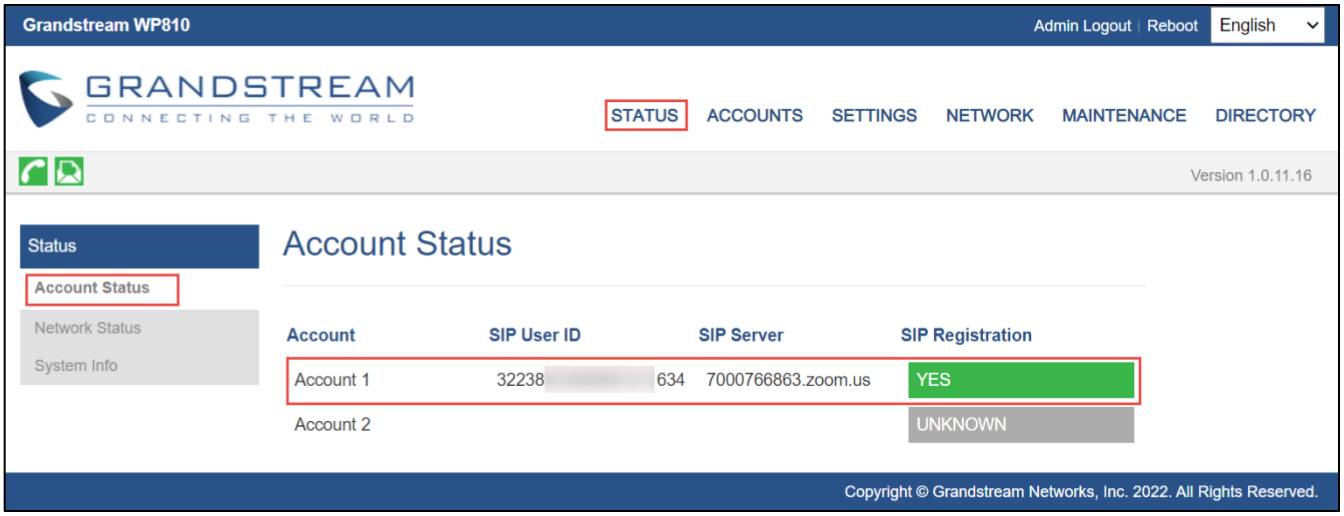

Figure 23 : Provisioning through a HTTP server - (Cont.)

3.4.2 Provisioning through Phone's Web Interface

This section explains how the device can be registered in Zoom portal using SIP Account details that is populated in Zoom portal (explained earlier in 2. Add SIP Devices -> Provisioning).

- WP810 configuration is done via web interface, enter the phone's IP address in the browser's address bar (https://ip-address) and enter. Input phone's username and password and click Login.
- Navigate to **Accounts -> Account 1.**
- Under Account 1 **-> General Settings**. Ensure the below parameters are set.
  1. Account Active: Set to **"Yes".**
  2. Account Name: Enter Account Name of your choice. Here, **6E2C** is used.
  3. SIP Server: **7000766863.zoom.us** (Provided by Zoom).
  4. Outbound Proxy: **us01sip0h.sc.zoom.us:5091** (Provided by Zoom).
  5. SIP User ID: **Provide the SIP User ID** (from Zoom provisioning).
  6. Authenticate ID: **Provide the Authenticate ID** (from Zoom provisioning).
  7. Authenticate Password: **Provide the Authenticate Password** (from Zoom provisioning).
  8. Name: Enter the Name of your choice. Here, **8028** is used.
  9. Click **Save and Apply.**

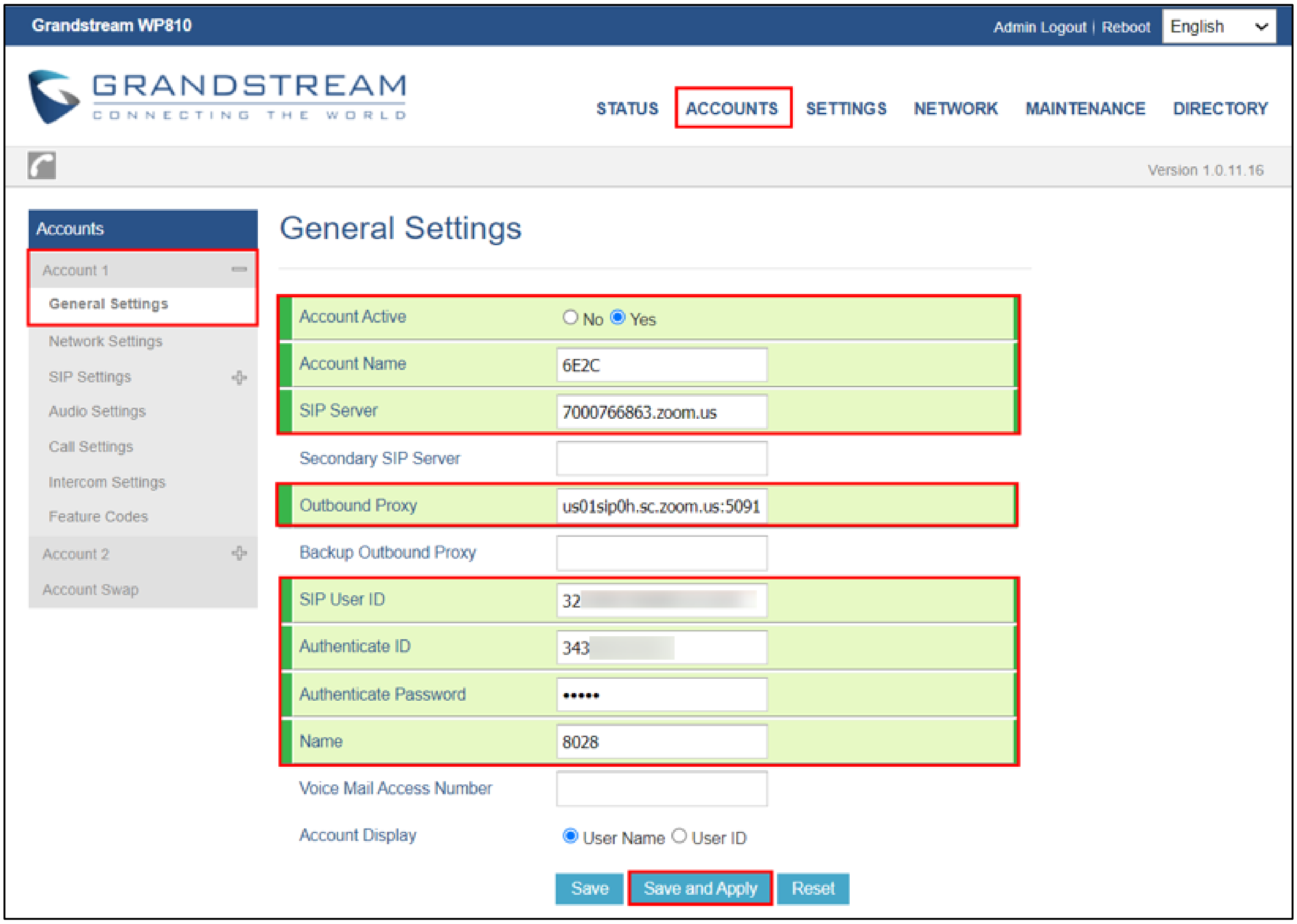

Figure 24 : General Settings

- Under Account 1 **-> Network Settings**. Ensure the below parameters are set.

  1. DNS Mode: **SRV**
  2. DNS SRV Failover Mode: **Saved one until no response**
  3. NAT Traversal: **No**
  4. Click **Save and Apply**

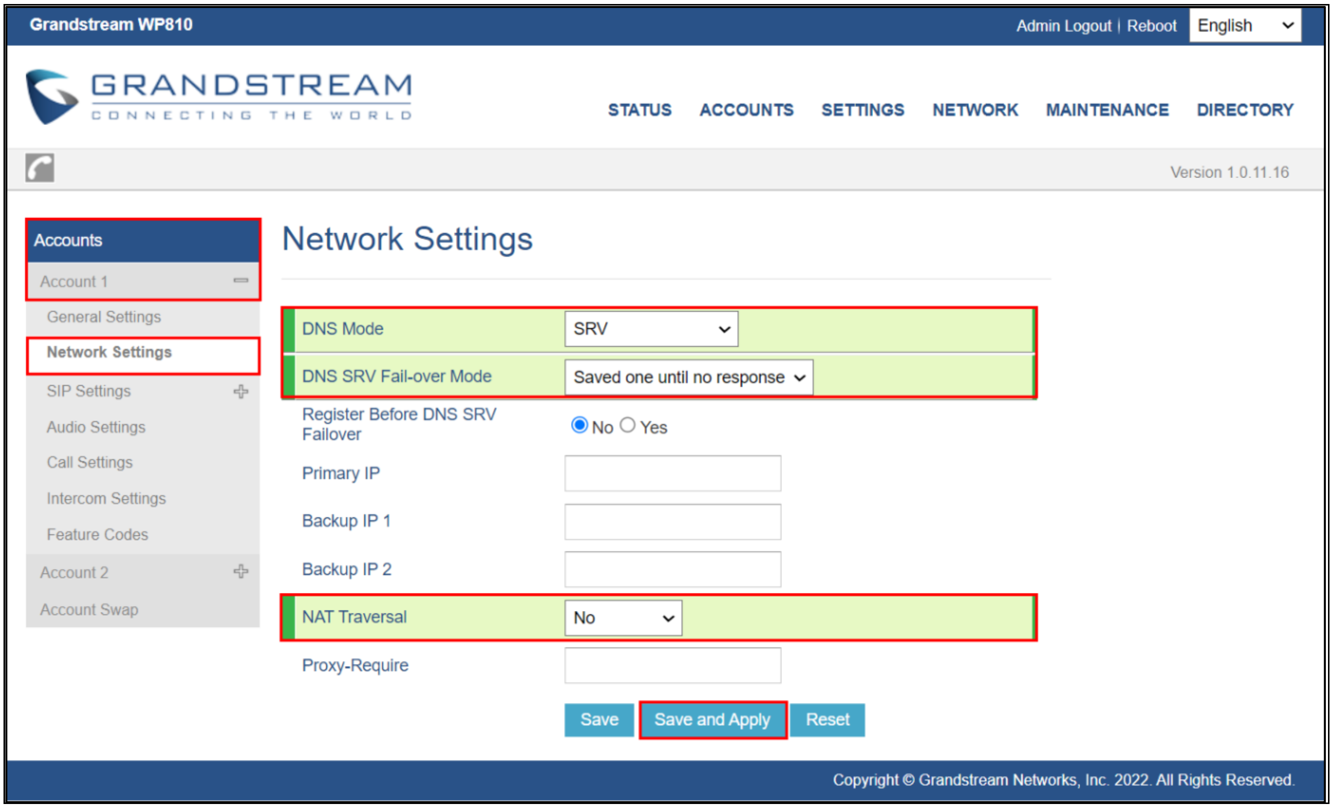

Figure 25 : Network Settings

- Under Account 1 **-> SIP Settings -> Basic Settings**. Ensure the below parameters are set.

    1. SIP Registration: Set to “**Yes**”
    2. Register Expiration: **60**
    3. SIP Transport: Set to “**TLS/TCP**”
    4. Caller ID Display: Set to **“Auto”**
    5. Click **Save and Apply**

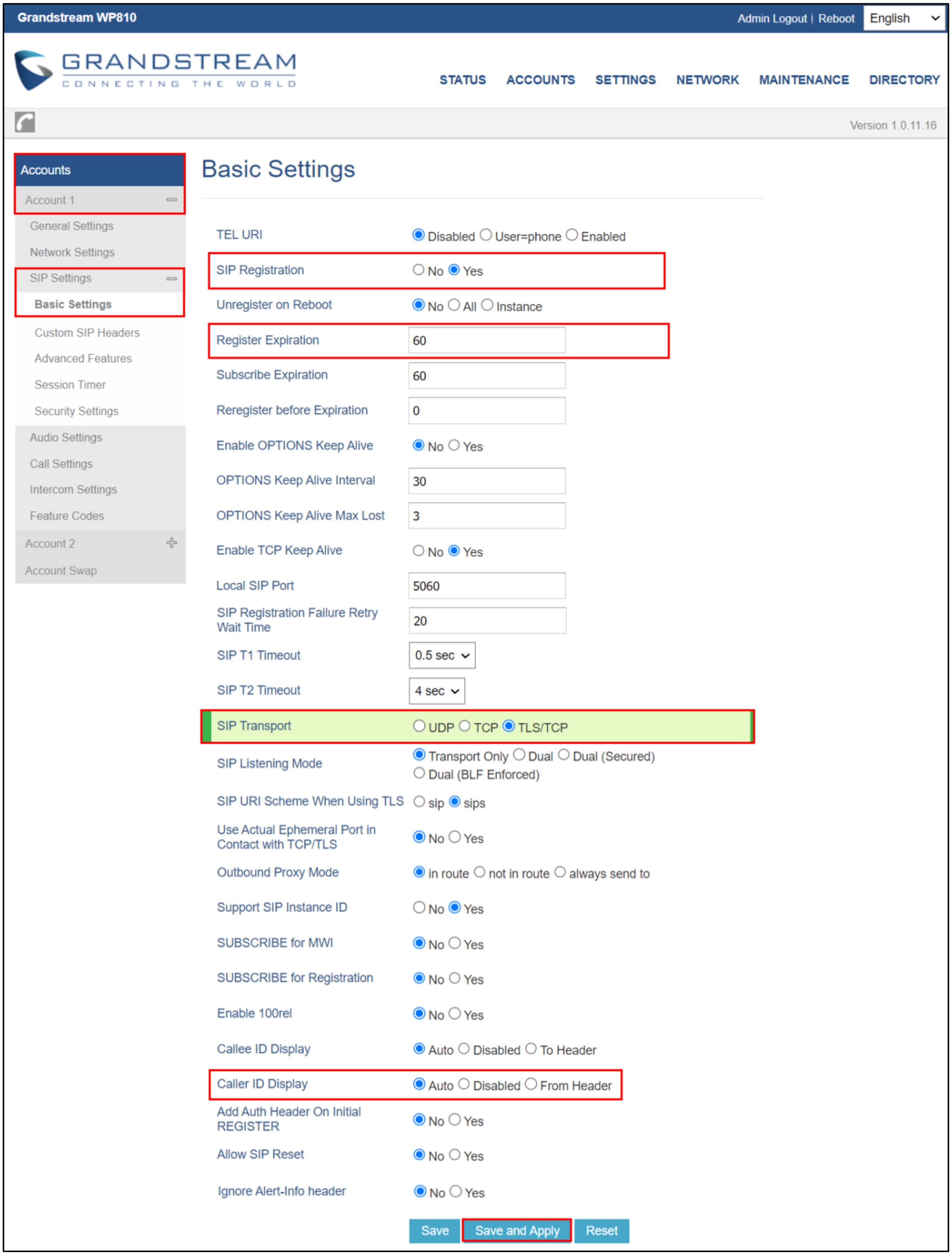

Figure 26 : SIP-Basic Settings

- Under Account 1 **-> SIP Settings -> Session Timer**. Ensure the below parameters are set.

    1. Enable Session Timer: Set to **"Yes".**
    2. Click **Save and Apply.**

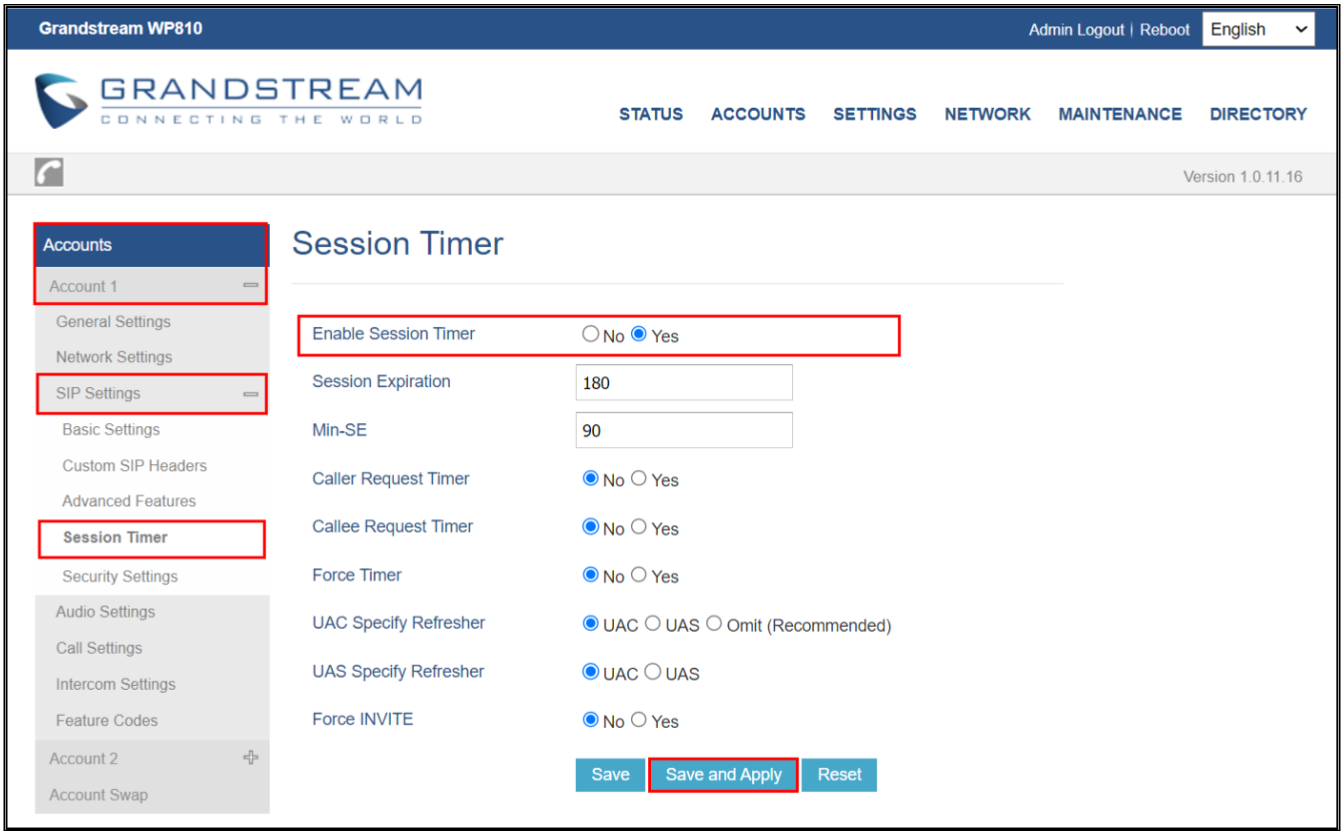

Figure 27 : SIP-Session Timer

- Under Account 1 **-> Audio Settings**. Ensure the below parameters are set.

    1. SRTP Mode: **Enabled and forced**
    2. SRTP Key Length: **AES 128&256 bit**
    3. Click **Save and Apply**

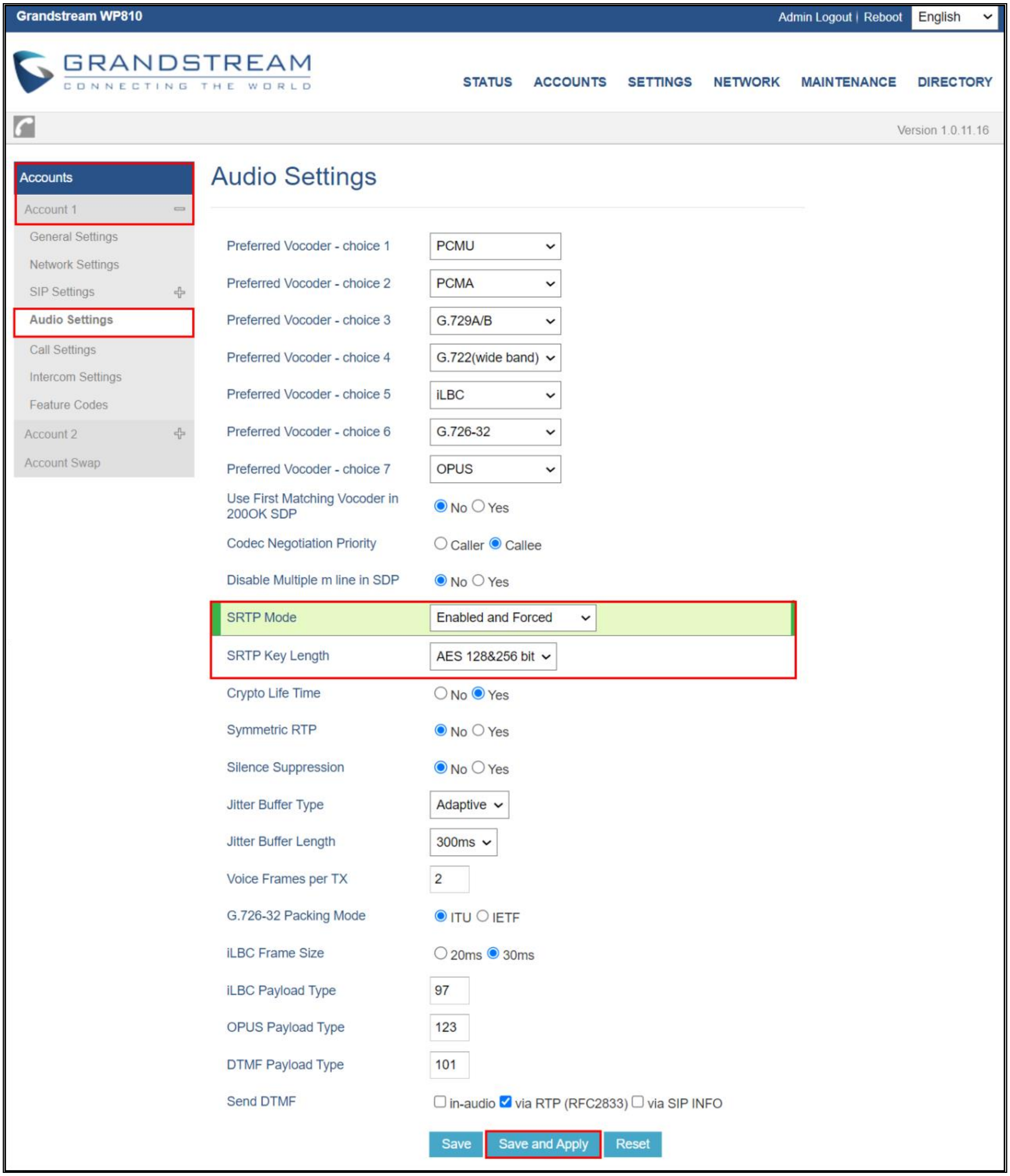

Figure 28: Audio Settings

- Under Account 1 **-> SIP Settings -> Advanced Features**. Ensure the below parameters are set.

    1. Special Feature: **Zoom**
    2. Click **Save and Apply**

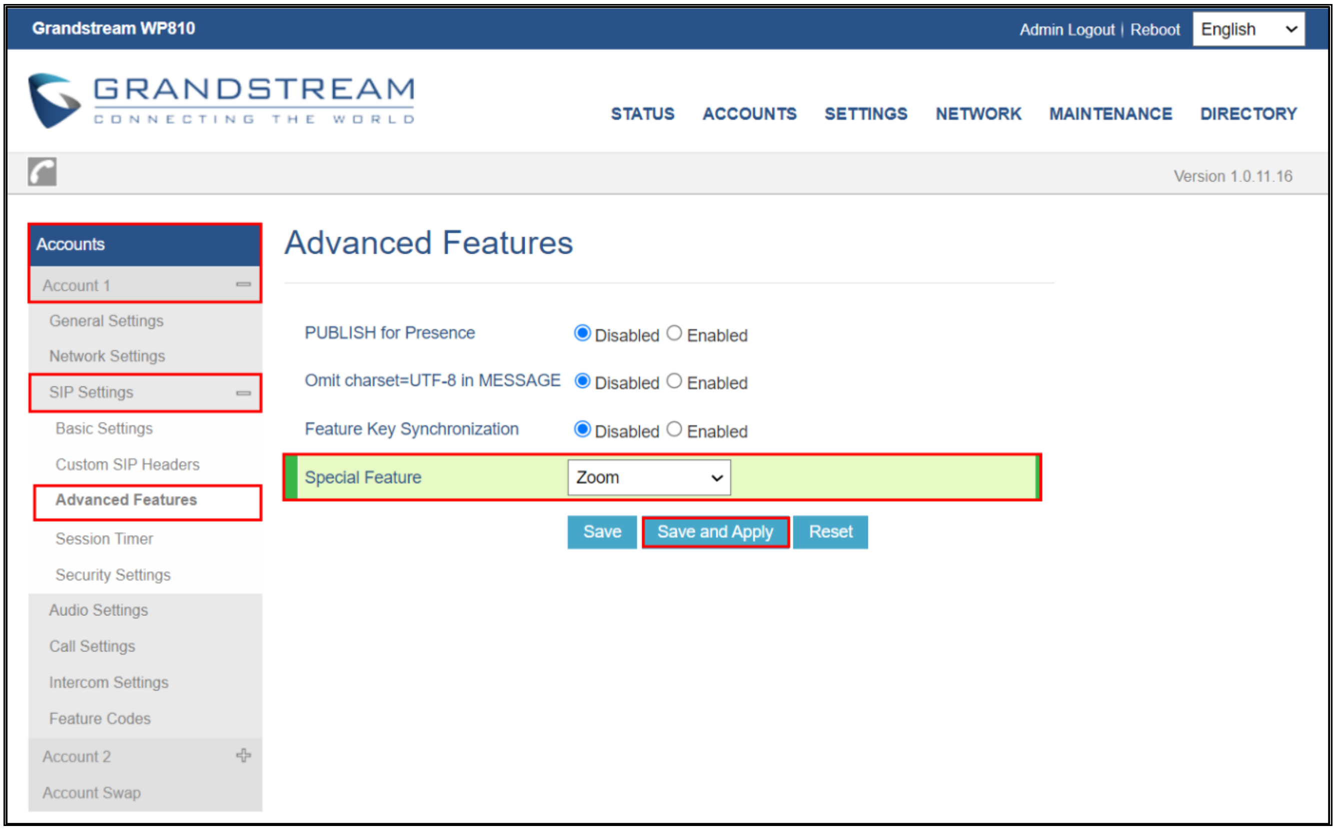

Figure 29: SIP-Advanced Features

- Under Account 1 **-> Feature Codes**. Ensure the below parameters are set.

  1. Enable Local Call Features: Set to **"Yes"**
  2. Click **Save and Apply**

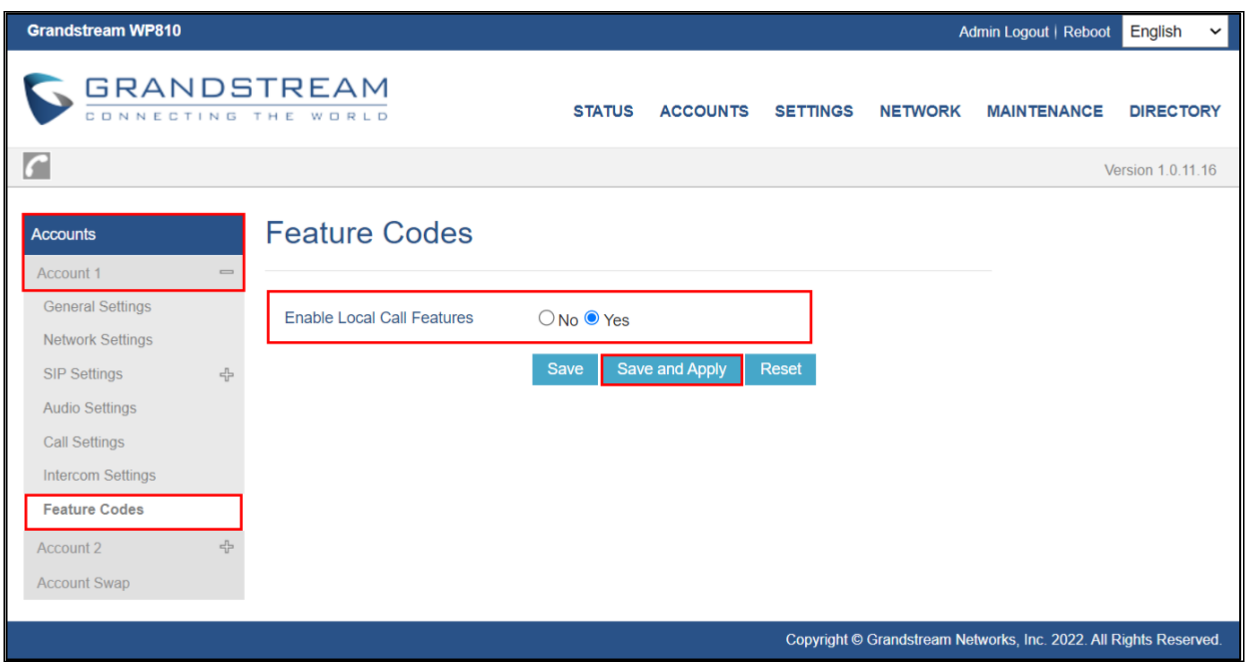

Figure 30: Feature Codes

- Navigate to **Settings -> General Settings**. Under **General Settings** header: Use Random Port is set to **"No" (**By Default this would be set to Yes).

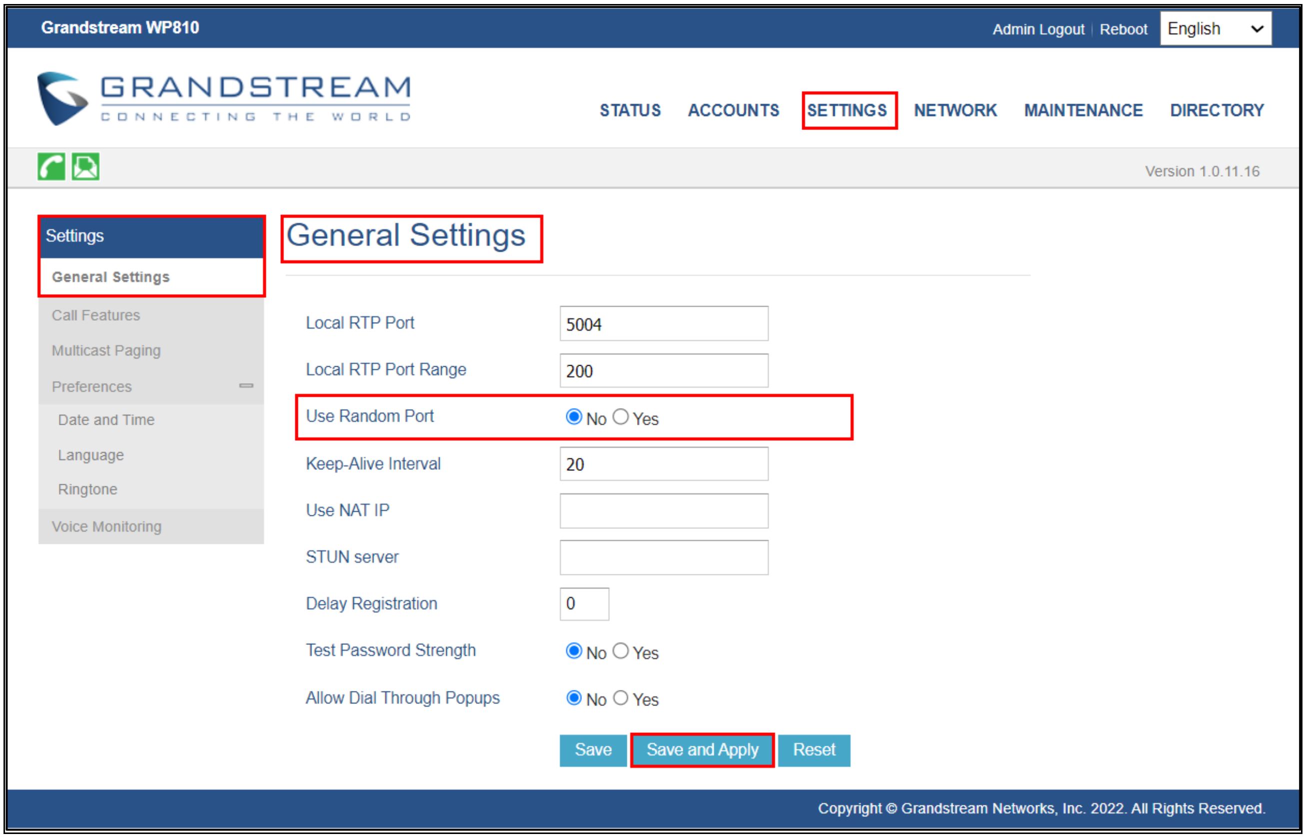

Figure 31: Settings-General Settings