

**zoom**phone

# Configuration Guide For Grandstream GRP261x series

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## Revision History

Version	Date	Change
1.0	03/29/2022	Created document for GRP2615 Configurations

## DUT and Zoom Software Versions

	Equipment	Software Version
Grandstream (Device Under Test)	GRP2615	1.0.7.19
Zoom	Zoom app Desktop	5.10.0 (4306)
	Zoom app Mobile	5.9.6 (4756)

## Features Supported by GRP261x 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
- Busy Lamp Field
- 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 Flip

- Shared Line - Call Delegation
- Call Monitoring
- Auto Receptionist IVR
- AES-256
- Call Park/Retrieve

## 1. Overview

This document outlines the configuration best practices for the Grandstream GRP261x as Zoom generic SIP phone.

## 2. Configuration Steps - Zoom Web Portal

This section provides instructions on how to configure Grandstream GRP2615 in Zoom Web Portal.

This section is mainly for adding phone devices (GRP2615) and assign Zoom users to the devices.

### Prerequisites:

- Zoom Phone account: a valid Zoom Phone subscription is required in order to assign an GRP2615 endpoint.
- Zoom approval for provisioning of GRP2615 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. Navigate to **User Management > Users**. Click **+ Add Users** to create new Zoom users.

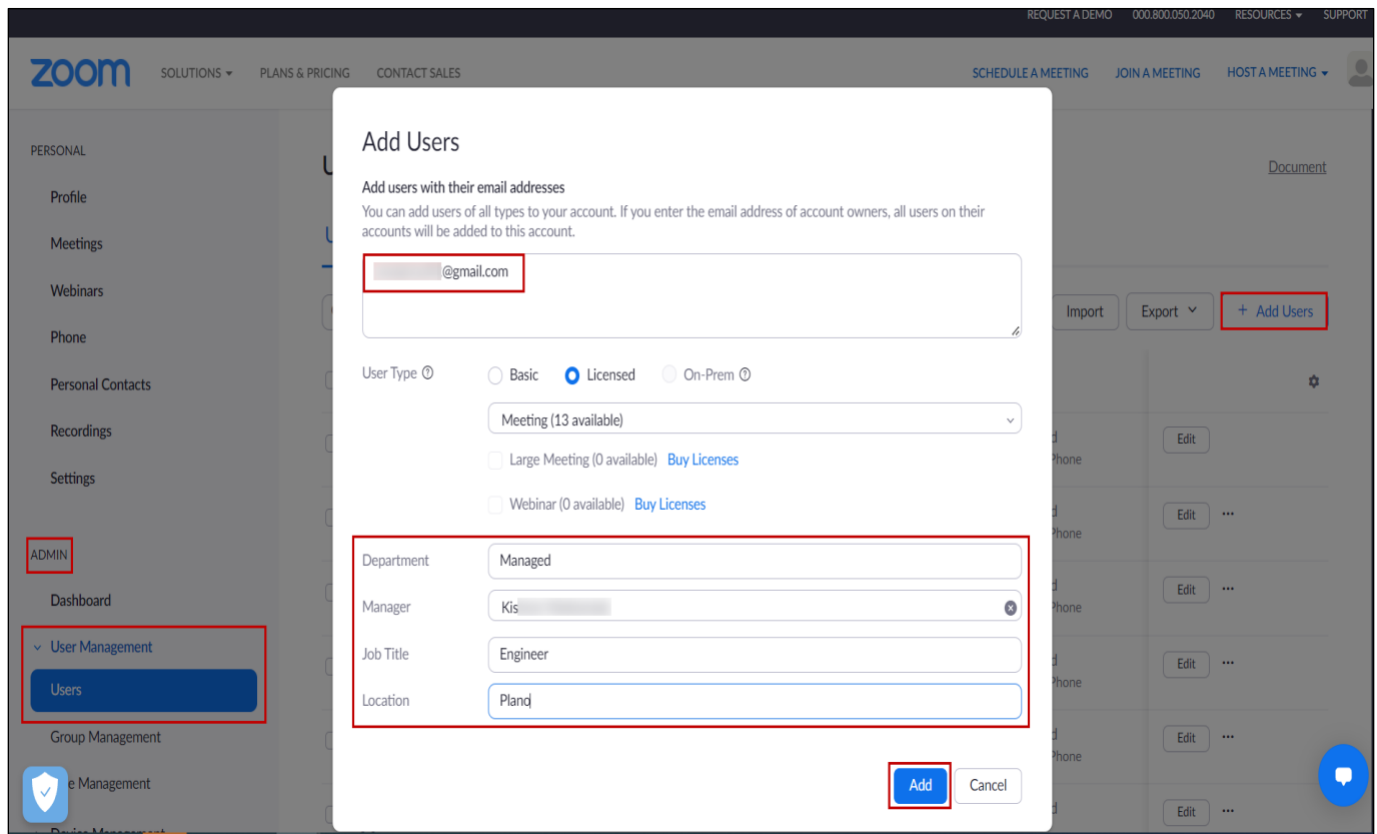
The screenshot shows the Zoom Admin console interface. On the left sidebar, the 'ADMIN' section is expanded, and 'User Management' is selected, with the 'Users' sub-option highlighted. The main content area displays the 'Add Users' modal. At the top of the modal, there's a text input field for email addresses, with a red box highlighting the '@gmail.com' placeholder. Below this, the 'User Type' section has three radio buttons: 'Basic', 'Licensed' (which is selected), and 'On-Prem'. Under 'Licensed', there are three options: 'Meeting (13 available)', 'Large Meeting (0 available)', and 'Webinar (0 available)', each with a 'Buy Licenses' link. A red box highlights the 'Department' field (set to 'Managed'), the 'Manager' dropdown (set to 'Kis'), the 'Job Title' field (set to 'Engineer'), and the 'Location' field (set to 'Pland'). At the bottom right of the modal, there are 'Add' and 'Cancel' buttons, with the 'Add' button highlighted by a red box. In the background, the 'Import', 'Export', and '+ Add Users' buttons are visible, with the '+ Add Users' button also highlighted by a red box.

Figure 1 : Add Users

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

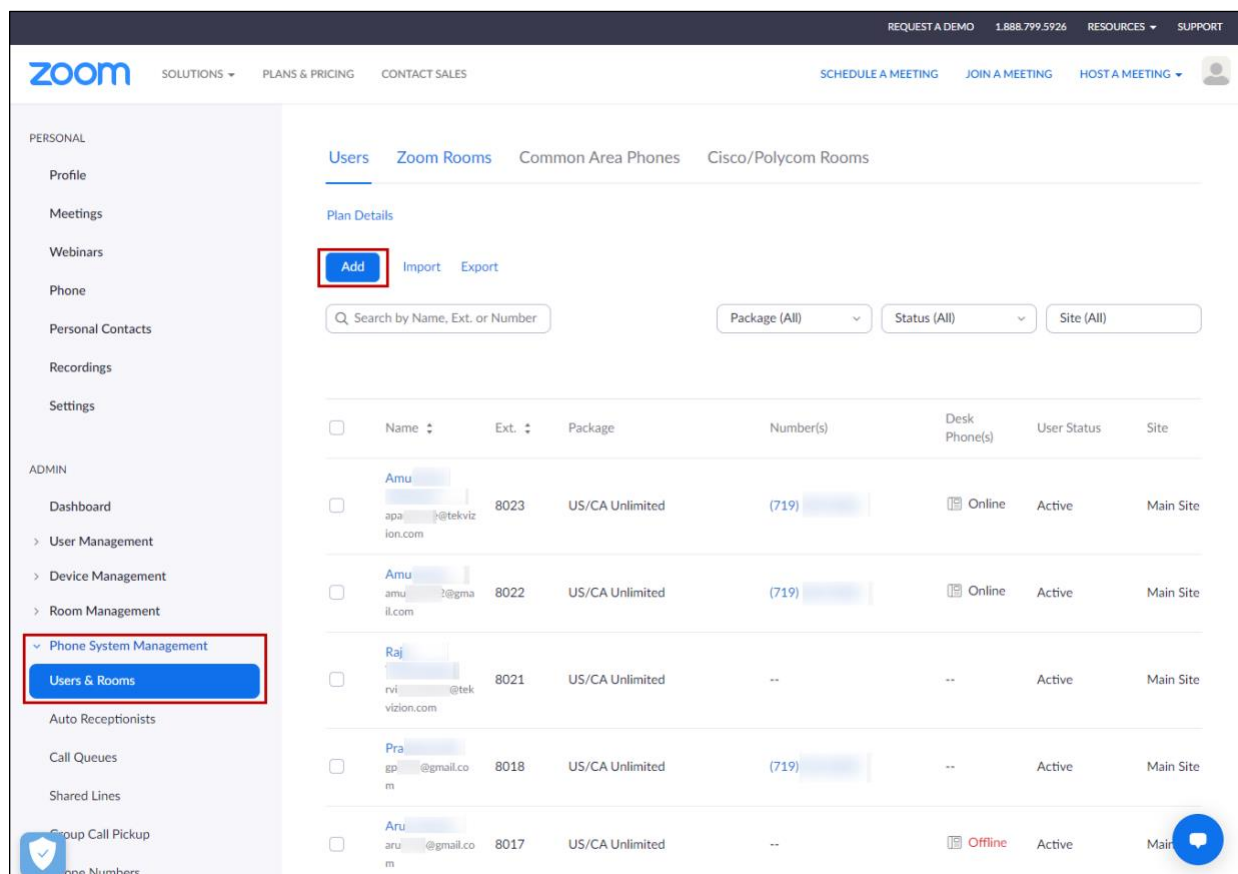


Figure 2 : Add Users and Rooms

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

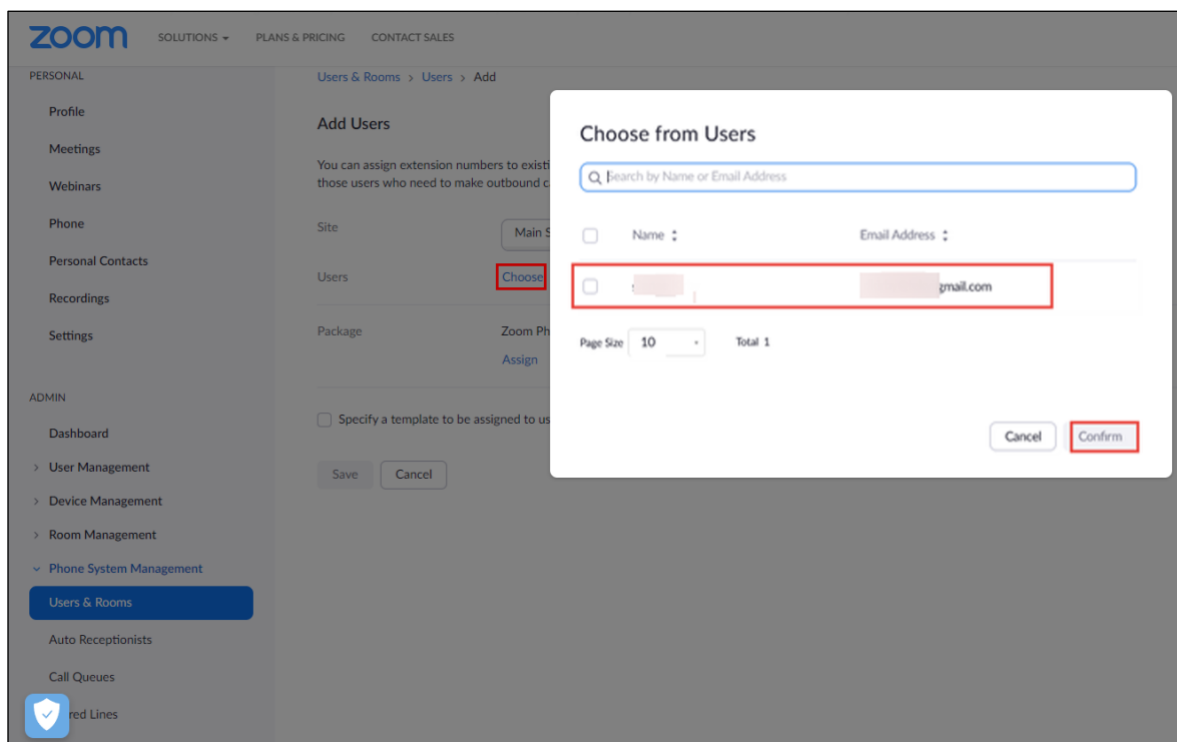


Figure 3 : Choose user

- 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.

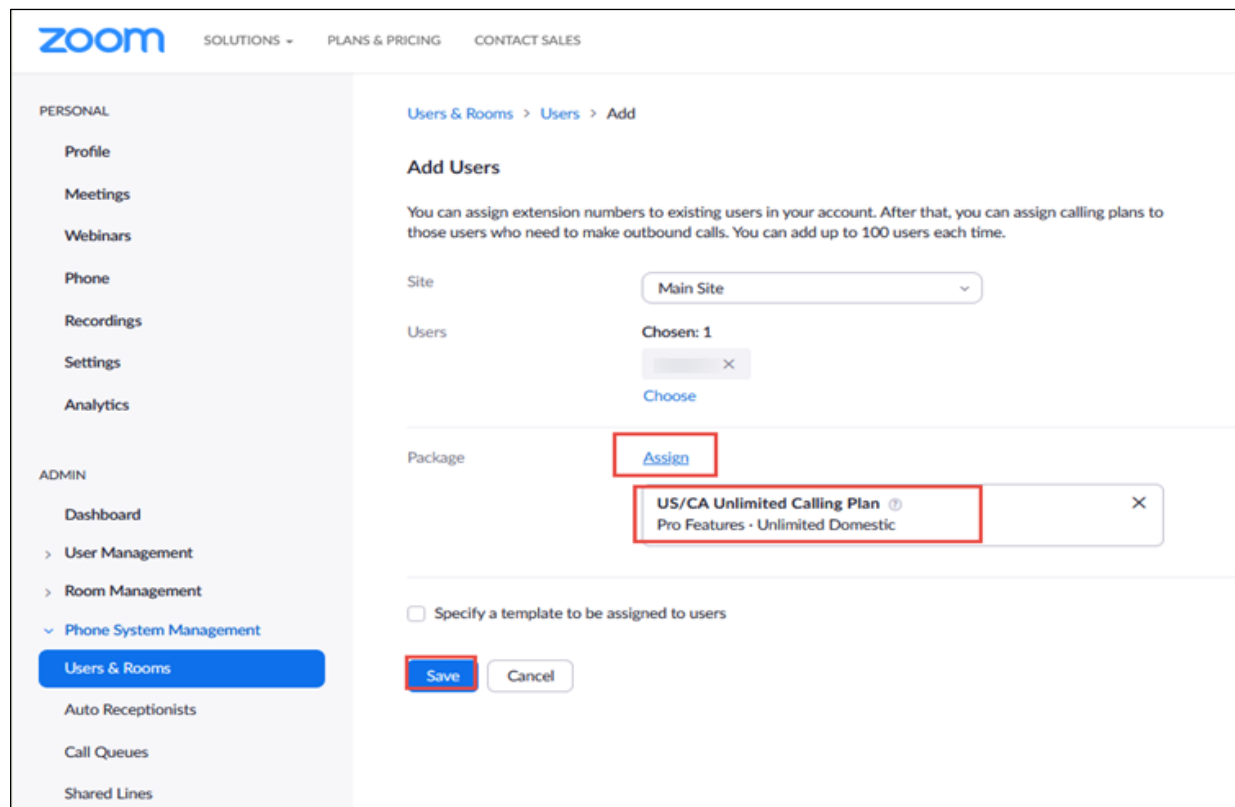


Figure 4 : Assign Calling Plan

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

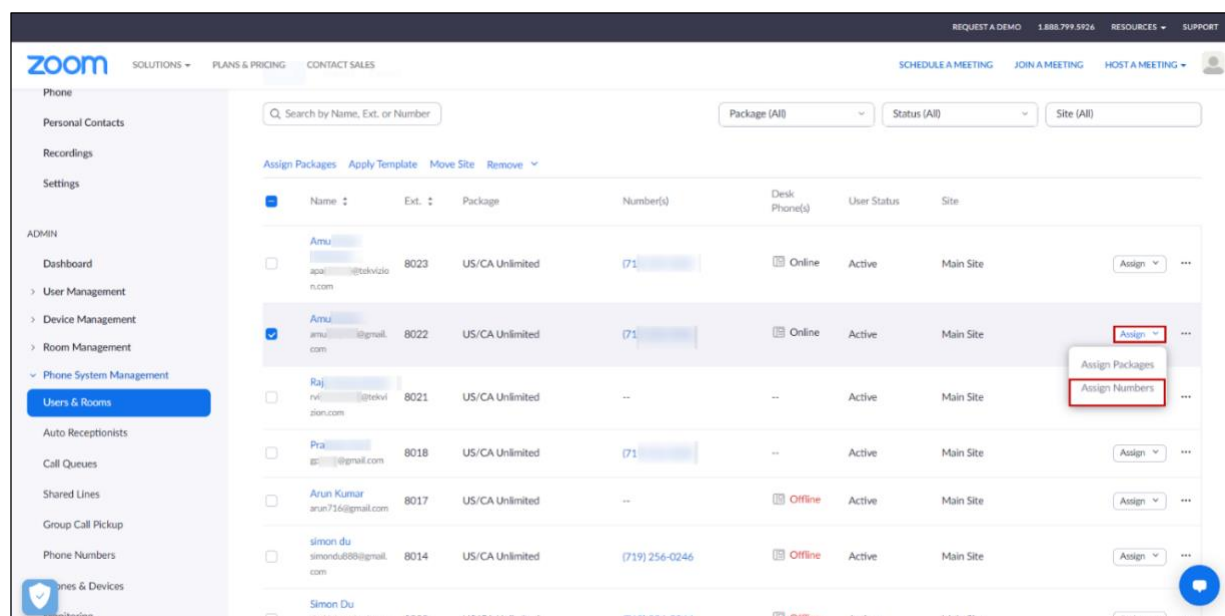


Figure 5 : Assign Number



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

**Assign Numbers**

Country/Region: United States

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

<input checked="" type="checkbox"/>	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

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

- Set **Display Name**: GRP2615-02 is set as an example
- Set **MAC Address**: add the GRP2615-02 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**

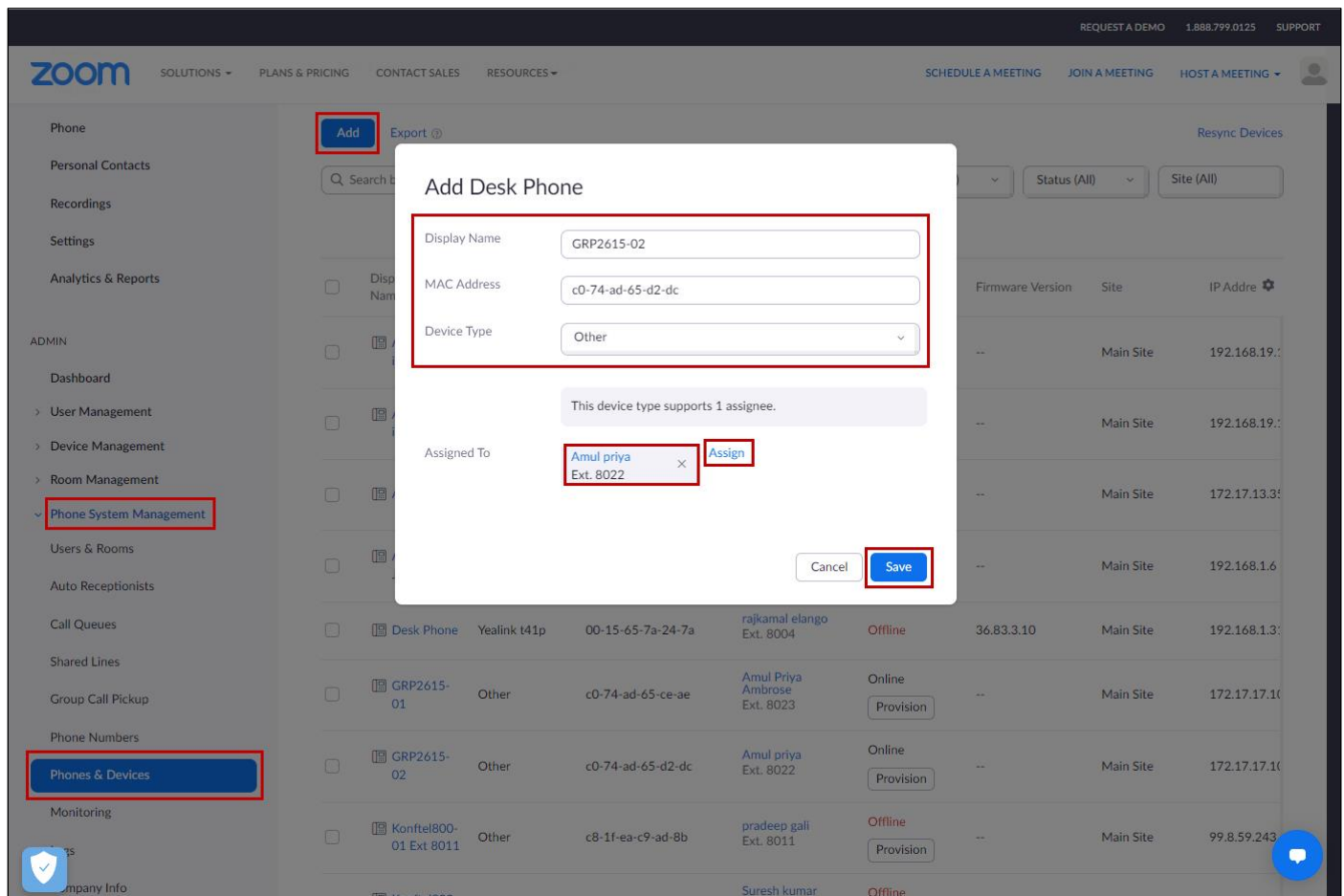


Figure 7 : Add Desk Phone

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 GRP2615-02 provisioning ([section Provisioning through Phone's Web Interface](#)).
3. Download the Certificates and import to the device, so that device will be considered in the trust list. **Note:** By default, GRP2615 natively supports DigiCert CA. So, uploading certificates manually is not required.

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PERSONAL

- Profile
- Meetings
- Webinars
- Phone
- Personal Contacts
- Recordings
- Settings
- Analytics & Reports

ADMIN

- Dashboard
- User Management
- Device Management
- Room Management
- Phone System Management
  - Users & Rooms
  - Auto Receptionists
  - Call Queues
  - Red Lines

Phones & Devices > Assigned > GRP2615-02

**GRP2615-02** Rename

Profile

Site Main Site

Assigned To Amul p Ext. 80

IP Address 172.17.1

Device Type Other

Firmware Version --

MAC Address c0-74-ad

Provision Template Unsuppor

Status Online

Provision Remove

**Provisioning**

MAC Address c0-74-ad-65-d2-dc

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: 31670
4. Authorization ID: 83326
5. Password: 8C iq

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 8 : Provisioning

### 3. Grandstream GRP2615 Provisioning

This section provides instructions on how to configure Grandstream GRP2615 to register to Zoom Phone Services.

### 3.1 Deployment Topology Diagram

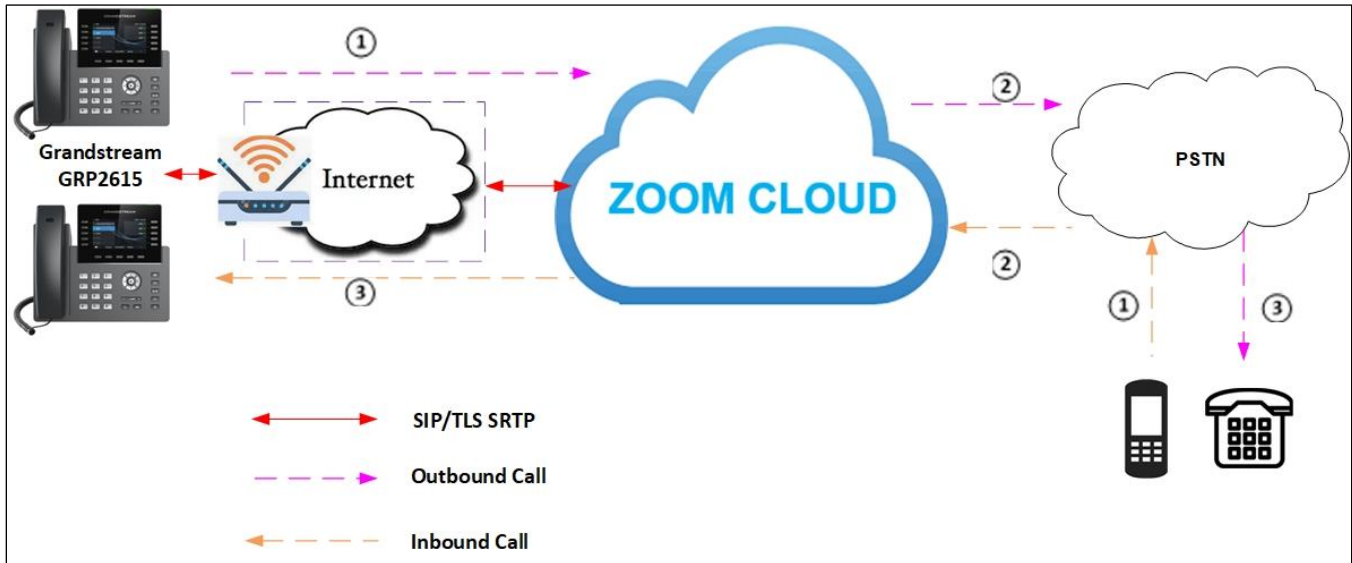


Figure 9 : Network Diagram

### 3.2 Network

By default, GRP2615 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 GRP2615 by navigating the physical phone: **Menu -> Status -> Network Status -> Ethernet**.

### 3.3 Firmware Upgrade

This section ensures the phone is upgraded with the required firmware. The firmware used for this test is **1.0.7.19**. From the phone's home page, Navigate to: **Maintenance -> Upgrade and Provisioning**.

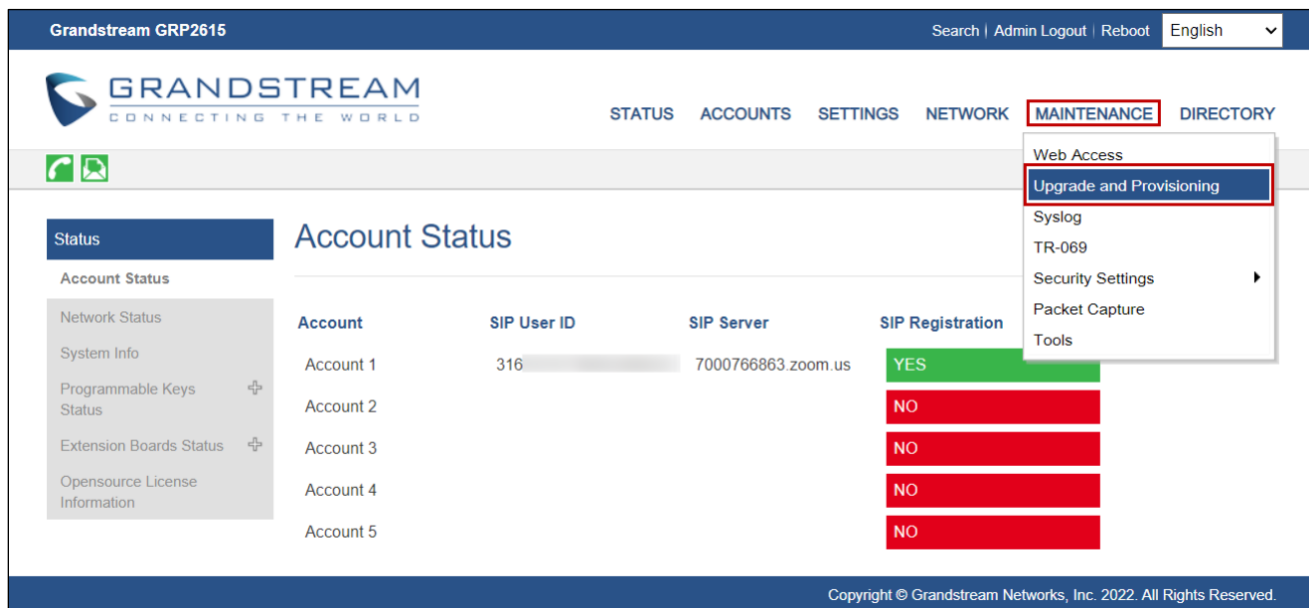


Figure 10 : Firmware Upgrade

- Click **Start** for Upgrade.
- Please select and upload the firmware file from the system and then click **Start** (Phone may have to restart to get applied with the changes).

The screenshot shows the Grandstream GRP2615 web interface. The top navigation bar includes 'Search', 'Admin Logout', 'Reboot', and a language dropdown set to 'English'. The main menu has 'STATUS', 'ACCOUNTS', 'SETTINGS', 'NETWORK', 'MAINTENANCE', and 'DIRECTORY'. The 'Maintenance' sidebar is active, showing 'Web Access', 'Upgrade and Provisioning', 'Syslog', 'TR-069', 'Security Settings', 'Security', 'Trusted CA Certificates', 'Keypad Lock', 'Packet Capture', and 'Tools'. The 'Upgrade and Provisioning' section is titled 'Upgrade and Provisioning'. It contains a 'Start' button (highlighted with a red box) and several settings: 'Firmware Upgrade and Provisioning' (radio buttons for 'Always check for new firmware', 'Check new firmware only when F/W pre/suffix changes', and 'Always skip the firmware check'); 'Always Authenticate Before Challenge' (radio buttons for 'No' and 'Yes'); 'Validate Hostname in Certificate' (radio buttons for 'No' and 'Yes'); 'Allow DHCP Option 43 and Option 66 to Override Server' (radio buttons for 'No', 'Yes', and 'Prefer, fallback when failed'); 'Additional Override DHCP Option' (a dropdown menu set to 'Option 150'); 'Allow DHCP Option 120 to Override SIP Server' (radio buttons for 'No' and 'Yes'); '3CX Auto Provision' (radio buttons for 'No' and 'Yes'); 'Automatic Upgrade' (radio buttons for 'No', 'Yes, check for upgrade every 1008 minute(s)', 'Yes, check for upgrade every day', and 'Yes, check for upgrade every week'); and 'Start Upgrade at Random Time' (radio buttons for 'No' and 'Yes').

Figure 11 : Firmware Upgrade (Cont.)

## 3.4 Provisioning

The GRP2615 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>5D2DC</P270>
<!-- SIP User ID -->
<P35>31670XXXXXXXXXXXX223</P35>
<!-- Authenticate ID -->
<P36>83326XXXXX10</P36>
<!-- Authenticate password -->
<P34>8GXXXXXq</P34>
<!-- Display Name (John Doe) -->
<P3>8022</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**.

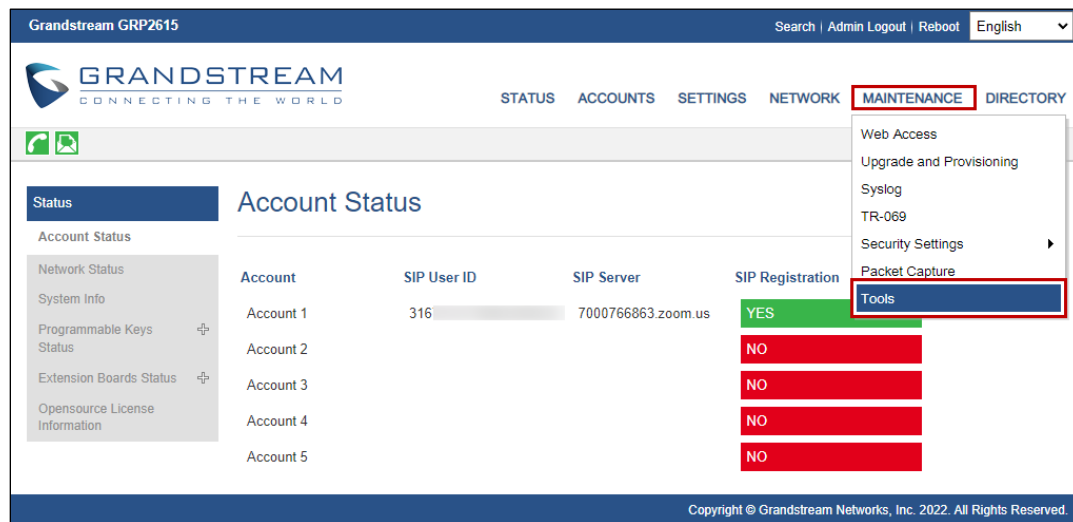


Figure 12 : Factory reset

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

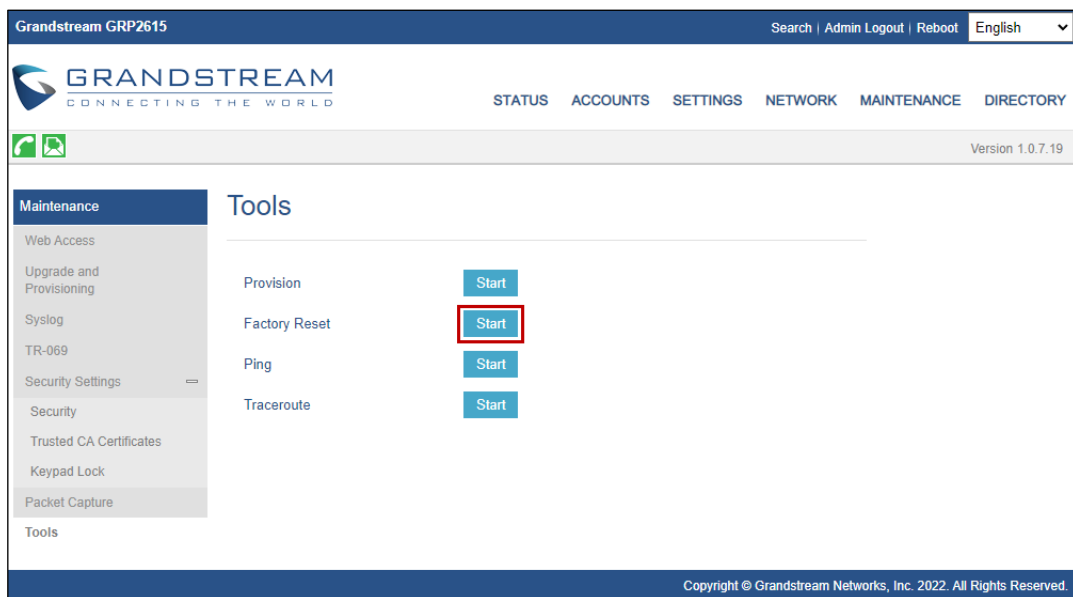


Figure 13 : Factory reset - (Cont.)

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

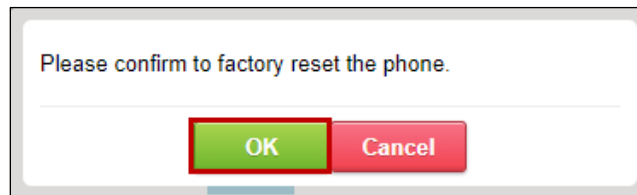


Figure 14 : Factory reset - (Cont.)

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 enter the **default password** displayed at the back of the unit and click **Login**.

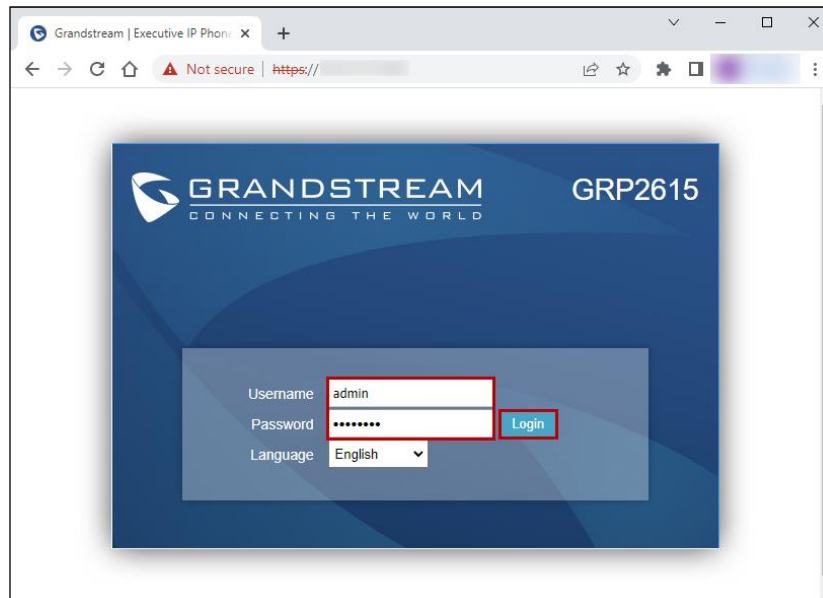


Figure 15 : Login details

6. Phone prompts to update with the **New Password** instead of Default password. Enter the **New Password** and **Confirm Password**. Click **Save**. On saving, a notification appears saying that Password has been changed successfully.

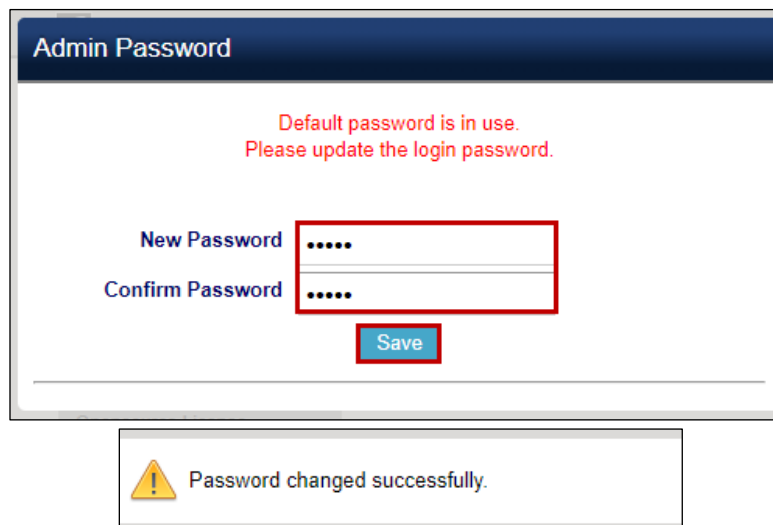


Figure 16 : Login details - (Cont.)



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

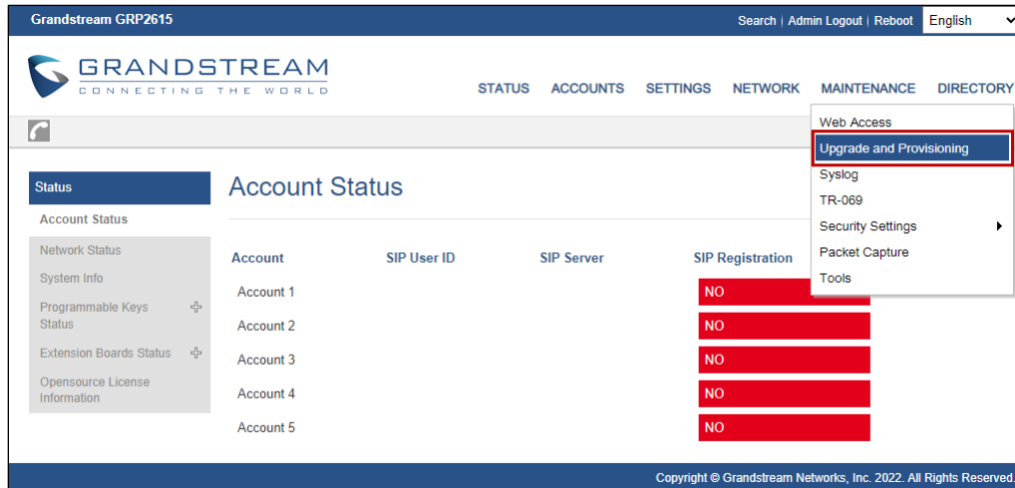


Figure 17 : Provisioning through a HTTP server

8. Scroll down and look for **Config** header. Set **HTTP** in Config Upgrade via and enter (**http://ipaddress:90/Folder Path**) in Config Server Path.
9. Click **Save and Apply**.

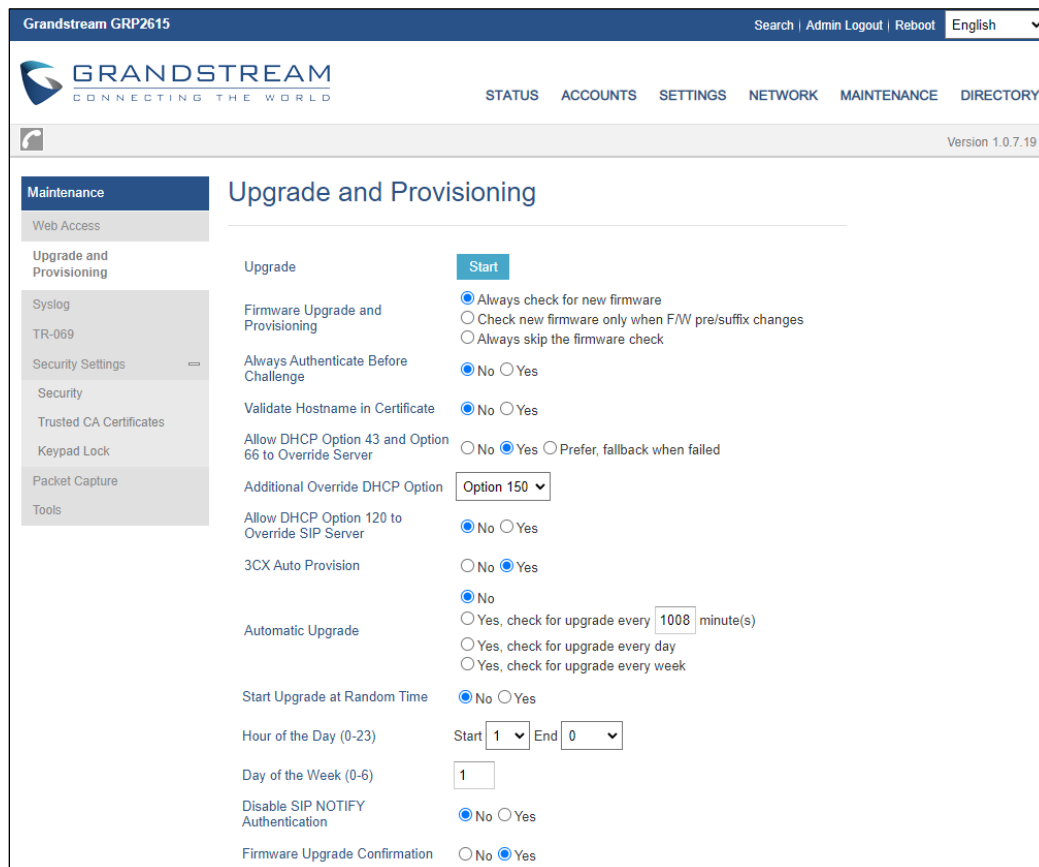


Figure 18 : Provisioning through a HTTP server - (Cont.)

**Config**

Config Upgrade via ☐ TFTP ☒ HTTP ☐ HTTPS ☐ FTP ☐ FTPS

Config Server Path **172.16.1.100:90/amul**

Config Server Username

Config Server Password

Config File Prefix

Config File Postfix

XML Config File Password

Authenticate Conf File ☒ No ☐ Yes

Download Device Configuration [Download](#)

Download Device Configuration (XML) [Download](#)

User Protection ☒ Off ☐ On

Download and Process All Available Config Files ☒ No ☐ Yes

Download User Configuration [Download](#)

Upload Device Configuration [Upload](#)

Export Backup Package [Download](#)

Restore from Backup Package [Upload](#)

**Firmware**

Firmware Upgrade via ☐ TFTP ☒ HTTP ☐ HTTPS ☐ FTP ☐ FTPS

Firmware Server Path **fm.grandstream.com/gs**

Firmware Server Username

Firmware Server Password

Firmware File Prefix

Firmware File Postfix

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

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Figure 19 : Provisioning through a HTTP server - (Cont.)

10. After the phone restarts, the phone will start downloading the necessary configuration files from the HTTP server and get provisioned.
11. Navigate to **Status** -> **Account Status** to verify the provision status.

Grandstream GRP2615 Search | Admin Logout | Reboot | English

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**GRANDSTREAM**  
CONNECTING THE WORLD

STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE DIRECTORY

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Version 1.0.7.19

**Status** **Account Status**

Account	SIP User ID	SIP Server	SIP Registration
Account 1	316	7000766863.zoom.us	YES
Account 2			NO
Account 3			NO
Account 4			NO
Account 5			NO

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Figure 20 : Account Status

### 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](#)).

- GRP2615 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, **5D2DC** 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. SIP Authentication ID: **Provide the SIP Authentication ID** (from Zoom provisioning).
  7. SIP Authentication Password: **Provide the SIP Authentication Password** (from Zoom provisioning).
  8. Name: Enter the Name of your choice. Here, **8022** is used.
  9. Click **Save and Apply**.

GRANDSTREAM  
CONNECTING THE WORLD

STATUS **ACCOUNTS** SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.7.19

Accounts

Account 1

General Settings

Dial Plan

Network Settings

SIP Settings

Audio Settings

Call Settings

Intercom Settings

Feature Codes

Account 2

Account 3

Account 4

Account 5

Account Swap

General Settings

Account Active ☐ No ☒ Yes

Account Name 5D2DC

SIP Server 7000766863.zoom.us

Secondary SIP Server

Outbound Proxy us01sip0h.sc.zoom.us:5091

Secondary Outbound Proxy

BLF Server

SIP User ID 31670747798852999223

SIP Authentication ID 833260556010

SIP Authentication Password .....

Name 8022

Voicemail Access Number

Picture Select

Account Display ☒ Username ☐ User ID

Save Save and Apply Reset

Figure 21 : 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 GRP2615

Search | Admin Logout | Reboot | English

GRANDSTREAM  
CONNECTING THE WORLD

STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.7.19

**Accounts**

Account 1

General Settings

Dial Plan

**Network Settings**

SIP Settings

Audio Settings

Call Settings

Intercom Settings

Feature Codes

Account 2

Account 3

Account 4

Account 5

Account Swap

## Network Settings

DNS Mode: SRV

DNS SRV Failover 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:

Use SBC: ☒ No ☐ Yes

Save Save and Apply Reset

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Figure 22 : 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 GRP2615 Search | Admin Logout | Reboot English

**GRANDSTREAM**  
CONNECTING THE WORLD

STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.7.19

Accounts

Account 1

General Settings

Dial Plan

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 3

Account 4

Account 5

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 Tries

Local SIP Port

Registration Retry Wait Time

SIP SUBSCRIBE Failure Retry Wait Time

Maximum Number of SIP Request Retries

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 MMI ☒ 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

Use Route Set In NOTIFY ☐ No ☒ Yes

Save Save and Apply Reset

Figure 23 : 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**

Grandstream GRP2615

Search | Admin Logout | Reboot English

GRANDSTREAM  
CONNECTING THE WORLD

STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.7.19

**Accounts**

- Account 1
- General Settings
- Dial Plan
- Network Settings
- SIP Settings**
- Basic Settings
- Custom SIP Headers
- Advanced Features
- Session Timer**
- Security Settings
- Audio Settings
- Call Settings
- Intercom Settings
- Feature Codes

## Session Timer

Enable Session Timer ☐ No ☒ Yes

Session Expiration

Min-SE

Caller Request Timer ☒ No ☐ Yes

Callee Request Timer ☒ No ☐ Yes

Force Timer ☒ No ☐ Yes

UAC Specify Refresher ☒ UAC ☐ UAS ☐ Omit (Recommended)

UAS Specify Refresher ☒ UAC ☐ UAS

Force INVITE ☒ No ☐ Yes

Figure 24 : 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**

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**GRANDSTREAM**  
CONNECTING THE WORLD

STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.7.19

**Accounts**

Account 1

General Settings

Dial Plan

Network Settings

SIP Settings

**Audio Settings**

Call Settings

Intercom Settings

Feature Codes

Account 2

Account 3

Account 4

Account 5

Account Swap

## Audio Settings

Preferred Vocoder - choice 1 PCMU

Preferred Vocoder - choice 2 PCMA

Preferred Vocoder - choice 3 G.723.1

Preferred Vocoder - choice 4 G.729A/B

Preferred Vocoder - choice 5 G.722 (wide band)

Preferred Vocoder - choice 6 iLBC

Preferred Vocoder - choice 7 G.726-32

Preferred Vocoder - choice 8 OPUS

Use First Matching Vocoder in 200OK SDP ☒ No ☐ Yes

Codec Negotiation Priority ☐ Caller ☒ Callee

Hide Vocoder ☒ No ☐ Yes

Configures to enable or disable multiple m lines in SDP. ☒ No ☐ Yes

RTCP Port Selection Default

**SRTP Mode** Enabled and Forced

**SRTP Key Length** AES 128&256 bit

Crypto Life Time ☐ No ☒ Yes

Symmetric RTP ☒ No ☐ Yes

Silence Suppression ☒ No ☐ Yes

Jitter Buffer Type Adaptive

Jitter Buffer Length 300ms

Voice Frames per TX 2

G723 Rate ☐ 6.3kbps encoding rate ☒ 5.3kbps encoding rate

G.726-32 Packing Mode ☒ ITU ☐ IETF

iLBC Frame Size ☐ 20ms ☒ 30ms

iLBC Payload Type 97

Opus Payload Type 123

DTMF Payload Type 101

Send DTMF ☐ in-audio ☒ via RTP (RFC2833) ☐ via SIP INFO

DTMF Delay 250

Save **Save and Apply** Reset

Figure 25: Audio Settings

- Under Account 1 -> **SIP Settings** -> **Advanced Features**. Ensure the below parameters are set.
  1. Special Feature: **Zoom**
  2. Click **Save and Apply**

**GRANDSTREAM**  
CONNECTING THE WORLD

STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.7.19

**Accounts**

Account 1

General Settings

Dial Plan

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 3

Account 4

Account 5

Account Swap

## Advanced Features

Line-seize Timeout: 15

Presence Eventlist URI:

Eventlist BLF URI:

Auto Provision Eventlists: ☒ Disabled ☐ BLF Eventlist ☐ Presence Eventlist

Conference URI:

Music On Hold URI:

BLF Call-pickup: ☒ Auto ☐ Force BLF Call-pickup by prefix ☐ Disabled

BLF Call-pickup Prefix: \*\*

Call Pickup Barge-in Code:

PUBLISH for Presence: ☒ Disabled ☐ Enabled

Omit charset=UTF-8 in MESSAGE: ☒ Disabled ☐ Enabled

Allow Unsolicited REFER: ☒ Disabled ☐ Enabled ☐ Enabled/Force Auth

Feature Key Synchronization: ☒ Disabled ☐ Enabled

**Special Feature**: Zoom

## BroadSoft

BroadSoft Call Center: ☒ Disabled ☐ Enabled

Hoteling Event: ☒ Disabled ☐ Enabled

Call Center Status: ☒ Disabled ☐ Enabled

BroadSoft Executive Assistant: ☒ Disabled ☐ Enabled

BroadSoft Call Park: ☒ Disabled ☐ Enabled

## VQ RTCP-XR

VQ RTCP-XR Collector Name:

VQ RTCP-XR Collector Address:

VQ RTCP-XR Collector Port: 5060

Enable RTCP: ☐ Disable ☐ RTCP ☒ RTCP-XR

Save Save and Apply Reset

Figure 26: 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**

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CONNECTING THE WORLD

STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE DIRECTORY

Version 1.0.7.19

**Accounts**

- Account 1
  - General Settings
  - Dial Plan
  - Network Settings
  - SIP Settings
  - Audio Settings
  - Call Settings
  - Intercom Settings
  - Feature Codes**
  - Account 2
  - Account 3
  - Account 4
  - Account 5
  - Account Swap

## Feature Codes

Enable Local Call Features ☐ No ☒ Yes ☐ Call Forwarding & DND only

### Do Not Disturb (DND)

On

Off

### Call Forward Always

On

Off

Target

### Call Forward Busy

On

Off

Target

### Call Forward No Answer

On

Off

Target

Call Forward No Answer Timeout (s)

Figure 27: 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).

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STATUS ACCOUNTS **SETTINGS** NETWORK MAINTENANCE DIRECTORY

Version 1.0.7.19

**Settings**

**General Settings**

BroadSoft  
External Service  
Call Features  
Multicast Paging  
Outbound Notification  
Preferences  
Programmable Keys  
Extension Boards  
Web Service  
XML Applications  
Voice Monitoring  
E911 Service

Local RTP Port: 5004  
Local RTP Port Range: 200  
**Use Random Port: ☒ No ☐ Yes**  
Keep-Alive Interval: 20  
Use NAT IP  
STUN server  
Delay Registration: 0  
Test Password Strength: ☒ No ☐ Yes

**Public Mode**

Enable Public Mode: ☒ Disabled ☐ Enabled  
Enable Fix For RTP Timestamp Jump: ☒ No ☐ Yes  
Public Mode Username Prefix  
Public Mode Username Suffix  
Enable Remote Synchronization: ☒ Disabled ☐ Enabled  
Server Type: ☒ TFTP ☐ FTP ☐ HTTP  
Server Path  
FTP/HTTP Username  
FTP/HTTP Password

**Outbound Notification**

Enable Outbound Notification: ☐ Disabled ☒ Enabled

**UCM Call Center**

Enable UCM Call Center Fast Login/Logout: ☒ Disabled ☐ Enabled

Save Save and Apply Reset

Figure 28: Settings-General Settings