Grandstream Networks, Inc.

UCM6200 Basic Configuration Guide

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OVERVIEW

This document will provide instructions on how to set up a UCM6200 series from an out of the box state to a fully functional state. This includes design considerations, creating user extensions, provisioning endpoints with Zero-Config, conferencing, auto attendant configuration, analog/VoIP trunks, routing inbound/outbound calls and voicemail/fax to email setup.

For detailed information in regard to parameters that are encountered in this guide, please check the UCM6200 series User Manual

SETUP GUIDE SCENARIO

The image below shows the most typical setup where you have the UCM6200 series WAN port connected to the Internet and the LAN side of the UCM6200 series providing DHCP.

We will use this scenario to setup a Grandstream GXP2135 which will be connected to the LAN side of the UCM6200 series. There can also be a network switch connected to the LAN port of the UCM6200 series so that there may be more than one device connected.

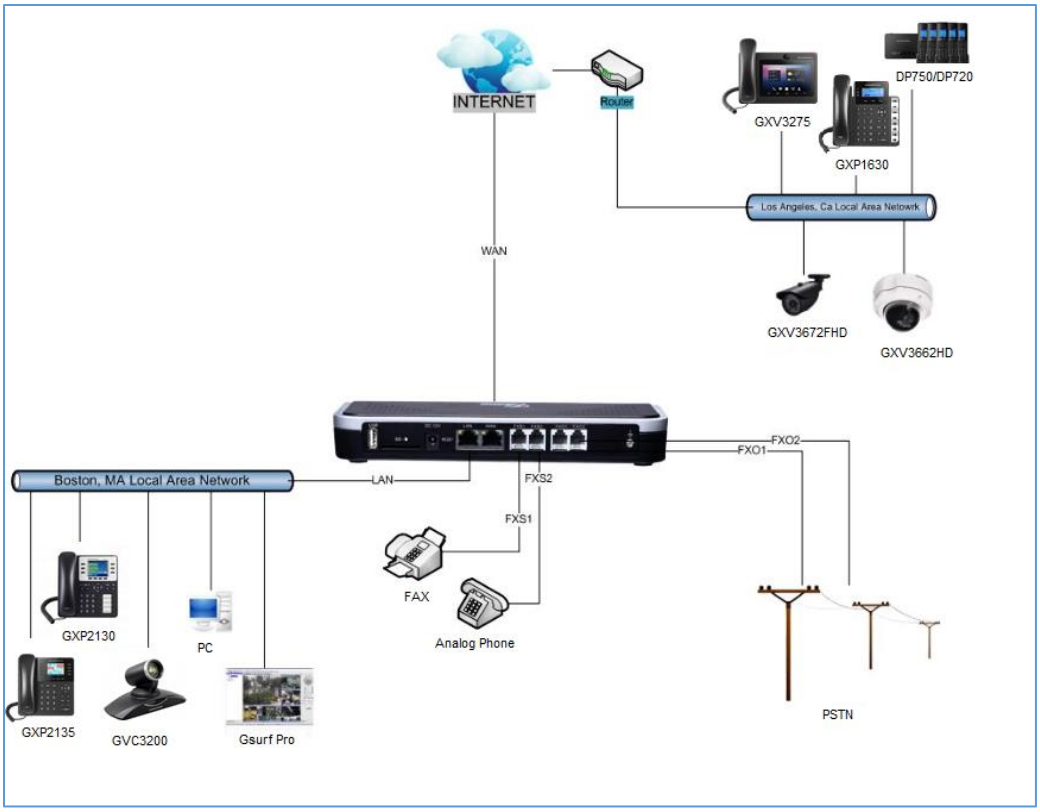
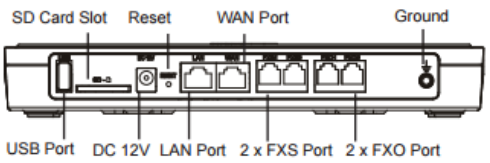


Figure 1: Typical UCM6200 Scenario

QUICK INSTALLATION

Connecting the UCM6200



1. Connect the UCM6202 **WAN** port to the uplink port of an Ethernet switch/hub with an RJ45 Ethernet cable.
2. Connect the power adapter.
3. Once the UCM6202 boots up and connects to the network, the LED indicator for WAN port will be in solid green and the LCD shows up the IP address.
4. (Optional) Connect PSTN lines from the wall jack to the FXO ports; connect analog lines (phone and fax) to the FXS ports.

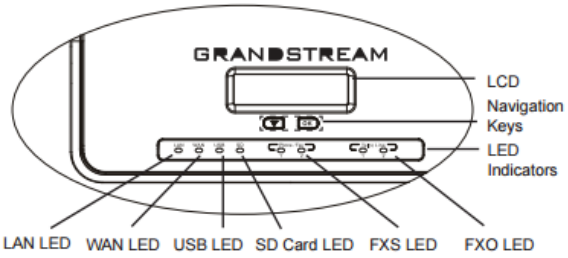


Figure 2: Quick Installation Guide for UCM6202

Access UCM6200 series Web Interface

Accessing the UCM6200 series web interface allows users to manage users and system settings locally as well as remotely.

The IP address used to access the web UI depends on where the user's computer is connected. If the computer is connected to the same switch/router that the UCM6200 series WAN port is connected, then browse to the IP address that is displayed on the UCM6200 series LCD. This address is the WAN IP. If the computer is connected to the LAN side of the UCM6200 series, then users would browse to the default IP of the UCM6200 series which is 192.168.2.1. If connected successfully, the UCM6200 series login page will be displayed as shown below.

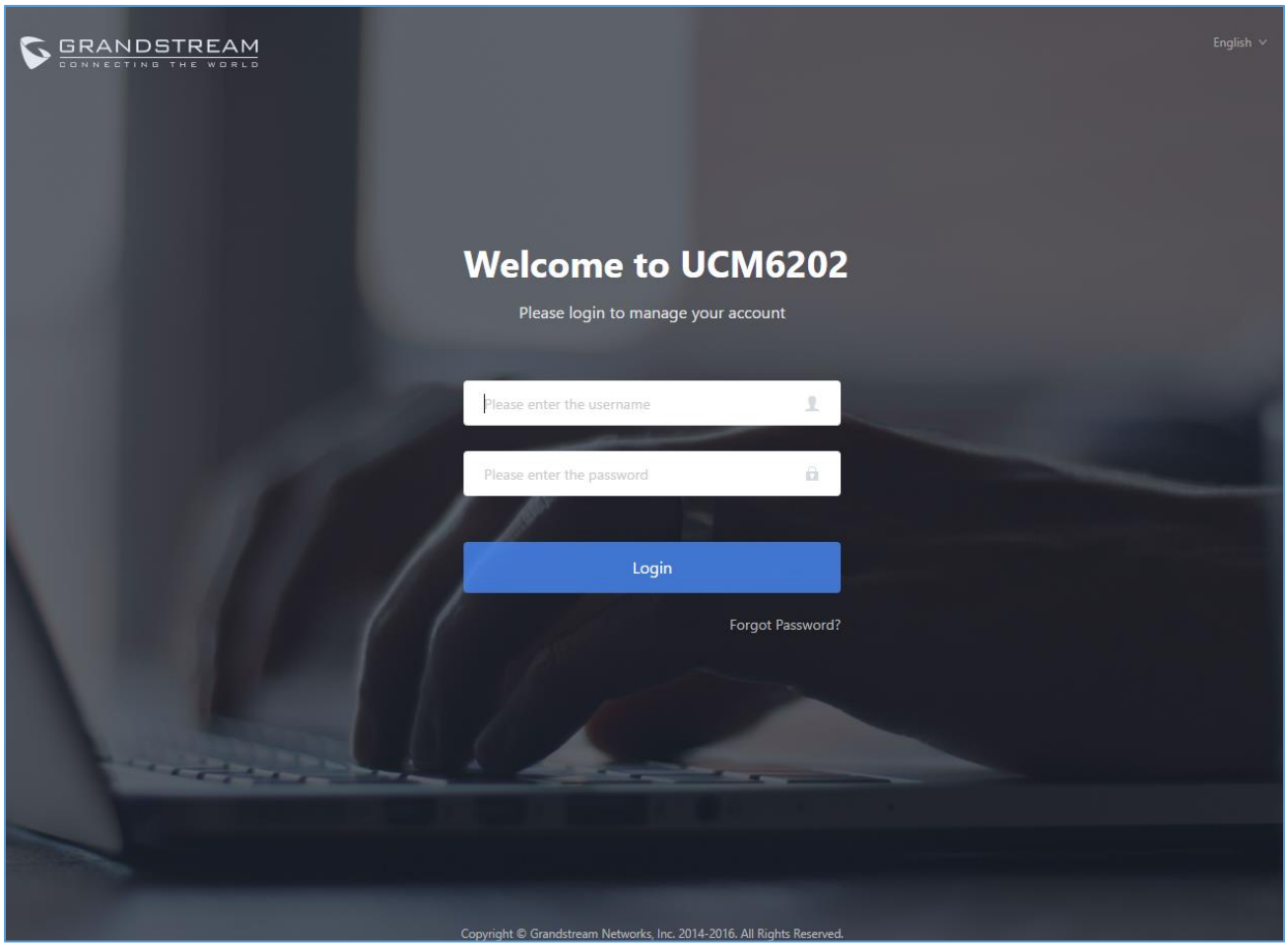


Figure 3: Quick Installation Guide – Login Screen

CREATE USER EXTENSION

Configure Extension Range

First part of configuring the UCM6200 series should be about planning for expansion. Here are some questions to think about when setting up extension ranges:

- How many users are in the office?
- Will there be departments within the office?
- Is this office interconnecting with another office?

Once there is a clear picture of how many users will be at each location, how departments are going to be segmented and what is the expected growth of the company, a user can then configure the extension range. Navigate to PBX Settings → General Settings → General.

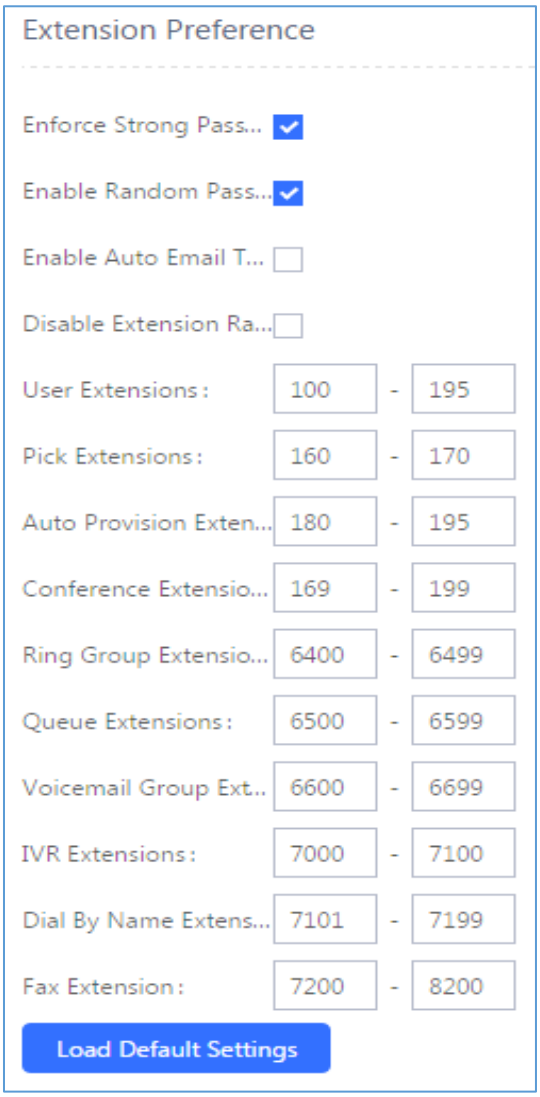


Figure 4: Create User Extension – Extension Range

In the figure above, the user extension range is set with a starting extension of 100 and ending at 195. This allows up to 196 extensions to assign to users. We have also created a conference extension range from 196 to 199, which gives 4 conference bridges. Users can configure any extension range as they desire. Here we've configured it to use three digit extensions with a leading 1.

Click "Save" at the bottom. Then remember to click on "Apply Changes" at the top so that our extension range will be ready for the next steps.

Batch & Single User Creation

So now that there are extension ranges configured, we can now begin creating users to prepare for the provisioning process. There are two methods of creating a user. One method would be creating a single user. The other method would be creating a batch of users.

Steps on Adding Single User

1. Navigate to Extension / Trunk→Extensions. For first time setups users will see "No Data".

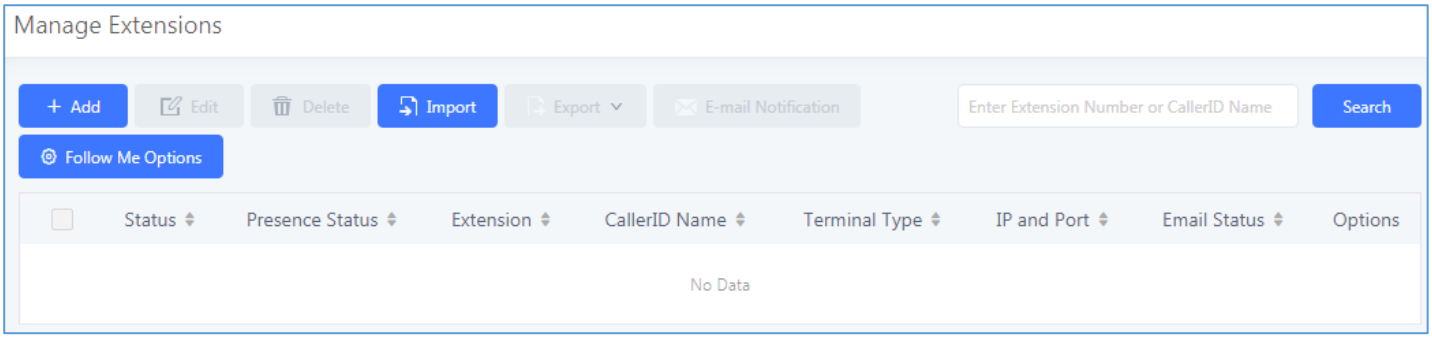


Figure 5: Create User Extension – Extensions

2. Click on "Add".
3. Choose an Extension Type from the "Select Extension Type" dropdown.
4. On the "Create New Extension" screen, users can enter in quite a few options, but they are not required for this tutorial. Click "Save" at the bottom of the page.

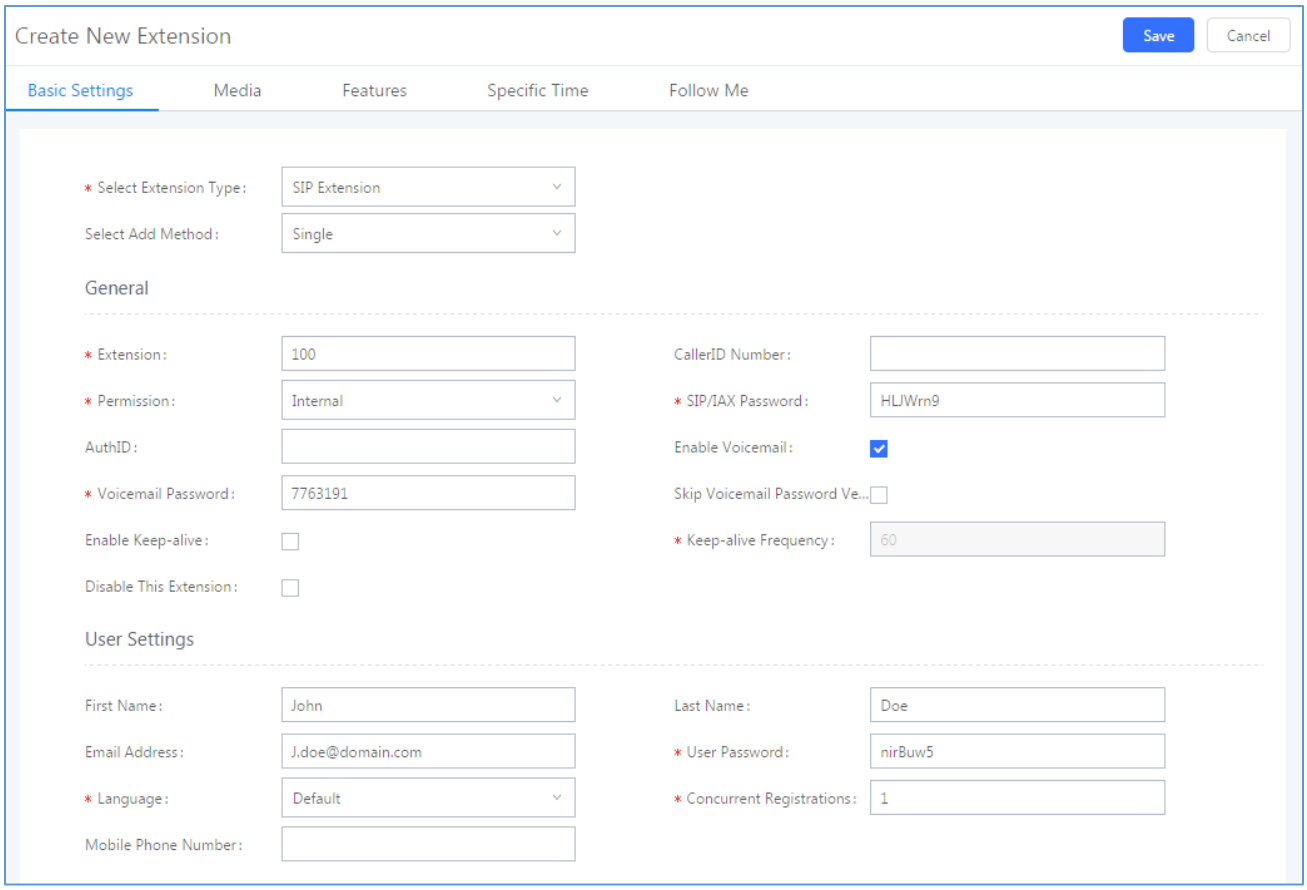


Figure 6: Create User Extension – Create New User

After clicking "Save" the Extensions page will display with the single SIP extension that was just created. Here the user has the ability to edit the extensions by clicking the at the right.

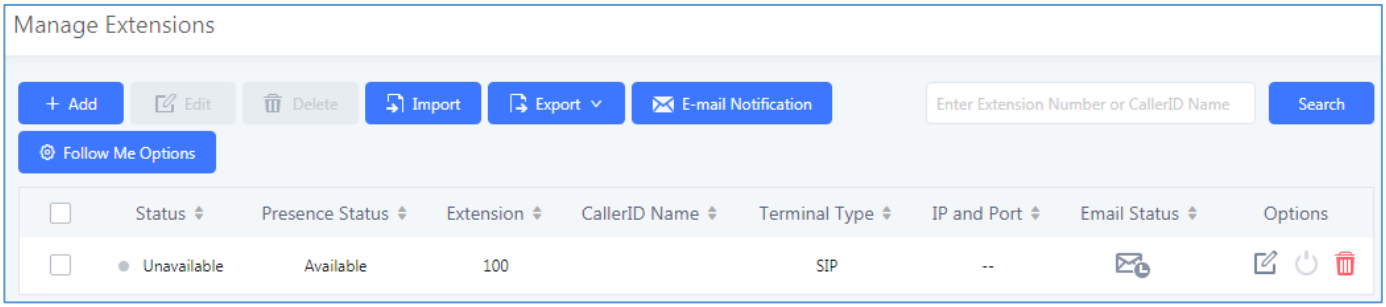


Figure 7: Create User Extension – Created Single User

Steps on Adding Batch of Users

1. Navigate to **Extension / Trunk→Extensions**.
2. Click on "Add".
3. Choose "Batch" from "Select Add Method".

4. At the “Create New Extension” dialog, the user can specify the “Extension” which is the start extension number and the number of extensions to generate by setting the “Create Number”. In this tutorial, we will have a starting extension of 101 and have the UCM6200 series generate 5 extensions with a password randomly generated. This would create extensions 101, 102, 103, 104 and 105.

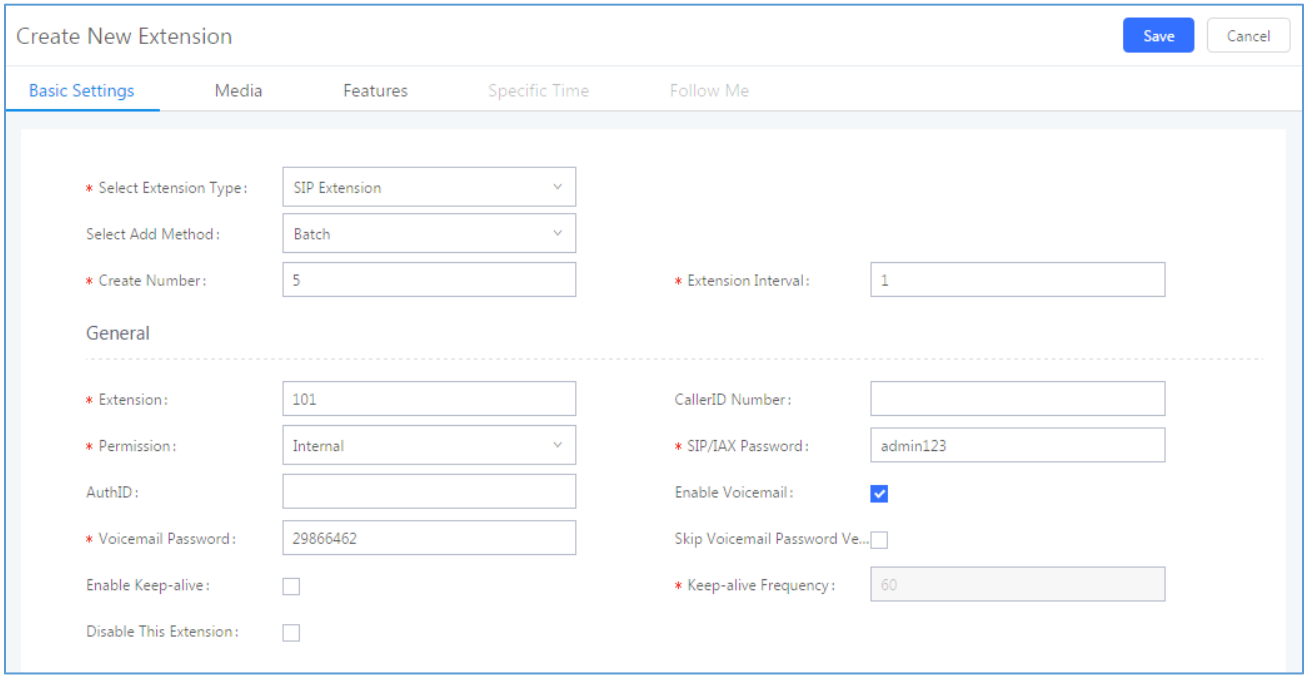


Figure 8: Create User Extension – Batch Add User

5. Click “Save” on top right to have the UCM6200 series generate the extensions.

6. Next, a prompt will appear asking “Are you sure you want to create users: 101, 102, 103, 104 105”. Click “OK”.

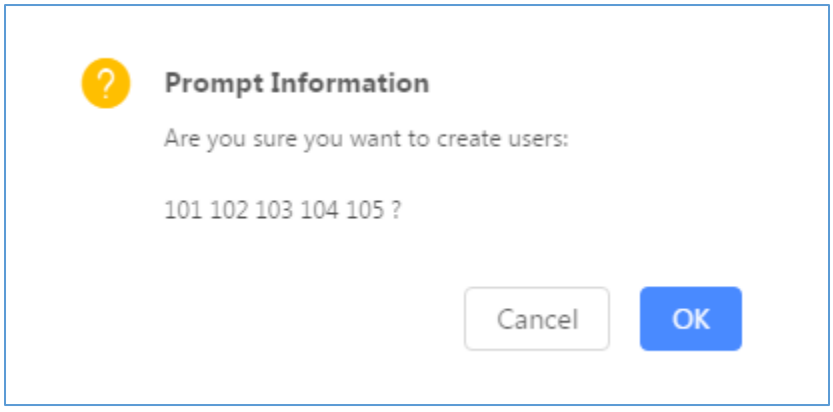


Figure 9: Create User Extension – Create User Prompt

At this stage, the Extensions page should look similar to this:

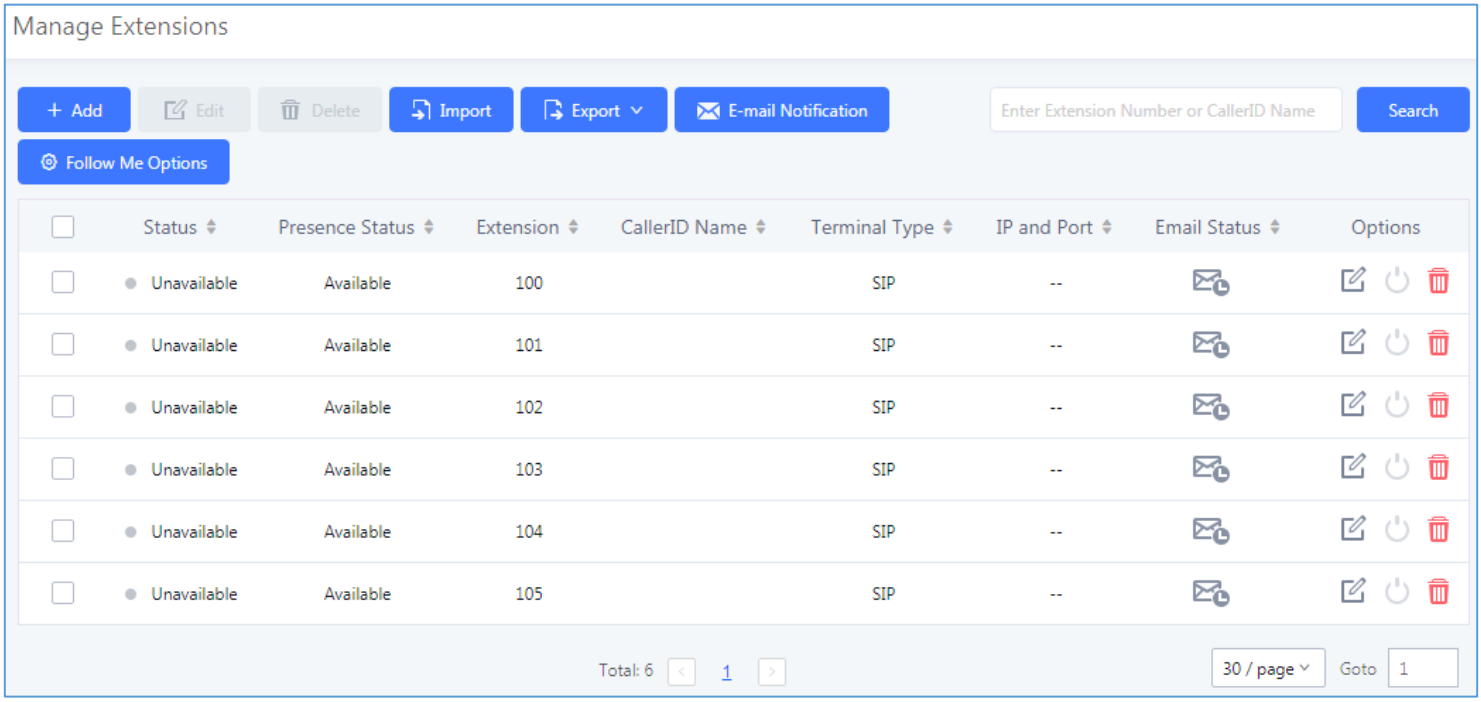


Figure 10: Create User Extension – Single and Batch Add Extensions Created

For more details on Extension, parameters please see the UCM6200 series User Manual.

PROVISIONING & ZERO CONFIG

Grandstream SIP Devices can be configured via Web interface as well as via configuration file through TFTP/HTTP/HTTPS download. All Grandstream SIP devices support an XML format configuration file. The UCM6200 provides a Plug and Play mechanism to auto-provision the Grandstream SIP devices in a zero configuration manner by generating XML config file and having the phone to download it. This allows users to finish the installation with ease and start using the SIP devices in a managed way.

To provision a phone, three steps are involved, i.e., discovery, assignment and provisioning. The UCM6200 creates XML config file to the detected/assigned Grandstream device and accomplishes the following configurations on the device after the provisioning:

- A UCM6200 extension will be assigned and registered on the phone.
- SIP-related network settings such as "NAT traversal" and "Use Random Port" are configured on the phone.
- Call settings such as "Dial Plan" and "Auto Answer".
- LDAP client configurations will be set up automatically on the phone to use the default LDAP directory generated in the UCM6200 LDAP server.

This section explains how zero config works on the UCM6200. The settings for this feature can be accessed via Web GUI→**Value-added Features**→**Zero Config**.

Auto Discovering

When referring back to our typical scenario in the beginning of this tutorial, we know that there is a GXP2135 connected to the LAN side of the UCM6200 series. The UCM6200 series LAN interface has DHCP enabled and has assigned the GXP2135 with an IP address. By default, the IP assigned would be within the 192.168.2.X range. When the UCM6200 series offers an IP address to the phone it also offers DHCP Option 66, which provides the phone with a config server path that points to the UCM6200 series. All of Grandstream VoIP devices have Option 66 turned on by default and this is how the GXP2135 was discovered by the UCM6200. Upon being discovered, the GXP2135 will provide details about its MAC, IP Address, Version(Firmware), Vendor, Model, Connection Status and Create Config.

By navigating to Value-added Features →Zero Config →Zero Config, a user will be able to see the discovered device(s).

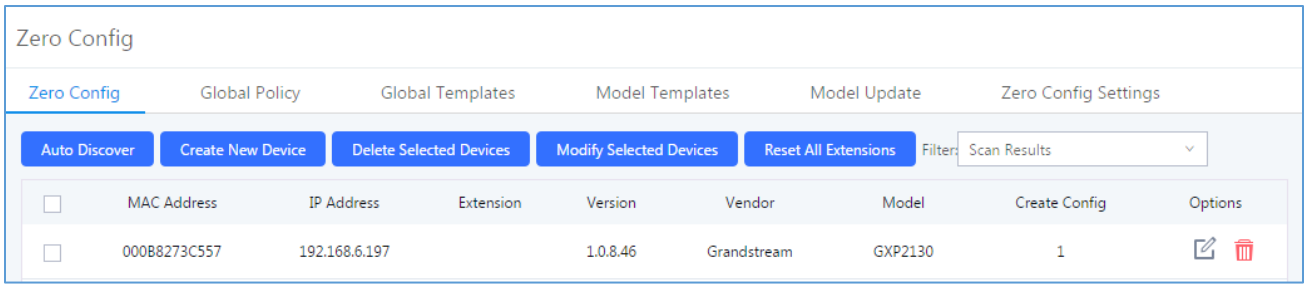


Figure 11: Provisioning with Zero Config – Auto Discovery

Extension Assignment

In the Auto Provision settings, users have the option to enable "Automatically Assign Extension". If enabled, an extension will be created and assigned to the new device detected. This is a great feature if specific extension assignment isn't required, but for our setup we'll work on manually assigning an extension to a discovered device.

Navigate to Value-added Features →Zero Config→Zero Config and click on for the discovered device. The next screen provides details of the device and also allows a user to assign an extension. Since our GXP2135 is a 4-line phone, the UCM6200 series gives us the option to assign 4 extensions to the phone.

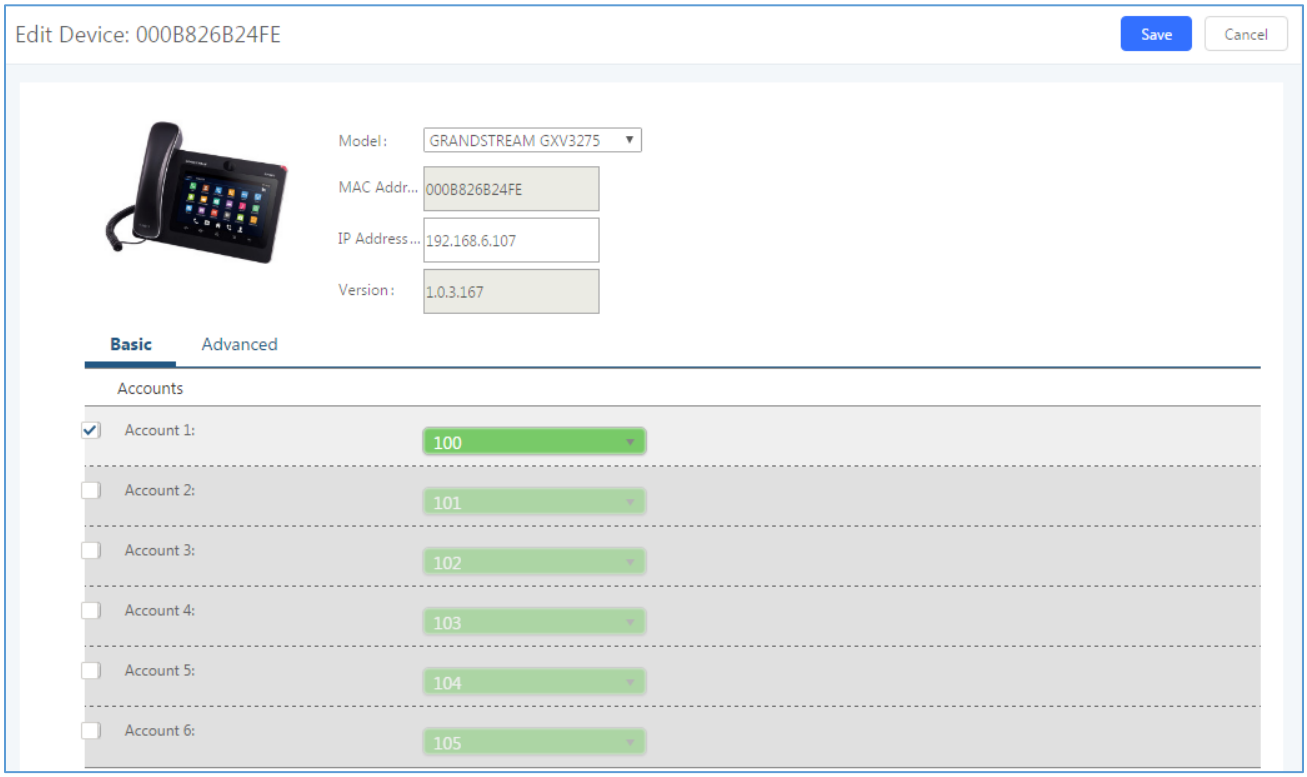


Figure 12: Provisioning with Zero Config – Extension Assignment

Check Account 1 and select the desired extension, then click "Save". Now the Zero Config page displays the device with an extension assigned to it on Account 1.

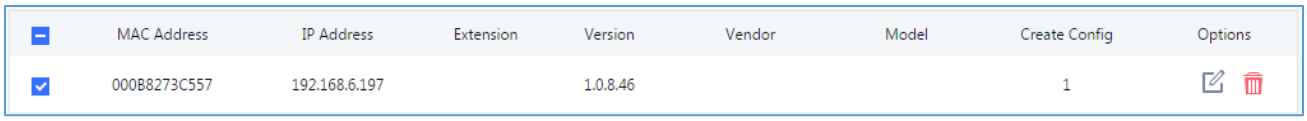


Figure 13: Provisioning with Zero Config – Manually Assigned Extension

After assigning an extension, the phone may be rebooted in order to pick up the configuration file from the UCM6200 series. During this process, the phone will boot up, request for config file, download the config file, then reboot once more in order to apply the changes.

After the phone is completely booted, navigate to PBX→Basic/Call Routes→Extensions page to see the extension status. The SIP status will show a green circle for a successful registration.

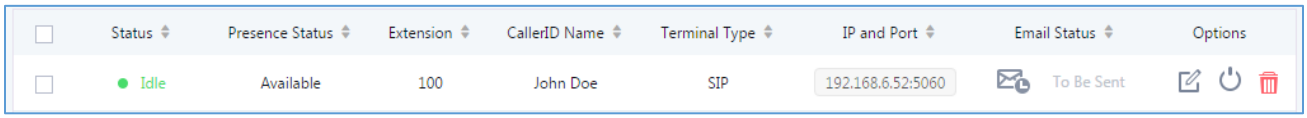


Figure 14: Provisioning with Zero Config – Extension Status

CONFERENCE BRIDGE

The UCM6200 supports conference bridge allowing multiple bridges used at the same time:

- UCM6202/6204 supports up to 3 conference bridges allowing up to 25 simultaneous PSTN or IP participants.
- UCM6208 supports up to 6 conference bridges allowing up to 32 simultaneous PSTN or IP participants.

The conference bridge configurations can be accessed under Web GUI->**Call Features→Conference→Conference**. In this page, users could create, edit, view, invite, manage the participants and delete conference bridges. The conference bridge status and conference call recordings (if recording is enabled) will be displayed in this web page as well.

How to Set up a Conference Bridge

1. Click on “Create New Conference Room”.
2. At the “Create New Conference Room” screen, the Extension is automatically populated since the extension range is in effect. Users can assign another number that is within the Conference Extension range.
3. Next, uncheck the option “Public Mode”. This will make the password options available. Enter a numeric password for an admin as well as the regular user password. In this example, the user enters 511511 as the user password and 510510 as the admin password.

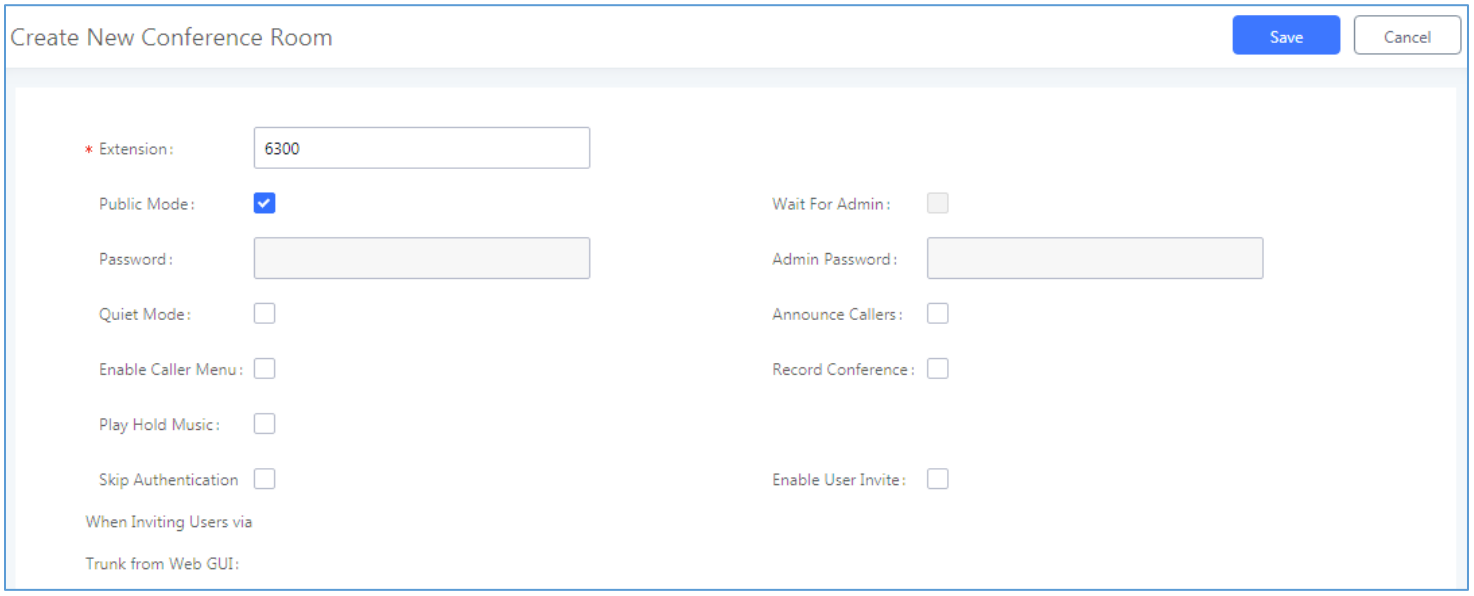


Figure 15: Conference Bridge – Create New Conference Room

4. Click “Save” at the bottom then “Apply Changes” at the top.

We should now have a conference room created and the status page will display the following:

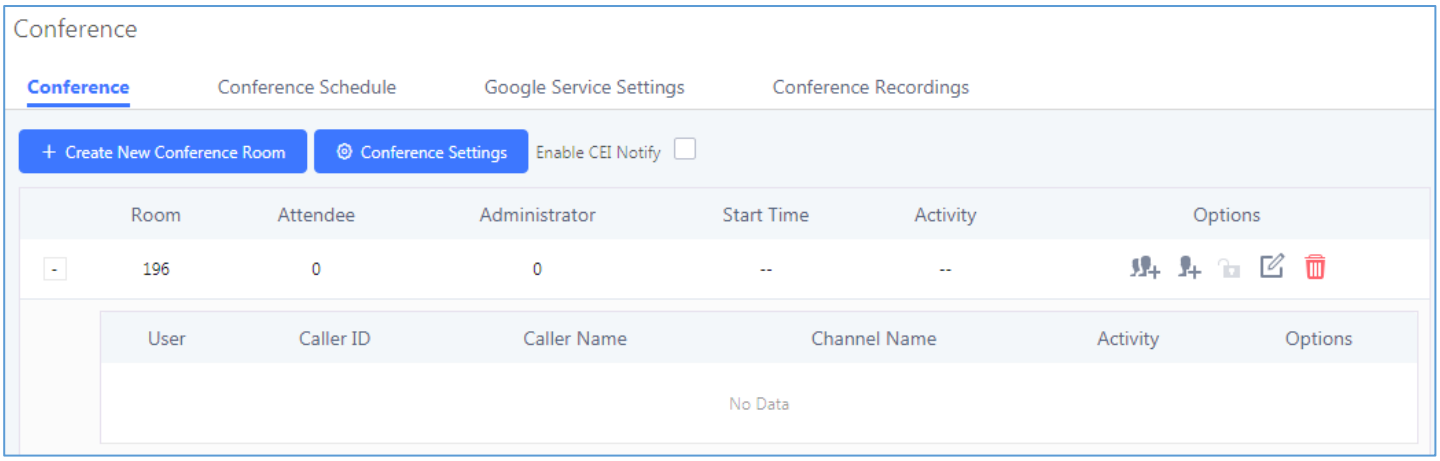


Figure 16: Conference Bridge – New Conference Room Created

With the phone that was registered to the UCM6200 series, dial the conference room extension. For this example, we will dial 196. The user will hear a prompt for the conference password. Depending on the user, the regular password or admin password can be entered.

During a conference call, the admin can log in to the UCM6200 series and view the conference room status. This provides the admin with details on which conference room is active, who the participants are and the conference call's current duration.

Besides status information, the admin has several conference management tools. By clicking the button, the admin can invite other users to the conference. Clicking will kick the selected user from the conference. When clicking button the admin will mute/unmute the specific user.

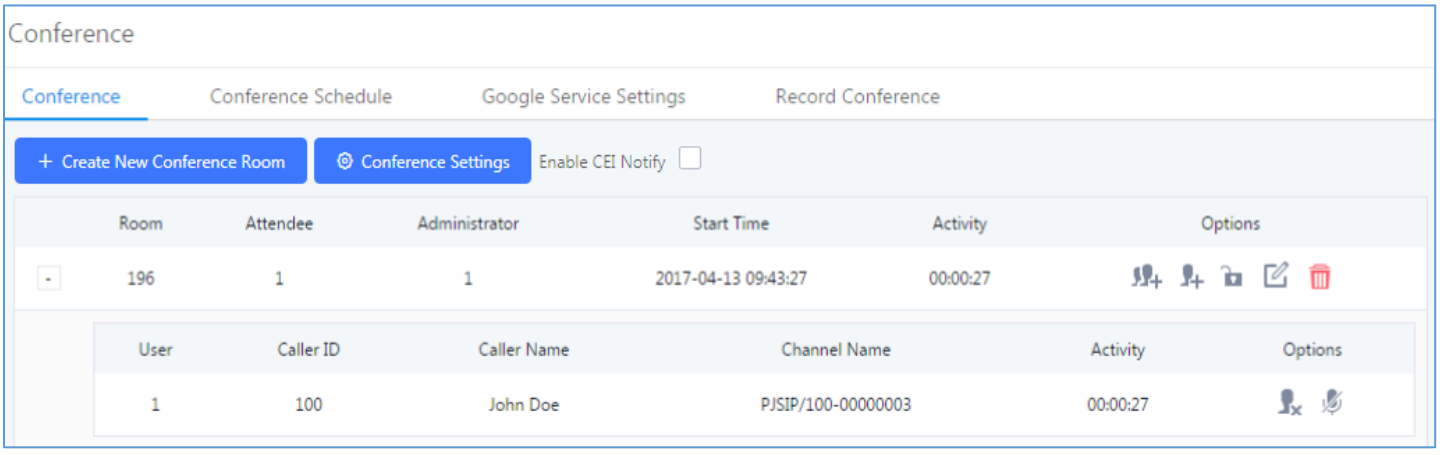


Figure 17: Conference Bridge – Conference in Progress

For more options and features for the conference room, please see the UCM6200 series User Manual.

IVR (INTERACTIVE VOICE RESPONSE)

Configure IVR

IVR configuration can be accessed under the UCM6200 Web GUI→**Call Features→IVR**. Users could create, edit, view and delete an IVR.

- Click on "Create New IVR" to add a new IVR.
- Click on to edit the IVR configuration.
- Click on to delete the IVR.

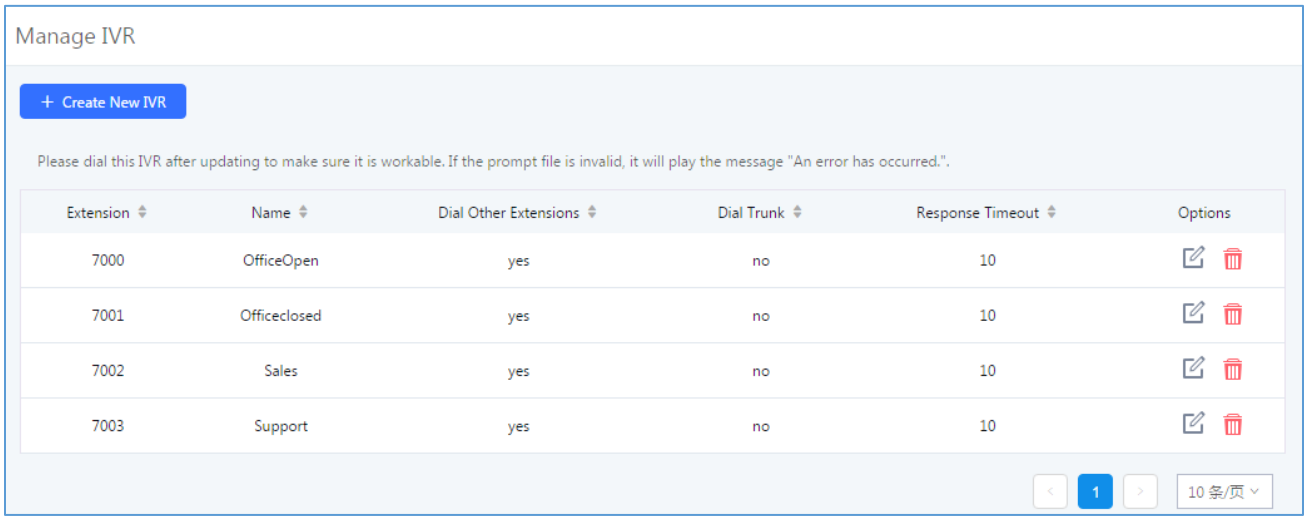


Figure 18: IVR – Manage IVR

Create New IVR

1. Click on “Create New IVR”.
2. Enter a Name for the IVR. This is used for referencing purposes. In this example, the user will create “OfficeHours2”.
3. “Extension” is populated automatically by the extension range that was created in the beginning of the tutorial. A user may choose another IVR extension within the range if desired.
4. Enable “Dial Other Extensions” to allow callers who reach the IVR to dial a direct internal extension without having to go through voice prompts.
5. “Welcome Prompt” is the recording that will be played when a caller reaches the IVR. Users can click on the “Prompt” link next to the dropdown box to configure a welcome prompt.

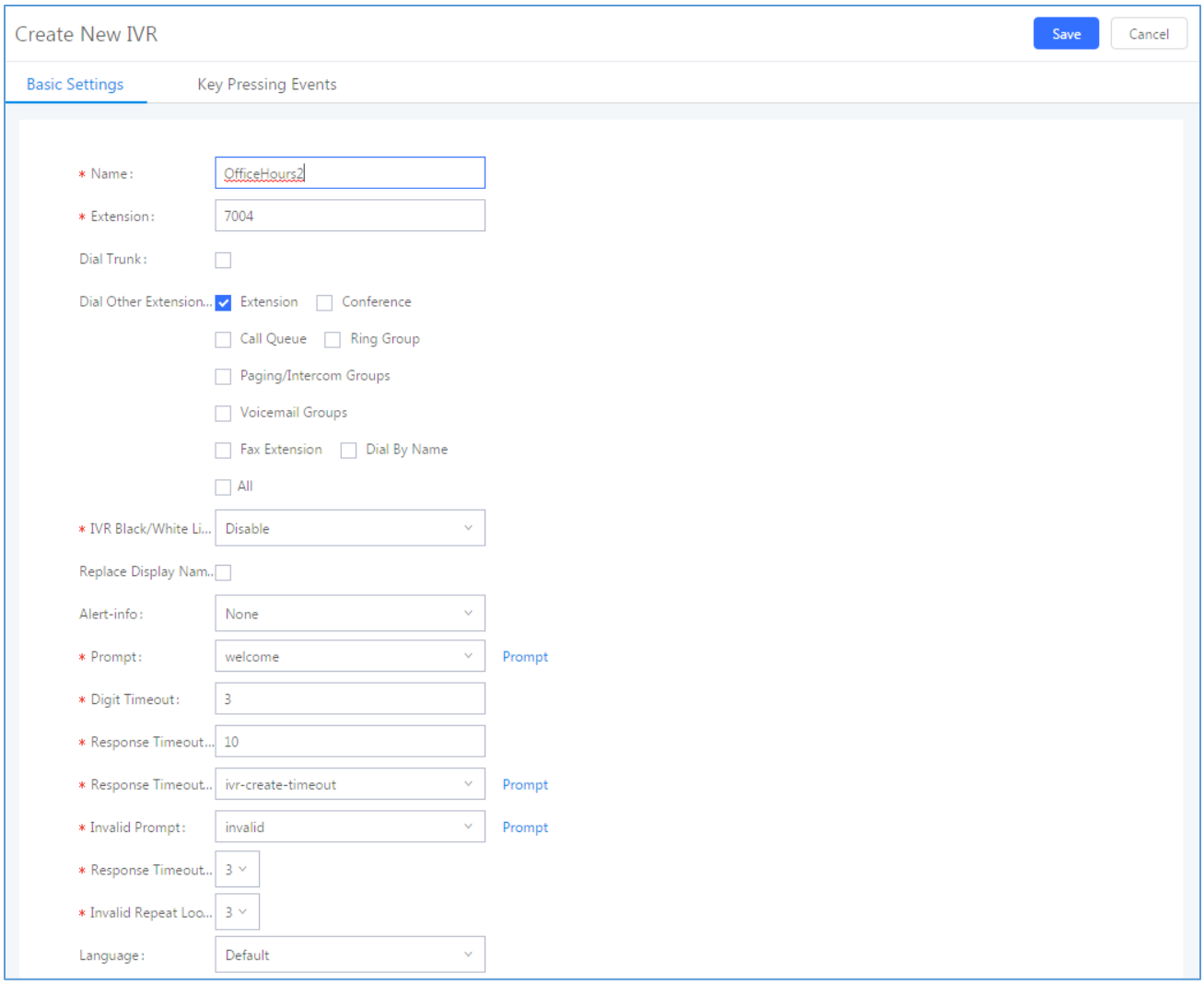


Figure 19: IVR – Create New IVR

6. The next section on configuring an IVR would be the Key Pressing Events. Users can direct calls based on a caller selection. For example, a caller reaches the IVR and the Welcome Prompt plays, “Thank you for calling Grandstream Networks. For support, please dial 0. For Sales, please dial 1.”

   Click on the drop down box for the first event, which is “Press 0” and select Extension. Another drop down box will appear and the user can then select an extension from the list. For this example, the user selects Extension 100.

7. For Key Pressing Event “Press 1” click on the drop down box and select Extension. User selects Extension 101. Besides from Extension as a Key Pressing Event, the user can select Voicemail, Conference Room, Voicemail Group, IVR, Ring Group and many more.

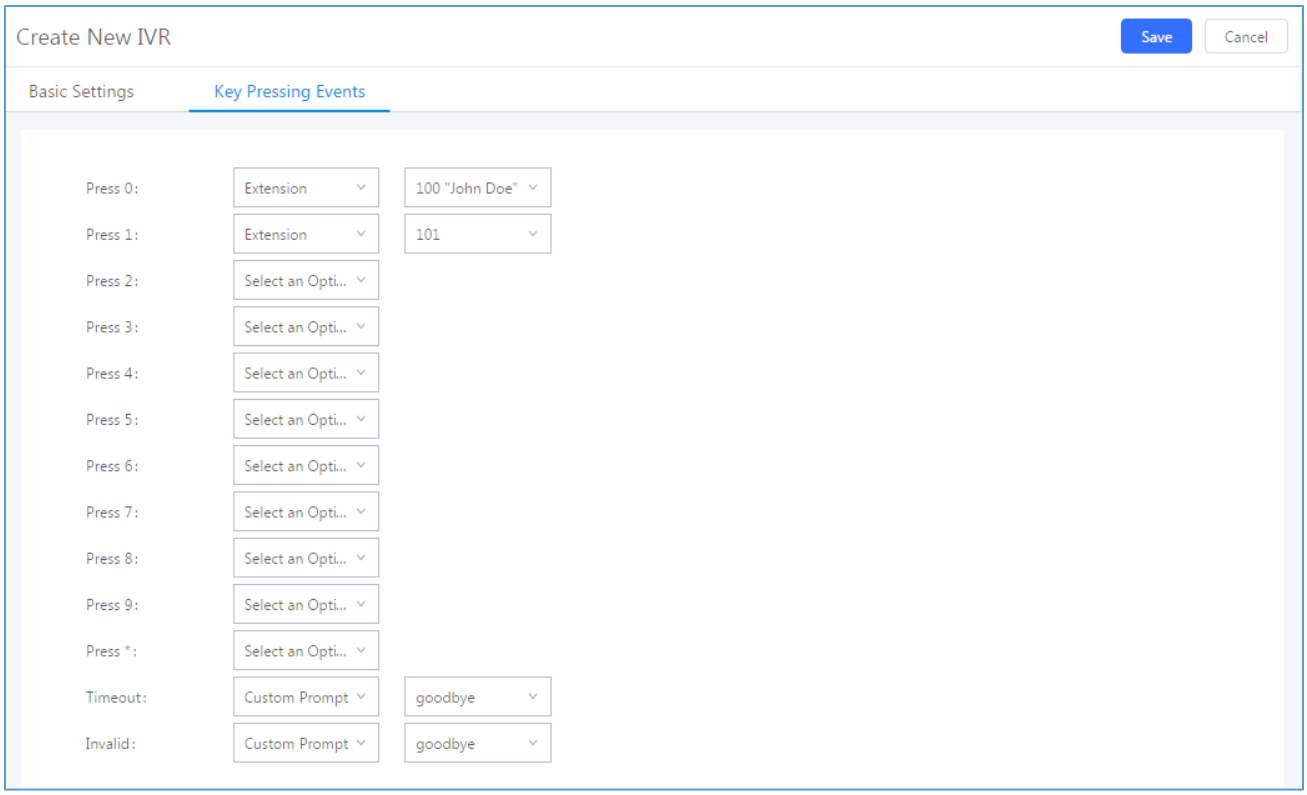


Figure 20: IVR – Key Pressing Events

8. Click “Save” at the bottom and then click on “Apply Changes” at the top.

To find out more options and parameter descriptions in regard to the IVR, please refer to the UCM6200 series User Manual.

TRUNKS

Analog Trunks

Go to Web GUI→**Extension/Trunk→Analog Trunks** to add and edit analog trunks.

- Click on "Create New Analog Trunk" to add a new analog trunk.
- Click on to edit the analog trunk.
- Click on to delete the analog trunk.

The UCM6200 series has built in FXO ports which allows it to pull in PSTN lines and provide analog trunk service. For this example, we will configure an analog line that is connected to FXO1.

Set up Analog Trunk

1. Click on “Create New Analog Trunk”.
2. For “Channels” select 1.
3. Trunk Name is set to “Analog1”.
4. “Caller ID Scheme” is set for “Bellcore/Telcordia”. Depending on the provider, users can select other schemes.
5. “Tone Country” is set to “United States of America (USA)” since this tutorial is set up in the North American region. Users can click the dropdown box to select an option more suited for their region.
6. Click “Save” and then click on “Apply Changes” at the top.

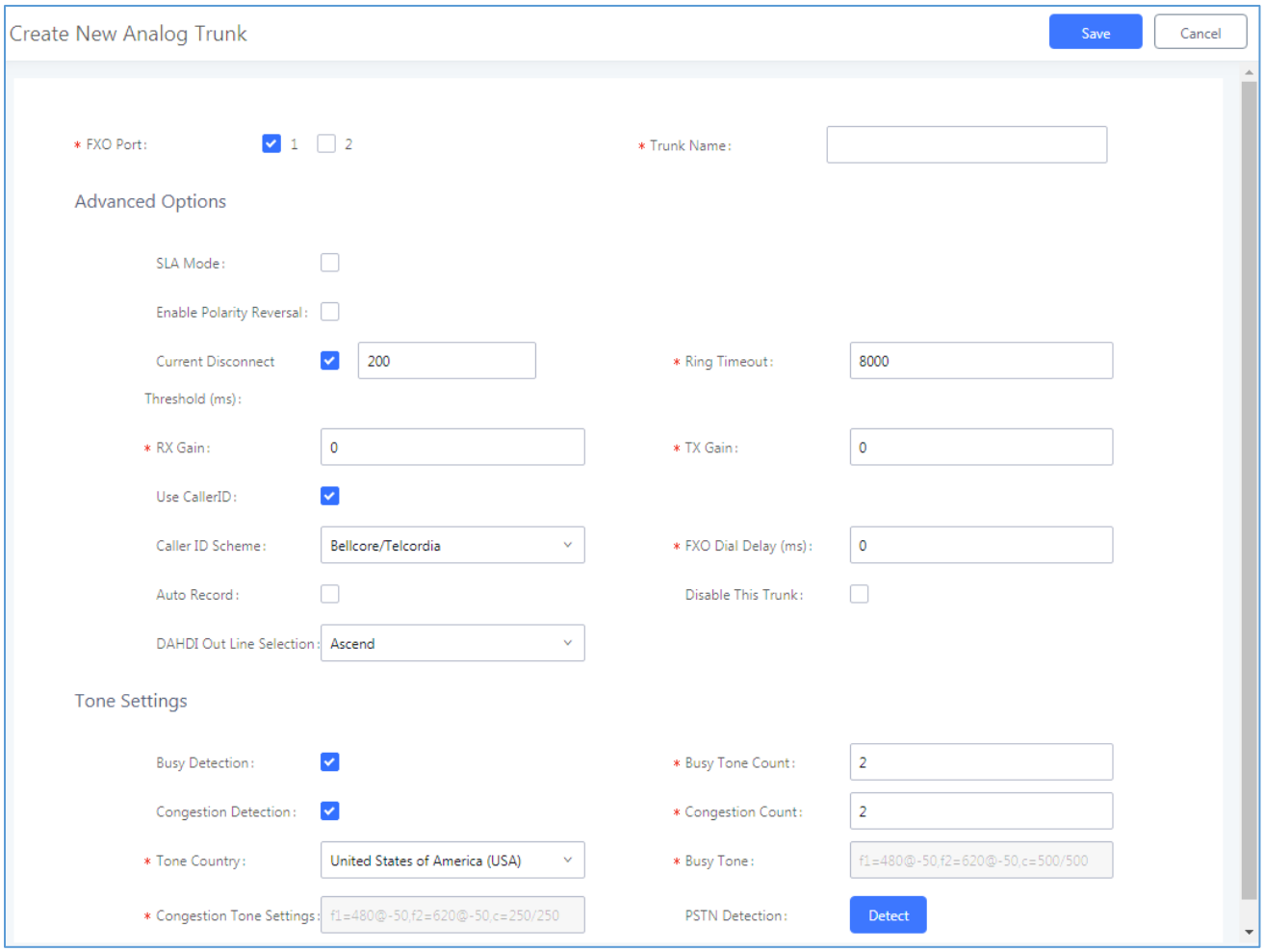


Figure 21: Analog Trunks – Create New Analog Trunk

Note: If there are reports of calls being dropped and lines are not disconnecting properly, this could mean that there are line settings mismatched. The UCM6200 series offers an auto detect feature that tests the line and provides the best possible settings. This option is called “PSTN Detection” which can be found under the Analog trunk configuration page.

VoIP Trunks

VoIP trunks can be configured under Web GUI→**PBX→Basic/Call Routes→VoIP Trunks**. Once created, the VoIP trunks will be listed with Provider Name, Type, Hostname/IP, Username and Options to edit/detect the trunk.

- Click on "Create New SIP Trunk" or “Create New IAX Trunk” to add a new VoIP trunk.
- Click on to configure detailed parameters for the VoIP trunk.
- Click on to configure Direct Outward Dialing (DOD) for the VoIP Trunk.
- Click on to start LDAP Sync.
- Click on to delete the VoIP trunk.

There are 4 different type of VoIP trunks that can be configured. The types are Peered/Registered SIP trunk or Peered/Registered IAX trunk. This tutorial will demonstrate how to configure a SIP Registered Trunk.

Set up VoIP Trunk

1. Click on “Create New SIP Trunk”.
2. Click the dropdown box for Type and select “Register SIP Trunk”.
3. Enter a name for “Provider Name” e.g., GSTrunk.
4. Configure the IP address or URL of the VoIP providers server as the “Host Name” e.g., sip.grandstream.com.
5. Input the username to authenticate with the VoIP provider as the “Username” e.g, 6266389172.
6. Enter password for username to authenticate with VoIP provider server.
7. Enter the AuthID. This is the SIP service subscriber’s ID used for authentication. If not configured, the CallerID will be used for authentication.
8. Enter an outbound proxy if the provider requires one.
9. Click “Save” and then “Apply Changes” at the top

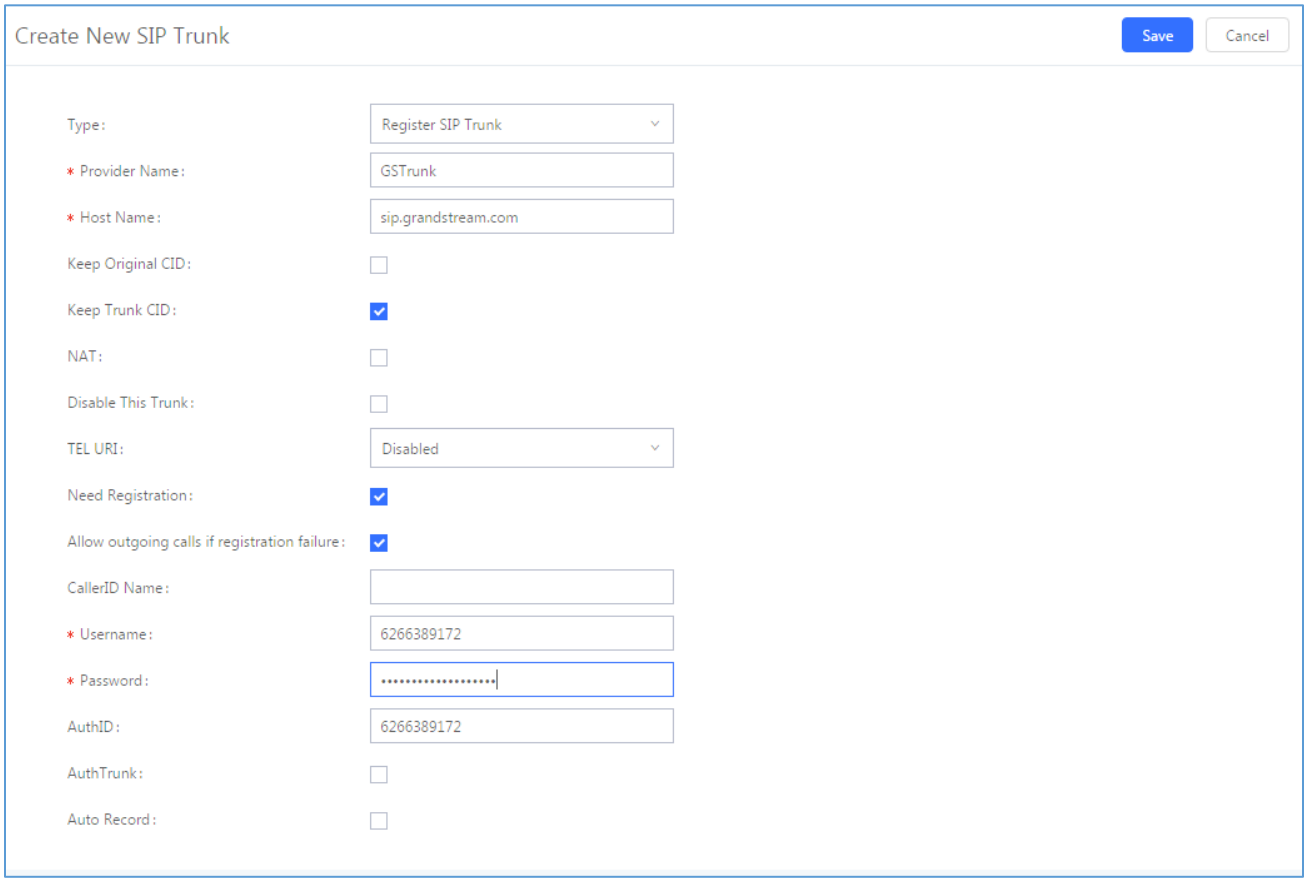


Figure 22: VoIP Trunks – Create New Register SIP Trunk

After creating the SIP trunk, users can click the on the right of the VoIP trunk for more configuration options.

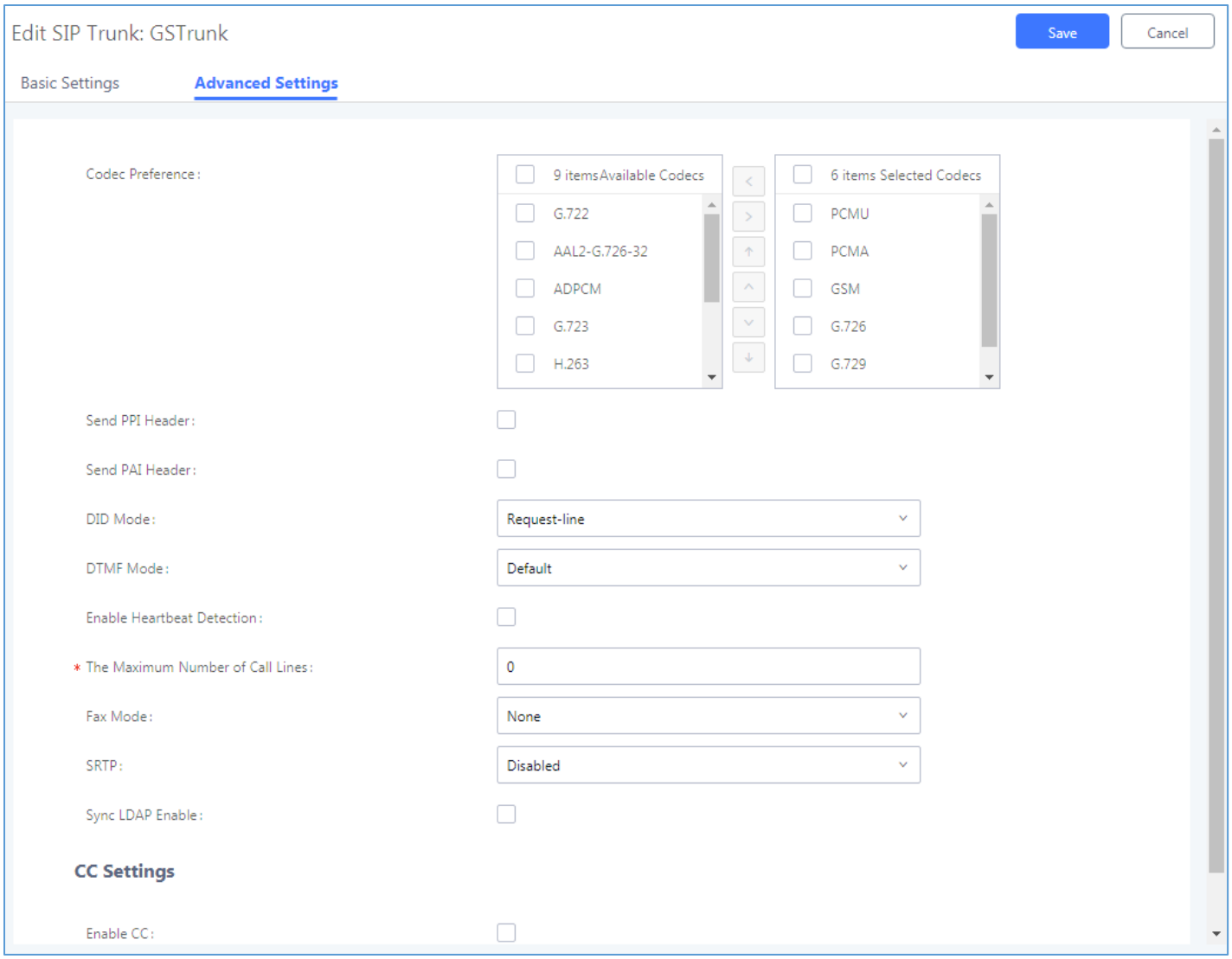


Figure 23: VoIP Trunks – Advanced Settings

To verify if the SIP trunk is registered, the user may navigate to Web GUI→Status.

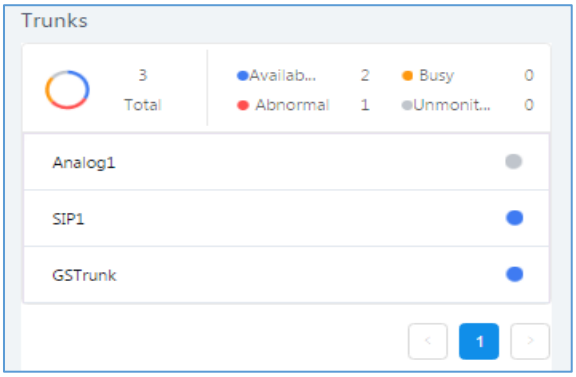


Figure 24: Status – Trunks

CALL ROUTES

Outbound Routes

In the UCM6200, an outgoing calling rule pairs an extension pattern with a trunk used to dial the pattern. This allows different patterns to be dialed through different trunks (e.g., "Local" 7-digit dials through a FXO while "Long distance" 10-digit dials through a low-cost SIP trunk). Users can also set up a failover trunk to be used when the primary trunk fails.

Go to Web GUI→**Extension/Trunk→Outbound Routes** to add and edit outbound rules.

- Click on "Add" to add a new outbound route.
- Click on to edit the outbound route.
- Click on to delete the outbound route.
- Click on to move the outbound route up/down to arrange the priority of the outbound rule. The outbound rule listed on the top has higher priority. When the dialing pattern matches two or more outbound rules (for example, the same pattern is configured for 2 different trunks; or dialing out 1000 matches pattern 1xxx for trunk 1 and pattern 100x for trunk 2), the one listed on the top will be used.

How to Configure an Outbound Route

1. Click “Create New Outbound Rule”.
2. Configure the name of the Calling Rule. For example, ‘Local’, ‘LongDistance’ and etc.
3. Create a dial Pattern. For 7 digit dialing (Local) the pattern will look like this: XXXXXXX
   This means that any 7 digits will use this route. For 10 digits dialing the pattern would look like this: NXXNXXXXXX. If an ‘1’ is required before dialing out, the dial pattern can be configured in this manner: 1NXXNXXXXXX.

   In this example, the dial pattern is going to be created for local 7 digits dialing so the pattern is XXXXXXX

4. Next, set a “Privilege Level”. There are 4 Privilege Levels; Internal, Local, National, International. Internal is the lowest level of security whereas international is the highest level of security.

   The way that Privilege Levels are used could be thought of as door locks. Extensions are granted with “Permissions” that can be seen as keys. If an Extension is configured with an Internal permission, it CANNOT access an outbound route with a Privilege level of Local, National and International. A user configured with National permission can access routes that have a Privilege Level of Internal, Local, National, but not International.
5. The pattern that was configured is a 7-digit dialing pattern. For “Use Trunk” select the Analog Trunk or select the desired trunk for this particular dial pattern. For long distance calls, it may be cost effective to route the calls to SIP trunk versus going through a local PSTN line.
6. Click “Save” and then click on “Apply Changes” at the top.

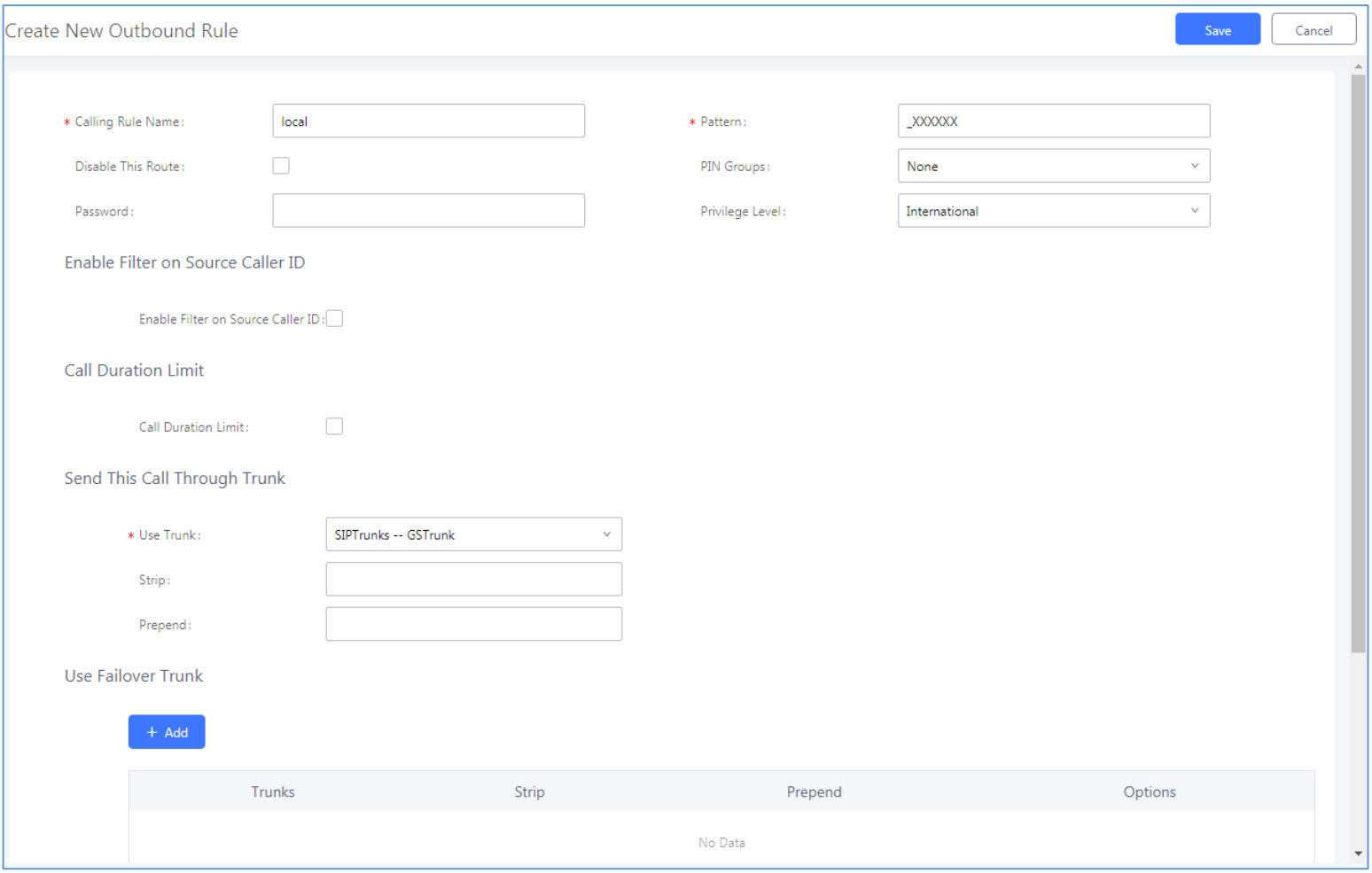


Figure 25: Routes – Create Outbound Route

Inbound Routes

In the UCM6200, an incoming calling rule allows for various inbound destinations. Users can route inbound calls by DID, to Extension, Voicemail, Conference Room, Queue, Ring Group, Page, Voicemail Group, FAX, DISA or IVR. These inbound routes can also be triggered based on a time condition as well.

Inbound routes can be configured via Web GUI→**Extension/Trunk→Inbound Routes**.

- Click on "Create New Inbound Rule" to add a new inbound route.
- Click on "DID Features" to configure DID features for the inbound route.
- Click on "Blacklist" to configure blacklist for all inbound routes.
- Click on to edit the inbound route.
- Click on to delete the inbound route.

This section of the tutorial will provide instructions on how to setup an inbound route for a Registered SIP Trunk. Inbound calls into this SIP trunk will have a default destination as well as a time conditioned destination.

Set up Inbound Route

1. Click on “Add”.
2. For “Trunks” select the SIP trunk that has been configured. (e.g. GS Trunk).
3. The DID pattern can be composed of two parts, divided by a ‘/’ character. The first part is used to specify the dialed number and the second part is used to specify the caller ID, which is optional. If caller ID section is entered it means only the extension with the specific caller ID is allowed to call into this trunk.
4. Click the dropdown box for “Default Destination” and select IVR. Then select the IVR extension.
5. Select a time condition from the “Time Condition” dropdown
6. For “Destination” select Extension and choose an extension number.

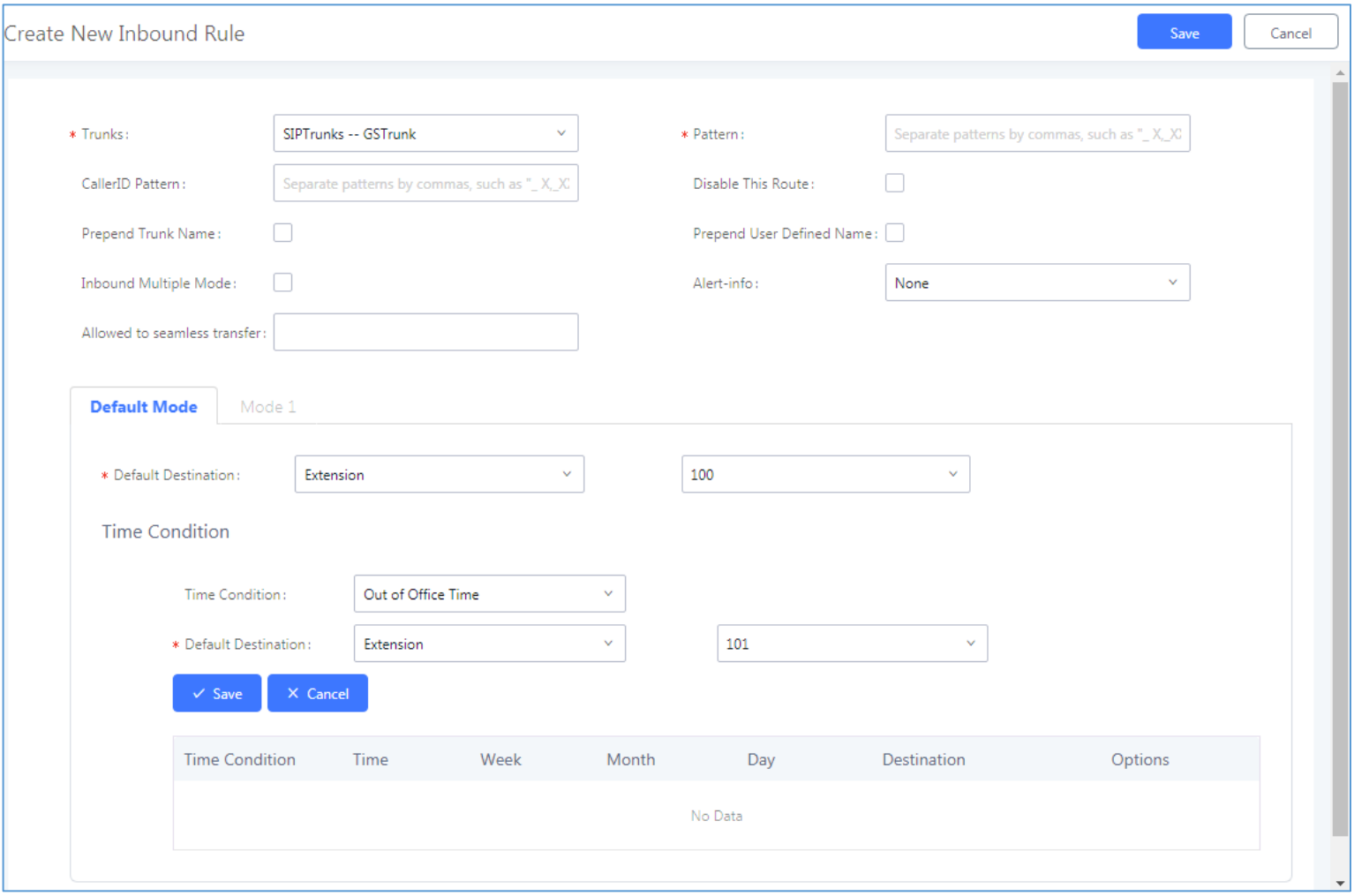


Figure 26: Routes – Create Inbound Route

When a call is received on this SIP trunk during the hours of 8am to 6pm it will be routed to extension 100. All calls outside of this condition will be routed to the other extension.