

zoomphone

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/>.

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.

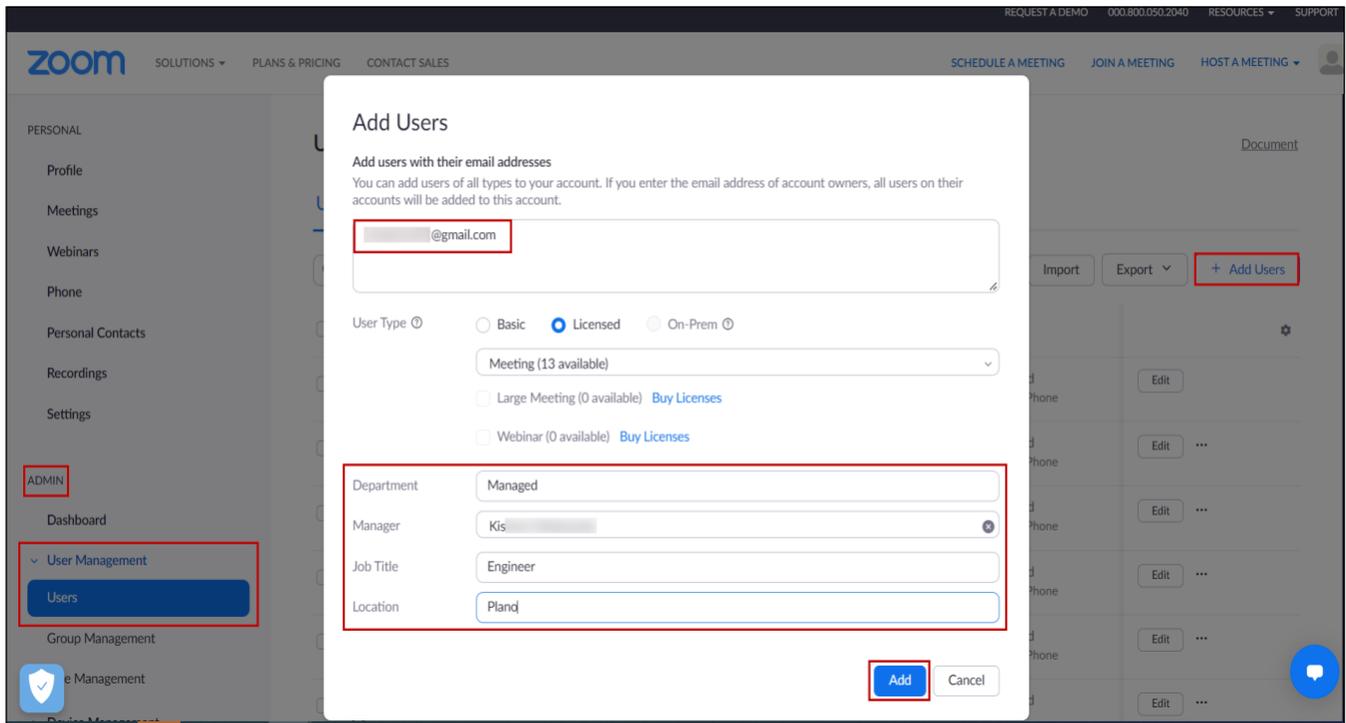


Figure 1 : Add Users

2. A Zoom activation email is sent to the email address while creating the user, follow the instruction to activate the zoom account.
3. Navigate to **Phone System Management -> Users & Rooms**. Click **Add**.

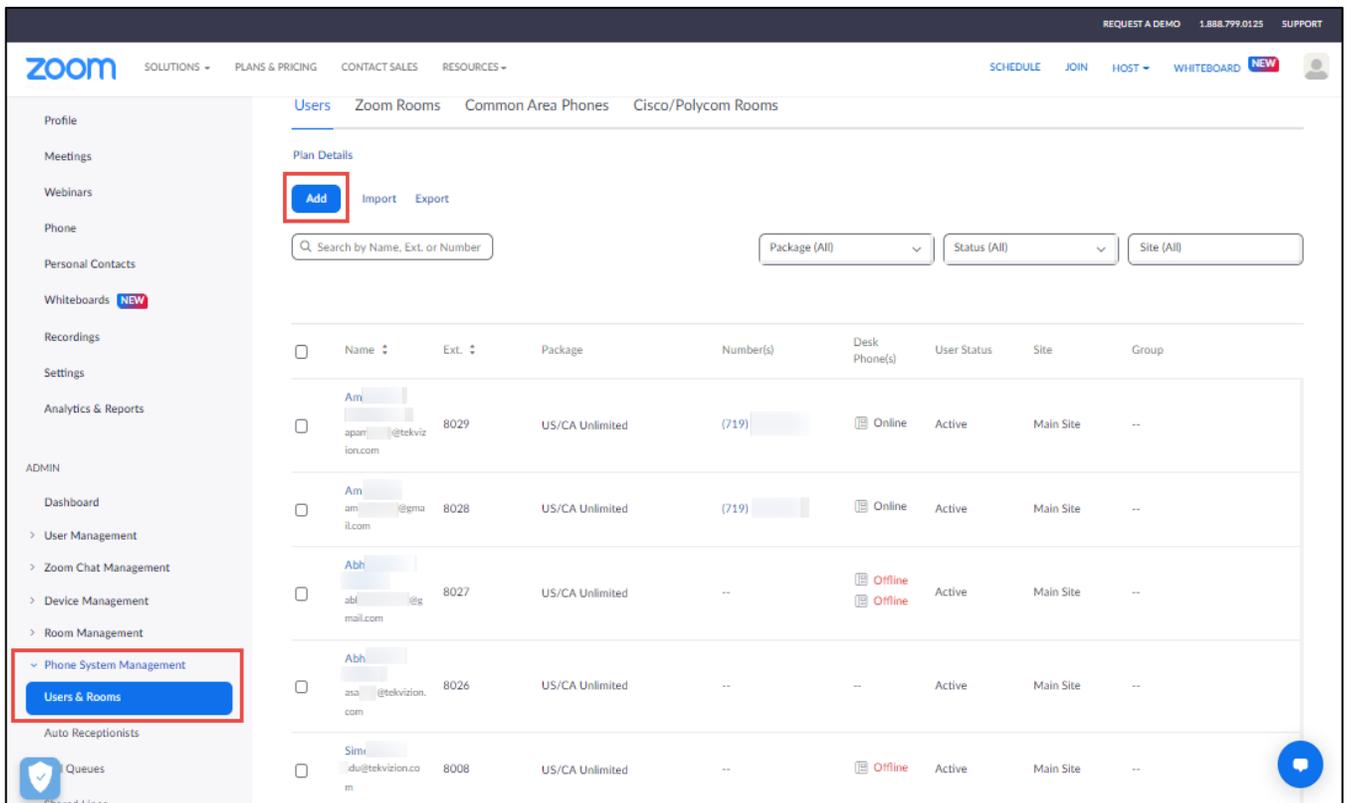


Figure 2 : Add Users and Rooms

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

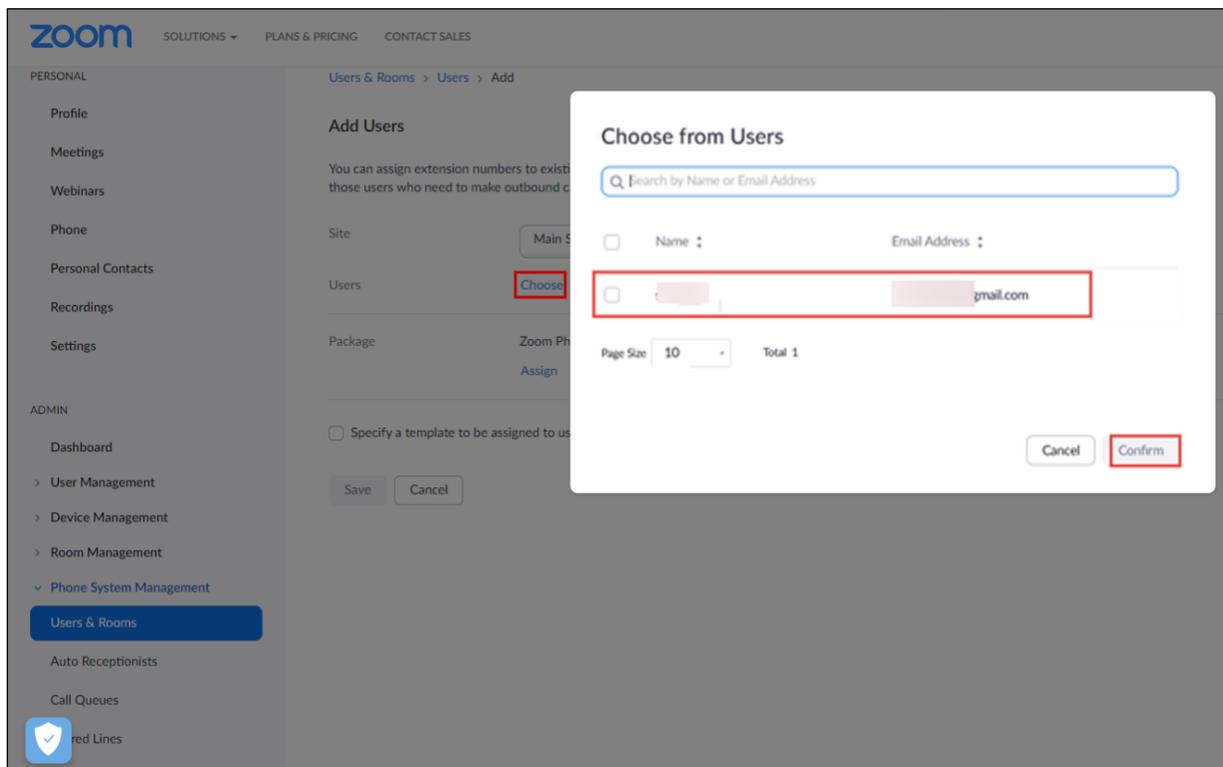


Figure 3 : Choose user

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.

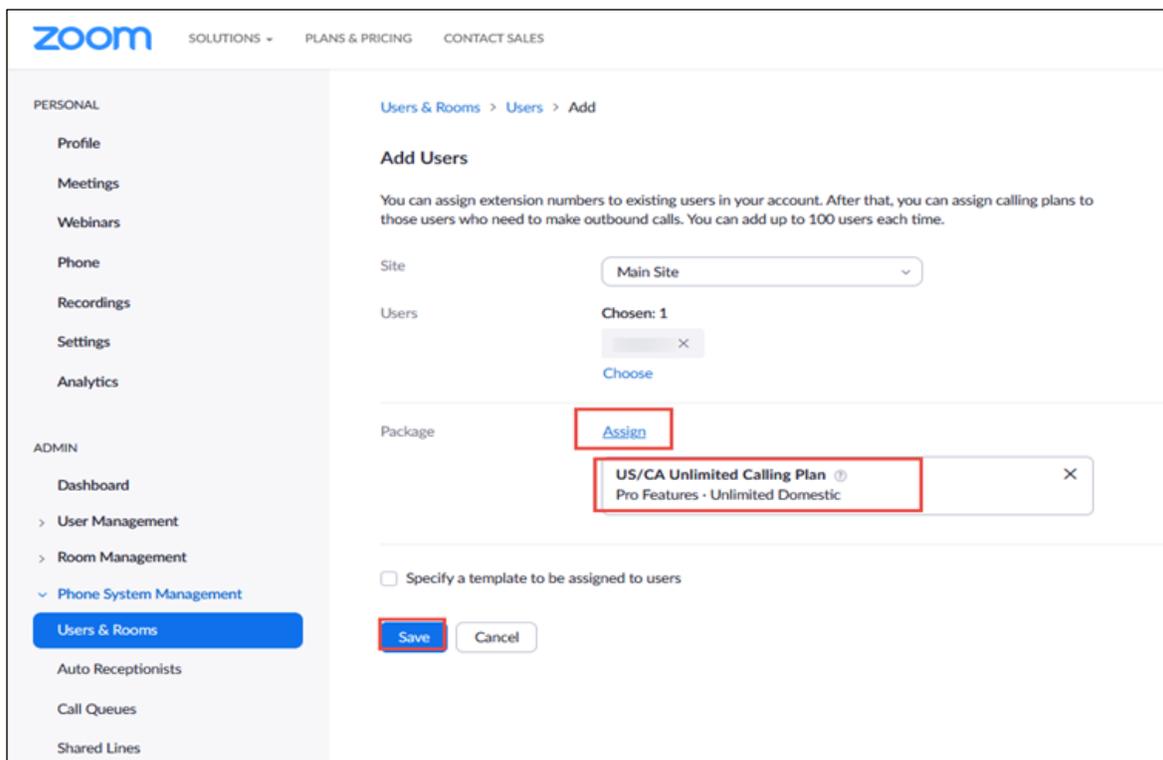


Figure 4 : Assign Calling Plan

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

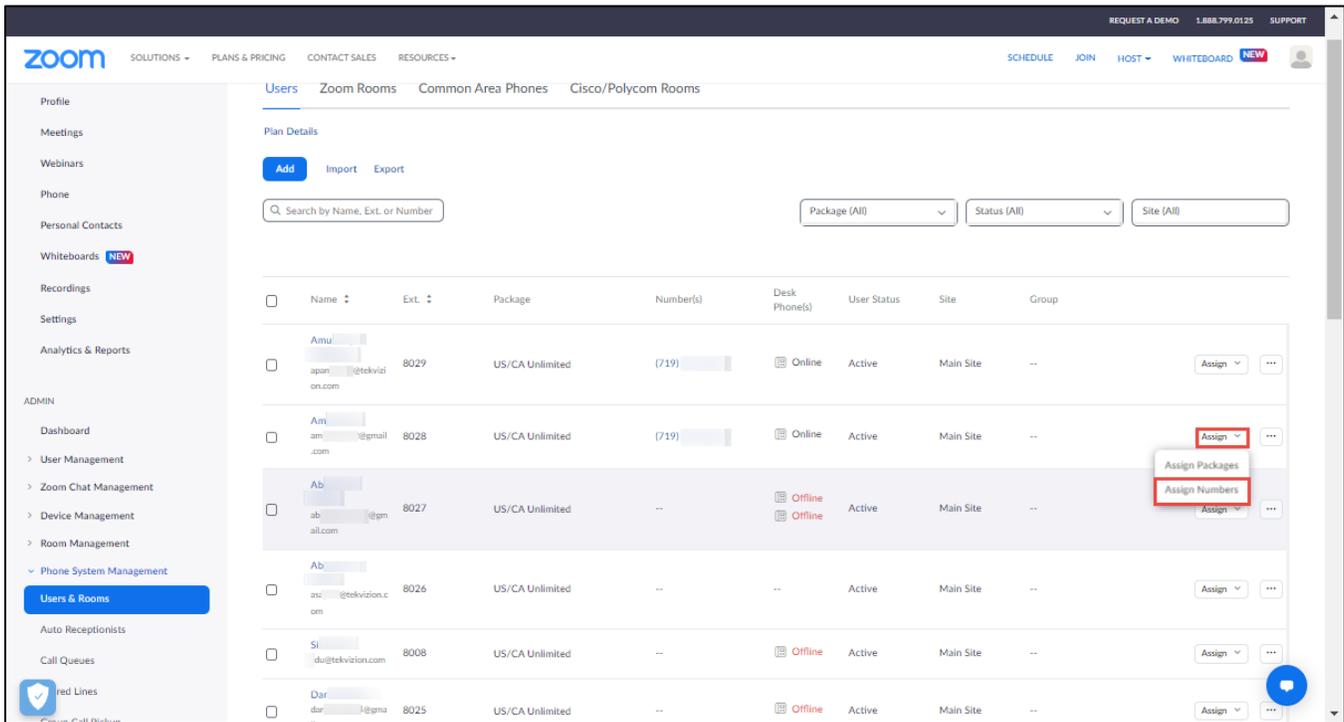


Figure 5 : Assign Number

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

Assign Numbers

Country/Region
United States

Q Search Site (All) Number Type (All) Get Numbers

Number	Area	Number Type	Capability	Site
<input checked="" type="checkbox"/> (719)	Canon City, Colorado, United States	Toll Number	Incoming & Outgoing	Main Site
<input type="checkbox"/>	United States	Toll Number	Incoming & Outgoing	Main Site
<input type="checkbox"/>	United States	Toll Number	Incoming & Outgoing	Main Site
<input type="checkbox"/>	United States	Toll Number	Incoming & Outgoing	Main Site
<input type="checkbox"/>	United States	Toll Number	Incoming & Outgoing	Main Site
<input type="checkbox"/>	United States	Toll Number	Incoming & Outgoing	Main Site
<input type="checkbox"/>	United States	Toll Number	Incoming & Outgoing	Main Site
<input type="checkbox"/>	United States	Toll Number	Incoming & Outgoing	Main Site
<input type="checkbox"/>	United States	Toll Number	Incoming & Outgoing	Main Site
<input type="checkbox"/>	United States	Toll Number	Incoming & Outgoing	Main Site

Page 1 of 2 Page Size 10 Total 12

Cancel Confirm

Figure 6 : Select DID Number

2.2 Add SIP Devices

- 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**.

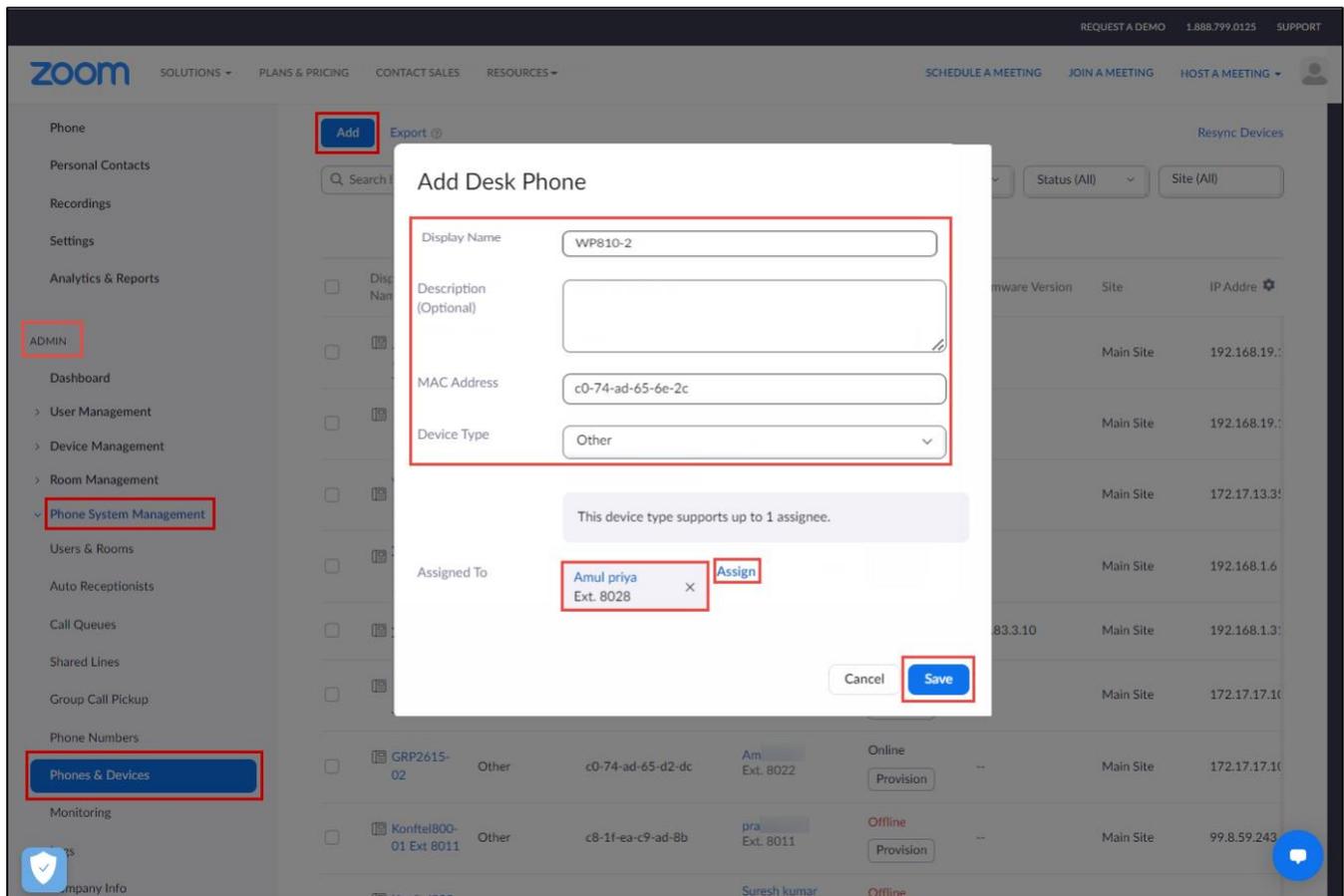


Figure 7 : Add Desk Phone

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](#)).

WP810-2 [Rename](#)

No description

Profile

Site	Main Site (Main Site)
Assigned To	Amul priya Ext. 8028
IP Address	172.16.31.209
Device Type	Other
Firmware Version	--
MAC Address	c0-74-ad-65-6e-2c
Provision Template	Unsupported ?
Status	Online

Actions Remove

- Provision
- Resync

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.

Provisioning

MAC Address c0-74-ad-65-6e-2c

Device Type Other

You will need to enable TLS1.2 for SIP registration and enable SRTP for secure calling on your IP phone. Please refer to your manufacturer's instructions for these processes.

You'll need following information for manual provisioning.

SIP Account 1:

1. SIP Domain: 7000766863.zoom.us
2. Outbound Proxy: us01sip0h.sc.zoom.us:5091
3. User Name: 3223
4. Authorization ID: 343
5. Password: A1 Wg

Please download [DigiCert Global Root CA](#), [DigiCert Global Root G2](#), [DigiCert Global Root G3](#) and import to your IP phone if they are not in the trust list of the device.

Note: Please note that Zoom support team will not be able to troubleshoot or configure IP phones that are provisioned in this manner. Some Zoom Phone features may not work on manually provisioned phones. It may vary depending on your desk phone model.

[Close](#)

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

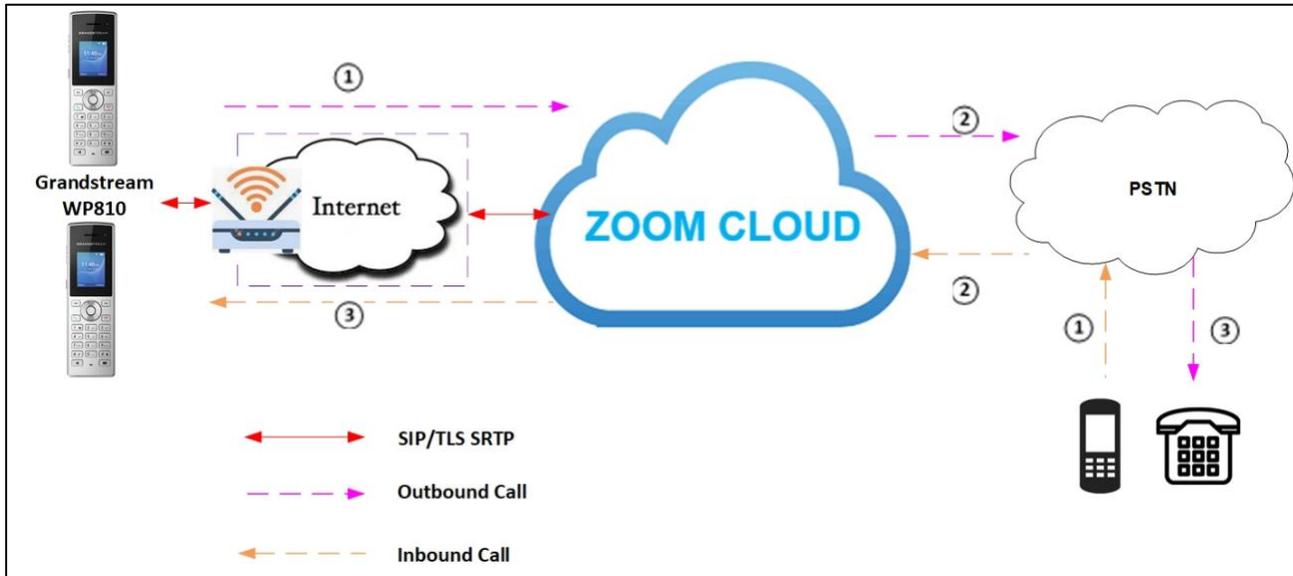


Figure 10 : Network Diagram

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**.

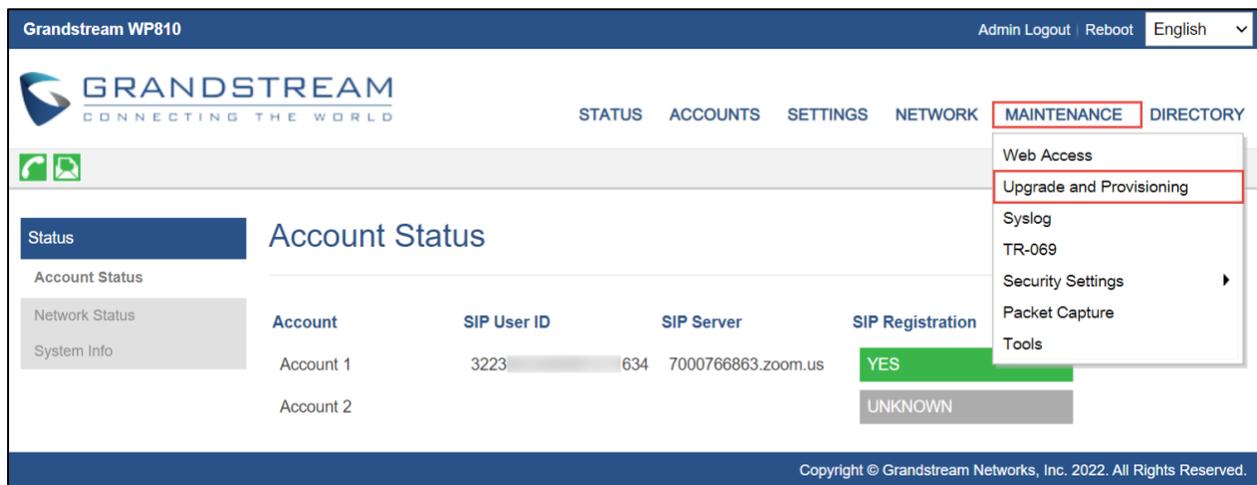


Figure 11 : Firmware Upgrade

- 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).

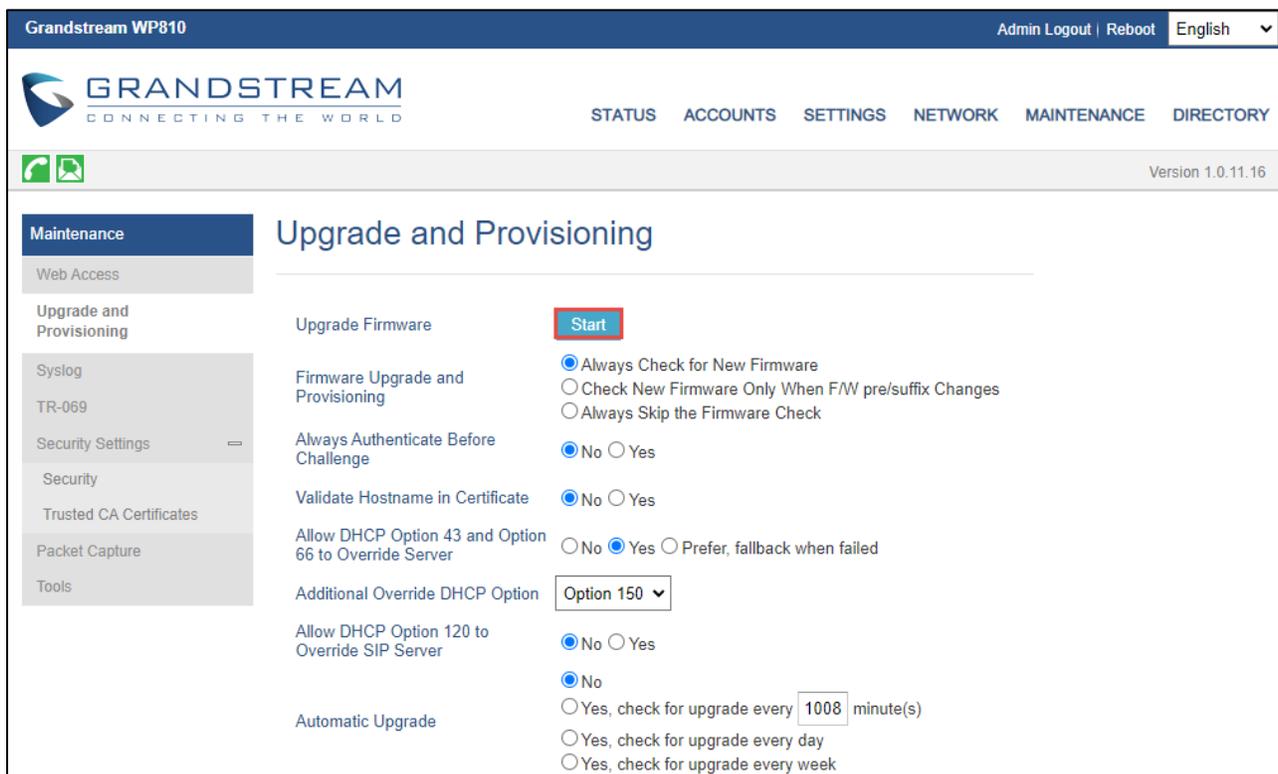


Figure 12 : Firmware Upgrade (Cont.)

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**.

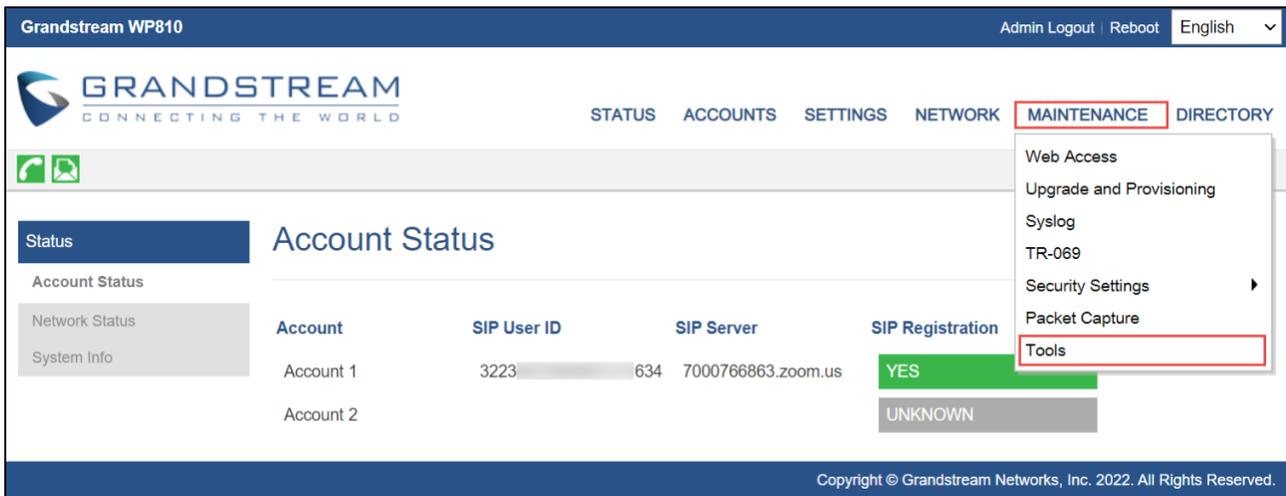


Figure 13 : Factory reset

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

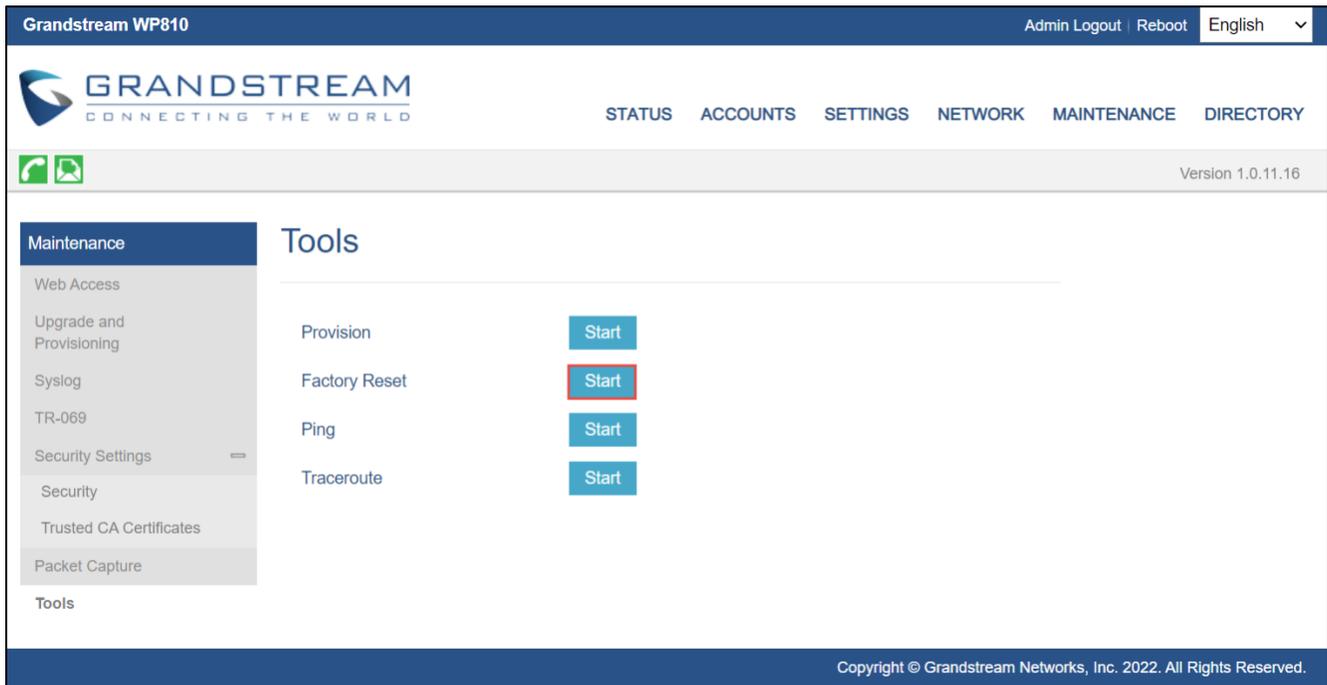


Figure 14 : Factory reset - (Cont.)

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.

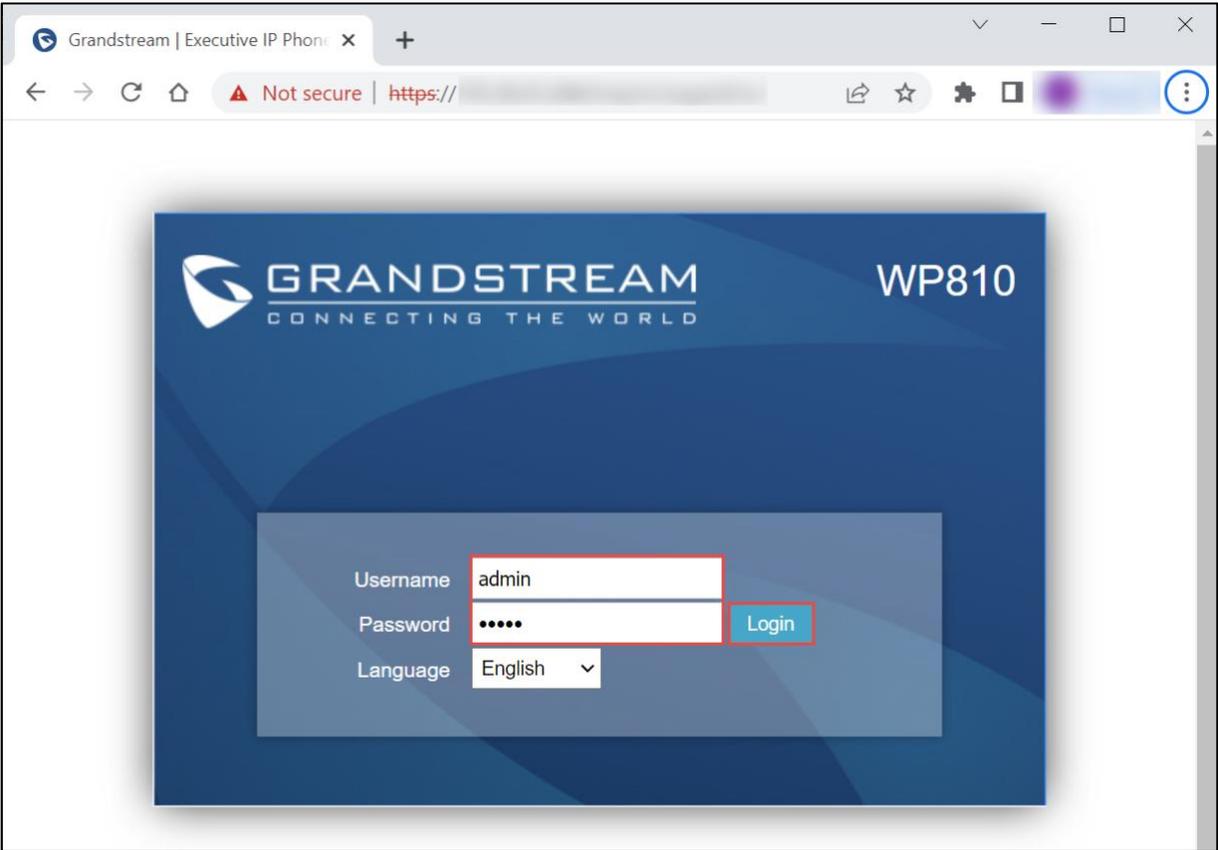


Figure 16 : Login details

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

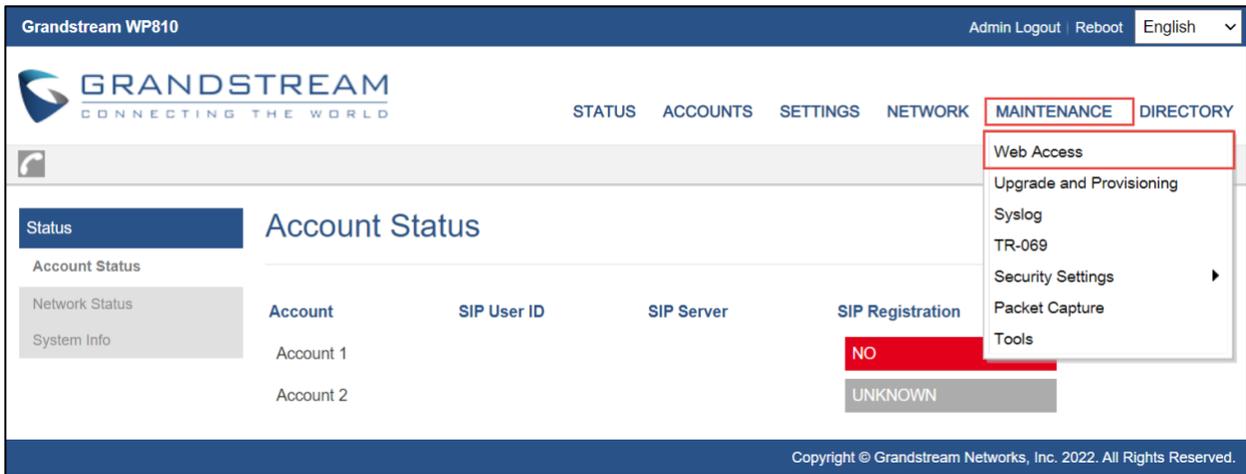


Figure 17 : Login details - (Cont.)

- 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.

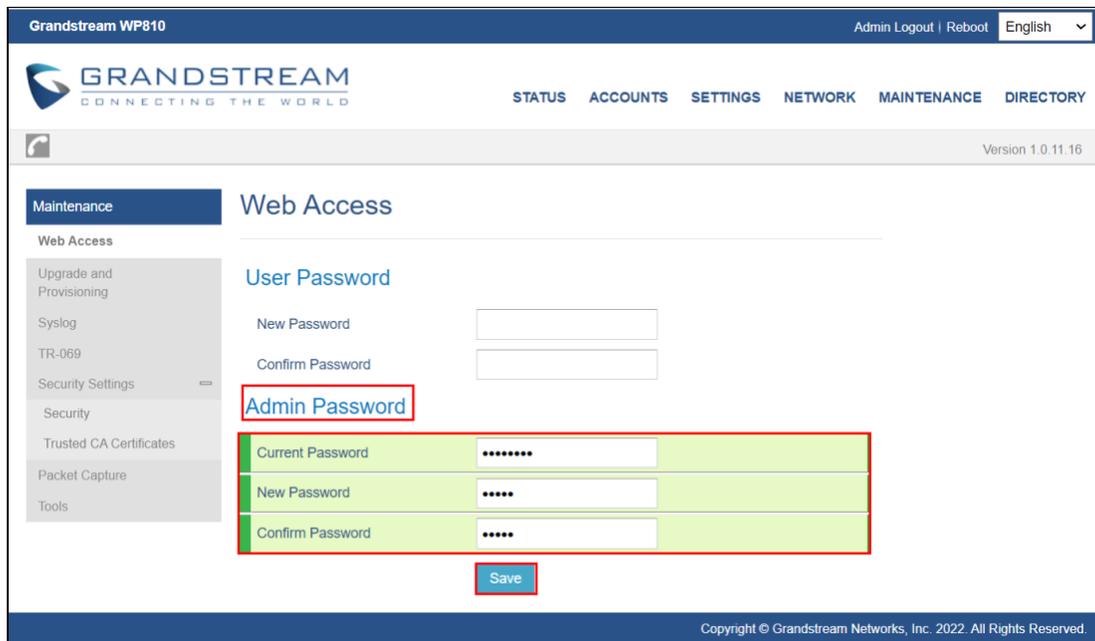


Figure 18 : Login details - (Cont.)

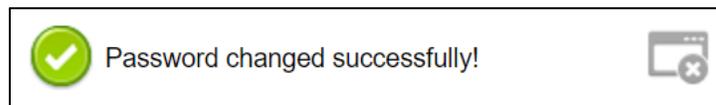


Figure 19 : Password change

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

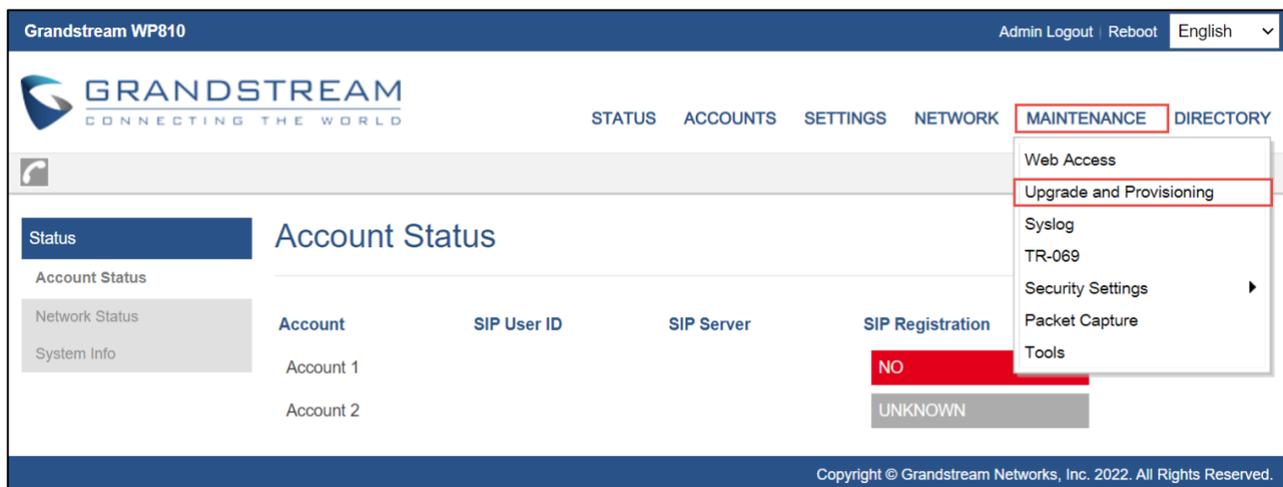


Figure 20 : Provisioning through a HTTP server

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**.

The screenshot displays the Grandstream WP810 web management interface. At the top, the header includes 'Grandstream WP810', 'Admin Logout | Reboot', and a language dropdown set to 'English'. The main navigation bar contains links for STATUS, ACCOUNTS, SETTINGS, NETWORK, MAINTENANCE, and DIRECTORY. The left sidebar shows a menu with 'Maintenance' selected, containing sub-items like Web Access, Upgrade and Provisioning, Syslog, TR-069, Security Settings, Security, Trusted CA Certificates, Packet Capture, and Tools. The main content area is titled 'Upgrade and Provisioning' and features a 'Start' button. Below this, several configuration options are listed with radio buttons or dropdown menus:

- Upgrade Firmware:** Includes a 'Start' button and three radio options: 'Always Check for New Firmware' (selected), 'Check New Firmware Only When F/W pre/suffix Changes', and 'Always Skip the Firmware Check'.
- Always Authenticate Before Challenge:** Radio buttons for 'No' (selected) and 'Yes'.
- Validate Hostname in Certificate:** Radio buttons for 'No' (selected) and 'Yes'.
- Allow DHCP Option 43 and Option 66 to Override Server:** Radio buttons for 'No', 'Yes' (selected), and 'Prefer, fallback when failed'.
- Additional Override DHCP Option:** A dropdown menu currently showing 'Option 150'.
- Allow DHCP Option 120 to Override SIP Server:** Radio buttons for 'No' (selected) and 'Yes'.
- Automatic Upgrade:** Radio buttons for 'No' (selected), 'Yes, check for upgrade every 1008 minute(s)', 'Yes, check for upgrade every day', and 'Yes, check for upgrade every week'.
- Randomized Automatic Upgrade:** Radio buttons for 'No' (selected) and 'Yes'.
- Hour of the Day(0-23):** Start dropdown set to '1' and End dropdown set to '0'.
- Day of the Week (0-6):** A text input field containing '1'.
- Disable SIP NOTIFY Authentication:** Radio buttons for 'No' (selected) and 'Yes'.

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

Config

Config Upgrade via	<input type="radio"/> TFTP <input checked="" type="radio"/> HTTP <input type="radio"/> HTTPS <input type="radio"/> FTP <input type="radio"/> FTPS
Config Server Path	172. . :90/amul
Config Server Username	<input type="text"/>
Config Server Password	<input type="text"/>
Config File Prefix	<input type="text"/>
Config File Postfix	<input type="text"/>
XML Config File Password	<input type="text"/>
Authenticate Conf File	<input checked="" type="radio"/> No <input type="radio"/> Yes
Download Device Configuration	Download
Download Device Configuration (XML)	Download
Download and Process All Available Config Files	<input checked="" type="radio"/> No <input type="radio"/> Yes
Download User Configuration	Download
Upload Device Configuration	Upload
Export Backup Package	Download
Restore from Backup Package	Upload

Firmware

Firmware Upgrade via	<input type="radio"/> TFTP <input checked="" type="radio"/> HTTP <input type="radio"/> HTTPS <input type="radio"/> FTP <input type="radio"/> FTPS
Firmware Server Path	<input type="text" value="fm.grandstream.com/gs"/>
Firmware Server Username	<input type="text"/>
Firmware Server Password	<input type="text"/>
Firmware File Prefix	<input type="text"/>
Firmware File Postfix	<input type="text"/>

[Save](#)
[Save and Apply](#)
[Reset](#)

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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.

Grandstream WP810 Admin Logout Reboot English

GRANDSTREAM
CONNECTING THE WORLD

STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.11.16

Status

Account Status

Network Status

System Info

Account	SIP User ID	SIP Server	SIP Registration
Account 1	32238	634 7000766863.zoom.us	YES
Account 2			UNKNOWN

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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**.

Grandstream WP810 Admin Logout | Reboot English

GRANDSTREAM
CONNECTING THE WORLD

STATUS **ACCOUNTS** SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.11.16

Accounts

Account 1

General Settings

Network Settings

SIP Settings

Audio Settings

Call Settings

Intercom Settings

Feature Codes

Account 2

Account Swap

General Settings

Account Active	<input type="radio"/> No <input checked="" type="radio"/> Yes
Account Name	6E2C
SIP Server	7000766863.zoom.us
Secondary SIP Server	
Outbound Proxy	us01sip0h.sc.zoom.us:5091
Backup Outbound Proxy	
SIP User ID	32
Authenticate ID	343
Authenticate Password	•••••
Name	8028
Voice Mail Access Number	
Account Display	<input checked="" type="radio"/> User Name <input type="radio"/> User ID

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**

Grandstream WP810 Admin Logout | Reboot English

GRANDSTREAM
CONNECTING THE WORLD

STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.11.16

Network Settings

- Accounts
 - Account 1
- General Settings
- Network Settings**
- SIP Settings
- Audio Settings
- Call Settings
- Intercom Settings
- Feature Codes
- Account 2
- Account Swap

DNS Mode: SRV

DNS SRV Fail-over Mode: Saved one until no response

Register Before DNS SRV Failover: No Yes

Primary IP:

Backup IP 1:

Backup IP 2:

NAT Traversal: No

Proxy-Require:

Save Save and Apply Reset

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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**

Grandstream WP810 Admin Logout | Reboot | English

GRANDSTREAM
CONNECTING THE WORLD

STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.11.16

Accounts

Account 1

General Settings

Network Settings

SIP Settings

Basic Settings

Custom SIP Headers

Advanced Features

Session Timer

Security Settings

Audio Settings

Call Settings

Intercom Settings

Feature Codes

Account 2

Account Swap

Basic Settings

TEL URI Disabled User=phone Enabled

SIP Registration No Yes

Unregister on Reboot No All Instance

Register Expiration

Subscribe Expiration

Reregister before Expiration

Enable OPTIONS Keep Alive No Yes

OPTIONS Keep Alive Interval

OPTIONS Keep Alive Max Lost

Enable TCP Keep Alive No Yes

Local SIP Port

SIP Registration Failure Retry Wait Time

SIP T1 Timeout

SIP T2 Timeout

SIP Transport UDP TCP TLS/TCP

SIP Listening Mode Transport Only Dual Dual (Secured)
 Dual (BLF Enforced)

SIP URI Scheme When Using TLS sip sips

Use Actual Ephemeral Port in Contact with TCP/TLS No Yes

Outbound Proxy Mode in route not in route always send to

Support SIP Instance ID No Yes

SUBSCRIBE for MWI No Yes

SUBSCRIBE for Registration No Yes

Enable 100rel No Yes

Callee ID Display Auto Disabled To Header

Caller ID Display Auto Disabled From Header

Add Auth Header On Initial REGISTER No Yes

Allow SIP Reset No Yes

Ignore Alert-Info header No Yes

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**.

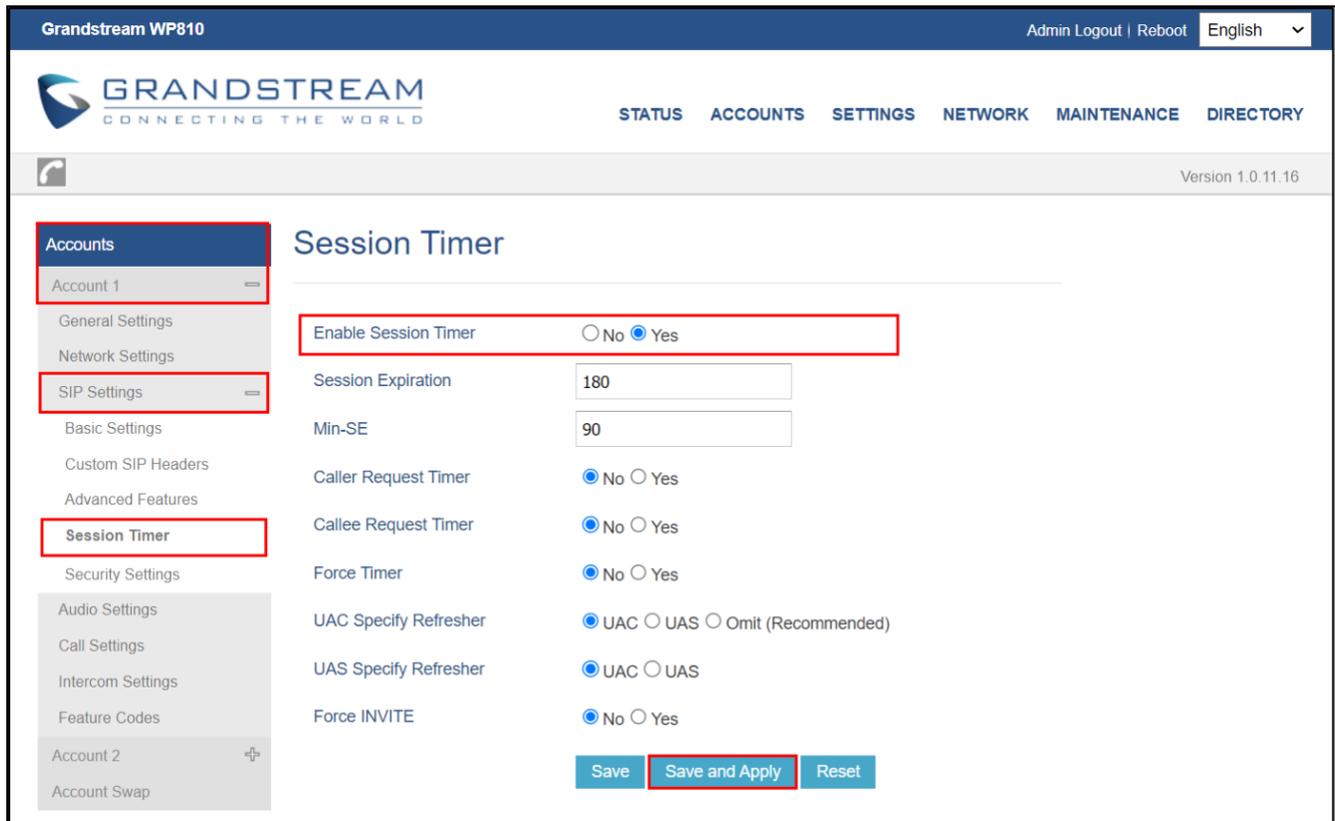


Figure 27 : SIP-Session Timer

- 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**

Grandstream WP810 Admin Logout | Reboot English

GRANDSTREAM
CONNECTING THE WORLD

STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.11.16

Accounts

Account 1

General Settings

Network Settings

SIP Settings

Audio Settings

Call Settings

Intercom Settings

Feature Codes

Account 2

Account Swap

Audio Settings

Preferred Vocoder - choice 1	<input type="text" value="PCMU"/>
Preferred Vocoder - choice 2	<input type="text" value="PCMA"/>
Preferred Vocoder - choice 3	<input type="text" value="G.729A/B"/>
Preferred Vocoder - choice 4	<input type="text" value="G.722(wide band)"/>
Preferred Vocoder - choice 5	<input type="text" value="iLBC"/>
Preferred Vocoder - choice 6	<input type="text" value="G.726-32"/>
Preferred Vocoder - choice 7	<input type="text" value="OPUS"/>
Use First Matching Vocoder in 200OK SDP	<input checked="" type="radio"/> No <input type="radio"/> Yes
Codec Negotiation Priority	<input type="radio"/> Caller <input checked="" type="radio"/> Callee
Disable Multiple m line in SDP	<input checked="" type="radio"/> No <input type="radio"/> Yes
SRTP Mode	<input type="text" value="Enabled and Forced"/>
SRTP Key Length	<input type="text" value="AES 128&256 bit"/>
Crypto Life Time	<input type="radio"/> No <input checked="" type="radio"/> Yes
Symmetric RTP	<input checked="" type="radio"/> No <input type="radio"/> Yes
Silence Suppression	<input checked="" type="radio"/> No <input type="radio"/> Yes
Jitter Buffer Type	<input type="text" value="Adaptive"/>
Jitter Buffer Length	<input type="text" value="300ms"/>
Voice Frames per TX	<input type="text" value="2"/>
G.726-32 Packing Mode	<input checked="" type="radio"/> ITU <input type="radio"/> IETF
iLBC Frame Size	<input type="radio"/> 20ms <input checked="" type="radio"/> 30ms
iLBC Payload Type	<input type="text" value="97"/>
OPUS Payload Type	<input type="text" value="123"/>
DTMF Payload Type	<input type="text" value="101"/>
Send DTMF	<input type="checkbox"/> in-audio <input checked="" type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO

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**

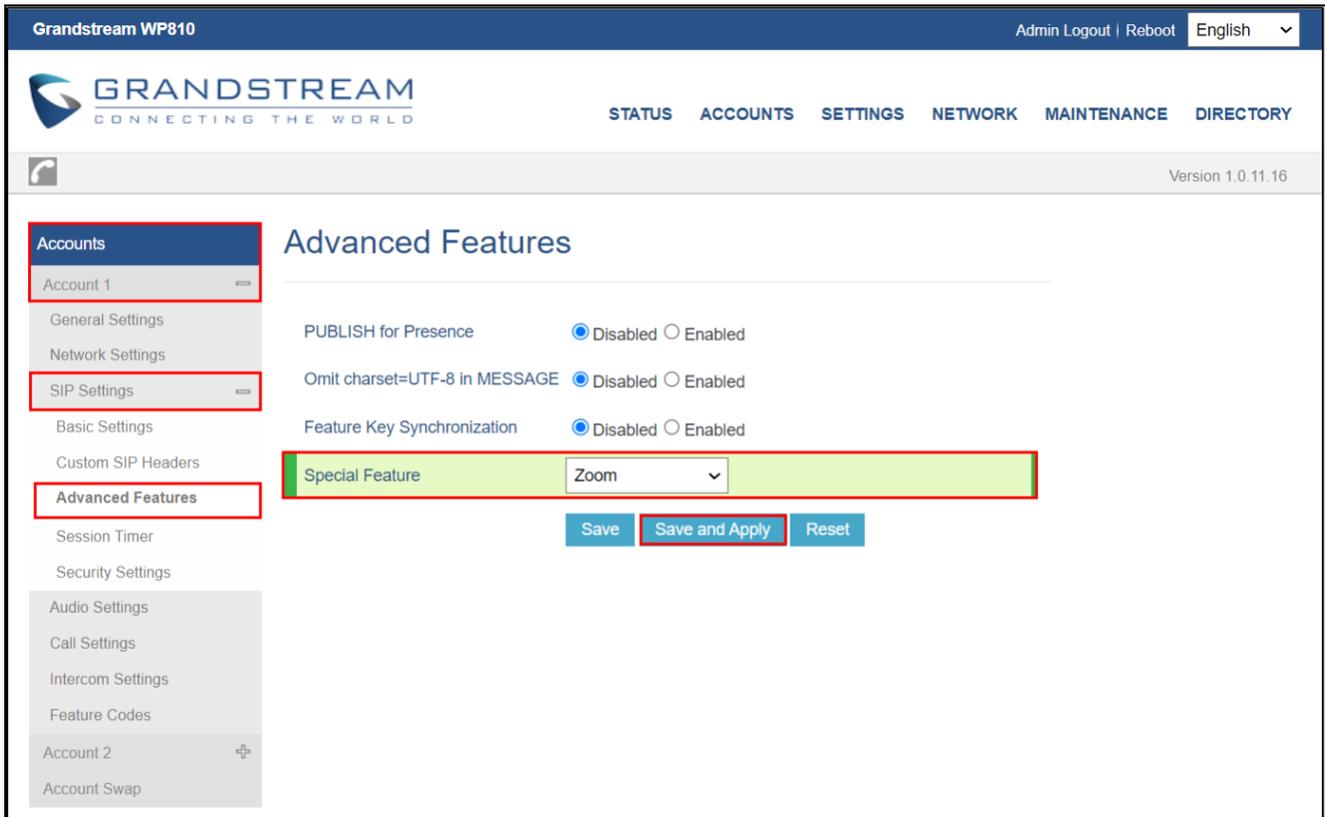


Figure 29: SIP-Advanced Features

- Under Account 1 -> **Feature Codes**. Ensure the below parameters are set.
 1. Enable Local Call Features: Set to **“Yes”**
 2. Click **Save and Apply**

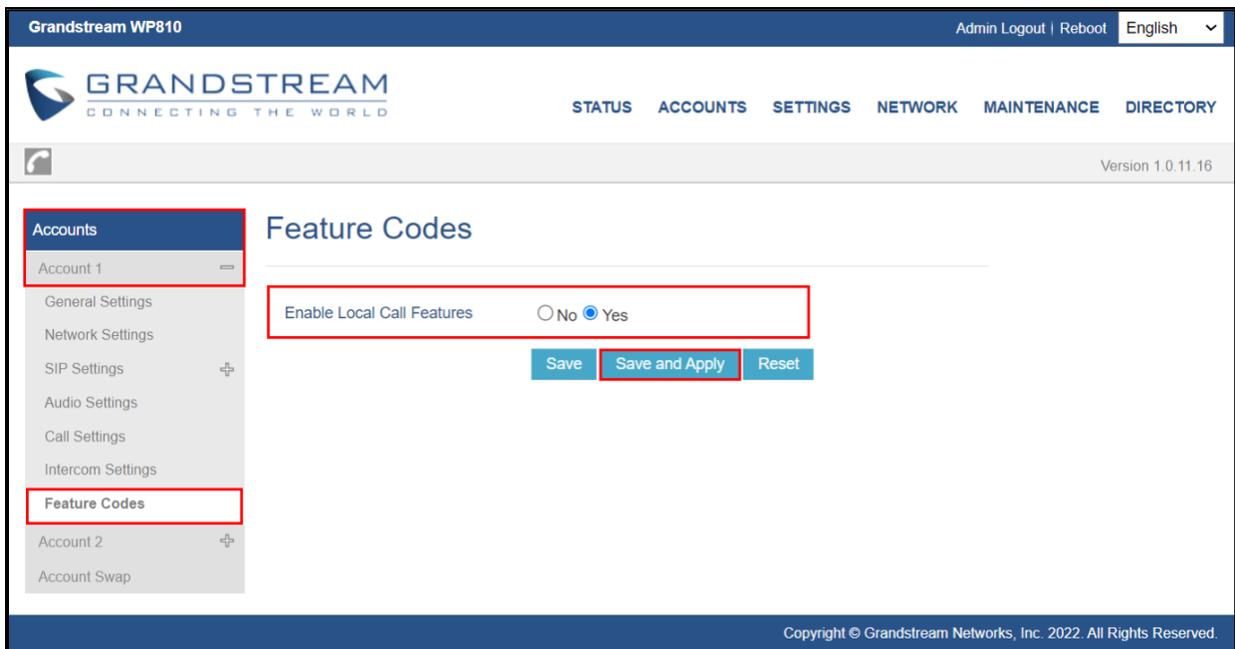


Figure 30: Feature Codes

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

The screenshot shows the Grandstream WP810 web interface. At the top, there is a header with 'Grandstream WP810' on the left, 'Admin Logout | Reboot' in the middle, and 'English' with a dropdown arrow on the right. Below the header is the Grandstream logo and a navigation menu with 'STATUS', 'ACCOUNTS', 'SETTINGS' (highlighted with a red box), 'NETWORK', 'MAINTENANCE', and 'DIRECTORY'. A secondary navigation bar shows 'Version 1.0.11.16' on the right and two green icons on the left. On the left side, there is a 'Settings' sidebar with 'General Settings' (highlighted with a red box) and other options like 'Call Features', 'Multicast Paging', 'Preferences', 'Date and Time', 'Language', 'Ringtone', and 'Voice Monitoring'. The main content area is titled 'General Settings' (highlighted with a red box) and contains several configuration fields: 'Local RTP Port' (5004), 'Local RTP Port Range' (200), 'Use Random Port' (radio buttons for 'No' and 'Yes', with 'No' selected and highlighted by a red box), 'Keep-Alive Interval' (20), 'Use NAT IP' (empty), 'STUN server' (empty), 'Delay Registration' (0), 'Test Password Strength' (radio buttons for 'No' and 'Yes', with 'No' selected), and 'Allow Dial Through Popups' (radio buttons for 'No' and 'Yes', with 'No' selected). At the bottom of the settings area are three buttons: 'Save', 'Save and Apply' (highlighted with a red box), and 'Reset'. The footer contains the text 'Copyright © Grandstream Networks, Inc. 2022. All Rights Reserved.'

Figure 31: Settings-General Settings