Grandstream Networks, Inc.

UCM6200 Series IP PBX

User Manual

UCM6200 Series IP PBX User Manual

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GNU GPL INFORMATION

UCM6200 firmware contains third-party software licensed under the GNU General Public License (GPL). Grandstream uses software under the specific terms of the GPL. Please see the GNU General Public License (GPL) for the exact terms and conditions of the license.

Grandstream GNU GPL related source code can be downloaded from Grandstream web site from:
http://www.grandstream.com/support/faq/gnu-general-public-license/gnu-gpl-information-download

CHANGE LOG

This section documents significant changes from previous versions of the UCM6200 user manuals. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

FIRMWARE VERSION 1.0.0.1

- This is the initial version.

WELCOME

Thank you for purchasing Grandstream UCM6200 series IP PBX appliance. The UCM6200 series IP PBX appliance is designed to bring enterprise-grade voice, video, data, and mobility features to small-to-medium businesses (SMBs) in an easy-to-manage fashion. This IP PBX series allows businesses to unify multiple communication technologies, such comprehensive voice, video calling, video conferencing, video surveillance, data tools and facility access management onto one common network that that can be managed and/or accessed remotely. The UCM6200 series supports a dual core 1GHz ARM Cortex^{TM} A9 and 400Mhz VINETIC^{TM} A8 processors, 1GB RAM and 4GB flash. The secure and reliable UCM6200 series delivers enterprise-grade features without any licensing fees, costs-per-feature or recurring fees.

⚠ **Caution:**

Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

⚠ **Warning:**

Please do not use a different power adaptor with the UCM6200 as it may cause damage to the products and void the manufacturer warranty.

This document is subject to change without notice. The latest electronic version of this user manual is available for download here:
http://www.grandstream.com/support

PRODUCT OVERVIEW

TECHNICAL SPECIFICATIONS

Table 1: Technical Specifications

Interfaces	
Analog Telephone FXS Ports	2 ports (both with lifeline capability in case of power outage)
PSTN Line FXO Ports	• UCM6202: 2 ports • UCM6204: 4 ports • UCM6208: 8 ports
Network Interfaces	• UCM6202/6204: Dual Gigabit RJ45 ports with integrated PoE Plus (IEEE 802.3at-2009) • UCM6208: Single Gigabit RJ45 port with integrated PoE Plus (IEEE 802.3at-2009)
NAT Router	Yes, UCM6204/UCM6208 only
Peripheral Ports	USB, SD
LED Indicators	Power/Ready, Network, PSTN Line, USB, SD
LCD Display	128x32 graphic LCD with DOWN and OK button
Reset Switch	Yes
Voice/Video Capabilities	
Voice-over-Packet Capabilities	LEC with NLP Packetized Voice Protocol Unit, 128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection and auto-switch to G.711
Voice and Fax Codecs	G.711 A-law/U-law, G.722, G.723.1 5.3K/6.3K, G.726, G.729A/B, iLBC (30ms only), GSM, AAL2-G.726-32, ADPCM; T.38
Video Codecs	H.264, H.263, H.263+, VP8
QoS	Layer 3 QoS, Layer 2 QoS
Signaling and Control	
DTMF Methods	In Audio, RFC2833, and SIP INFO
Provisioning Protocol and Plug-and-Play	TFTP/HTTP/HTTPS, auto-discovery and auto-provisioning of Grandstream IP endpoints via ZeroConfig (DHCP Option 66/multicast SIP SUBSCRIBE/mDNS), eventlist between local and remote trunk
Network Protocols	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, SIP (RFC3261), STUN, SRTP, TLS
Disconnect Methods	Call Progress Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect, Busy Tone

Security	
Media	SRTP, TLS, HTTPS, SSH
Physical	
Universal Power Supply	• Output: 12VDC, 1.5A • Input: 100-240VAC, 50-60Hz
Dimensions	• UCM6202/6204: 226mm (L) x 155mm (W) x 34.5mm (H) • UCM6208: 440mm (L) x 185mm (W) x 44mm (H)
Environmental	• Operating: 32 - 104^{o}F / 0 - 40^{o}C, 10-90% (non-condensing) • Storage: 14 - 140^{o}F / -10 - 60^{o}C
Mounting	• UCM6202/6204: Wall mount and Desktop • UCM6208: Rack mount and Desktop
Weight	• UCM6202: Unit weight 0.51kg, Package weight 0.94kg • UCM6204: Unit weight 0.51kg, Package weight 0.94kg • UCM6208: Unit weight 2.23kg, Package weight 3.09kg
Additional Features	
Multi-language Support	English/Simplified Chinese/Traditional Chinese/Spanish/French/ Portuguese/German/Russian/Italian/Polish/Czech for Web UI; Customizable IVR/voice prompts for English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic; Customizable language pack to support any other languages
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 - BT
Polarity Reversal/ Wink	Yes, with enable/disable option upon call establishment and termination
Call Center	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability busy level, in-queue announcement
Customizable Auto Attendant	Up to 5 layers of IVR (Interactive Voice Response)
Maximum Call Capacity	• UCM6202: Concurrent audio calls up to 50, concurrent WebRTC calls up to 25 • UCM6204: Concurrent audio calls up to 75, concurrent WebRTC calls up to 35 • UCM6208: Concurrent audio calls up to 100, concurrent WebRTC calls up to 50 Or up to 66% performance if calls are SRTP encrypted
Conference Bridges	• UCM6202/6204: Up to 3 password-protected conference bridges allowing up to 25 simultaneous PSTN or IP participants • UCM6208: Up to 6 password-protected conference bridges allowing up to 32 simultaneous PSTN or IP participants

Call Features	Call park, call forward, call transfer, DND, ring/hunt group, paging/intercom and etc
Compliance	• FCC: Part 15 (CFR 47) Class B, Part 68 • CE: EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1, TBR21, RoHS • A-TICK: AS/NZS CISPR 22 Class B, AS/NZS CISPR 24, AS/NZS 60950, AS/ACIF S002 and ITU-T K.21 (Basic Level) • UL 60950 (power adapter)

⚠ **Note:**

- UCM6200 FXS ports lifeline functionality:
  The UCM6200 FXS interfaces are metallic through to the FXO interfaces. If there is power outage, FXS1 port will fail over to FXO 1 port, FXS 2 port will fail over to FXO 2 port. The user can still access the PSTN connected with the FXO interfaces from FXS interfaces.

INSTALLATION

Before deploying and configuring the UCM6200 series, the device needs to be properly powered up and connected to network. This section describes detailed information on installation, connection and warranty policy of the UCM6200 series.

EQUIPMENT PACKAGING

Table 2: UCM6202/UCM6204 Equipment Packaging

Main Case	Yes (1)
Power Adaptor	Yes (1)
Ethernet Cable	Yes (1)
Quick Installation Guide	Yes (1)
GPL License	Yes (1)

CONNECT YOUR UCM6200

CONNECT THE UCM6202

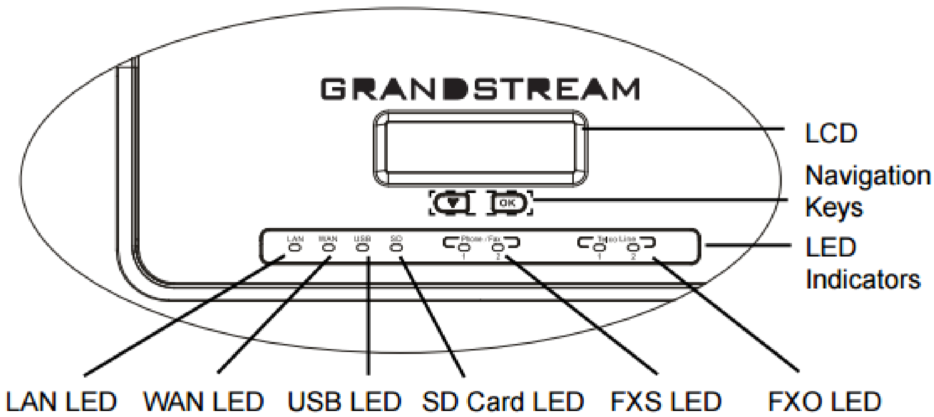


Figure 1: UCM6202 Front View

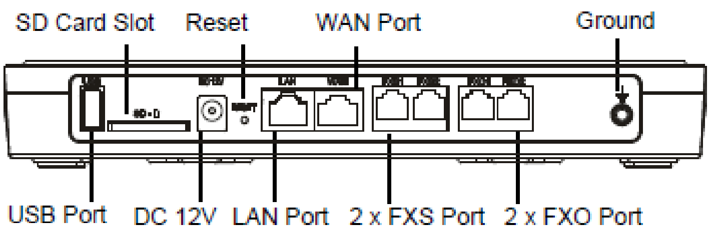


Figure 2: UCM6202 Back View

To set up the UCM6202, follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the WAN port of the UCM6202.
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub.
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the UCM6202. Insert the main plug of the power adapter into a surge-protected power outlet.
4. Wait for the UCM6202 to boot up. The LCD in the front will show the device hardware information when the boot process is done.
5. Once the UCM6202 is successfully connected to network, the LED indicator for WAN in the front will be in solid green and the LCD shows up the IP address.
6. (Optional) Connect PSTN lines from the wall jack to the FXO ports; connect analog lines (phone and Fax) to the FXS ports.

CONNECT THE UCM6204

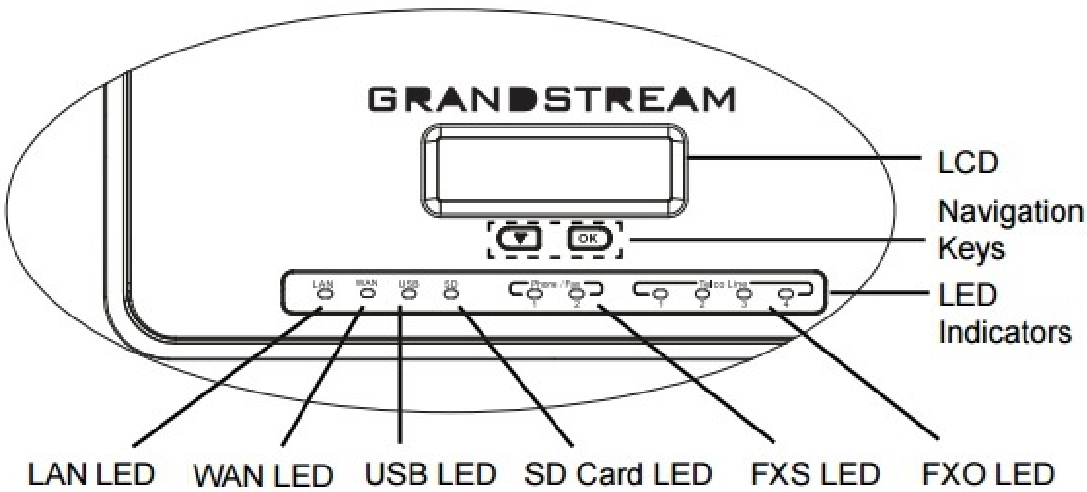


Figure 3: UCM6204 Front View

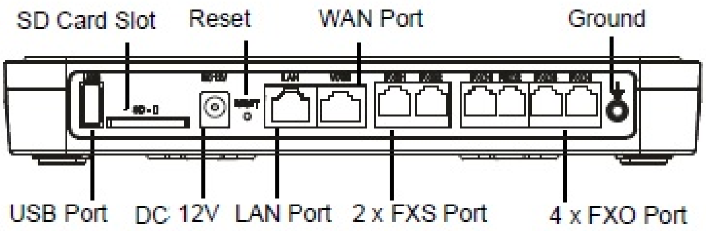


Figure 4: UCM6204 Back View

To set up the UCM6204, follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the WAN port of the UCM6204.
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub.
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the UCM6204. Insert the main plug of the power adapter into a surge-protected power outlet.
4. Wait for the UCM6204 to boot up. The LCD in the front will show the device hardware information when the boot process is done.
5. Once the UCM6204 is successfully connected to network, the LED indicator for WAN in the front will be in solid green and the LCD shows up the IP address.
6. (Optional) Connect PSTN lines from the wall jack to the FXO ports; connect analog lines (phone and Fax) to the FXS ports.

CONNECT THE UCM6208

To set up the UCM6208, follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the LAN port of the UCM6208.
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub.
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the UCM6208. Insert the main plug of the power adapter into a surge-protected power outlet.
4. Wait for the UCM6208 to boot up. The LCD in the front will show the device hardware information when the boot process is done.
5. Once the UCM6208 is successfully connected to network, the LED indicator for NETWORK in the front will be in solid green and the LCD shows up the IP address.

6. (Optional) Connect PSTN lines from the wall jack to the FXO ports; connect analog lines (phone and Fax) to the FXS ports.

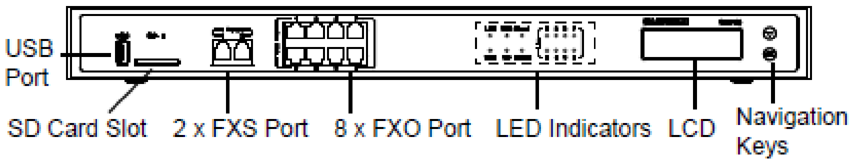


Figure 5: UCM6208 Front View

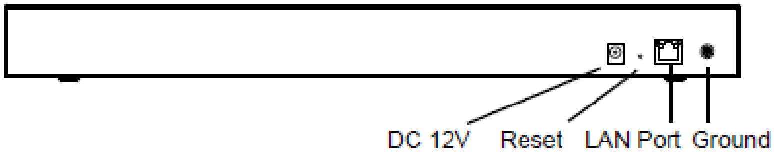


Figure 6: UCM6208 Back View

SAFETY COMPLIANCES

The UCM6200 series IP PBX complies with FCC/CE and various safety standards. The UCM6200 power adapter is compliant with the UL standard. Use the universal power adapter provided with the UCM6200 package only. The manufacturer's warranty does not cover damages to the device caused by unsupported power adapters.

WARRANTY

If the UCM6200 series IP PBX was purchased from a reseller, please contact the company where the device was purchased for replacement, repair or refund. If the device was purchased directly from Grandstream, contact our Technical Support Team for a RMA (Return Materials Authorization) number before the product is returned. Grandstream reserves the right to remedy warranty policy without prior notification.

⚠ **Warning:**

Use the power adapter provided with the UCM6200 series IP PBX. Do not use a different power adapter as this may damage the device. This type of damage is not covered under warranty.

GETTING STARTED

The UCM6200 series provides LCD interface, LED indication and web GUI configuration interface.

- The LCD displays hardware, software and network information. Users could also navigate in the LCD menu for device information and basic network configuration.
- The LED indication at the front of the device provides interface connection and activity status.
- The web GUI gives users access to all the configurations and options for UCM6200 series setup.

This section provides step-by-step instructions on how to use the LCD menu, LED indicators and Web GUI of the UCM6200 series. Once the basic settings are done, users could start making calls from UCM6200 extension registered on a SIP phone as described at the end of this section.

USE THE LCD MENU

- **Default LCD Display**
  When the device is powered up, the LCD will show device model (e.g., UCM6204), hardware version (e.g., V1.0A) and IP address. Press "Down" button and the system time will be displayed as well.

- **Menu Access**
  Press "OK" button to start browsing menu options. Please see menu options in [Table 3: LCD Menu Options].

- **Menu Navigation**
  Press the "Down" arrow key to browser different menu options. Press the "OK" button to select an entry.

- **Exit**
  If "Back" option is available in the menu, select it to go back to the previous menu. For "Device Info" "Network Info" and "Web Info" which do not have "Back" option, simply press the "OK" button to go back to the previous menu. Also, the LCD will display default idle screen after staying in menu option for 15 seconds.

- **LCD Backlight**
  The LCD backlight will be on upon key pressing. The backlight will go off after the LCD stays in idle for 30 seconds.

Table 3: LCD Menu Options

View Events	• **Critical Events** • **Other Events**
Device Info	• **Hardware**: Hardware version number • **Software**: Software version number • **P/N**: Part number • **WAN MAC**: WAN side MAC address (UCM6202/UCM6204 only) • **LAN MAC**: LAN side MAC address • **Uptime**: System up time
Network Info	For UCM6208: • **LAN Mode**: DHCP, Static IP, or PPPoE • **LAN IP**: IP address • **LAN Subnet Mask** For UCM6202/UCM6204: • **WAN Mode**: DHCP, Static IP, or PPPoE • **WAN IP**: IP address • **WAN Subnet Mask** • **LAN IP**: IP address • **LAN Subnet Mask**
Network Menu	For UCM6208: • **LAN Mode**: Select LAN mode as DHCP, Static IP or PPPoE • **Static Route Reset**: Click to reset the static route setting For UCM6202/UCM6204: • **WAN Mode**: Select WAN mode as DHCP, Static IP or PPPoE **Static Route Reset**: Click to reset the static route setting
Factory Menu	• **Reboot** • **Factory Reset** • **LCD Test Patterns** Press "OK" to start. Then press "Down" button to test different LCD patterns. When done, press "OK" button to exit. • **Fan Mode** Select "Auto" or "On". • **LED Test Patterns**

	Select "All On" "All Off" or "Blinking" and check LED status. • **RTC Test Patterns** Select "2022-02-22 22:22" or "2011-01-11 11:11" to start the RTC (Real-Time Clock) test pattern. Then check the system time from LCD idle screen by pressing "DOWN" button, or from web GUI->**System Status**->**General** page. Reboot the device manually after the RTC test is done. • **Hardware Testing** Select "Test SVIP" to perform SVIP test on the device. This is mainly for factory testing purpose which verifies the hardware connection inside the device. The diagnostic result will display in the LCD after the test is done.
Web Info	• **Protocol**: Web access protocol. HTTP or HTTPS. By default it's HTTPS • **Port**: Web access port number. By default it's 8089
SSH Switch	• **Enable SSH:** Enable SSH access. • **Disable SSH:** Disable SSH access. By default the SSH access is disabled.

USE THE LED INDICATORS

The UCM6200 has LED indicators in the front to display connection status. The following table shows the status definitions.

Table 4: UCM6202/UCM6204 LED INDICATORS

LED Indicator	LED Status
LAN WAN USB SD FXS (Phone/Fax) FXO (Telco Line)	**Solid**: Connected **Flashing**: Data Transferring **OFF**: Not Connected

Table 5: UCM6208 LED INDICATORS

LED	LED Status
NETWORK	**Solid**: Connected **OFF**: Not Connected
ACT USB SD Phone (FXS) Line (FXO)	**Solid:** Connected **Flashing**: Data Transferring **OFF**: Not Connected

USE THE WEB GUI

ACCESS WEB GUI

The UCM6200 embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow users to configure the device through a Web browser such as Microsoft IE, Mozilla Firefox, Google Chrome and etc.

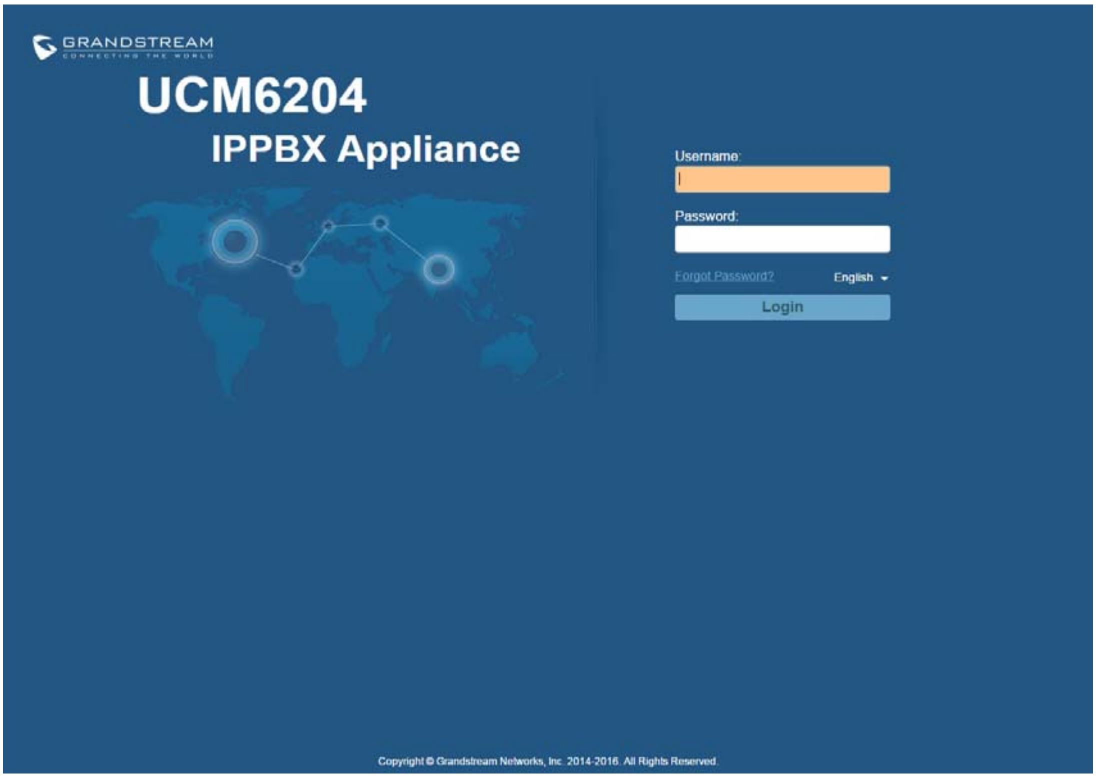


Figure 7: UCM6204 Web GUI Login Page

To access the Web GUI:

1. Connect the computer to the same network as the UCM6200.
2. Ensure the device is properly powered up and shows its IP address on the LCD.
3. Open a Web browser on the computer and enter the web GUI URL in the following format:

   http(s)://IP-Address:Port

   where the ***IP-Address*** is the IP address displayed on the UCM6200 LCD.

   By default, the protocol is HTTPS and the Port number is 8089.

   For example, if the LCD shows 192.168.40.167, please enter the following in your web browser:

   https://192.168.40.167:8089
4. Enter the administrator’s login and password to access the Web Configuration Menu. The default administrator's username and password is "admin" and "admin". It is highly recommended to change the default password after login for the first time.

⚠ **Note:**

By default, the UCM6200 has "Redirect From Port 80" enabled. Therefore, if users type in the UCM6200 IP address in the web browser, the web page will be automatically redirected to the page using HTTPS and port 8089. For example, if the LCD shows 192.168.40.167, please enter 192.168.40.167 in your web browser and the web page will be redirected to:

https://192.168.40.167:8089

The option "Redirect From Port 80" can be configured under the UCM6200 web GUI->**Settings**->**HTTP Server**.

SETUP WIZARD

When the user logs in the UCM6200 web UI for the first time, a setup wizard will guide the user to set up basic configuration. Configurations in setup wizard includes: **Time zone, Change password, Network settings, Extensions, Trunk and routes.**

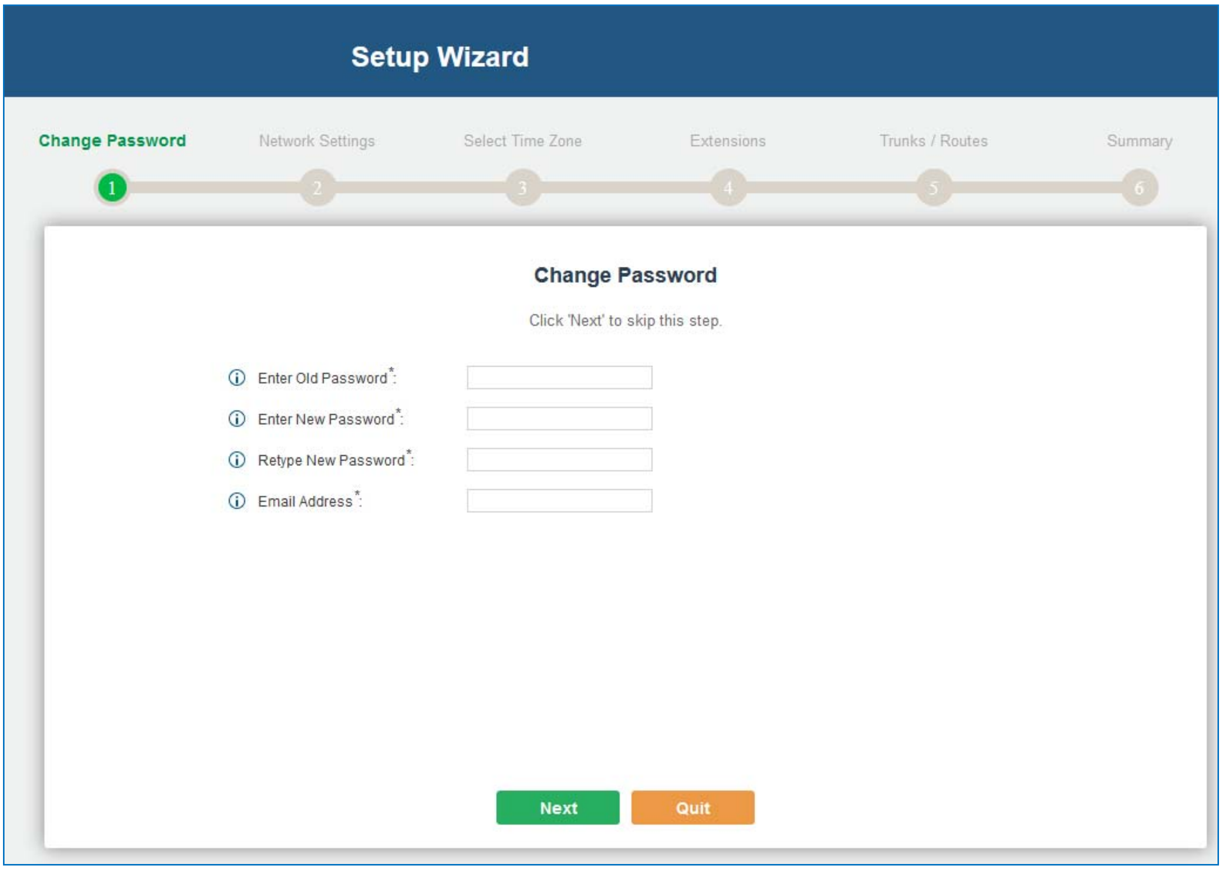


Figure 8: UCM6200 Setup Wizard

During the wizard, the user can quit the setup wizard at any time to start over with manual configuration. At the last step of the wizard, the user will be provided with summary for review, before the configuration is loaded. Once the setup is completed, the system is ready to go.

WEB GUI CONFIGURATIONS

There are four main sections in the Web GUI for users to view the PBX status, configure and manage the PBX.

- **Status:** Displays PBX status, System Status, System Events and CDR.
- **PBX:** To configure extensions, trunks, call routes, zero config for auto provisioning, call features, internal options, IAX settings and SIP settings.
- **Settings:** To configure user management, network settings, firewall settings, change password, LDAP Server, HTTP Server, Email Settings, Time Settings, NTP server, recording storage and login timeout.
- **Maintenance**: To perform firmware upgrade, backup configurations, cleaner setup, reset/reboot, syslog setup and troubleshooting.

WEB GUI LANGUAGES

Currently the UCM6200 series web GUI supports ***English, Simplified Chinese, Traditional Chinese, Spanish, French, Portuguese, Russian, Italian, Polish, German and etc.***

Users can select the displayed language in web GUI login page, or at the upper right of the web GUI after logging in.

Figure 9: UCM6200 Web GUI Language

SAVE AND APPLY CHANGES

Click on "Save" button after configuring the web GUI options in one page. After saving all the changes, make sure click on "Apply Changes" button on the upper right of the web page to submit all the changes. If the change requires reboot to take effect, a prompted message will pop up for you to reboot the device.

MAKE YOUR FIRST CALL

Power up the UCM6200 and your SIP end point phone. Connect both devices to the network. Then follow the steps below to make your first call.

1. Log in the UCM6200 web GUI, go to **PBX**->**Basic/Call Routes**->**Extensions**.
2. Click on "Create New SIP Extension" to create a new extension. You will need User ID, Password and Voicemail Password information to register and use the extension later.
3. Register the extension on your phone with the SIP User ID, SIP server and SIP Password information. The SIP server address is the UCM6200 IP address.
4. When your phone is registered with the extension, dial *97 to access the voicemail box. Enter the Voicemail Password once you hear "Password" voice prompt.
5. Once successfully logged in to the voicemail, you will be prompted with the Voice Mail Main menu.
6. You are successfully connected to the PBX system now.

SYSTEM SETTINGS

This section explains configurations for system-wide parameters on the UCM6200. System settings are under “Settings” tag on UCM6200 web GUI. System settings include User Management, Network Settings, Firewall, Change Password, LDAP server, HTTP server, Email settings, Time Settings, NTP Server, Recordings Storage and Login Timeout settings.

USER MANAGEMENT

User management is on web GUI->**Settings**->**User Management** page. User could create multiple accounts for different administrators to log in the UCM6200 web GUI. Additionally, the system will automatically create user accounts along with creating new extensions for extension users to login to the web UI using their extension number and password. All existing user accounts for web UI login will be displayed on User Management page as shown in the following figure.

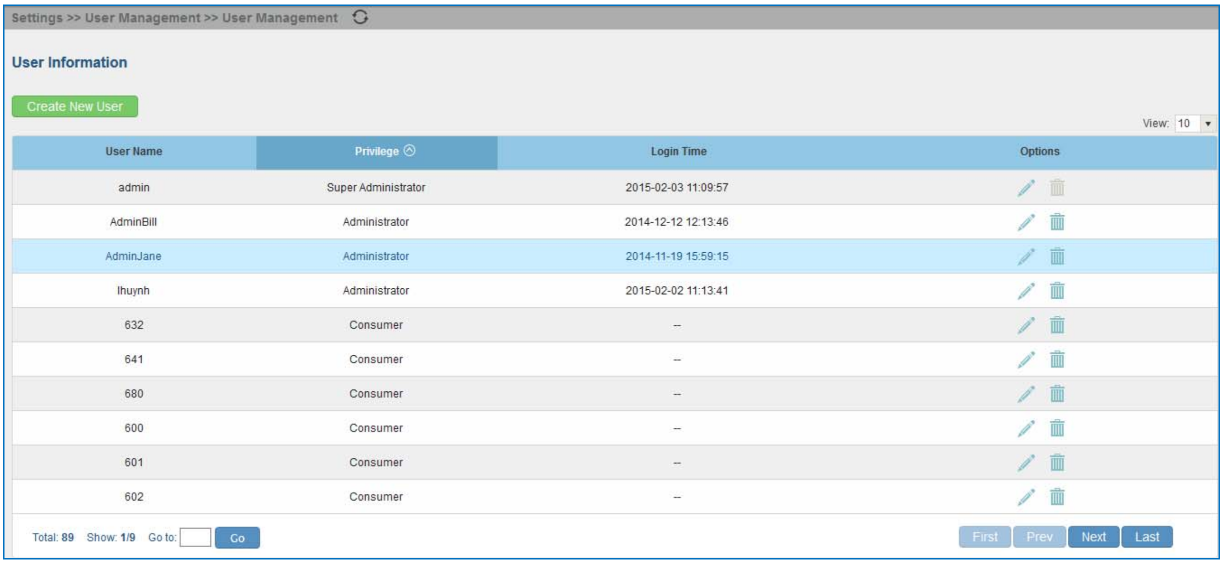


Figure 10: User Management Page Display

USER PRIVILEGES

Three privilege levels are supported:

- **Super Admin**

- This is the highest privilege. Super Admin can access all pages on UCM6200 web GUI, change configuration for all options and execute all the operations.
- Super Admin can create, edit and delete one or more users with “Admin” privilege
- Super Admin can edit and delete one or more users with “Consumer” privilege
- Super Admin can view operation logs generated by all users.
- By default, the user account “admin” is configured with “Super Admin” privilege and it’s the only user with “Super Admin” privilege. The User Name and Privilege level cannot be changed or deleted.
- Super Admin could change its own login password on web UI->**Settings**->**Change Password** page.
- Super Admin could view operations done by all the users in web UI->**Settings**->**User Management**->**Operation Log**.

- **Admin**

  - Users with “Admin” privilege can only be created by “Super Admin” user.
  - “Admin” privilege users are not allowed to access the following pages:
    Maintenance->**Upgrade**
    Maintenance->**Backup**
    Maintenance->**Cleaner**
    Maintenance->**Reset/Reboot**
    Settings->**User Management**->**Operation Log**
  - “Admin” privilege users cannot create new users for login.

- **Consumer**

  - A user account for web UI login is created automatically by the system when a new extension is created.
  - The user could log in the web UI with the extension number and password to access user information, extension configuration and CDR of that extension.

CREATE NEW WEB UI USER

When logged in as Super Admin, click on 

to create a new account for web UI user. The following dialog will prompt. Configure the parameters as shown in below table.

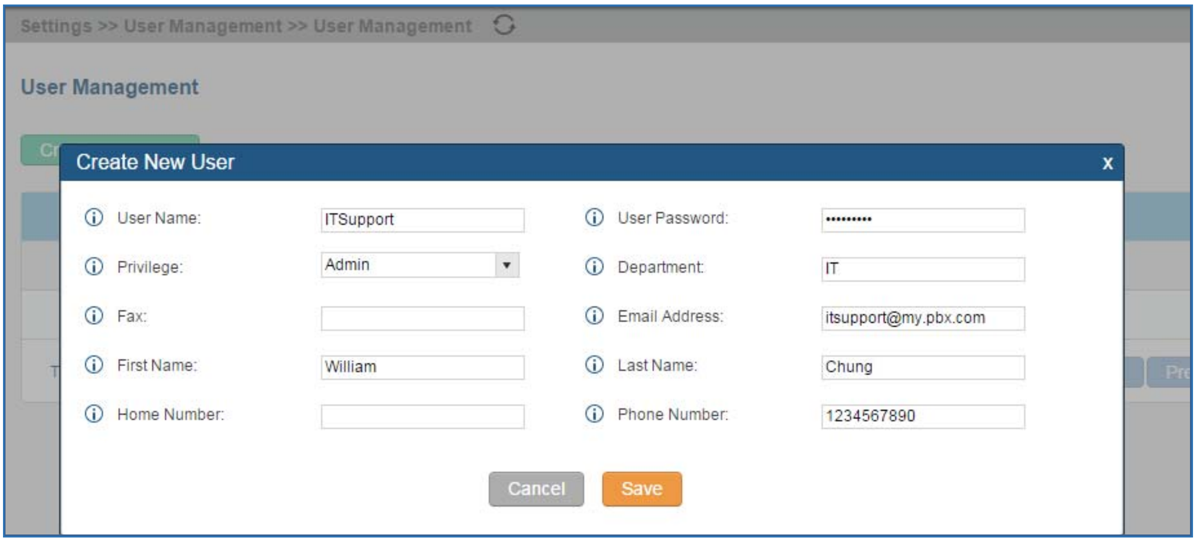


Figure 11: Create New User

Table 6: User Management->Create New User

User Name	Configure a username to identify the user which will be required in web UI login. Letters, digits and underscore are allowed in the user name.
User Password	Configure a password for this user which will be required in web UI login. Letters, digits and underscore are allowed.
Privilege	This is the role of the web UI user. Currently only "Admin" is supported when Super Admin creates a new user.
Department	Enter the necessary information to keep a record for this user.
Fax	
Email Address	
First Name	
Last Name	
Home Number	
Phone Number	

Once created, the Super Admin can edit the users by clicking on  or delete the user by clicking on .

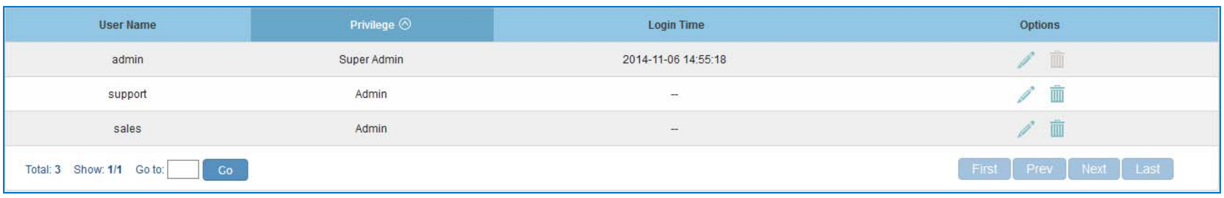


Figure 12: User Management – New Users

USER PORTAL

The user could log in web UI user portal using the extension number and password. When there is an extension created in the UCM6200, the corresponding user account for the extension is automatically created. The user portal allows limited access including user information, extension configuration and CDR information of the extension. The login username is the extension number and the password is configured by Super Admin. The following figure shows the dialog of editing the account information by Super Admin. The User Name must be the extension number and it's not configurable.

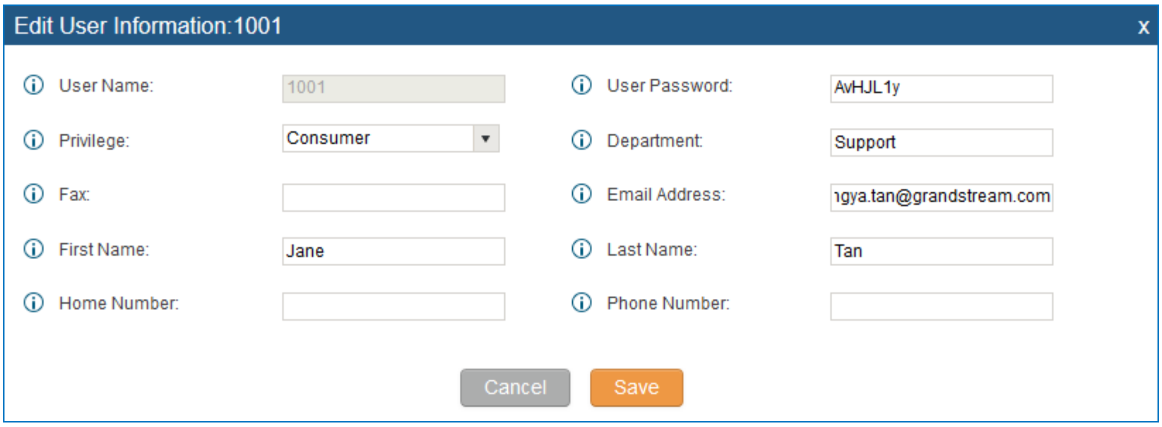


Figure 13: Edit User Information by Super Admin

The following figure shows an example of login page using extension number 1000 as the username.

Figure 14: User Portal Login

After login, the web UI displays is shown as below.

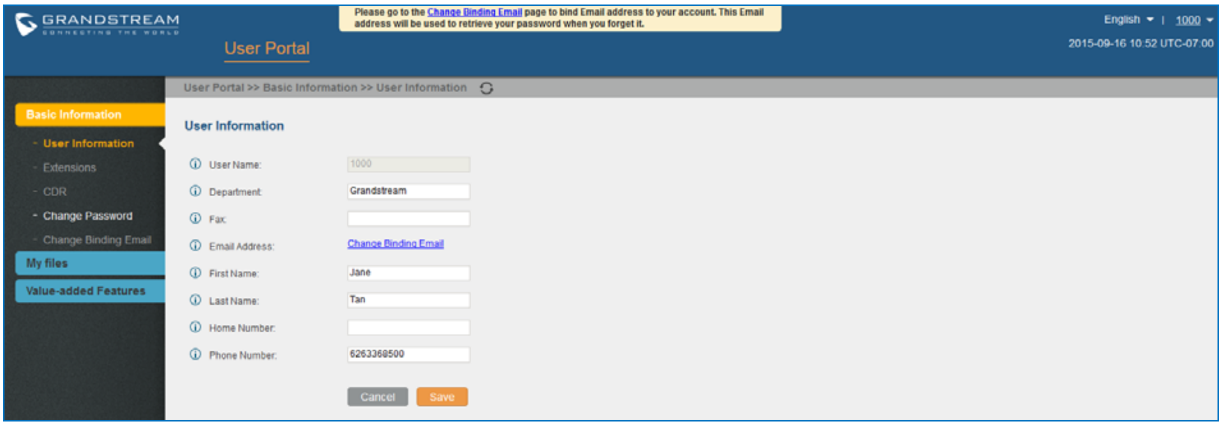


Figure 15: User Portal Layout

For the configuration parameter information in each page, please refer to ***[Table 6: User Management->Create New User]*** for options in **User Portal**->**Basic Information**->**User Information** page; please refer to ***[EXTENSIONS]*** for options in **User Portal**->**Basic Information**->**Extension** page; please refer to ***[CDR]*** for **User Portal**->**Basic Information**->**CDR** page.

CONCURRENT MULTI-USER LOGIN

When there are multiple web UI users created, concurrent multi-user login is supported on the UCM6200. Multiple users could edit options and have configurations take effect simultaneously. However, if different users are editing the same option or making the same operation (by clicking on "Apply Changes"), a prompt will pop up as shown in the following figure.

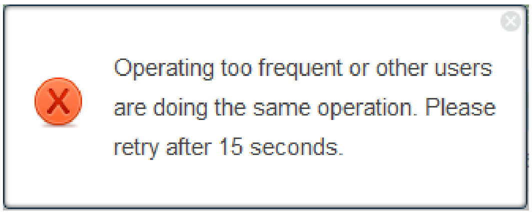
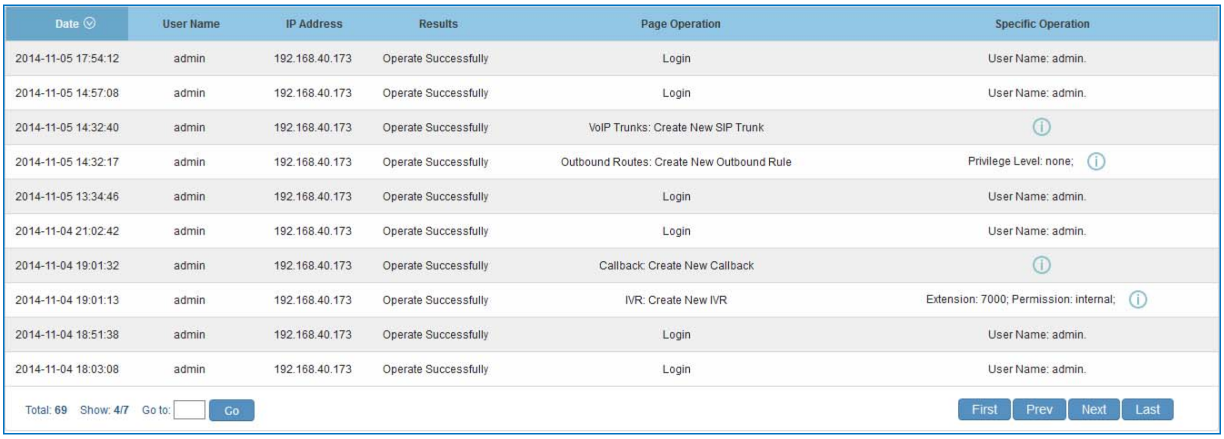


Figure 16: Multiple User Operation Error Prompt

OPERATION LOG

Super Admin has the authority to view operation logs on UCM6200 web GUI->**Settings**->**User Management**->**Operation Log** page. Operation logs list operations done by all the web UI users, for example, web UI login, creating trunk, creating outbound rule and etc. There are 6 columns to record the operation details "Date", "User Name", "IP Address", "Results", "Page Operation" and "Specific Operation".



Date	User Name	IP Address	Results	Page Operation	Specific Operation
2014-11-05 17:54:12	admin	192.168.40.173	Operate Successfully	Login	User Name: admin.
2014-11-05 14:57:08	admin	192.168.40.173	Operate Successfully	Login	User Name: admin.
2014-11-05 14:32:40	admin	192.168.40.173	Operate Successfully	VoIP Trunks: Create New SIP Trunk	
2014-11-05 14:32:17	admin	192.168.40.173	Operate Successfully	Outbound Routes: Create New Outbound Rule	Privilege Level: none;
2014-11-05 13:34:46	admin	192.168.40.173	Operate Successfully	Login	User Name: admin.
2014-11-04 21:02:42	admin	192.168.40.173	Operate Successfully	Login	User Name: admin.
2014-11-04 19:01:32	admin	192.168.40.173	Operate Successfully	Callback: Create New Callback	
2014-11-04 19:01:13	admin	192.168.40.173	Operate Successfully	IVR: Create New IVR	Extension: 7000; Permission: internal;
2014-11-04 18:51:38	admin	192.168.40.173	Operate Successfully	Login	User Name: admin.
2014-11-04 18:03:08	admin	192.168.40.173	Operate Successfully	Login	User Name: admin.

Total: 69 Show: 4/7 Go to: Go First Prev Next Last

Figure 17: Operation Logs

The operation log can be sorted and filtered for easy access. Click on the header of each column to sort. For example, clicking on "Date" will sort the logs according to operation date and time. Clicking on "Date" again will reverse the order.

Table 7: Operation Log Column Header

Date	The date and time when the operation is executed.
User Name	The username of the user who performed the operation.
IP Address	The IP address from which the operation is made.
Results	The result of the operation.
Page Operation	The page where the operation is made. For example, login, logout, delete user, create trunk and etc.
Specific Operation	Click on to view the options and values configured by this operation.

User could also filter the operation logs by time condition, IP address and/or username. Configure these conditions and then click on View Operation Logs.

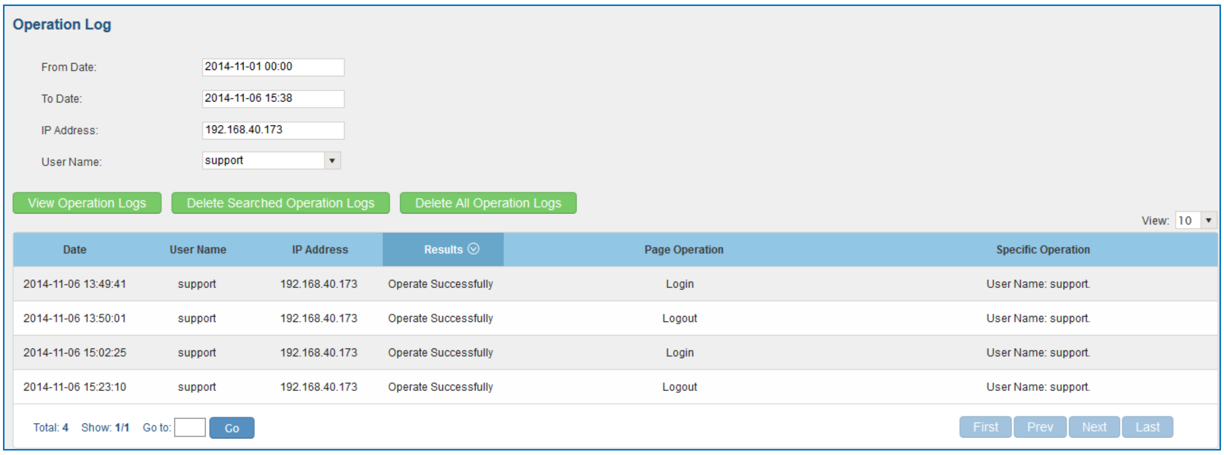


Figure 18: Operation Logs Filter

The above figure shows an example that operations made by user "support" on device with IP 192.168.40.173 from 2014-11-01 00:00 to 2014-11-06 15:38 are filtered out and displayed.

To delete operation logs, users can perform filtering first and then click on Delete Searched Operation Logs to delete the filtered result of operation logs. Or users can click on Delete All Operation Logs to delete all operation logs at once.

CHANGE PASSWORD

After logging in the UCM6200 web UI for the first time, it is highly recommended for users to change the default password "admin" to a more complicated password for security purpose. Follow the steps below to change the Web UI access password.

1. Go to Web UI->**Settings->User Management-> Change Password** page.
2. Enter the old password first.
3. Enter the new password and re-type the new password to confirm. The new password has to be at least 4 characters. The maximum length of the password is 30 characters.
4. Configure the Email Address that is used when login credential is lost.
5. Click on "Save" and the user will be automatically logged out.
6. Once the web page comes back to the login page again, enter the username "admin" and the new password to login.

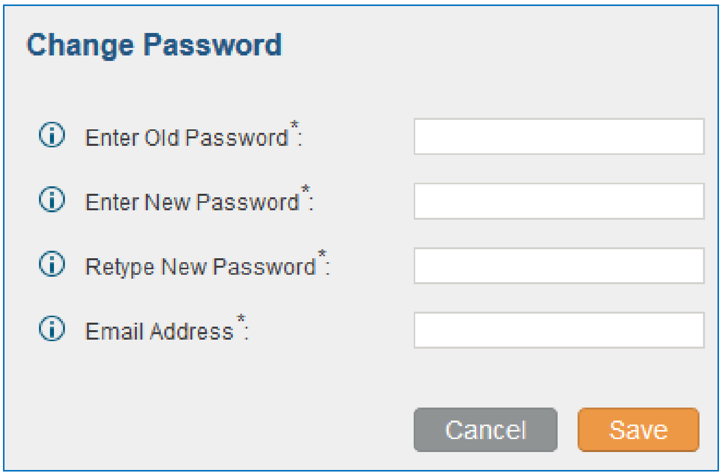


Figure 19 : Change Password

Enter Old Password	Enter the Old Password for UCM6200
Enter New Password	Enter the New Password for UCM6200
Retype New Password	Retype the New Password for UCM6200
Email Address	Configure the Email address for UCM6200. In case login credential is lost, Email address is used to retrieve login credential

CHANGE BINDNG EMAIL

UCM6200 allows user to configure binding email in case login password is lost. UCM6200 login credential will be sent to the designated email address. The feature can be found under web UI->Settings->User Management->Change Binding Email.

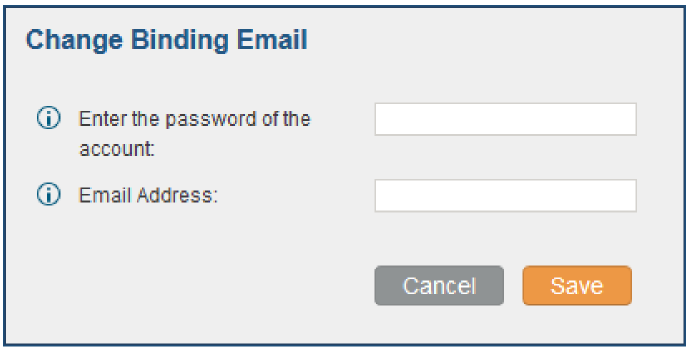


Figure 20: Change Binding Email

Table 8: Change Binding Email option

Enter the password of the account	Enter the current login user credential for UCM6200
Email Address	Email Address is used to retrieve password when password is lost

NETWORK SETTINGS

After successfully connecting the UCM6200 to the network for the first time, users could login the Web GUI and go to **Settings->Network Settings** to configure the network parameters for the device.

Additional network functions and settings are available for UCM6202 and UCM6204:

- UCM6202/UCM6204 supports Route/Switch/Dual mode functions.

In this section, all the available network setting options are listed for all models. Select each tab in web GUI->**Settings**->**Network Settings** page to configure LAN settings, WAN settings (UCM6202/UCM6204 only), 802.1X and Port Forwarding (UCM6202/UCM6204 only).

BASIC SETTINGS

Please refer to the following tables for basic network configuration parameters on UCM6202/UCM6204, and UCM6208 respectively.

Table 9: UCM6202/UCM6204 Network Settings->Basic Settings

Method	Select "Route", "Switch" or "Dual" mode on the network interface of UCM6200. The default setting is "Route". • **Route** WAN port interface will be used for uplink connection. LAN port interface will be used to serve as router. • **Switch** WAN port interface will be used for uplink connection. LAN port interface will be used as bridge for PC connection. • **Dual** Both ports can be used for uplink connection. Users will need assign LAN 1 or LAN 2 as the default interface in option "Default Interface" and configure "Gateway IP" for this interface.

Preferred DNS Server	Enter the preferred DNS server address. If Preferred DNS is used, UCM will try to use it as Primary DNS server.
WAN (when "Method" is set to "Route")	
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
Subnet Mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
Gateway IP	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
DNS Server 1	Enter the DNS server 1 address for static IP settings. The default setting is 0.0.0.0.
DNS Server 2	Enter the DNS server 2 address for static IP settings.
User Name	Enter the user name to connect via PPPoE.
Password	Enter the password to connect via PPPoE.
Layer 2 QoS 802.1Q/VLAN Tag	Assign the VLAN tag of the layer 2 QoS packets for WAN port. The default value is 0.
Layer 2 QoS 802.1p Priority Value	Assign the priority value of the layer 2 QoS packets for WAN port. The default value is 0.
LAN (when Method is set to "Route")	
IP Address	Enter the IP address assigned to LAN port. The default setting is 192.168.2.1.
Subnet Mask	Enter the subnet mask. The default setting is 255.255.255.0.
DHCP Server Enable	Enable or disable DHCP server capability. The default setting is "Yes".
DNS Server 1	Enter DNS server address 1. The default setting is 8.8.8.8.
DNS Server 2	Enter DNS server address 2. The default setting is 208.67.222.222.
Allow IP Address From	Enter the DHCP IP Pool starting address. The default setting is 192.168.2.100.
Allow IP Address To	Enter the DHCP IP Pool ending address. The default setting is 192.168.2.254.
Default IP Lease Time	Enter the IP lease time (in seconds). The default setting is 43200.
LAN (when Method is set to "Switch")	
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
Subnet Mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
Gateway IP	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
DNS Server 1	Enter the DNS server 1 address for static IP settings. The default setting is 0.0.0.0.
DNS Server 2	Enter the DNS server 2 address for static IP settings.

User Name	Enter the user name to connect via PPPoE.
Password	Enter the password to connect via PPPoE.
Layer 2 QoS 802.1Q/VLAN Tag	Assign the VLAN tag of the layer 2 QoS packets for LAN port. The default value is 0.
Layer 2 QoS 802.1p Priority Value	Assign the priority value of the layer 2 QoS packets for LAN port. The default value is 0.
LAN 1 / LAN 2 (when Method is set to "Dual")	
Default Interface	If "Dual" is selected as "Method", users will need assign the default interface to be LAN 1 (mapped to UCM6202 WAN port) or LAN 2 (mapped to UCM6202 LAN port) and then configure network settings for LAN 1/LAN 2. The default interface is LAN 2.
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
Subnet Mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
Gateway IP	Enter the gateway IP address for static IP settings when the port is assigned as default interface. The default setting is 0.0.0.0.
DNS Server 1	Enter the DNS server 1 address for static IP settings. The default setting is 0.0.0.0.
DNS Server 2	Enter the DNS server 2 address for static IP settings.
User Name	Enter the user name to connect via PPPoE.
Password	Enter the password to connect via PPPoE.
Layer 2 QoS 802.1Q/VLAN Tag	Assign the VLAN tag of the layer 2 QoS packets for LAN port. The default value is 0.
Layer 2 QoS 802.1p Priority Value	Assign the priority value of the layer 2 QoS packets for LAN port. The default value is 0.

Table 10: UCM6208 Network Settings->Basic Settings

Preferred DNS Server	Enter the preferred DNS server address. If Preferred DNS is used, UCM will try to use it as Primary DNS server.
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings.
Subnet Mask	Enter the subnet mask address for static IP settings.
Gateway IP	Enter the gateway IP address for static IP settings.

DNS Server 1	Enter the DNS server 1 address for static IP settings.
DNS Server 2	Enter the DNS server 2 address for static IP settings.
User Name	Enter the user name to connect via PPPoE.
Password	Enter the password to connect via PPPoE.
Layer 2 QoS 802.1Q/VLAN Tag	Assign the VLAN tag of the layer 2 QoS packets for LAN port. The default value is 0.
Layer 2 QoS 802.1p Priority Value	Assign the priority value of the layer 2 QoS packets for LAN port. The default value is 0.

- **Method: Route**

When the UCM6200 has method set to Route in network settings, WAN port interface is used for uplink connection and LAN port interface is used as a router. Please see a sample diagram below.

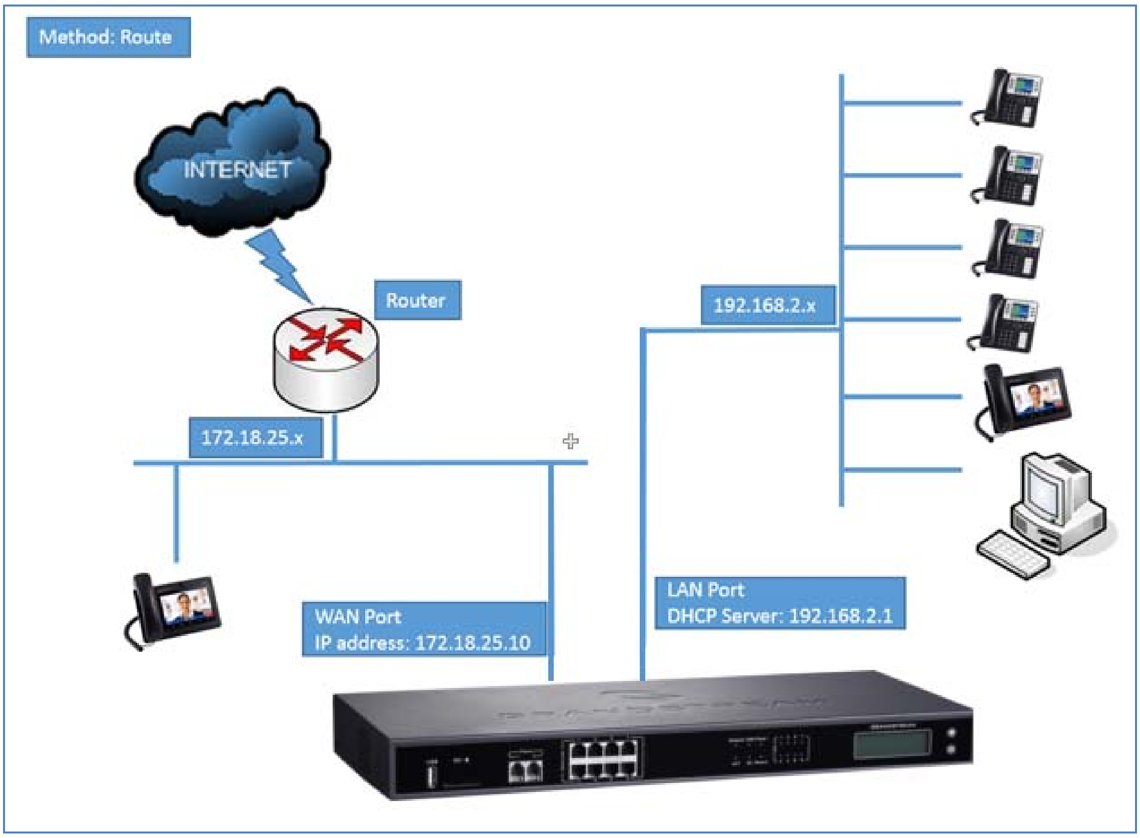


Figure 21: UCM6200 Network Interface Method: Route

- **Method: Switch**

WAN port interface is used for uplink connection; LAN port interface is used as bridge for PC connection.

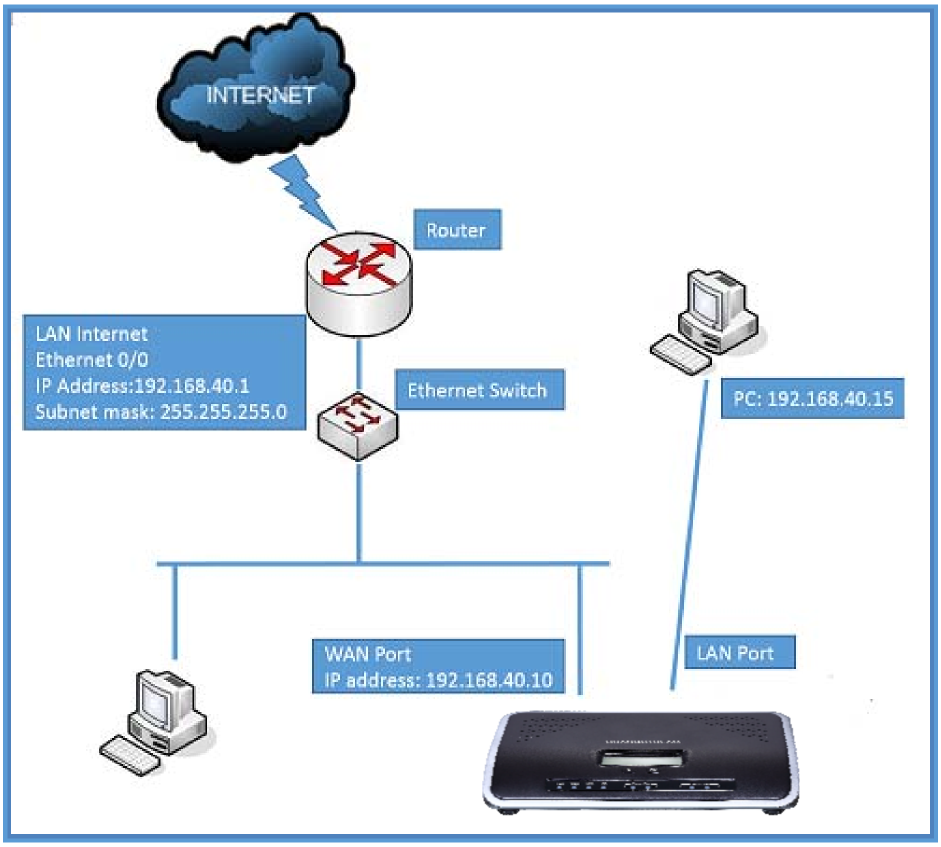


Figure 22: UCM6200 Network Interface Method: Switch

- **Method: Dual**

Both WAN port and LAN port are used for uplink connection. Users will need assign LAN 1 or LAN 2 as the default interface in option "Default Interface" and configure "Gateway IP" if static IP is used for this interface.

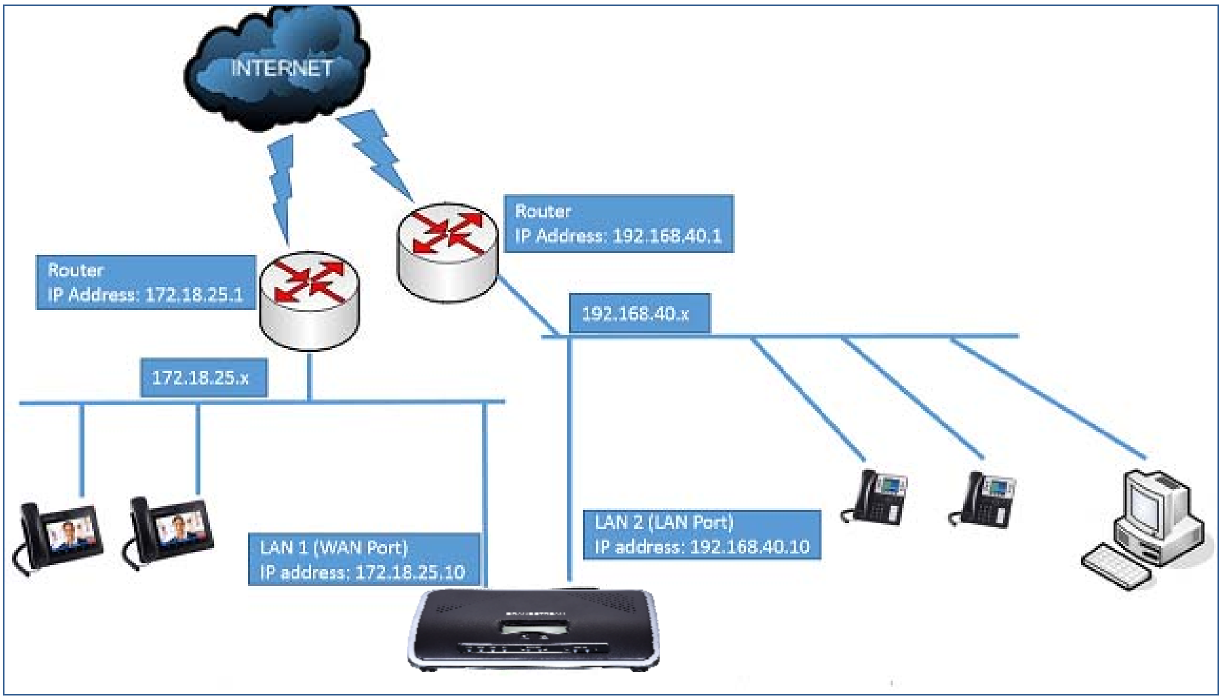


Figure 23: UCM6200 Network Interface Method: Dual

802.1X

IEEE 802.1X is an IEEE standard for port-based network access control. It provides an authentication mechanism to device before the device is allowed to access Internet or other LAN resources. The UCM6200 supports 802.1X as a supplicant/client to be authenticated. The following diagram and figure show UCM6200 uses 802.1X mode “EAP-MD5” on WAN port as client in the network to access Internet.

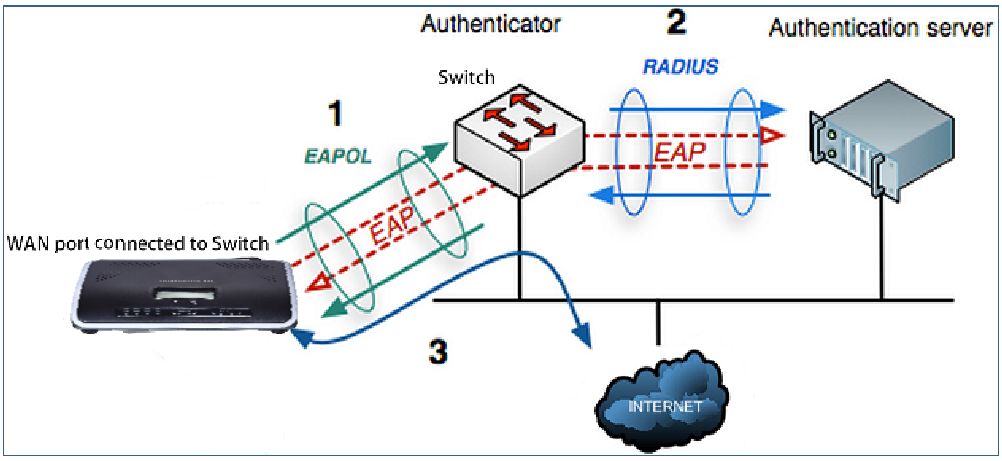


Figure 24: UCM6200 Using 802.1X as Client

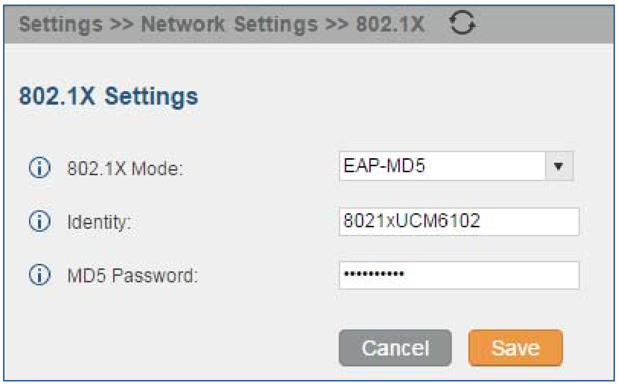


Figure 25: UCM6200 Using 802.1X EAP-MD5

The following table shows the configuration parameters for 802.1X on UCM6200. Identity and MD5 password are required for authentication, which should be provided by the network administrator obtained from the RADIUS server. If "EAP-TLS" or "EAP-PEAPv0/MSCHAPv2" is used as the 802.1X mode, users will also need upload 802.1X CA Certificate and 802.1X Client Certificate, which should be also generated from the RADIUS server.

Table 11: UCM6200 Network Settings->802.1X

802.1X Mode	Select 802.1X mode. The default setting is "Disable". The supported 802.1X mode are: • EAP-MD5 • EAP-TLS • EAP-PEAPv0/MSCHAPv2
Identity	Enter 802.1X mode identity information.
MD5 Password	Enter 802.1X mode MD5 password information.
802.1X Certificate	Select 802.1X certificate from local PC and then upload.
802.1X Client Certificate	Select 802.1X client certificate from local PC and then upload.

STATIC ROUTES

The UCM6200 provides users static routing capability that allows the device to use manually configured routes, rather than information only from dynamic routing or gateway configured in the UCM6200 web GUI->**Network Settings**->**Basic Settings** to forward traffic. It can be used to define a route when no other routes are available or necessary, or used in complementary with existing routing on the UCM6200 as a failover backup, and etc.

- Click on Create New Static Route to create a new static route. The configuration parameters are listed in the table below.
- Once added, users can select to edit the static route.
- Select to delete the static route.

Table 12: UCM6200 Network Settings->Static Routes

Destination	Configure the destination IP address or the destination IP subnet for the UCM6200 to reach using the static route. Example: IP address - ***192.168.66.4*** IP subnet - ***192.168.66.0***
Netmask	Configure the subnet mask for the above destination address. If left blank, the default value is 255.255.255.255. Example: ***255.255.255.0***
Gateway	Configure the gateway address so that the UCM6200 can reach the destination via this gateway. Gateway address is optional. Example: ***192.168.40.5***
Interface	Specify the network interface on the UCM6200 to reach the destination using the static route. LAN interface is eth0; WAN interface is eth1.

Static routes configuration can be reset from LCD menu->Network Menu.

The following diagram shows a sample application of static route usage on UCM6204.

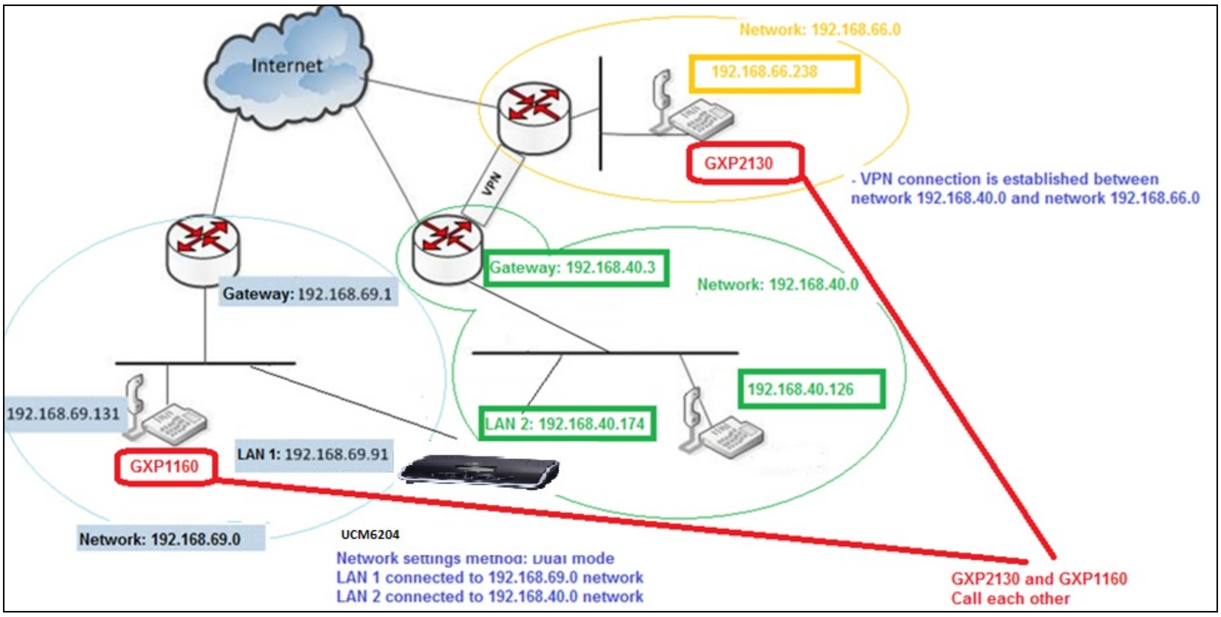


Figure 26: UCM6204 Static Route Sample

The network topology of the above diagram is as below:

- Network 192.168.69.0 has IP phones registered to UCM6204 LAN 1 address
- Network 192.168.40.0 has IP phones registered to UCM6204 LAN 2 address
- Network 192.168.66.0 has IP phones registered to UCM6204 via VPN
- Network 192.168.40.0 has VPN connection established with network 192.168.66.0

In this network, by default the IP phones in network 192.168.69.0 are unable to call IP phones in network 192.168.66.0 when registered on different interfaces on the UCM6204. Therefore, we need configure a static route on the UCM6204 so that the phones in isolated networks can make calls between each other.

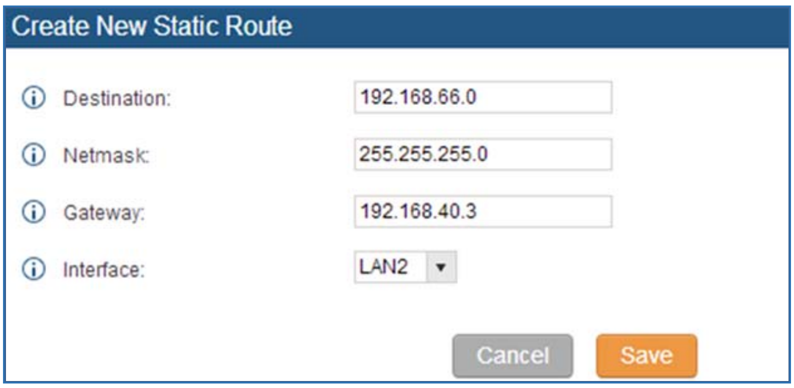


Figure 27: UCM6204 Static Route Configuration

PORT FORWORDING

The UCM network interface supports router function which provides users the ability to do port forwarding. If the UCM6202/UCM6204 LAN mode is set to "Route" under web GUI->**Settings**->**Network Settings**->**Basic Settings** page, port forwarding is available for configuration.

The port forwarding configuration is under web GUI->**Settings**->**Network Settings**->**Port Forwarding** page. Please see related settings in the table below.

Table 13: UCM6202/UCM6204 Network Settings->Port Forwarding

WAN Port	Specify the WAN port number or a range of WAN ports. Up to 8 ports can be configured. **Note:** When it is set to a range, WAN port and LAN port must be configured with the same range, such as WAN port: 1000-1005 and LAN port: 1000-1005, and access from WAN port will be forwarded to the LAN port with the same port number, for example, WAN port 1000 will be port forwarding to LAN port 1000.
LAN IP	Specify the LAN IP address.
LAN Port	Specify the LAN port number or a range of LAN ports. **Note:** When it is set to a range, WAN port and LAN port must be configured with the same range, such as WAN port: 1000-1005 and LAN port: 1000-1005, and access from WAN port will be forwarded to the LAN port with the same port number, for example, WAN port 1000 will be port forwarding to LAN port 1000.
Protocol Type	Select protocol type "UDP Only", "TCP Only" or "TCP/UDP" for the forwarding in the selected port. The default setting is "UDP Only".

The following figures demonstrate a port forwarding example to provide phone’s web UI access to public side.

- The UCM6202/UCM6204 network mode is set to “Route”
- The UCM6202/UCM6204 WAN port is connected to uplink switch, with a public IP address configured, e.g. 1.1.1.1.
- The UCM6202/UCM6204 LAN port provides DHCP pool that connects to multiple phone devices in the LAN network 192.168.2.x. The UCM6202/UCM6204 is used as a router, with gateway address 192.168.2.1

- There is a GXP2160 connected under the LAN interface network of the UCM6202/UCM6204. It obtains IP address 192.168.2.100 from UCM6200 DHCP pool
- On the UCM6202/UCM6204 web UI->**Settings**->**Network Settings**->**Port Forwarding**, configure a port forwarding entry as the figure shows below.
  WAN Port: This is the port opened up on the WAN side for access purpose.
  LAN IP: This is the GXP2160 IP address, under the LAN interface network of the UCM6202/UCM6204.
  Protocol Type: We select TCP here for web UI access using HTTP.

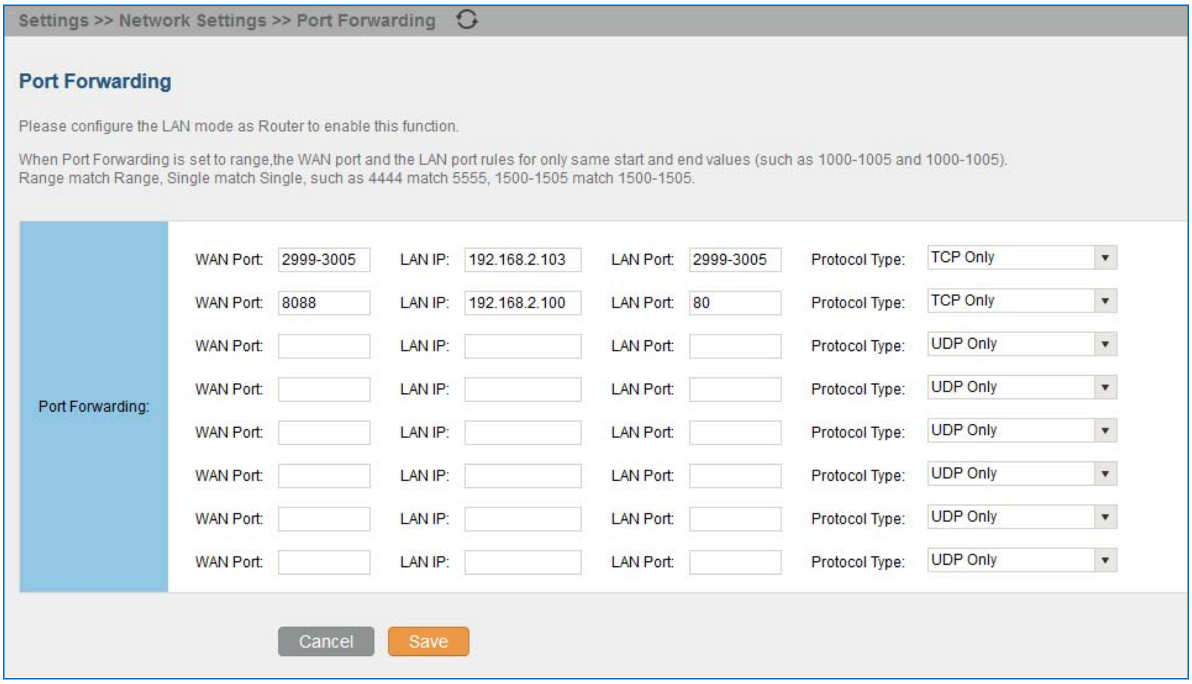


Figure 28: UCM6202/UCM6204 Port Forwarding Configuration

This will allow users to access the GXP2160 web UI from public side, by typing in address “1.1.1.1:8088”.

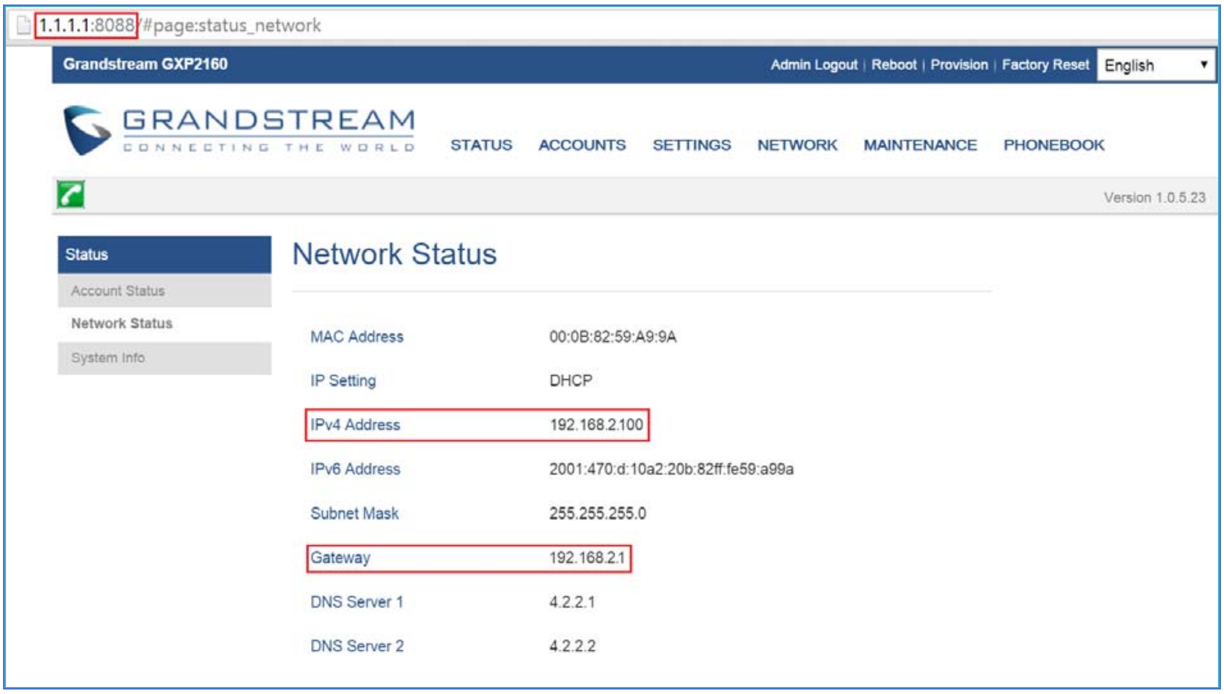


Figure 29: GXP2160 Web Access Using UCM6202 Port Forwarding

DDNS SETTINGS

DDNS setting allows user to access UCM6200 via domain name instead of IP address.
The UCM supports DDNS service from the following DDNS provider:

- dydns.org
- noip.com
- freedns.afraid.org
- zoneedit.com
- oray.net

Here is an example of using noip.com for DDNS.

1. Register domain in DDNS service provider. Please note the UCM6200 needs to have public IP access.

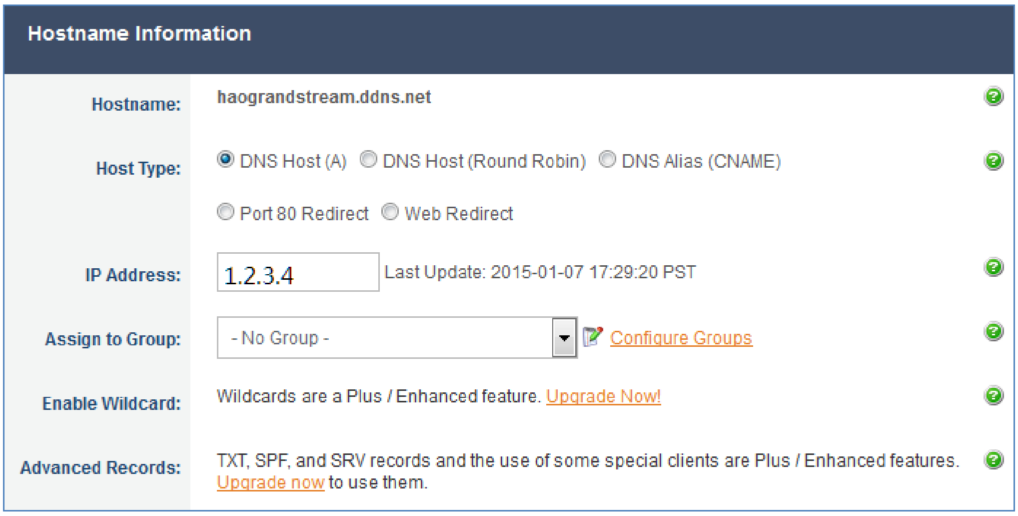


Figure 30: Register Domain Name on noip.com

2. On **web UI->Settings->Network Settings->DDNS Settings,** enable DDNS service and configure username, password and host name.

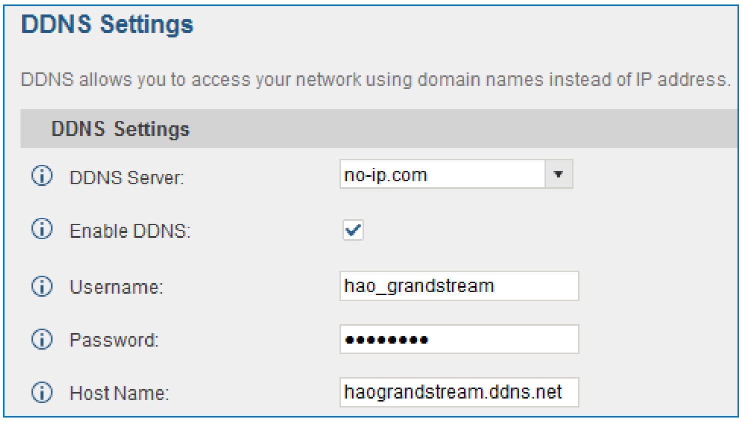


Figure 31: UCM6200 DDNS Setting

3. Now you can use domain name instead of IP address to connect to the UCM6200 web UI.

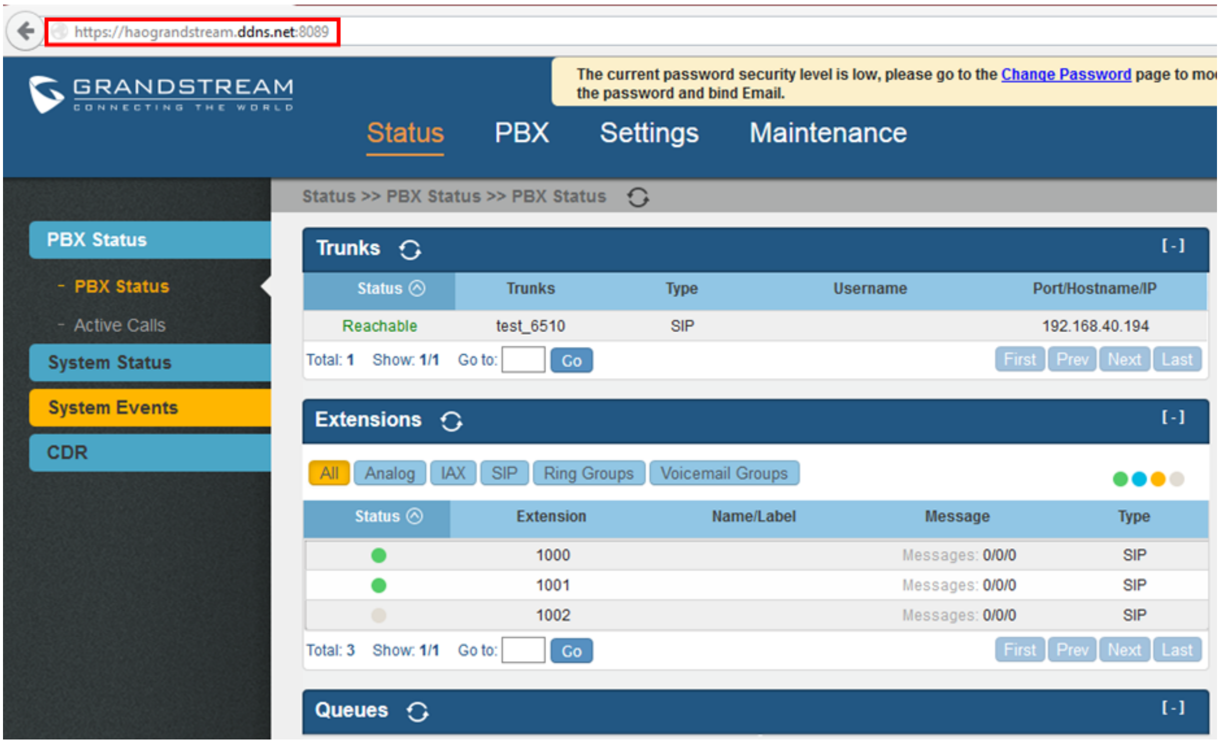


Figure 32: Using Domain Name to Connect to UCM6200

FIREWALL

The UCM6200 provides users firewall configurations to prevent certain malicious attack to the UCM6200 system. Users could configure to allow, restrict or reject specific traffic through the device for security and bandwidth purpose. The UCM6200 also provides Fail2ban feature for authentication errors in SIP REGISTER, INVITE and SUBSCRIBE. To configure firewall settings in the UCM6200, go to Web UI->**Settings**->**Firewall** page.

STATIC DEFENSE

Under Web GUI->**Settings**->**Firewall->Static Defense** page, users will see the following information:

- Current service information with port, process and type.
- Typical firewall settings.
- Custom firewall settings.

The following table shows a sample current service status running on the UCM6200.

Table 14: UCM6200 Firewall->Static Defense->Current Service

Port	Process	Type	Protocol or Service
7777	Asterisk	tcp/IPv4	SIP
389	Slapd	tcp/IPv4	LDAP
22	Dropbear	tcp/IPv4	SSH
80	Lighthttpd	tcp/IPv4	HTTP
8089	Lighthttpd	tcp/IPv4	HTTPS
69	Opentftpd	udp/IPv4	TFTP
9090	Asterisk	udp/IPv4	SIP
6060	zero_config	udp/IPv4	UCM6200 zero_config service
5060	Asterisk	udp/IPv4	SIP
4569	Asterisk	udp/IPv4	SIP
5353	zero_config	udp/IPv4	UCM6200 zero_config service
37435	Syslogd	udp/IPv4	Syslog

For typical firewall settings, users could configure the following options on the UCM6200.

Table 15: Typical Firewall Settings

Ping Defense Enable	If enabled, ICMP response will not be allowed for Ping request. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (UCM6202/UCM6204) interface.
Ping-of-Death Defense Enable	Enable to prevent Ping-of-Death attack to the device. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (UCM6202/UCM6204) interface.

Under "Custom Firewall Settings", users could create new rules to accept, reject or drop certain traffic going through the UCM6200. To create new rule, click on "Create New Rule" button and a new window will pop up for users to specify rule options.

Right next to "Create New Rule" button, there is a checkbox for option "Reject Rules". If it’s checked, all the rules will be rejected except the firewall rules listed below. In the firewall rules, only when there is a rule that meets all the following requirements, the option "Reject Rules" will be allowed to check:

- Action: "Accept"
- Type "In"
- Destination port is set to the system login port (e.g., by default 8089)
- Protocol is not UDP

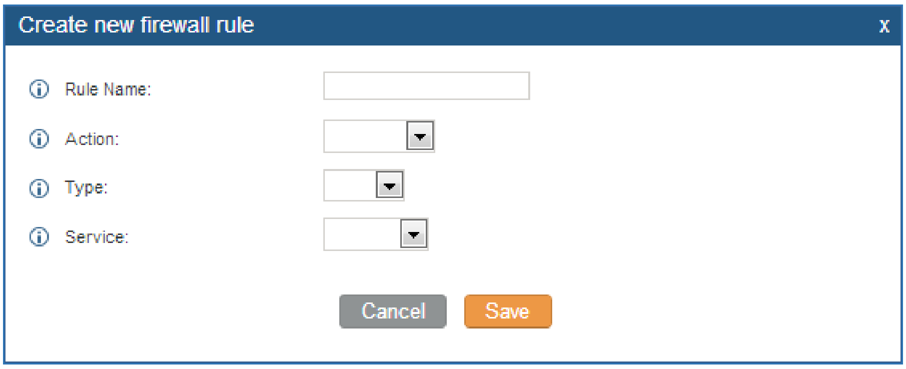


Figure 33: Create New Firewall Rule

Table 16: Firewall Rule Settings

Rule Name	Specify the Firewall rule name to identify the firewall rule.
Action	Select the action for the Firewall to perform. • ACCEPT • REJECT • DROP
Type	Select the traffic type. • IN If selected, users will need specify the network interface "LAN" or "WAN" (for UCM6202/UCM6204) for the incoming traffic. • OUT
Service	Select the service type. • FTP • SSH • Telnet • TFTP • HTTP • LDAP • Custom If "Custom" is selected, users will need specify Source (IP and port), Destination (IP and port) and Protocol (TCP, UDP or Both) for the service. Please note if the source or the destination field is left blank, it will be used as "Anywhere".

Save the change and click on "Apply" button. Then submit the configuration by clicking on "Apply Changes" on the upper right of the web page. The new rule will be listed at the bottom of the page with sequence number, rule name, action, protocol, type, source, destination and operation. More operations below:

- Click on to edit the rule
- Click on to delete the rule

DYNAMIC DEFENSE

Dynamic defense is supported on the UCM6200 series. It can blacklist hosts dynamically when the LAN mode is set to "Route" under web GUI->**Settings**->**Network Settings**->**Basic Settings** page. If enabled, the traffic coming into the UCM6200 can be monitored, which helps prevent massive connection attempts or brute force attacks to the device. The blacklist can be created and updated by the UCM6200 firewall, which will then be displayed in the web page. Please refer to the following table for dynamic defense options on the UCM6200.

Table 17: UCM6200 Firewall Dynamic Defense

Dynamic Defense Enable	Enable dynamic defense. The default setting is disabled.
Periodical Time Interval	Configure the dynamic defense periodic time interval (in minutes). If the number of TCP connections from a host exceeds the connection threshold within this period, this host will be added into Blacklist. The valid value is between 1 and 59 when dynamic defense is turned on. The default setting is 59.
Blacklist Update Interval	Configure the blacklist update time interval (in seconds). The default setting is 120.
Connection Threshold	Configure the connection threshold. Once the number of connections from the same host reaches the threshold, it will be added into the blacklist. The default setting is 100.
Dynamic Defense Whitelist	Configure the dynamic defense whitelist. For example, ***192.168.1.3*** ***192.168.1.4***

The following figure shows a configuration example like this:

- If a host at IP address 192.168.40.7 initiates more than 20 TCP connections to the UCM6200 within 1 minute, it will be added into UCM6200 blacklist.
- This host 192.168.40.7 will be blocked by the UCM6200 for 300 seconds.

- Since IP address 192.168.40.5 is in whitelist, if the host at IP address 192.168.40.5 initiates more than 20 TCP connections to the UCM6200 within 1 minute, it will not be added into UCM6200 blacklist. It can still establish TCP connection with the UCM6200.

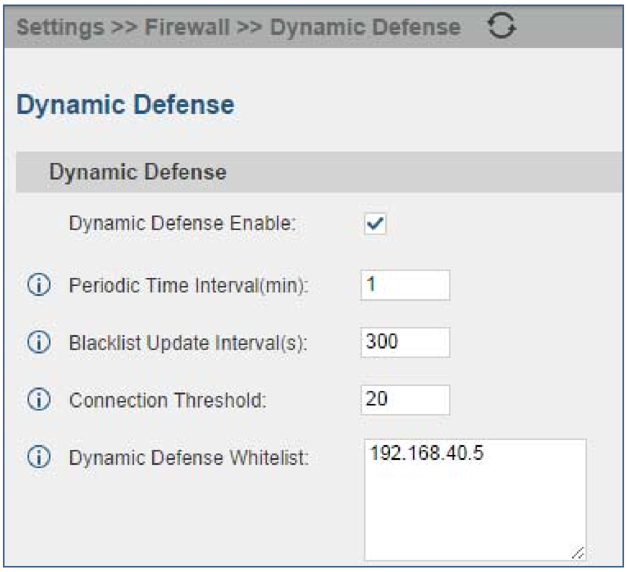


Figure 34: Configure Dynamic Defense

FAIL2BAN

Fail2Ban feature on the UCM6200 provides intrusion detection and prevention for authentication errors in SIP REGISTER, INVITE and SUBSCRIBE. Once the entry is detected within "Max Retry Duration", the UCM6200 will take action to forbid the host for certain period as defined in "Banned Duration". This feature helps prevent SIP brute force attacks to the PBX system.

Table 18: Fail2Ban Settings

Global Settings	
Enable Fail2Ban	Enable Fail2Ban. The default setting is disabled. Please make sure both "Enable Fail2Ban" and "Asterisk Service" are turned on in order to use Fail2Ban for SIP authentication on the UCM6200.
Banned Duration	Configure the duration (in seconds) for the detected host to be banned. The default setting is 300. If set to -1, the host will be always banned.
Max Retry Duration	Within this duration (in seconds), if a host exceeds the max times of retry as defined in "MaxRetry", the host will be banned. The default setting is 5.
MaxRetry	Configure the number of authentication failures during "Max Retry Duration" before the host is banned. The default setting is 10.
Fail2Ban Whitelist	Configure IP address, CIDR mask or DNS host in the whiltelist. Fail2Ban will not

	ban the host with matching address in this list. Up to 5 addresses can be added into the list.
Local Settings	
Asterisk Service	Enable Asterisk service for Fail2Ban. The default setting is disabled. Please make sure both "Enable Fail2Ban" and "Asterisk Service" are turned on in order to use Fail2Ban for SIP authentication on the UCM6200.
Protocol	Configure the listening port number for the service. Currently only 5060 (for UDP) is supported.
MaxRetry	Configure the number of authentication failures during "Max Retry Duration" before the host is banned. The default setting is 10. Please make sure this option is properly configured as it will override the "MaxRetry" value under "Global Settings".

LDAP SERVER

The UCM6200 has an embedded LDAP server for users to manage corporate phonebook in a centralized manner.

- By default, the LDAP server has generated the first phonebook with **PBX DN** "ou=pbx,dc=pbx,dc=com" based on the UCM6200 user extensions already.
- Users could add new phonebook with a different **Phonebook DN** for other external contacts. For example, "ou=people,dc=pbx,dc=com".
- All the phonebooks in the UCM6200 LDAP server have the same **Base DN** "dc=pbx,dc=com".

Term Explanation:
cn= Common Name
ou= Organization Unit
dc= Domain Component
These are all parts of the LDAP data Interchange Format, according to RFC 2849, which is how the LDAP tree is filtered.

If users have the Grandstream phone provisioned by the UCM6200, the LDAP directory will be set up on the phone and can be used right away for users to access all phonebooks.

Additionally, users could manually configure the LDAP client settings to manipulate the built-in LDAP server on the UCM6200. If the UCM6200 has multiple LDAP phonebooks created, in the LDAP client configuration, users could use "dc=pbx,dc=com" as Base DN to have access to all phonebooks on the

UCM6200 LDAP server, or use a specific phonebook DN, for example "ou=people,dc=pbx,dc=com", to access to phonebook with Phonebook DN "ou=people,dc=pbx,dc=com " only.

UCM can also act as a LDAP client to download phonebook entries from other LDAP server.

To access LDAP server and client settings, go to **Web GUI->Settings->LDAP Server**.

LDAP SERVER CONFIGURATIONS

The following figure shows the default LDAP server configurations on the UCM6200.

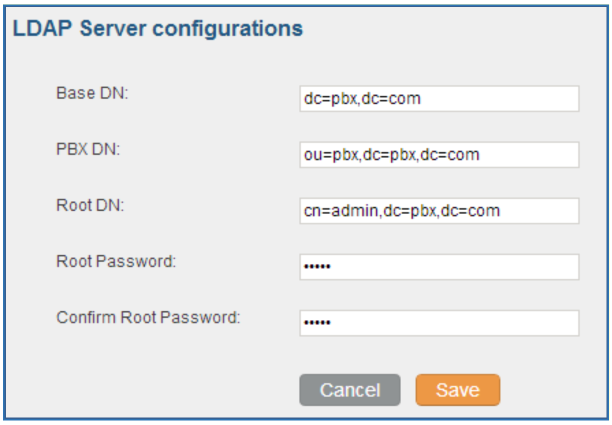
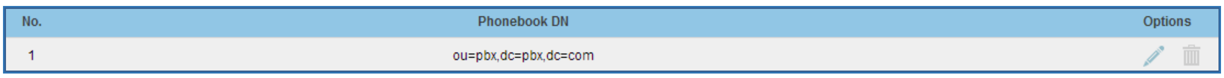


Figure 35: LDAP Server Configurations

The UCM6200 LDAP server supports anonymous access (read-only) by default. Therefore the LDAP client doesn't have to configure username and password to access the phonebook directory. The "Root DN" and "Root Password" here are for LDAP management and configuration where users will need provide for authentication purpose before modifying the LDAP information.

The default phonebook list in this LDAP server can be viewed and edited by clicking on for the first phonebook under LDAP Phonebook.



No.	Phonebook DN	Options
1	ou=pbx,dc=pbx,dc=com	

Figure 36: Default LDAP Phonebook DN

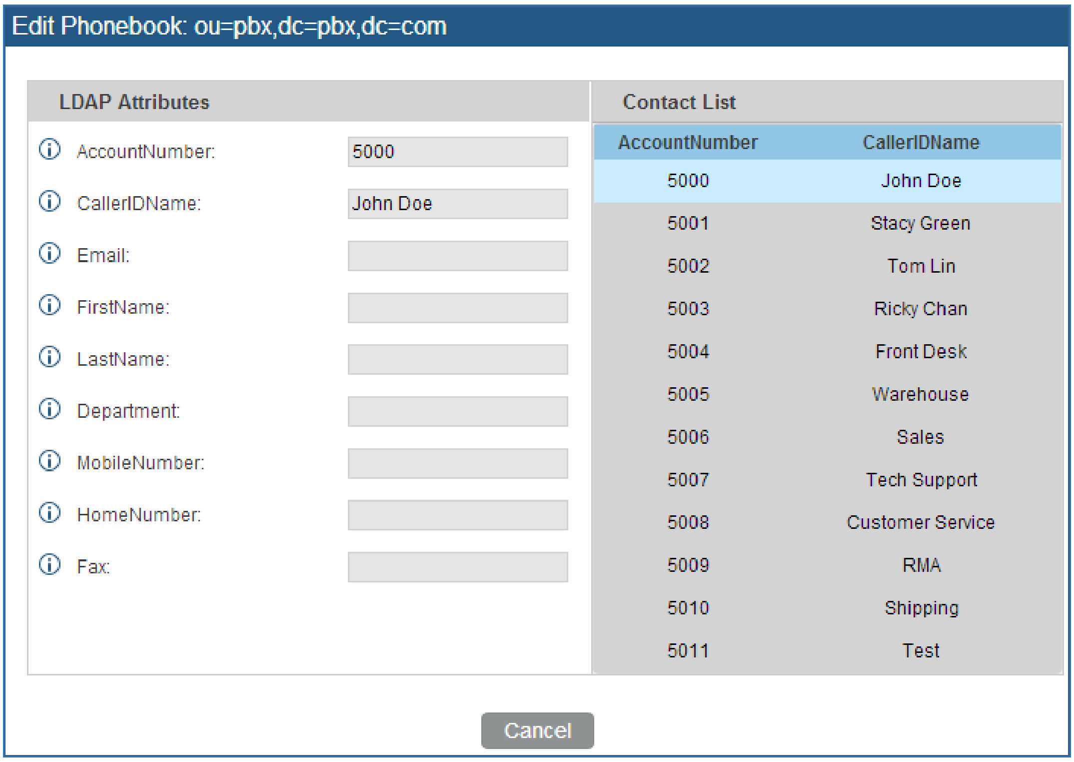


Figure 37: Default LDAP Phonebook Attributes

LDAP PHONEBOOK

Users could use the default phonebook, edit the default phonebook, add new phonebook, import phonebook on the LDAP server as well as export phonebook from the LDAP server. The first phonebook with default phonebook dn "ou=pbx,dc=pbx,dc=com" displayed on the LDAP server page is for extensions in this PBX. Users cannot add or delete contacts directly. The contacts information will need to be modified via Web GUI->**PBX**->**Basic/Call Routes**->**Extensions** first. The default LDAP phonebook will then be updated automatically.

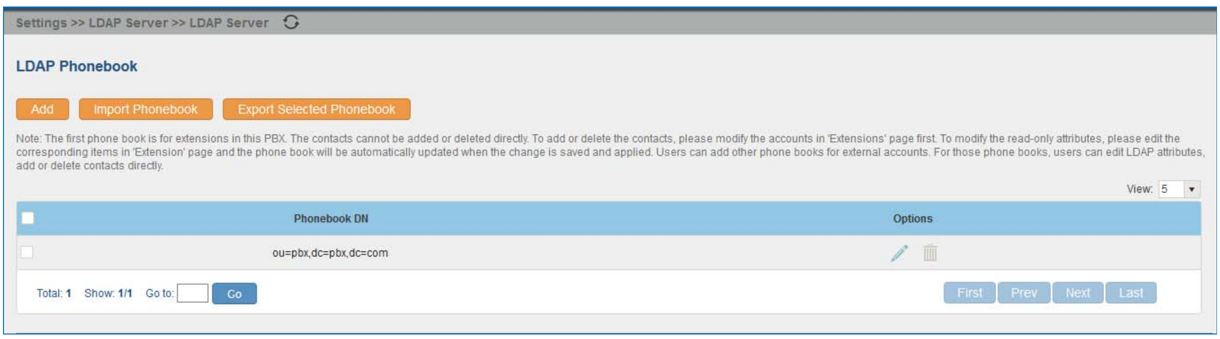


Figure 38: LDAP Server->LDAP Phonebook

- **Add new phonebook**

  A new sibling phonebook of the default PBX phonebook can be added by clicking on "Add" under "LDAP Phonebook" section.

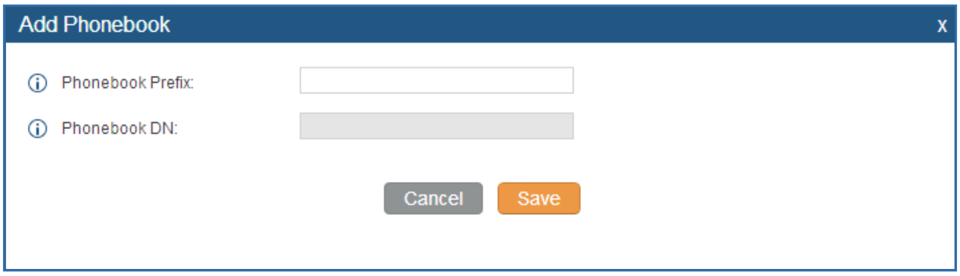


Figure 39: Add LDAP Phonebook

Configure the "Phonebook Prefix" first. The "Phonebook DN" will be automatically filled in. For example, if configuring "Phonebook Prefix" as "people", the "Phonebook DN" will be filled with "ou=people,dc=pbx,dc=com".

Once added, users can select to edit the phonebook attributes and contact list (see figure below), or select to delete the phonebook.

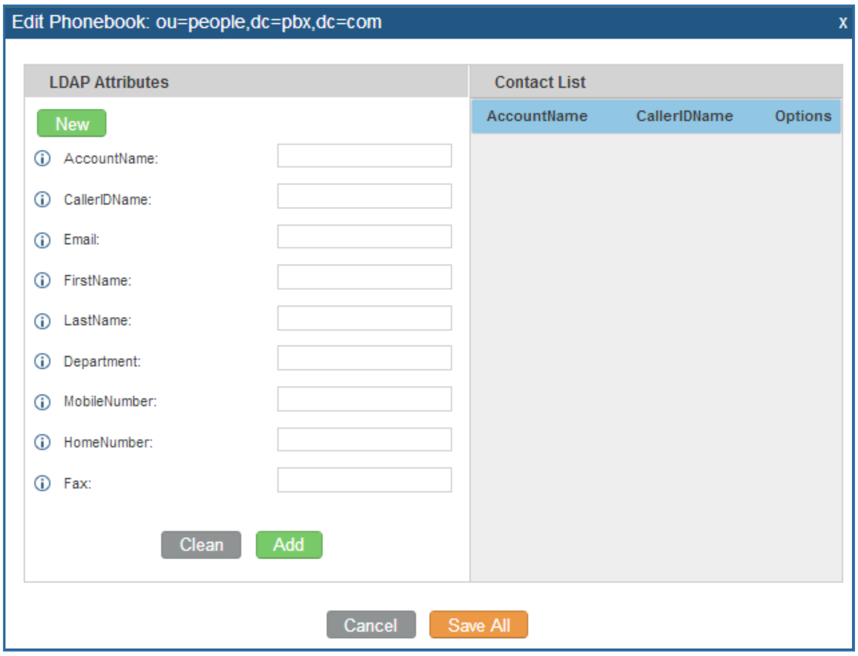


Figure 40: Edit LDAP Phonebook

- **Import phonebook from your computer to LDAP server**

Click on “Import Phonebook” and a dialog will prompt as shown in the figure below.

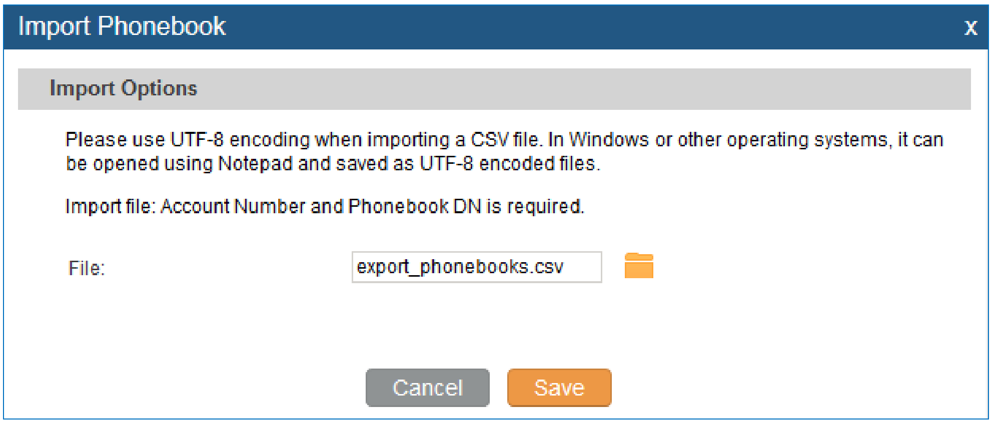


Figure 41: Import Phonebook

The file to be imported must be a CSV file with UTF-8 encoding. Users can open the CSV file with Notepad and save it with UTF-8 encoding.

Here is how a sample file looks like. Please note “Account Number” and “Phonebook DN” fields are required. Users could export a phonebook file from the UCM6200 LDAP phonebook section first and use it as a sample to start with.

	A	B	C	D	E	F	G	H	I	J
1	First Nam	Last Name	Account Number	CallerID Name	Email	Department	Mobile Number	Home Number	Fax	Phonebook DN
2	John	Doe	1001	1001		IT	1001000000			phonebook
3	Jane	Doe	1002	1002		Sales	1002000000			phonebook
4	William	Chung	1003	1003		Marketing	1003000000			phonebook
5	Linda	Kuo	1004	1004		Accounting	1004000000			phonebook
6	Steve	Chang	1005	1005		Support	1005000000			others

Figure 42: Phonebook CSV File Format

The Phonebook DN field is the same “Phonebook Prefix” entry as when the user clicks on “Add” to create a new phonebook. Therefore, if the user enters “phonebook” in “Phonebook DN” field in the CSV file, the actual phonebook DN “ou=phonebook,dc=pbx,dc=com” will be automatically created by the UCM6200 once the CSV file is imported.

In the CSV file, users can specify different phonebook DN fields for different contacts. If the phonebook DN already exists on the UCM6200 LDAP Phonebook, the contacts in the CSV file will be added into the existing phonebook. If the phonebook DN doesn’t exist on the UCM6200 LDAP Phonebook, a new phonebook with this phonebook DN will be created.

The sample phonebook CSV file in above picture will result in the following LDAP phonebook in the

UCM6200.

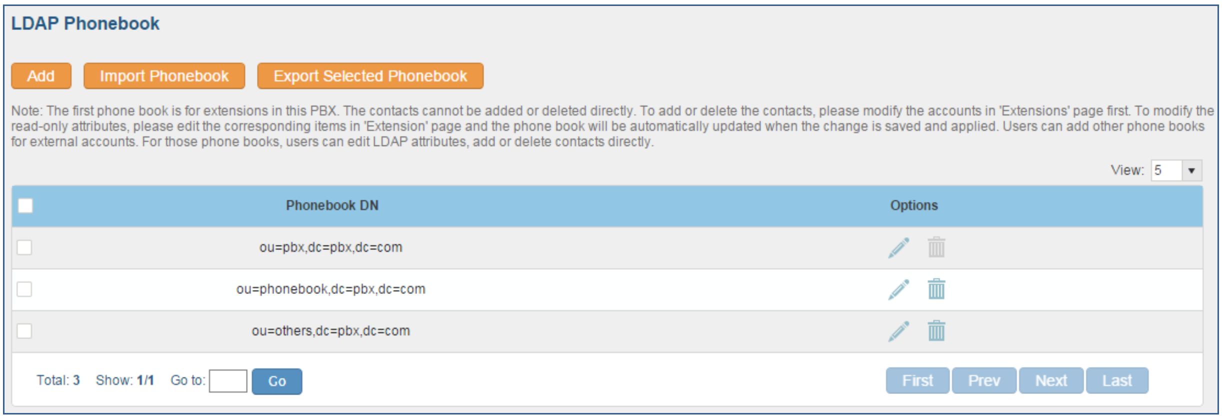


Figure 43: LDAP Phonebook After Import

As the default LDAP phonebook with DN "ou=pbx,dc=pbx,dc=com" cannot be edited or deleted in LDAP phonebook section, users cannot import contacts with Phonebook DN field "pbx" if existed in the CSV file.

- **Export phonebook to your computer from UCM6200 LDAP server**

Select the checkbox for the LDAP phonebook and then click on "Export Selected Phonebook" to export the selected phonebook. The exported phonebook can be used as a record or a sample CSV file for the users to add more contacts in it and import to the UCM6200 again.

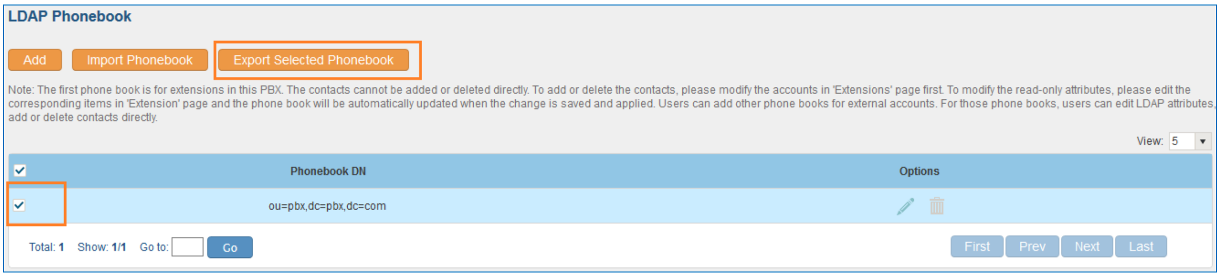


Figure 44: Export Selected LDAP Phonebook

LDAP CLIENT CONFIGURATIONS

The configuration on LDAP client is similar when you use other LDAP servers. Here we provide an example on how to configure the LDAP client on the SIP end points to use the default PBX phonebook.

Assuming the server base dn is "dc=pbx,dc=com", configure the LDAP clients as follows (case insensitive):

Server Address: LDAP server IP address

Base DN: dc=pbx,dc=com
User Name: cn= “LDAP server login name”, dc=pbx, dc=com [matching LDAP server format]
Password: “LDAP server login password”
Filter: (|(CallerIDName=%)(AccountNumber=%))
Port: 389

The following figure gives a sample configurations for UCM6200 acting as a LDAP client.

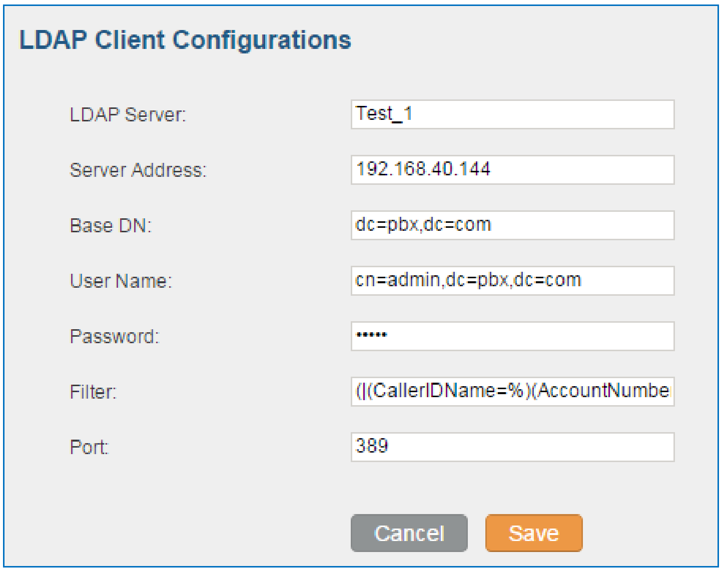


Figure 45: LDAP Client Configurations

To configure Grandstream IP phones as the LDAP client, please refer to the following example:

Server Address: The IP address or domain name of the UCM6200
Base DN: dc=pbx,dc=com
User Name: Please leave this field empty
Password: Please leave this field empty
LDAP Name Attribute: CallerIDName Email Department FirstName LastName
LDAP Number Attribute: AccountNumber MobileNumber HomeNumber Fax
LDAP Number Filter: (AccountNumber=%)
LDAP Name Filter: (CallerIDName=%)
LDAP Display Name: AccountNumber CallerIDName
LDAP Version: If existed, please select LDAP Version 3
Port: 389

The following figure shows the configuration information on a Grandstream GXP2200 to successfully use the LDAP server as configured in ***Figure 35: LDAP Server Configurations***.

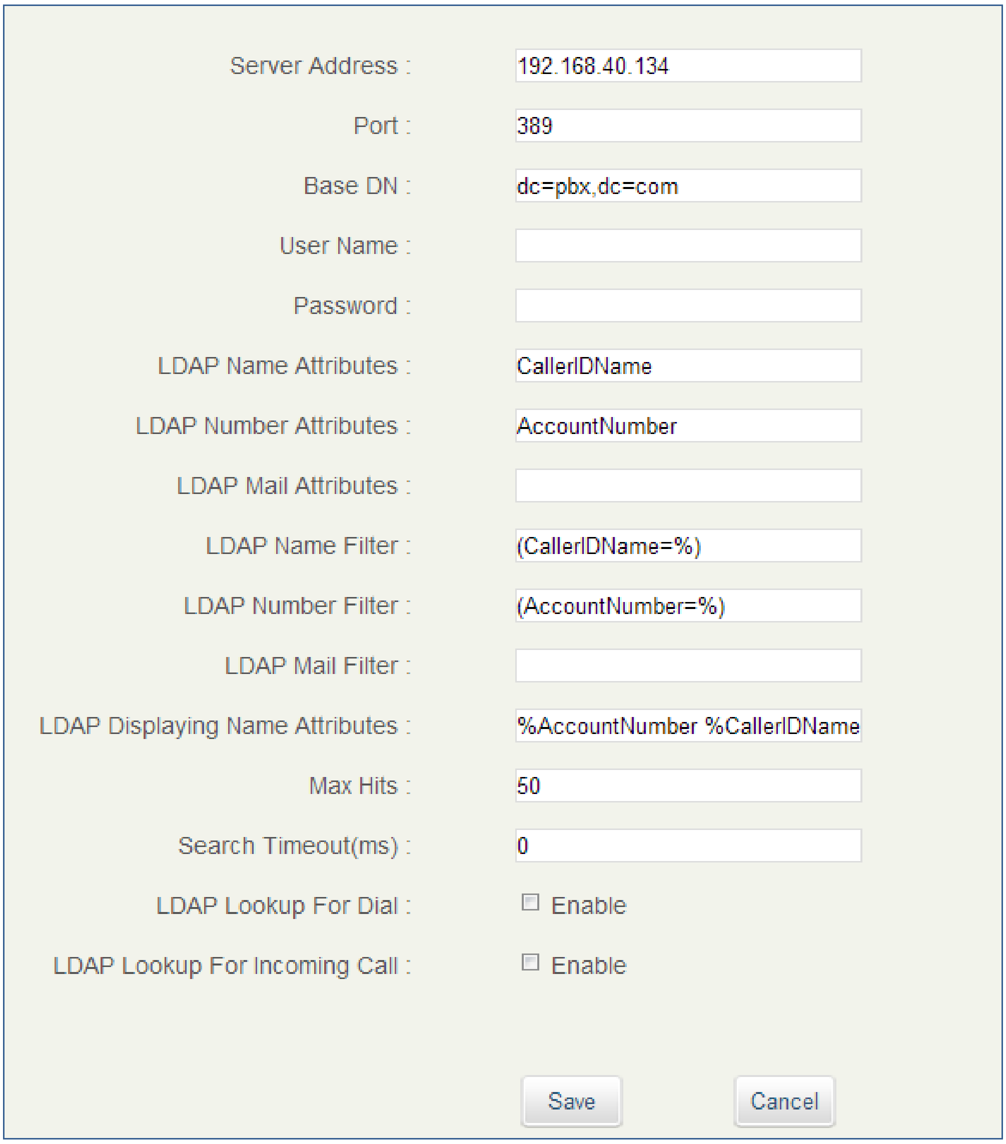


Figure 46: GXP2200 LDAP Phonebook Configuration

HTTP SERVER

The UCM6200 embedded web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow the users to configure the PBX through a Web browser such as Microsoft IE, Mozilla Firefox and Google Chrome. By default, the PBX can be accessed via HTTPS using Port 8089 (e.g., https://192.168.40.50:8089). Users could also change the access protocol and port as preferred under Web GUI->**Settings**->**HTTP Server**.

Table 19: HTTP Server Settings

Redirect From Port 80	Enable or disable redirect from port 80. On the PBX, the default access protocol is HTTPS and the default port number is 8089. When this option is enabled, the access using HTTP with Port 80 will be redirected to HTTPS with Port 8089. The default setting is "Enable".
Protocol Type	Select HTTP or HTTPS. The default setting is "HTTPS". This is also the protocol used for zero config when the end point device downloads the config file from the UCM6200.
Port	Specify port number to access the HTTP server. The default port number is 8089.

Once the change is saved, the web page will be redirected to the login page using the new URL. Enter the username and password to login again.

EMAIL SETTINGS

The Email application on the UCM6200 can be used to send out alert event Emails, Fax (Fax-To-Email), Voicemail (Voicemail-To-Email) and etc. The configuration parameters can be accessed via Web GUI->**Settings**->**Email Settings**.

Table 20: Email Settings

TLS Enable	Enable or disable TLS during transferring/submitting your Email to other SMTP server. The default setting is "Yes".
Type	Select Email type. • MTA: Mail Transfer Agent. The Email will be sent from the configured domain. When MTA is selected, there is no need to set up SMTP server for it or no user login is required. However, the Emails sent from MTA might be considered as spam by the target SMTP server. • Client: Submit Emails to the SMTP server. A SMTP server is required and users need login with correct credentials.
Domain	Specify the domain name to be used in the Email when using type "MTA".
Server	Specify the SMTP server when using type "Client".
Username	Username is required when using type "Client". Normally it's the Email address.
Password	Password to login for the above Username (Email address) is required when using type "Client".
Display Name	Specify the display name in the FROM header in the Email.
Sender	Specify the sender's Email address. For example, pbx@example.mycompany.com.

The following figure shows a sample Email settings on the UCM6200, assuming the Email is using *smtp.gmail.com* as the SMTP server.

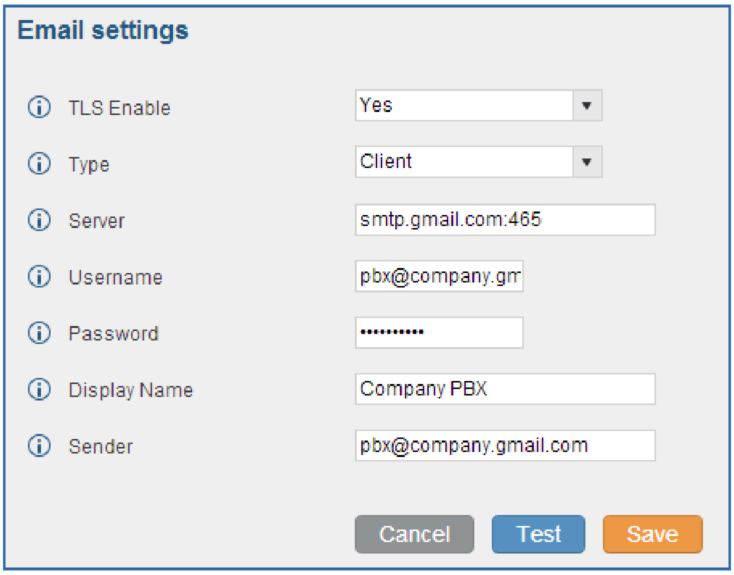


Figure 47: UCM6200 Email Settings

Once the configuration is finished, click on "Test". In the prompt, fill in a valid Email address to send a test Email to verify the Email settings on the UCM6200.

TIME SETTINGS

AUTO TIME UPDATING

The current system time on the UCM6200 is displayed on the upper right of the web page. It can also be found under Web GUI->**Status**->**System Status**->**General**.

To configure the UCM6200 to update time automatically, go to Web GUI->**Settings**->**Time Settings**->**Time Auto Updating**.

⚠ **Note:**

The configurations under Web GUI->**Settings**->**Time Settings**->**Time Auto Updating** page require reboot to take effect. Please consider configuring auto time updating related changes when setting up the UCM6200 for the first time to avoid service interrupt after installation and deployment in production.

Table 21: Time Auto Updating

Remote NTP Server	Specify the URL or IP address of the NTP server for the UCM6200 to synchronize the date and time. The default NTP server is ntp.ipvideotalk.com.
Enable DHCP Option 2	If set to "Yes", the UCM6200 is allowed to get provisioned for Time Zone from DHCP Option 2 in the local server automatically. The default setting is "Yes".
Enable DHCP Option 42	If set to "Yes", the UCM6200 is allowed to get provisioned for NTP Server from DHCP Option 42 in the local server automatically. This will override the manually configured NTP Server. The default setting is "Yes".
Time Zone	Select the proper time zone option so the UCM6200 can display correct time accordingly. If "Self-Defined Tome Zone" is selected, please specify the time zone parameters in "Self-Defined Time Zone" field as described in below option.
Self-Defined Time Zone	If "Self-Defined Time Zone" is selected in "Time Zone" option, users will need define their own time zone following the format below. The syntax is: std offset dst [offset], start [/time], end [/time] Default is set to: MTZ+6MDT+5,M4.1.0,M11.1.0 **MTZ+6MDT+5** This indicates a time zone with 6 hours offset and 1 hour ahead for DST, which is U.S central time. If it is positive (+), the local time zone is west of the Prime Meridian (A.K.A: International or Greenwich Meridian); If it is negative (-), the local time zone is east. **M4.1.0,M11.1.0** The 1st number indicates Month: 1,2,3.., 12 (for Jan, Feb, .., Dec). The 2nd number indicates the nth iteration of the weekday: (1st Sunday, 3rd Tuesday…). Normally 1, 2, 3, 4 are used. If 5 is used, it means the last iteration of the weekday. The 3rd number indicates weekday: 0,1,2,..,6 (for Sun, Mon, Tues, ... ,Sat). Therefore, this example is the DST which starts from the First Sunday of April to the 1st Sunday of November.

SET TIME MANUALLY

To manually set the time on the UCM6200, go to Web UI->**Settings**->**Time Settings**->**Set Time Manually**. The format is YYYY-MM-DD HH:MI:SS.

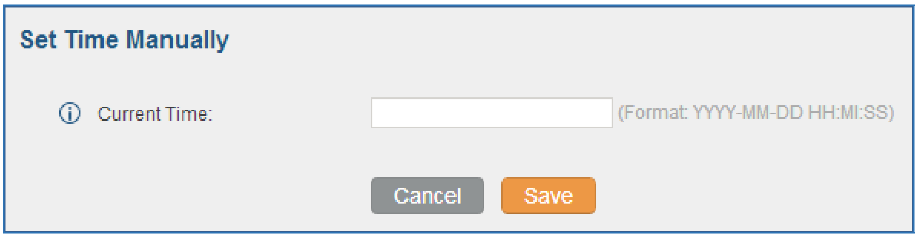


Figure 48: Set Time Manually

Note:

Manually setup time will take effect immediately after saving and applying change in the web UI. If users would like to reboot the UCM6200 and keep the manually setup time setting, please make sure "Remote NTP Server", "Enable DHCP Option 2" and "Enable DHCP Option 42" options under Web GUI->**Settings**->**Time Settings**->**Time Auto Updating** page are unchecked or set to empty. Otherwise, time auto updating settings in this page will take effect after reboot.

OFFICE TIME

On the UCM6200, the system administrator can define "office time", which can be used to configure time condition for extension call forwarding schedule and inbound rule schedule. To configure office time, go to Web UI->**Settings**->**Time Settings**->**Office Time**. Click on "Create New Office Time" to create an office time.

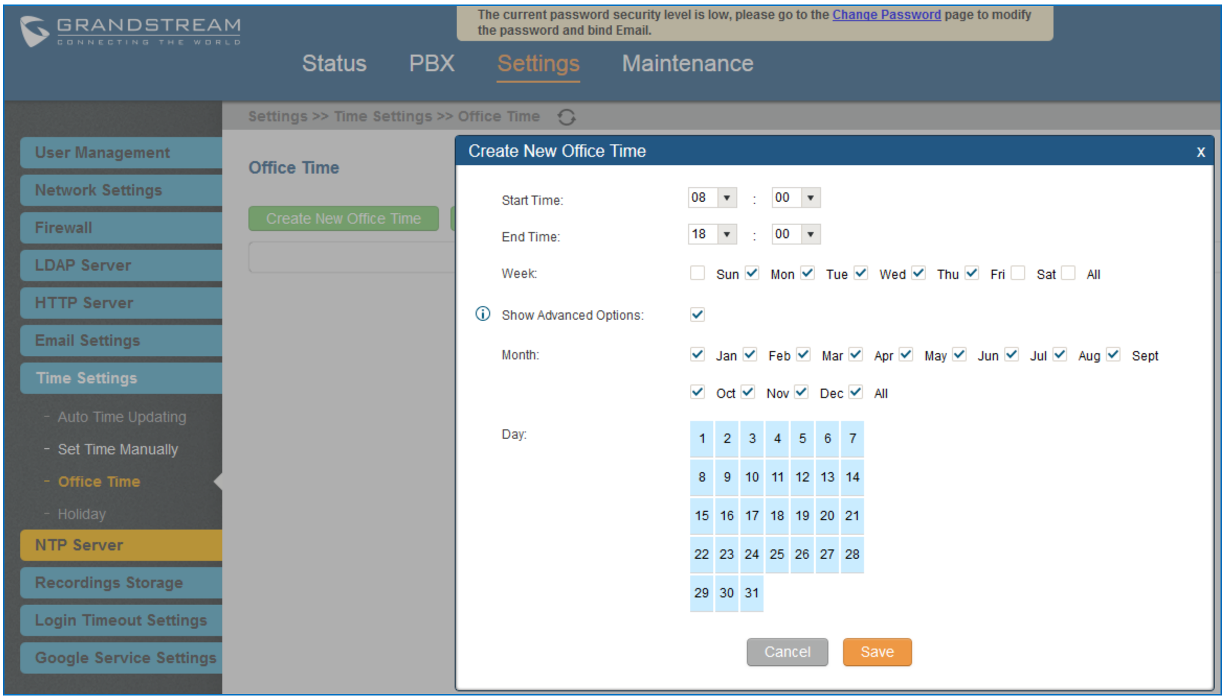


Figure 49: Create New Office Time

Table 22: Create New Office Time

Start Time	Configure the start time for office hour.
End Time	Configure the end time for office hour
Week	Select the work days in one week.
Show Advanced Options	Check this options to show advanced options. Once selected, please specify "Month" and "Day" below.
Month	Select the months for office time.
Day	Select the work days in one month.

Select "Start Time", "End Time" and the day for the "Week" for the office time. The system administrator can also define month and day of the month as advanced options. Once done, click on "Save" and then "Apply Change" for the office time to take effect. The office time will be listed in the web page as the figure shows below.

Figure 50: Settings->Time Settings->Office Time

- Click on ✎ to edit the office time.
- Click on 🗑 to delete the office time.
- Click on "Delete Selected Office Times" to delete multiple selected office times at once.

HOLIDAY

On the UCM6200, the system administrator can define "holiday", which can be used to configure time condition for extension call forwarding schedule and inbound rule schedule. To configure holiday, go to Web UI->**Settings**->**Time Settings**->**Holiday**. Click on "Create New Holiday" to create holiday time.

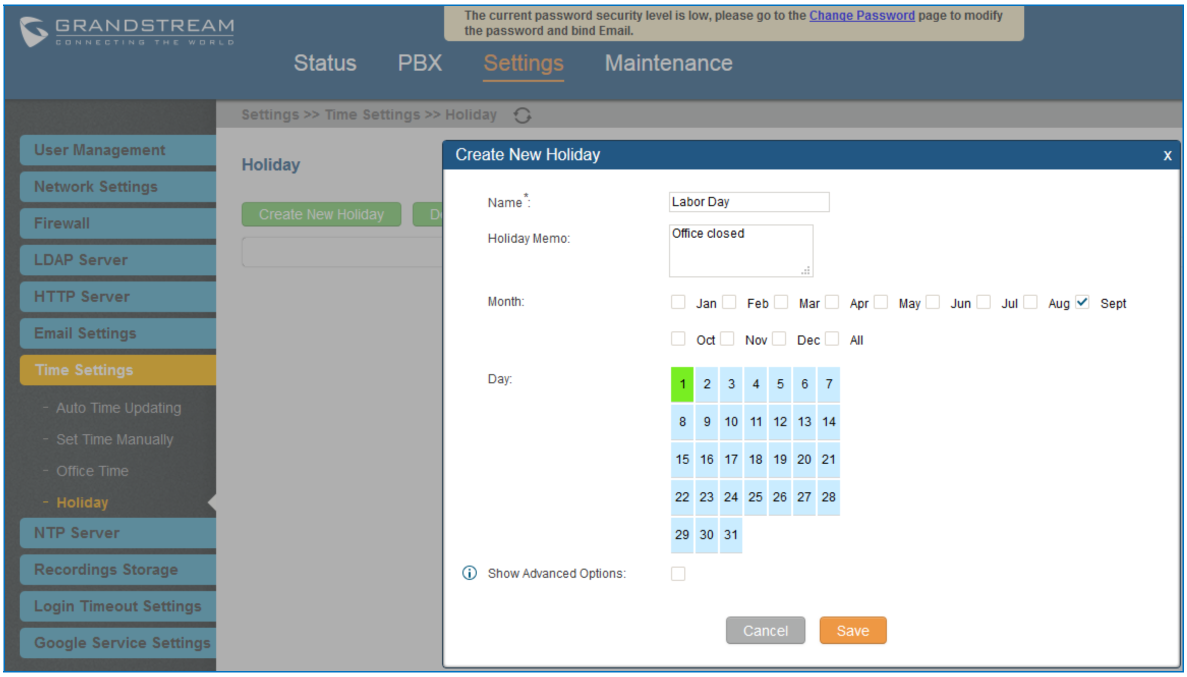


Figure 51: Create New Holiday

Table 23: Create New Holiday

Name	Specify the holiday name to identify this holiday.
Holiday Memo	Create a note for the holiday.
Month	Select the month for the holiday.
Day	Select the day for the holiday.
Show Advanced Options	Check this option to show advanced options. If selected, please specify the days as holiday in one week below.
Week	Select the days as holiday in one week.

Enter holiday "Name" and "Holiday Memo" for the new holiday. Then select "Month" and "Day". The system administrator can also define days in one week as advanced options. Once done, click on "Save" and then "Apply Change" for the holiday to take effect. The holiday will be listed in the web page as the figure shows below.

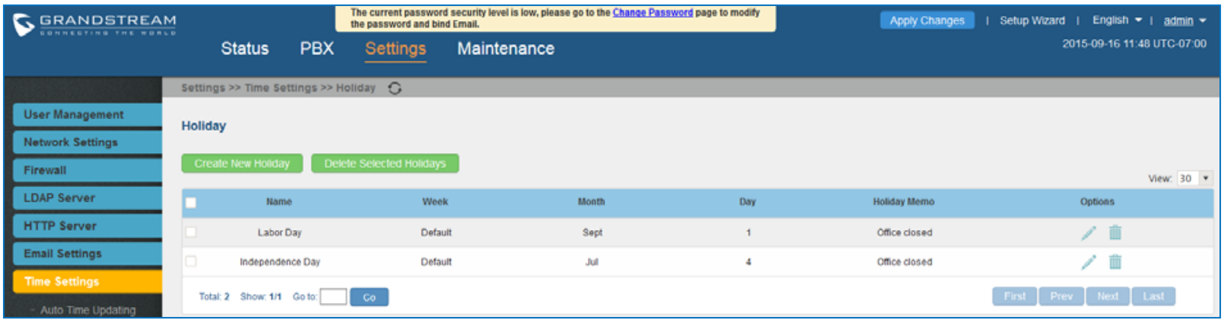


Figure 52: Settings->Time Settings->Holiday

- Click on to edit the holiday.
- Click on to delete the holiday.
- Click on "Delete Selected Holidays" to delete multiple selected holidays at once.

⚠ **Note:**

For more details on how to use office time and holiday, please refer to the link below:
http://www.grandstream.com/sites/default/files/Resources/How_to_use_office_time_and_holiday_UCM6200.pdf

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NTP SERVER

The UCM6200 can be used as a NTP server for the NTP clients to synchronize their time with. To configure the UCM6200 as the NTP server, set "Enable NTP server" to "Yes" under web GUI->**Settings**->**NTP Server**. On the client side, point the NTP server address to the UCM6200 IP address or host name to use the UCM6200 as the NTP server.

RECORDINGS STORAGE

The UCM6200 supports call recordings automatically or manually and the recording files can be saved in external storage plugged in the UCM6200 or on the UCM6200 locally. To manage the recording storage, users can go to UCM6200 web GUI->**Settings**->**Recordings Storage** page and select whether to store the recording files in USB Disk, SD card or locally on the UCM6200.

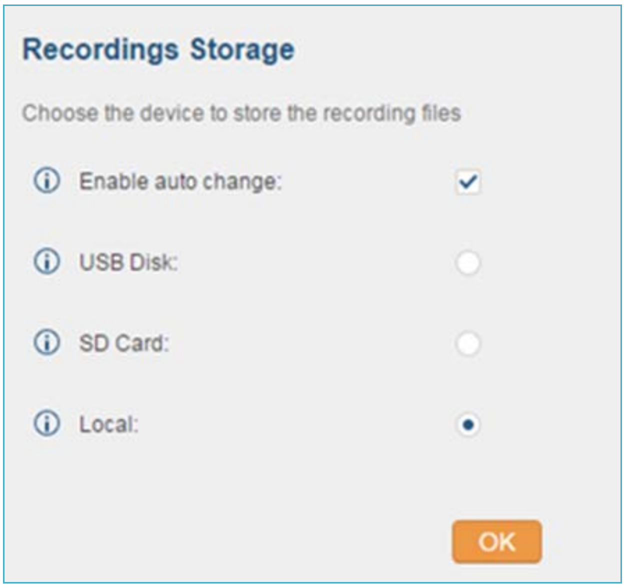


Figure 53: Settings->Recordings Storage

- If “Enable Auto Change” is selected, the recording files will be automatically saved in the available USB Disk or SD card plugged into the UCM6200. If both USB Disk and SD card are plugged in, the recording files will be always saved in the USB Disk.
- If “Local” is selected, the recordings will be stored in UCM6200 internal storage.
- If “USB Disk” or “SD Card” is selected, the recordings will be stored in the corresponding plugged in external storage device. Please note the options “USB Disk” and “SD Card” will be displayed only if they are plugged into the UCM6200.

Once “USB Disk” or “SD Card” is selected, click on “OK”. The user will be prompted to confirm to copy the local files to the external storage device.

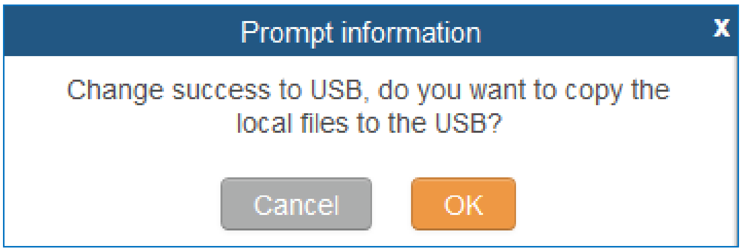


Figure 54: Recordings Storage Prompt Information

Click on “OK” to continue. The users will be prompted a new dialog to select the categories for the files to be copied over.

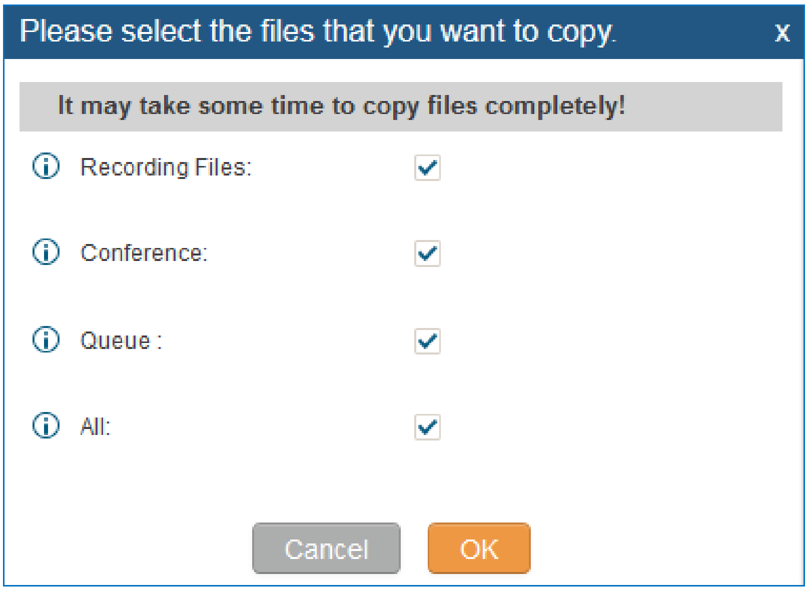


Figure 55: Recording Storage Category

On the UCM6200, recording files are generated and exist in 3 categories: normal call recording files, conference recording files, and call queue recording files. Therefore users have the following options when select the categories to copy the files to the external device:

- Recording Files: Copy the normal recording files to the external device.
- Conference: Copy the conference recording files to the external device.
- Queue: Copy the call queue recording files to the external device.
- All: Copy all recording files to the external device.

LOGIN TIMEOUT SETTINGS

After the user logs in the UCM6200 web UI, the user will be automatically logged out after certain timeout. This timeout value can be specified under UCM100 web GUI->**Settings**->**Login Timeout Settings** page.

The “User Login Timeout” value is in minute and the default setting is 10 minutes. If the user doesn’t make any operation on web UI within the timeout, the user will be logged out automatically. After that, the web UI will be redirected to the login page and the user will need to enter username and password to log in.

If set to 0, there is no timeout for the web UI login session and the user will not be automatically logged out.

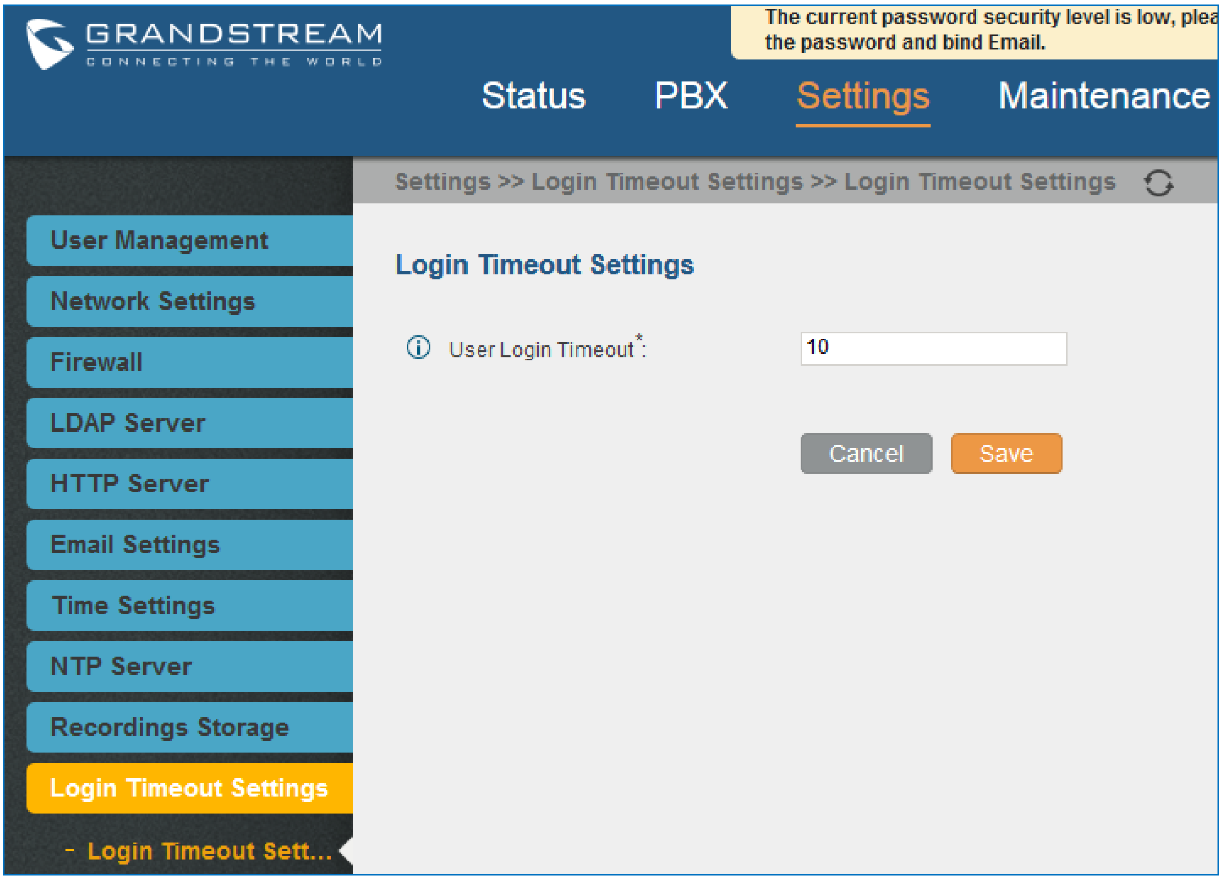


Figure 56: Login Timeout Settings

GOOGLE SERVICE SETTINGS SUPPORT

UCM6200 now supports Google OAuth 2.0 authentication. This feature is used for supporting UCM6200 conference scheduling system. Once OAuth 2.0 is enabled, UCM6200 conference system can access Google calendar to schedule or update conference.

Google Service Settings can be found under web GUI**-> Settings-> Google Service Settings-> Google Service Settings**.

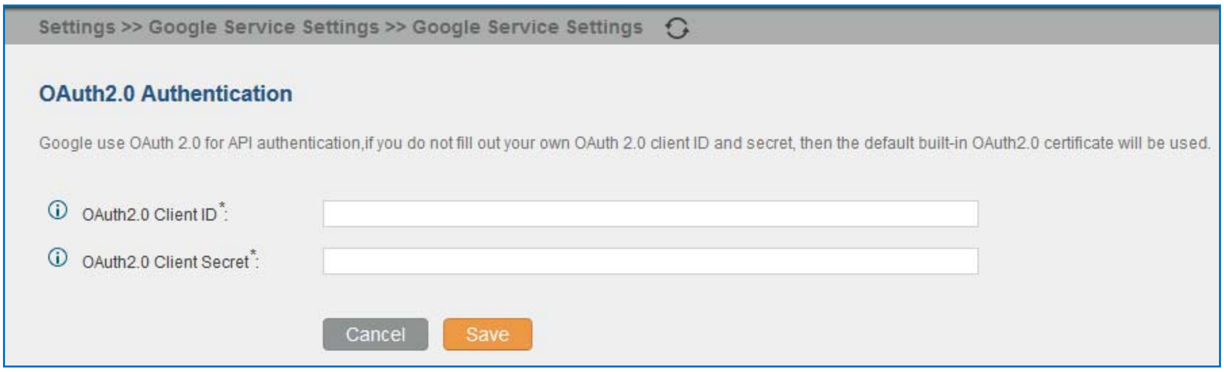


Figure 57: Google Service Settings->OAuth2.0 Authentication

If you already have OAuth2.0 project set up on **Google Developers** web page, please use your existing login credential for "OAuth2.0 Client ID" and "OAuth2.0 Client Secret" in the above figure for the UCM6200 to access Google Service.

If you do not have OAuth2.0 project set up yet, please following the steps below to create new project and obtain credentials:

1. Go to Google Developers page https://console.developers.google.com/start Create a New Project in Google Developers page.

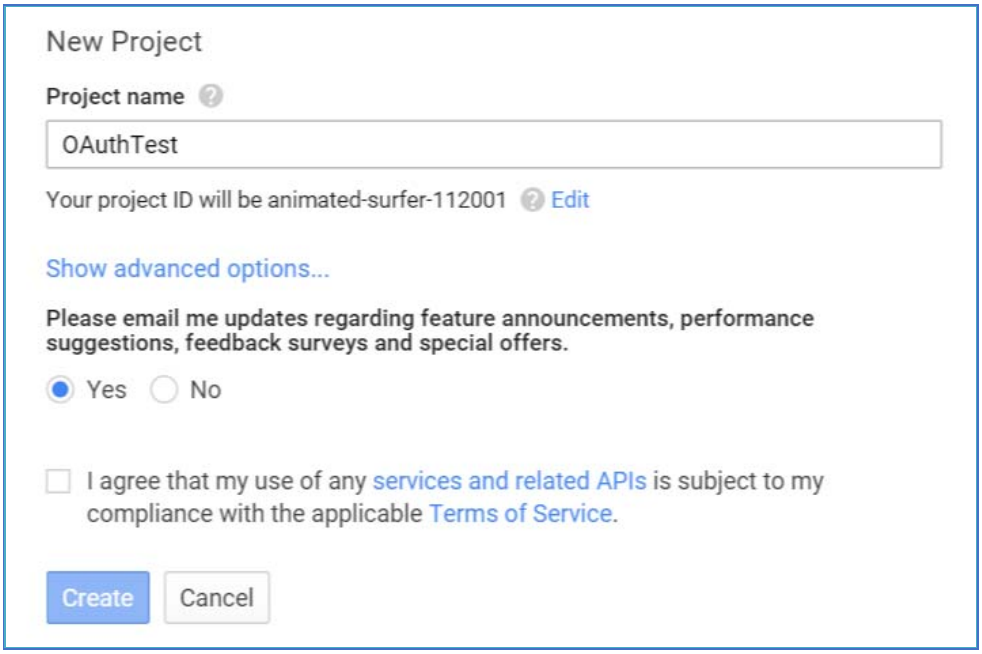


Figure 58: Google Service->New Project

2. Enable Calendar API from API Library.
3. Click "Credentials" on the left drop down menu to create new OAuth2.0 login credentials.

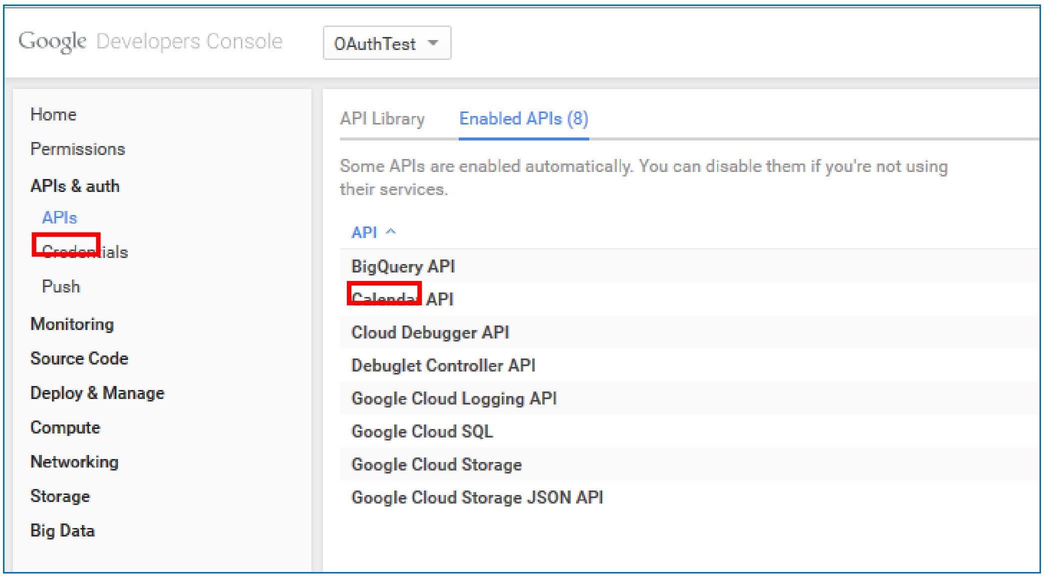


Figure 59: Google Service->Create New Credential

4. Use the newly created login credential to fill in "OAuth2.0 Client ID" and "OAuth2.0 Client Secret".
5. Click "Get Authentication Code" to obtain authentication code from Google Service.

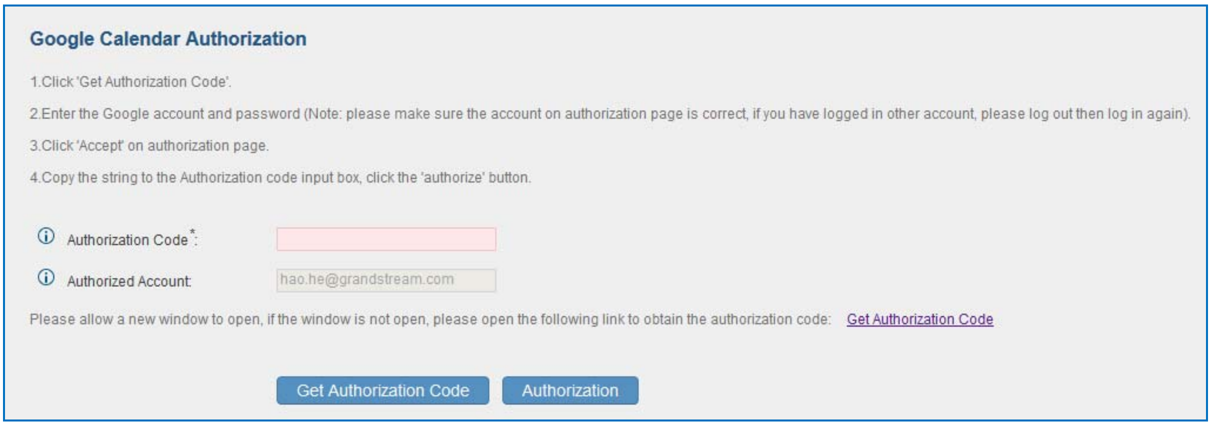


Figure 60: Google Service->OAuth2.0 Login

6. Now UCM6200 is connected with Google Service.

PROVISIONING

OVERVIEW

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 a proprietary binary format configuration file and 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 within LAN area. 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, configuration and provisioning. This section explains how Zero Config works on the UCM6200. The settings for this feature can be accessed via Web UI->**PBX**->**Zero Config**.

CONFIGURATION ARCHITECTURE FOR END POINT DEVICE

Started from firmware version 1.0.7.10, the end point device configuration in zero config is divided into the following three layers with priority from the lowest to the highest:

- Global
  This is the lowest layer. Users can configure the most basic options that could apply to all Grandstream SIP devices during provisioning via Zero config.

- Model
  In this layer, users can define model-specific options for the configuration template.

- Device
  This is the highest layer. Users can configure device-specific options for the configuration for individual device here.

Each layer also has its own structure in different levels. Please see figure below. The details for each layer are explained in sections ***[GLOBAL CONFIGURATION]***, ***[MODEL CONFIGURATION]*** and ***[DEVICE CONFIGURATION]***.

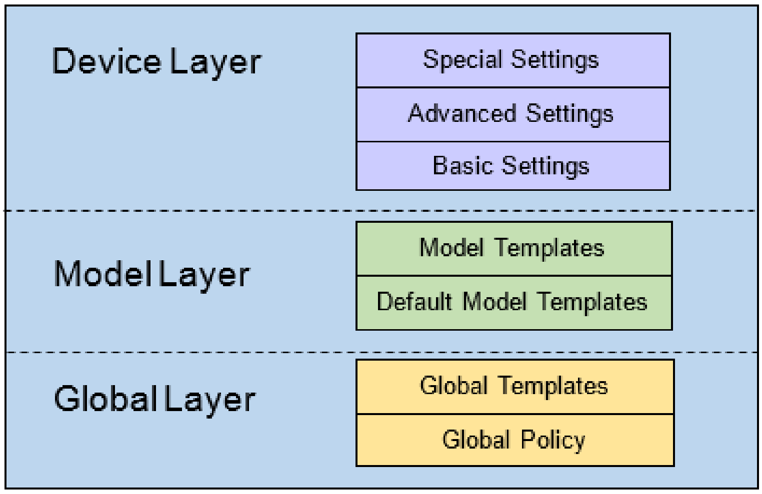


Figure 61: Zero Config Configuration Architecture for End Point Device

The configuration options in model layer and device layer have all the option in global layers already, i.e., the options in global layer is a subset of the options in model layer and device layer. If an option is set in all three layers with different values, the highest layer value will override the value in lower layer. For example, if the user selects English for Language setting in Global Policy and Spanish for Language setting in Default Model Template, the language setting on the device to be provisioned will use Spanish as model layer has higher priority than global layer. To sum up, **configurations in higher layer will always override the configurations for the same options/fields in the lower layer when presented at the same time.**

After understanding the zero config configuration architecture, users could configure the available options for end point devices to be provisioned by the UCM6200 by going through the three layers. This configuration architecture allows users to set up and manage the Grandstream end point devices in the same LAN area in a centralized way.

AUTO PROVISIONING SETTINGS

By default, the Zero Config feature is enabled on the UCM6200 for auto provisioning. Three methods of auto provisioning are used.

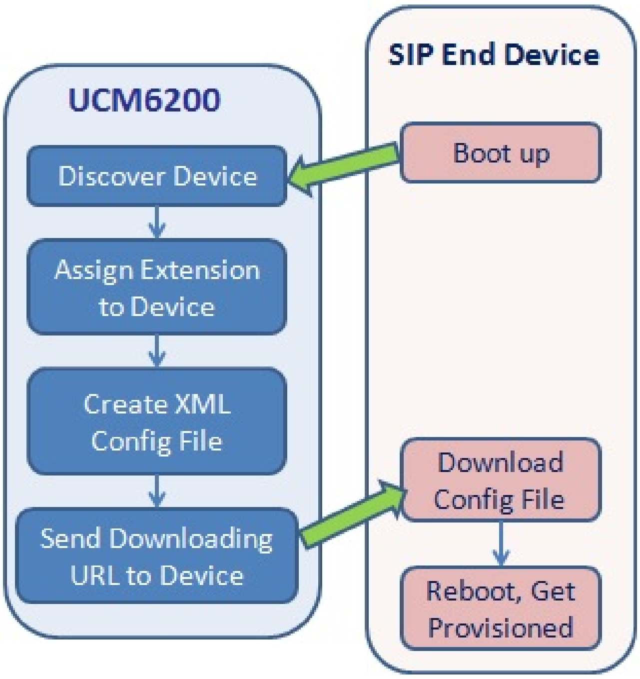


Figure 62: UCM6200 Zero Config

- **SIP SUBSCRIBE**

  When the phone boots up, it sends out SUBSCRIBE to a multicast IP address in the LAN. The UCM6200 discovers it and then sends a NOTIFY with the XML config file URL in the message body. The phone will then use the path to download the config file generated in the UCM6200 and take the new configuration.

- **DHCP OPTION 66**

  This method should be used on the UCM6202/UCM6204 because only the UCM6202/UCM6204 have WAN and LAN port with LAN port supporting the router function. When the phone restarts (by default DHCP Option 66 is turned on), it will send out a DHCP DISCOVER request. The UCM6202 receives it and returns DHCP OFFER with the config server path URL in Option 66, for example, https://192.168.2.1:8089/zccgi/. The phone will then use the path to download the config file generated in the UCM6200.

- **mDNS**

When the phone boots up, it sends out mDNS query to get the TFTP server address. The UCM6200 will respond with its own address. The phone will then send TFTP request to download the XML config file from the UCM6200.

To start the auto provisioning process, under Web GUI->**PBX**->**Zero Config**->**Zero Config Settings**, fill in the auto provision information.

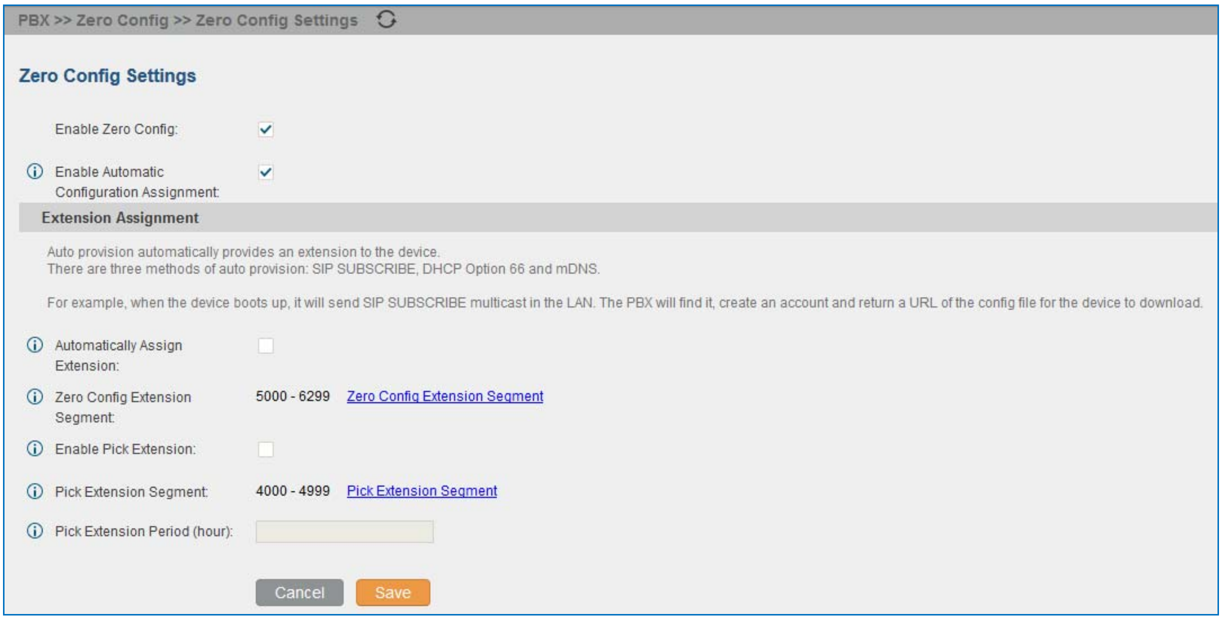


Figure 63: Auto Provision Settings

Table 24: Auto Provision Settings

Enable Zero Config	Enable or disable the zero config feature on the PBX. The default setting is enabled.
Enable Automatic Configuration Assignment	By default, this is disabled. If disabled, when SIP device boots up, the UCM6200 will not send the SIP device the URL to download the config file and therefore the SIP device will not be automatically provisioned by the UCM6200. **Note:** When disabled, SIP devices can still be provisioned by manually sending NOTIFY from the UCM6200 which will include the XML config file URL for the SIP device to download.
Automatically Assign Extension	If enabled, when the device is discovered, the PBX will automatically assign an extension within the range defined in "Zero Config Extension Segment" to the device. The default setting is disabled.
Zero Config Extension	Click on the link "Zero Config Extension Segment" to specify the

Segment	extension range to be assigned if "Automatically Assign Extension" is enabled. The default range is 5000-6299. Zero Config Extension Segment range can be defined in web UI->**PBX**->**Internal Options**->**General** page->Extension Preference section: "Auto Provision Extensions".
Enable Pick Extension	If enabled, the extension list will be sent out to the device after receiving the device's request. This feature is for the GXP series phones that support selecting extension to be provisioned via phone's LCD. The default setting is disabled.
Pick Extension Segment	Click on the link "Pick Extension Segment" to specify the extension list to be sent to the device. The default range is 4000 to 4999. Pick Extension Segment range can be defined in web UI->**PBX**->**Internal Options**->**General** page->Extension Preference section: "Pick Extensions".
Pick Extension Period (hour):	Specify the number of minutes to allow the phones being provisioned to pick extensions.

Please make sure an extension is manually assigned to the phone or "Automatically Assign Extension" is enabled during provisioning. After the configuration on the UCM6200 web GUI, click on "Save" and "Apply Changes". Once the phone boots up and picks up the config file from the UCM6200, it will take the configuration right away.

DISCOVERY

Users could manually discover the device by specifying the IP address or scanning the entire LAN network. Three methods are supported to scan the devices.

- PING
- ARP
- SIP Message (NOTIFY)

Click on "Auto Discover" under web **UI-> PBX-> Zero Config->Zero Config**, fill in the "Scan Method" and "Scan IP". The IP address segment will be automatically filled in based on the network mask detected on the UCM6200. If users need scan the entire network segment, enter 255 (for example, 192.168.40.255) instead of a specific IP address. Then click on "Save" to start discovering the devices within the same network. To successfully discover the devices, "Zero Config" needs to be enabled on the UCM6200 web GUI->**PBX**->**Zero Config**->**Auto Provisioning Settings**.

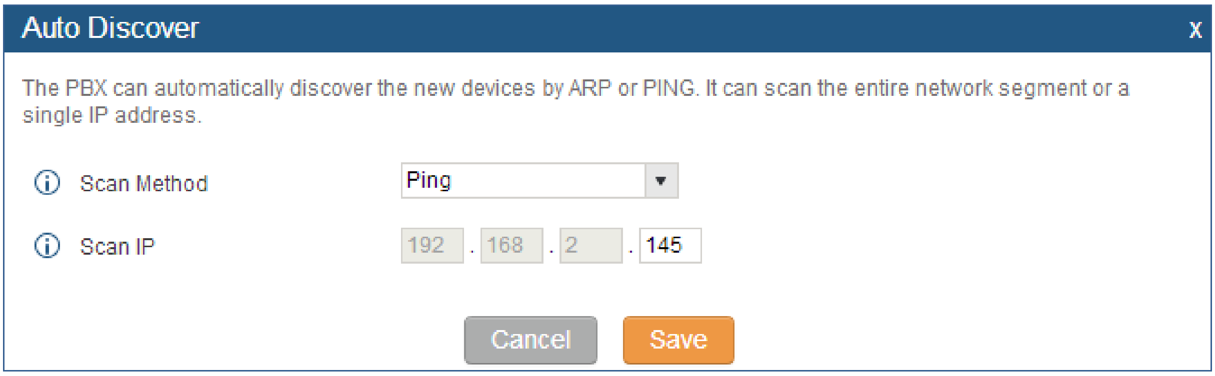
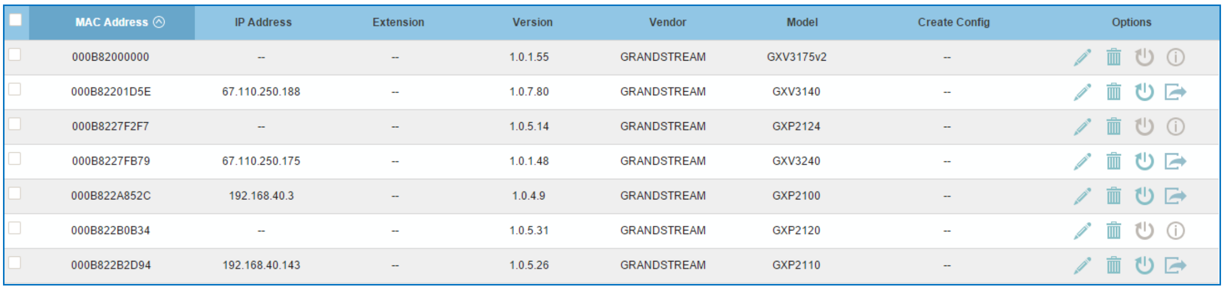


Figure 64: Auto Discover

The following figure shows a list of discovered phones. The MAC address, IP Address, Extension (if assigned), Version, Vendor, Model, Connection Status, Create Config, Options (Edit/Delete/Update) are displayed in the list.



MAC Address	IP Address	Extension	Version	Vendor	Model	Create Config	Options
000B82000000	--	--	1.0.1.55	GRANDSTREAM	GXV3175v2	--	
000B82201D5E	67.110.250.188	--	1.0.7.80	GRANDSTREAM	GXV3140	--	
000B8227F2F7	--	--	1.0.5.14	GRANDSTREAM	GXP2124	--	
000B8227FB79	67.110.250.175	--	1.0.1.48	GRANDSTREAM	GXV3240	--	
000B822A852C	192.168.40.3	--	1.0.4.9	GRANDSTREAM	GXP2100	--	
000B822B0B34	--	--	1.0.5.31	GRANDSTREAM	GXP2120	--	
000B822B2D94	192.168.40.143	--	1.0.5.26	GRANDSTREAM	GXP2110	--	

Figure 65: Discovered Devices

GLOBAL CONFIGURATION

GLOBAL POLICY

Global configuration will apply to all the connected Grandstream SIP end point devices in the same LAN with the UCM6200 no matter what the Grandstream device model it is. It is divided into two levels:

- Web UI->**PBX->Zero Config->Global Policy**
- Web UI->**PBX->Zero Config->Global Templates**.
- **Global Templates** configuration has higher priority to **Global Policy** configuration.

Global Policy can be accessed in web GUI->**PBX->Zero Config->Global Policy** page. On the top of the configuration table, users can select category in the "Options" dropdown list to quickly navigate to the category. The categories are:

- **Localization**: configure display language, data and time.
- **Phone Settings**: configure dial plan, call features, NAT, call progress tones and etc.
- **Contact List**: configure LDAP and XML phonebook download.
- **Maintenance**: configure upgrading, web access, Telnet/SSH access and syslog.
- **Network Settings**: configure IP address, QoS and STUN settings.
- **Customization**: customize LCD screen wallpaper for the supported models.

Select the checkbox on the left of the parameter you would like to configure to active the dropdown list for this parameter.

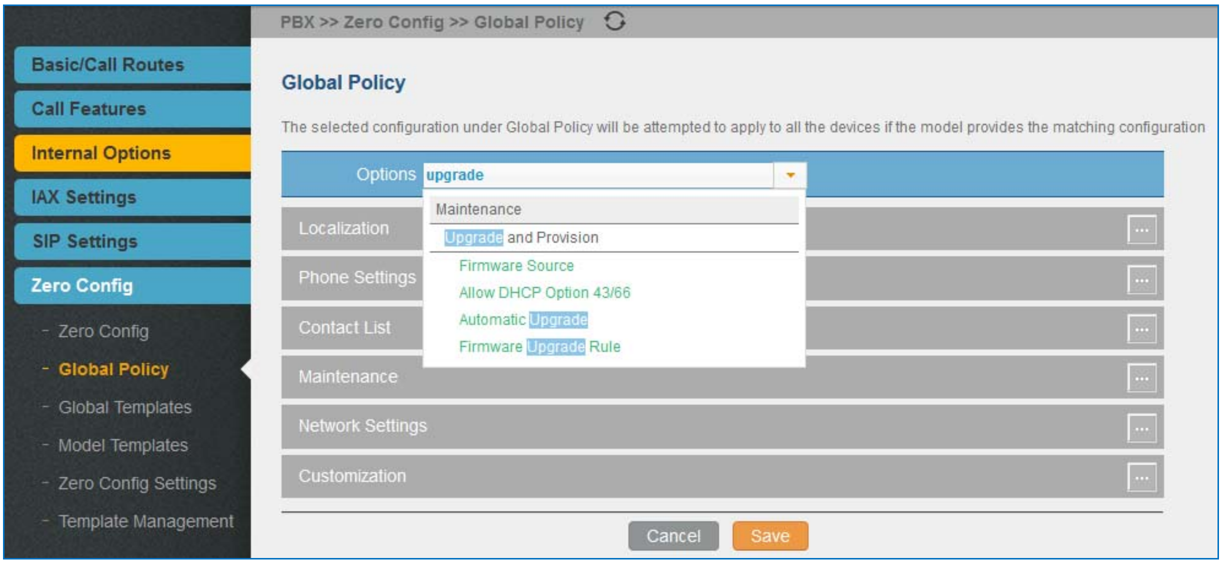


Figure 66: Global Policy Categories

The following tables list the Global Policy configuration parameters for the SIP end device.

Table 25: Global Policy Parameters->Localization

Language settings	
Language	Select the LCD display language on the SIP end device.
Date and Time	
Date Format	Configure the date display format on the SIP end device's LCD.
Time Format	Configure the time display in 12-hour or 24-hour format on the SIP end device's LCD.
NTP Server	Configure the URL or IP address of the NTP server. The SIP end device may obtain the date and time from the server.
Time Zone	Configure the time zone used on the SIP end device.

Table 26: Global Policy Parameters->Phone Settings

Default Call Settings	
Dial Plan	Configure the default dial plan rule. For syntax and examples, please refer to user manual of the SIP devices to be provisioned for more details.
Enable Call Features	When enabled, "Do Not Disturb", "Call Forward" and other call features can be used via the local feature code on the phone. Otherwise, the ITSP feature code will be used.
Use # as Dial Key	If set to "Yes", pressing the number key "#" will immediately dial out the input digits.
Auto Answer by Call-info	If set to "Yes", the phone will automatically turn on the speaker phone to answer incoming calls after a short reminding beep, based on the SIP Call-Info header sent from the server/proxy. The default setting is enabled.
NAT Traversal	Configure which NAT traversal mechanism will be enabled on the endpoint device. If set to "STUN" and STUN server is configured, the phone system will periodically send STUN message to the SUTN server to get the public IP address of its NAT environment and keep the NAT port open. STUN will not work if the NAT is symmetric type. If set to "Keep-alive", the phone system will send the STUN packets to maintain the connection that is first established during registration of the phone. The "Keep-alive" packets will fool the NAT device into keeping the connection open and this allows the host server to send SIP requests directly to the registered phone. If it needs to use OpenVPN to connect host server, it needs to set it to "VPN". If the firewall and the SIP device behind the firewall are both able to use UPNP, it can be set to "UPNP". The both parties will negotiate to use which port to allow SIP through. The default setting is "Keep-alive".
Use Random Port	Configure whether to allow the endpoint device to use random ports for both SIP and RTP messages. This is usually necessary when multiple phones are behind the same full cone NAT. The default setting is "No". Note: This parameter must be set to "No" for Direct IP Calling to work.
General Settings	
Call Progress Tones	Configure call progress tones including ring tone, dial tone, second dial tone, message waiting tone, ring back tone, call waiting tone, busy tone and reorder tone using the following syntax: f1=val, f2=val[, c=on1/ off1[- on2/ off2[- on3/ off3]]];

	• Frequencies are in Hz and cadence on and off are in 10ms). • “on” is the period (in ms) of ringing while “off” is the period of silence. Up to three cadences are supported. • Please refer to user manual of the SIP devices to be provisioned for more details
HEADSET Key Mode	Select “Default Mode” or “Toggle Headset/Speaker” for the Headset key. Please refer to user manual of the SIP devices to be provisioned for more details.

Table 27: Global Policy Parameters->Contact List

LDAP Phonebook	
Source	Select "Manual" or "PBX" as the LDAP configuration source. • If "Manual" is selected, the LDAP configuration below will be applied to the SIP end device. • If "PBX" is selected, the LDAP configuration built-in from UCM6200 web UI->**Settings**->**LDAP Server** will be applied.
Address	Configure the IP address or DNS name of the LDAP server.
Port	Configure the LDAP server port. The default value is 389.
Base DN	This is the location in the directory where the search is requested to begin. Example: • dc=grandstream, dc=com • ou=Boston, dc=grandstream, dc=com
User Name	Configure the bind "Username" for querying LDAP servers. The field can be left blank if the LDAP server allows anonymous binds.
Password	Configure the bind "Password" for querying LDAP servers. The field can be left blank if the LDAP server allows anonymous binds.
Number Filter	Configure the filter used for number lookups. Please refer to user manual for more details.
Name Filter	Configure the filter used for name lookups. Please refer to user manual for more details.
Version	Select the protocol version for the phone to send the bind requests. The default value is 3.
Name Attribute	Specify the "name" attributes of each record which are returned in the LDAP search result. Example: • gn • cn sn description
Number Attribute	Specify the "number" attributes of each record which are returned in the

	LDAP search result. Example: • telephoneNumber • telephoneNumber Mobile
Display Name	Configure the entry information to be shown on phone's LCD. Up to 3 fields can be displayed. Example: • %cn %sn %telephoneNumber
Max Hits	Specify the maximum number of results to be returned by the LDAP server. Valid range is 1 to 3000. The default value is 50.
Search Timeout	Specify the interval (in seconds) for the server to process the request and client waits for server to return. Valid range is 0 to 180. The default value is 30.
Sort Results	Specify whether the searching result is sorted or not. The default setting is No.
Incoming Calls	Configure to enable LDAP number searching when receiving calls. The default setting is No.
Outgoing Calls	Configure to enable LDAP number searching when making calls. The default setting is No.
Lookup Display Name	Configures the display name when LDAP looks up the name for incoming call or outgoing call. It must be a subset of the LDAP Name Attributes.
XML Phonebook	
Phonebook XML Server	Select the source of the phonebook XML server. • Disable Disable phonebook XML downloading. • Manual Once selected, users need specify downloading protocol HTTP, HTTPS or TFTP and the server path to download the phonebook XML file. The server path could be IP address or URL, with up to 256 characters. • Local UCM Server Once selected, click on the Server Path field to upload the phonebook XML file. Please note: after uploading the phonebook XML file to the server, the original file name will be used as the directory name and the file will be renamed as phonebook.xml under that directory.
Phonebook Download Interval	Configure the phonebook download interval (in Minute). If set to 0, automatic download will be disabled. Valid range is 5 to 720.
Remove manually-edited entries on download	If set to "Yes", when XML phonebook is downloaded, the entries added manually will be automatically removed.

Table 28: Global Policy Parameters->Maintenance

Upgrade and Provision	
Firmware Source	Firmware source via ZeroConfig provisioning could a URL for external server address, local UCM directory or USB media if plugged in to the UCM6200. Select a source to get the firmware file: • URL If select to use URL to upgrade, complete the configuration for the following four parameters: "Upgrade Via", "Server Path", "File Prefix" and "File Postfix". • Local UCM Server Firmware can be uploaded to the UCM6200 internal storage for firmware upgrade. If selected, click on "Manage Storage" icon next to "Directory" option, upload firmware file and select directory for the end device to retrieve the firmware file. • Local USB Media If selected, the USB storage device needs to be plugged into the UCM6200 and the firmware file must be put under a folder named "ZC_firmware" in the USB storage root directory. • Local SD Card Media If selected, an SD card needs to be plugged into the UCM6200 and the firmware file must be put under a folder named "ZC_firmware" in the USB storage root directory.
Upgrade via	When URL is selected as firmware source, configure upgrade via TFTP, HTTP or HTTPS.
Server Path	When URL is selected as firmware source, configure the firmware upgrading server path.
File Prefix	When URL is selected as firmware source, configure the firmware file prefix. If configured, only the firmware with the matching encrypted prefix will be downloaded and flashed into the phone, if URL is selected as firmware source.
File Postfix	When URL is selected as firmware source, configure the firmware file postfix. If configured, only the configuration file with the matching encrypted postfix will be downloaded and flashed into the phone.
Allow DHCP Option 43/66	If DHCP option 43 or 66 is enabled on the LAN side, the TFTP server can be redirected.
Automatic Upgrade	If enabled, the endpoint device will automatically upgrade if a new firmware is detected. Users can select automatic upgrading by day, by week or by minute.

	• By week Once selected, specify the day of the week to check HTTP/TFTP server for firmware upgrades or configuration files changes. • By day Once selected, specify the hour of the day to check the HTTP/TFTP server for firmware upgrades or configuration files changes. • By minute Once selected, specify the interval **X** that the SIP end device will request for new firmware every **X** minutes.
Firmware Upgrade Rule	Specify how firmware upgrading and provisioning request to be sent.
Web Access	
Admin Password	Configure the administrator password for admin level login.
End-User Password	Configure the end-user password for the end user level login.
Web Access Mode	Select HTTP or HTTPS as the web access protocol.
Web Server Port	Configure the port for web access. The valid range is 1 to 65535.
Security	
Disable Telnet/SSH	Enable Telnet/SSH access for the SIP end device. If the SIP end device supports Telnet access, this option controls the Telnet access of the device; if the SIP end device supports SSH access, this option controls the SSH access of the device.
Syslog	
Syslog Server	Configure the URL/IP address for the syslog server.
Syslog Level	Select the level of logging for syslog.
Send SIP Log	Configure whether the SIP log will be included in the syslog message.

Table 29: Global Policy Parameters->Network Settings

Basic Settings	
IP Address	Configure how the SIP end device shall obtain the IP address. DHCP or PPPoE can be selected. • DHCP Once selected, users can specify the Host Name (option 12) of the SIP end device as DHCP client, and Vendor Class ID (option 60) used by the client and server to exchange vendor class ID information. • PPPoE Once selected, users need specify the Account ID, Password and Service Name for PPPoE.
Advanced Setting	

Layer 3 QoS	Define the Layer 3 QoS parameter. This value is used for IP Precedence, Diff-Serv or MPLS. Valid range is 0-63.
Layer 2 QoS Tag	Assign the VLAN Tag of the Layer 2 QoS packets. Valid range is 0 -4095.
Layer 2 QoS Priority Value	Assign the priority value of the Layer 2 QoS packets. Valid range is 0-7.
STUN Server	Configure the IP address or Domain name of the STUN server. Only non-symmetric NAT routers work with STUN.
Keep Alive Interval	Specify how often the phone will send a blank UDP packet to the SIP server in order to keep the "ping hole" on the NAT router to open. Valid range is 10-160.

Table 30: Global Policy Parameters->Customization

Wallpaper	
Screen Resolution 1024 x 600	Check this option if the SIP end device shall use 1024 x 600 resolution for the LCD screen wallpaper. • Source Configure the location where wallpapers are stored. • File If "URL" is selected as source, specify the URL of the wallpaper file. If "Local UCM Server" is selected as source, click to upload wallpaper file to the UCM6200.
Screen Resolution 800 x 400	Check this option if the SIP end device shall use 800 x 400 resolution for the LCD screen wallpaper. • Source Configure the location where wallpapers are stored. • File If "URL" is selected as source, specify the URL of the wallpaper file. If "Local UCM Server" is selected as source, click to upload wallpaper file to the UCM6200.
Screen Resolution 480 x 272	Check this option if the SIP end device shall use 480 x 272 resolution for the LCD screen wallpaper. • Source Configure the location where wallpapers are stored. • File If "URL" is selected as source, specify the URL of the wallpaper file. If "Local UCM Server" is selected as source, click to upload wallpaper file to the UCM6200.
Screen Resolution 320 x 240	Check this option if the SIP end device supports 320 x 240 resolution for the LCD screen wallpaper. • Source

	Configure the location where wallpapers are stored. • File If "URL" is selected as source, specify the URL of the wallpaper file. If "Local UCM Server" is selected as source, click to upload wallpaper file to the UCM6200.

GLOBAL TEMPLATES

Global Templates can be accessed in web GUI->**PBX**->**Zero Config**->**Global Templates**. Users can create multiple global templates with different sets of configurations and save the templates. Later on, when the user configures the device in Edit Device dialog->Advanced Settings, the user can select to use one of the global template for the device. Please refer to section ***[MANAGE DEVICES]*** for more details on using the global templates.

When creating global template, users can select the categories and the parameters under each category to be used in the template. The global policy and the selected global template will both take effect when generating the config file. However, the selected global template has higher priority to the global policy when it comes to the same setting option/field. If the same option/field has different value configured in the global policy and the selected global template, the value for this option/field in the selected global template will override the value in global policy.

Click on "Create New Template" to add a global template. Users will see the following configurations.

Table 31: Create New Template

Template Name	Create a name to identify this global template.
Description	Provide a description for the global template. This is optional.
Active	Check this option to enable the global template.

- Click on to edit the global template.

  The window for editing global template is shown in the following figure. In the “Options” field, after entering the option name key word, the options containing the key word will be listed. Users could then select the options to be modified and click on “Add Option” to add it into the global template.

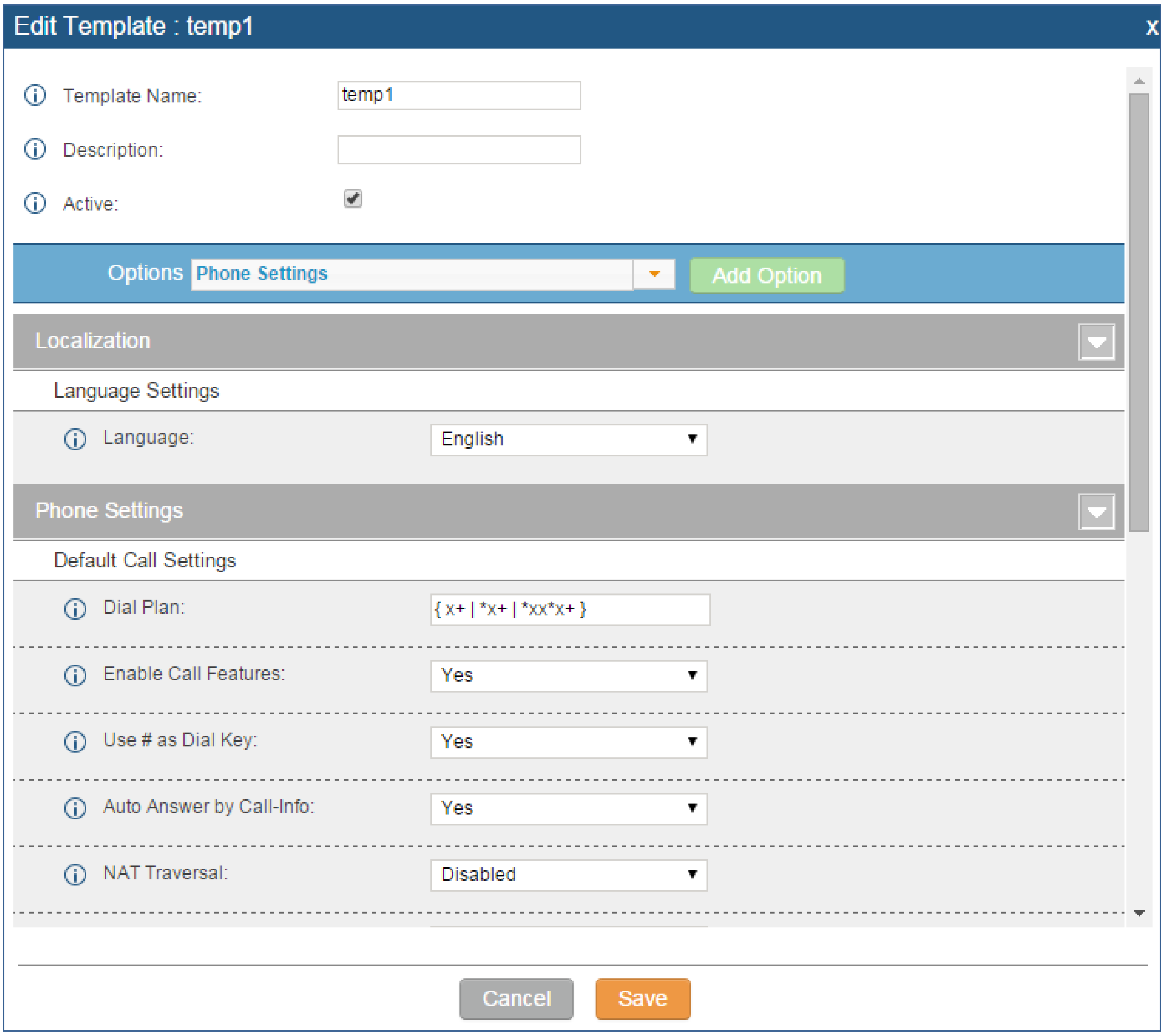


Figure 67: Edit Global Template

The added options will show in the list. Users can then enter or select value for each option to be used in the global template. On the left side of each added option, users can click on to remove this option from the template. On the right side of each option, users can click on to reset the option value to the default value.

Click on “Save” to save this global template.

- The created global templates will show in the web UI->**PBX**->**Zero Config**->**Global Templates** page. Users can click on to delete the global template or click on “Delete Selected Templates” to delete multiple selected templates at once.

- Click on “Toggle Selected Template(s)” to toggle the status between enabled/disabled for the selected templates.

MODEL CONFIGURATION

MODEL TEMPLATES

Model layer configuration allows users to apply model-specific configurations to different devices. Users could create/edit/delete a model template by accessing web GUI, page **PBX->Zero Config->Model Templates**. If multiple model templates are created and enabled, when the user configures the device in Edit Device dialog->Advanced Settings, the user can select to use one of the model template for the device. Please refer to section ***[MANAGE DEVICES]*** for more details on using the model template.

For each created model template, users can assign it as default model template. If assigned as default model template, the values in this model template will be applied to all the devices of this model. There is always only one default model template that can be assigned at one time on the UCM6200.

The selected model template and the default model template will both take effect when generating the config file for the device. However, the model template has higher priority to default model template when it comes to the same setting option/field. If the same option/field has different value configured in the default model template and the selected model template, the value for this option/field in the selected model template will override the value in default model template.

- Click on “Create New Template” to add a model template.

Table 32: Create New Model Template

Model	Select a model to apply this template. The supported Grandstream models are listed in the dropdown list for selection.
Template Name	Create a name for the model template.
Description	Enter a description for the model template. This is optional.
Default Model Template	Select to assign this model template as the default model template. The value of the option in default model template will be overridden if other selected model template has a different value for the same option.
Active	Check this option to enable the model template.

- Click on to edit the model template.

The editing window for model template is shown in the following figure. In the “Options” field, enter the option name key word, the option that contains the key word will be listed. User could then select the option and click on “Add Option” to add it into the model template.
Once added, the option will be shown in the list below. On the left side of each option, users can click on to remove this option from the model template. On the right side of each option, users can click on to reset the option to the default value.

User could also click on “Add New Field” to add a P value number and the value to the configuration. The following figure shows setting P value “P1362” to “en”, which means the display language on the LCD is set to English. For P value information of different models, please refer to configuration template here http://www.grandstream.com/support/tools.

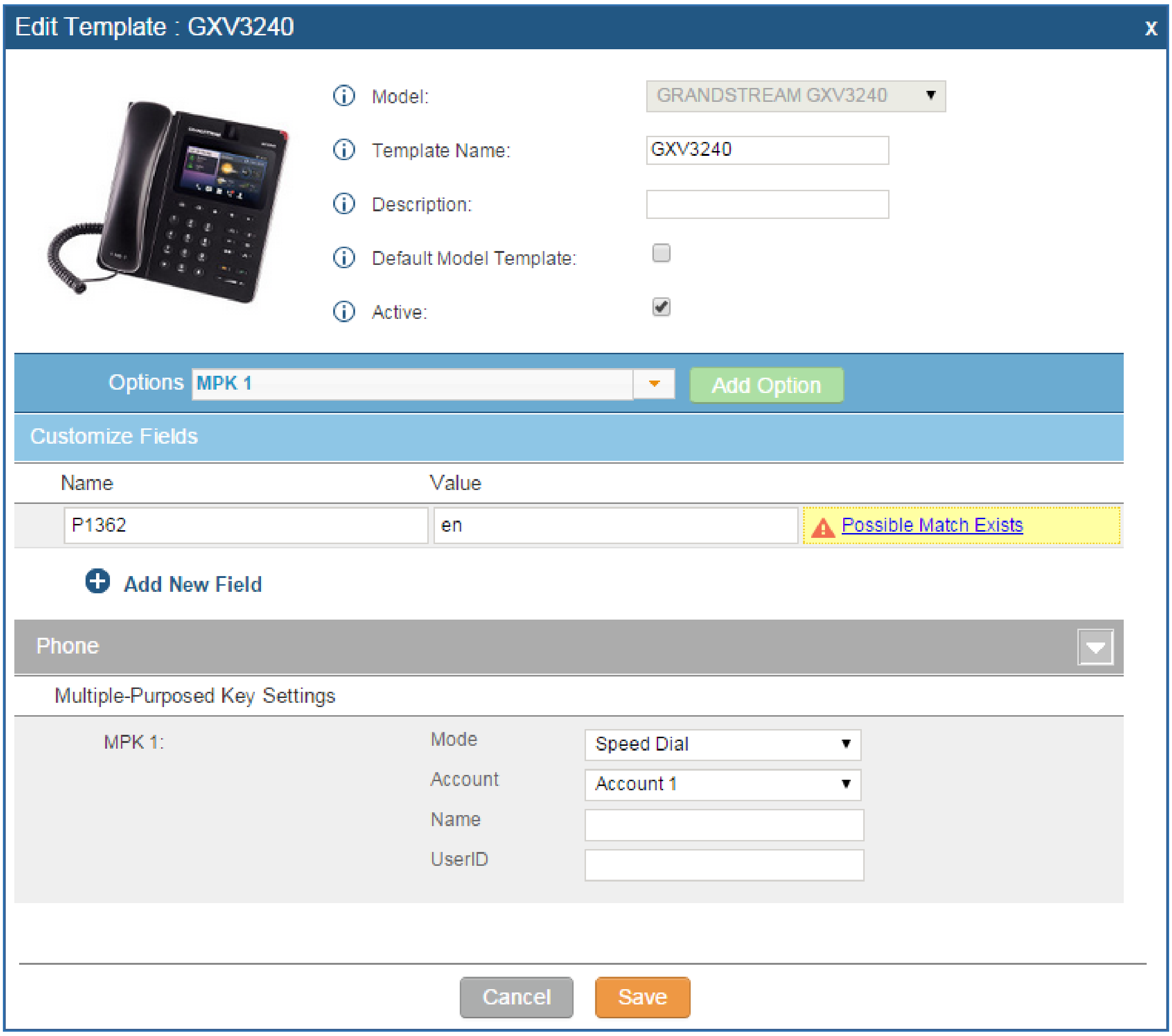


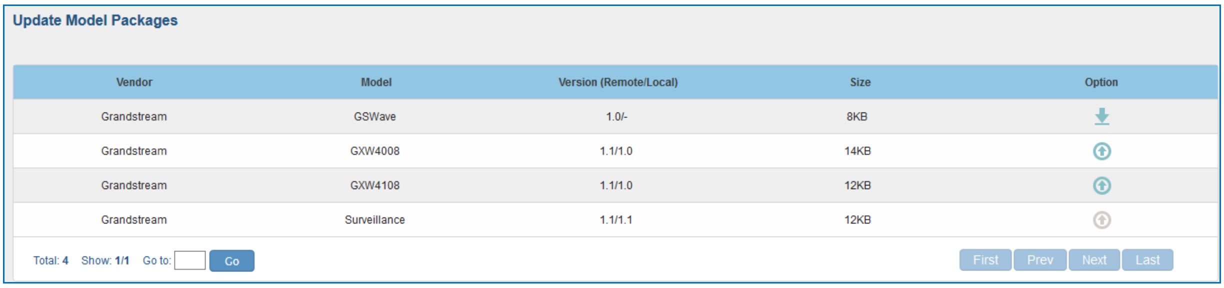
Figure 68: Edit Model Template

- Click on Save when done. The model template will be displayed on web UI->**PBX->Zero Config->Model Templates** page.
- Click on to delete the model template or click on “Delete Selected Templates” to delete multiple selected templates at once.
- Click on “Toggle Selected Template(s)” to toggle the status between enabled/disabled for the selected model templates.

MODEL UPDATE

UCM6200 zero config feature supports provisioning all models of Grandstream SIP end devices. Templates for most of the Grandstream models are built in with the UCM6200 already. Templates for GS Wave and Grandstream surveillance products require users to download and install under web UI->**PBX**->**Zero Config**->**Model Update** first before they are available in the UCM6200 for selection. After downloading and installing the model template to the UCM6200, it will show in the dropdown list for “Model” selection when editing the model template.

- Click on to download the template.
- Click on to upgrade the model template. Users will see this icon available if the device model has template updated in the UCM6200.


Update Model Packages

Vendor	Model	Version (Remote/Local)	Size	Option
Grandstream	GSWave	1.0/-	8KB	
Grandstream	GXW4008	1.1/1.0	14KB	
Grandstream	GXW4108	1.1/1.0	12KB	
Grandstream	Surveillance	1.1/1.1	12KB	

Total: 4 Show: 1/1 Go to: Go First Prev Next Last

Figure 69: Template Management

In case the UCM6200 is placed in the private network and Internet access is restricted, users will not be able to get packages by downloading and installing from the remote server. Model template package can be manually uploaded from local device through web UI. Please contact Grandstream customer support if the model package is needed for manual uploading.

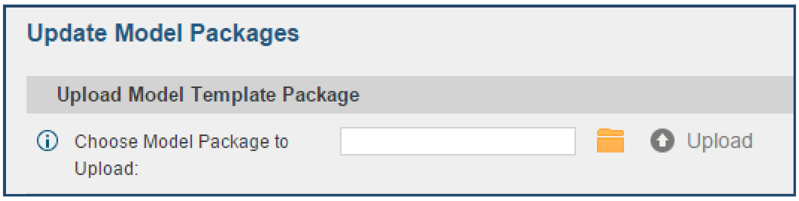


Figure 70: Upload Model Template Manually

DEVICE CONFIGURATION

On web GUI, page **PBX->Zero Config->Zero Config**, users could create new device, delete existing device(s), make special configuration for a single device, or send NOTIFY to existing device(s).

CREATE NEW DEVICE

Besides configuring the device after the device is discovered, users could also directly create a new device and configure basic settings before the device is discovered by the UCM6200. Once the device is plugged in, it can then be discovered and provisioned. This gives the system administrator adequate time to set up each device beforehand.

Click on "Create New Device" and the following dialog will show. Follow the steps below to create the configurations for the new device.

1. Firstly, select a model for the device to be created and enter its MAC address, IP address and firmware version (optional) in the corresponding field.
2. Basic settings will show a list of settings based on the model selected in step 1. Users could assign extensions to accounts, assign functions to Line Keys and Multiple-Purposed Keys if supported on the selected model.
3. Click on “Create New Device” to save the configuration for this device.

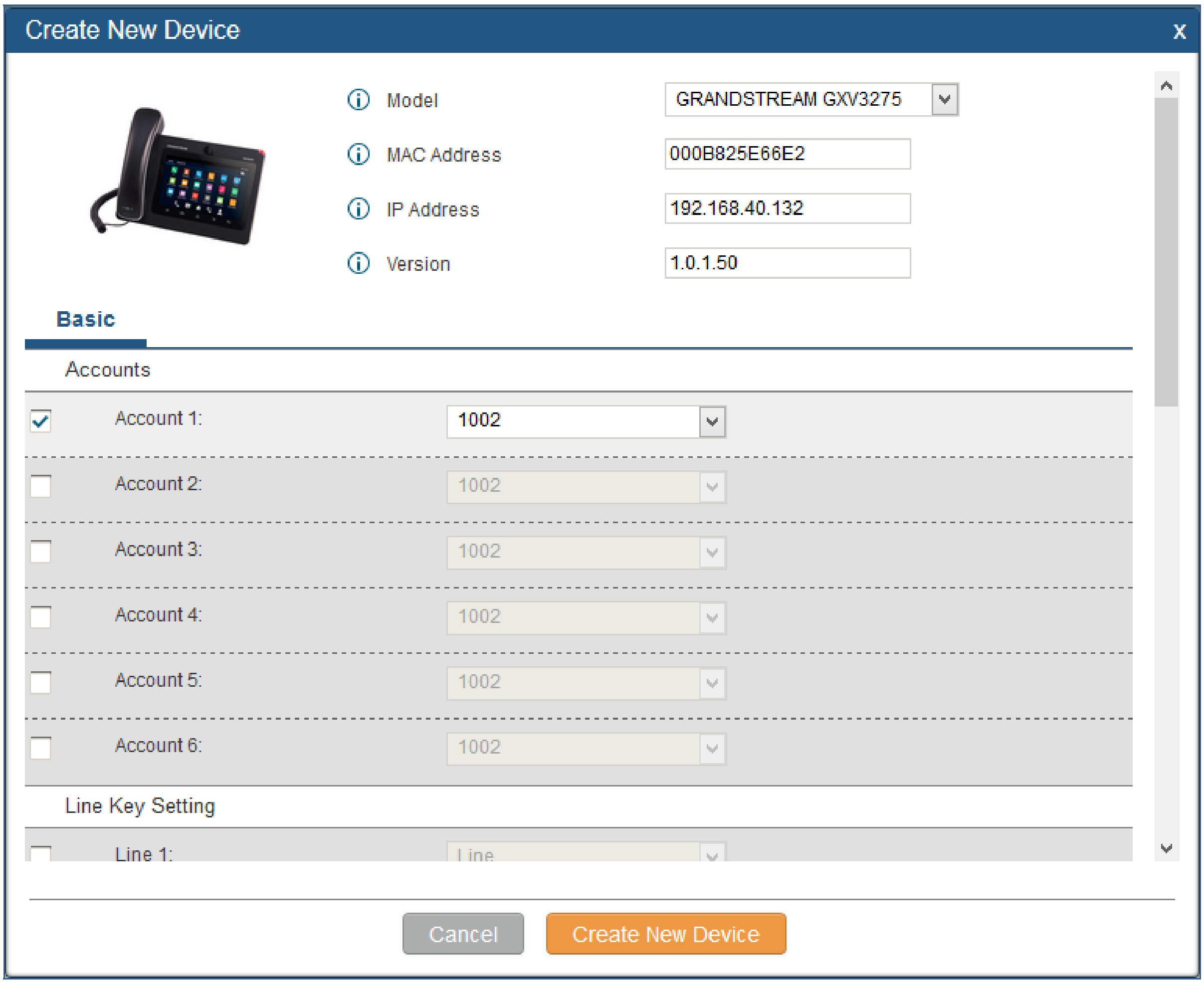


Figure 71: Create New Device

MANAGE DEVICES

The device manually created or discovered from Auto Discover will be listed in the web UI->**PBX->Zero Config->Zero Config** page. Users can see the devices with their MAC address, IP address, vendor, model and etc.

000B822A852C	192.168.40.3	--	1.0.4.9	GRANDSTREAM	GXP2100	--
000B822B0B34	--	--	1.0.5.31	GRANDSTREAM	GXP2120	--
000B822B2D94	192.168.40.143	--	1.0.5.26	GRANDSTREAM	GXP2110	--

Figure 72: Manage Devices

- Click on to access the web UI of the phone.
- Click on to edit the device configuration.

A new dialog will be displayed for the users to configure “Basic” settings and “Advanced” settings. “Basic” settings have the same configurations as displayed when manually creating a new device, i.e., account, line key and MPK settings; “Advanced” settings allow users to configure more details in a five-level structure.

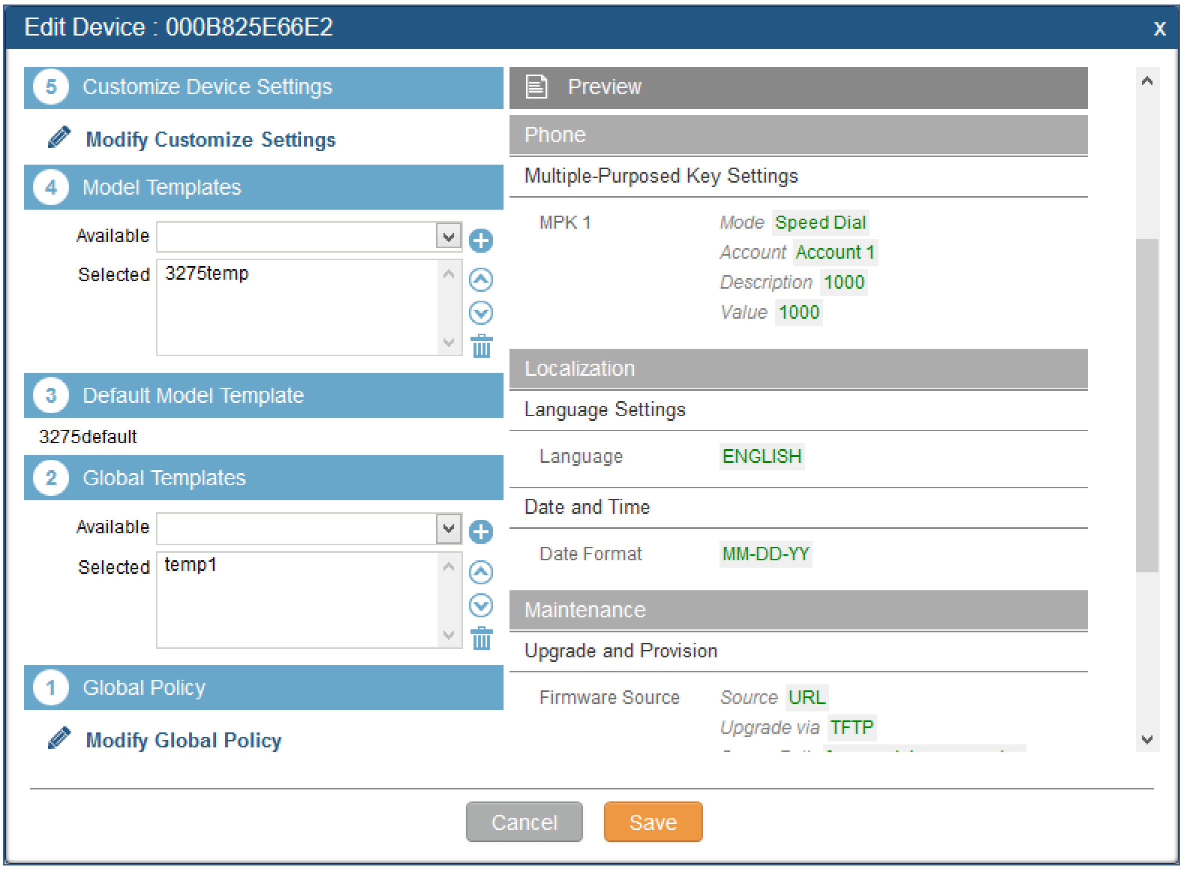


Figure 73: Edit Device

A preview of the “Advanced” settings is shown in the above figure. There are five levels configurations as described in (1) (2) (3) (4) (5) below, with priority from the lowest to the highest. The configurations in all levels will take effect for the device. If there are same options existing in different level configurations with different value configured, the higher level configuration will override the lower level configuration.

(1) Global Policy

This is the lowest level configuration. The global policy configured in web UI->**PBX->Zero Config->Global Policy** will be applied here. Clicking on “Modify Global Policy” to redirect to page **PBX->Zero Config->Global Policy**.

(2) Global Templates

Select a global template to be used for the device and click on to add. Multiple global templates can be selected and users can arrange the priority by adjusting orders via and . All the selected global templates will take effect. If the same option exists on multiple selected global templates, the value in the template with higher priority will override the one in the template with lower priority. Click on to remove the global template from the selected list.

(3) Default Model Template

Default Model Template will be applied to the devices of this model. Default model template can be configured in model template under web UI->**PBX**->**Zero Config**->**Model Templates** page. Please see default model template option in ***[Table 32: Create New Model Template].***

(4) Model Templates

Select a model template to be used for the device and click on to add. Multiple global templates can be selected and users can arrange the priority by adjusting orders via and . All the selected model templates will take effect. If the same option exists on multiple selected model templates, the value in the template with higher priority will override the one in the template with lower priority. Click on to remove the model template from the selected list.

(5) Customize Device Settings

This is the highest level configuration for the device. Click on "Modify Customize Device Settings" and following dialog will show.

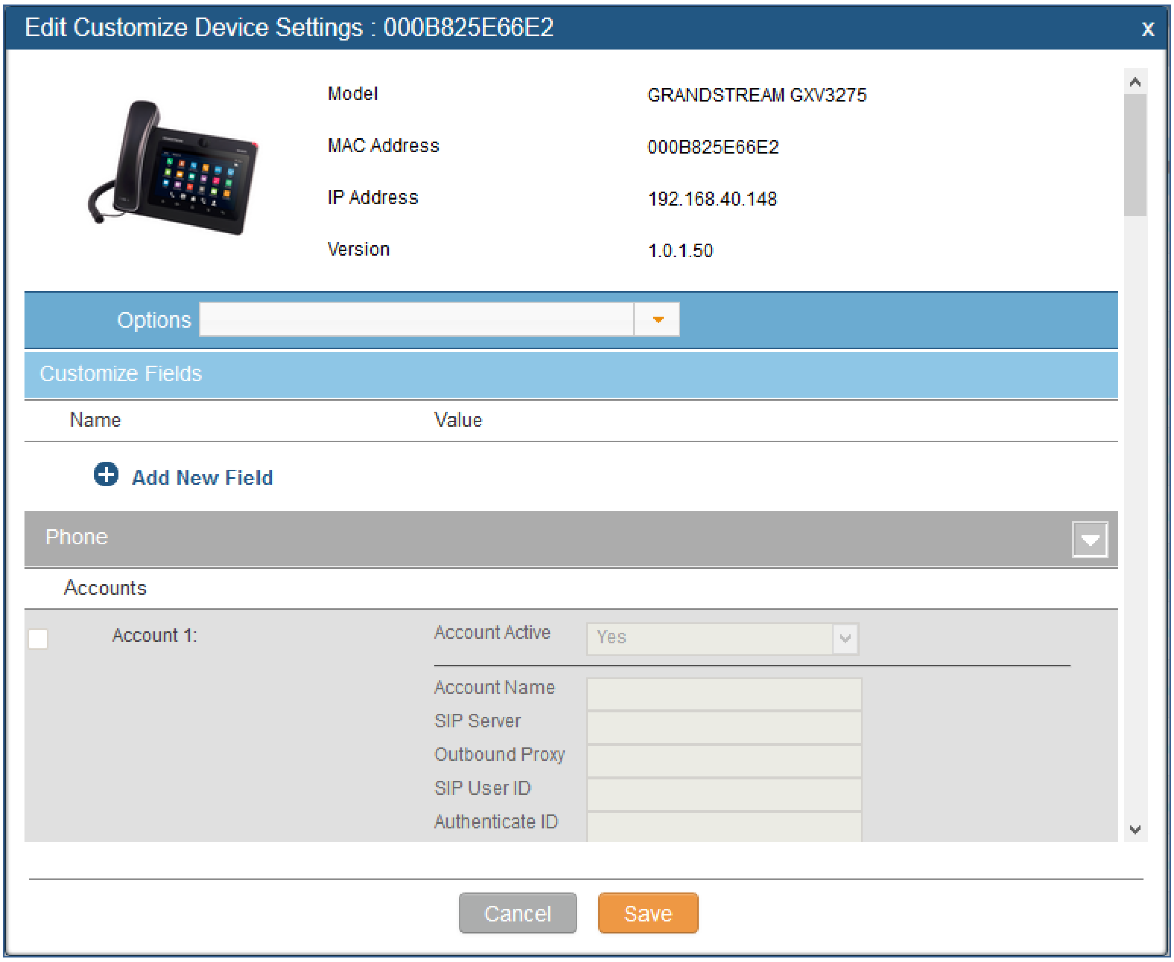


Figure 74: Edit Customize Device Settings

Scroll down in the dialog to view and edit the device-specific options. If the users would like to add more options which are not in the pre-defined list, click on "Add New Field" to add a P value number and the value to the configuration. The following figure shows setting P value "P1362" to "en", which means the display language on the LCD is set to English. The warning information on right tells that the option matching the P value number exists and clicking on it will lead to the matching option. For P value information of different models, please refer to configuration template here http://www.grandstream.com/sites/default/files/Resources/config-template.zip.

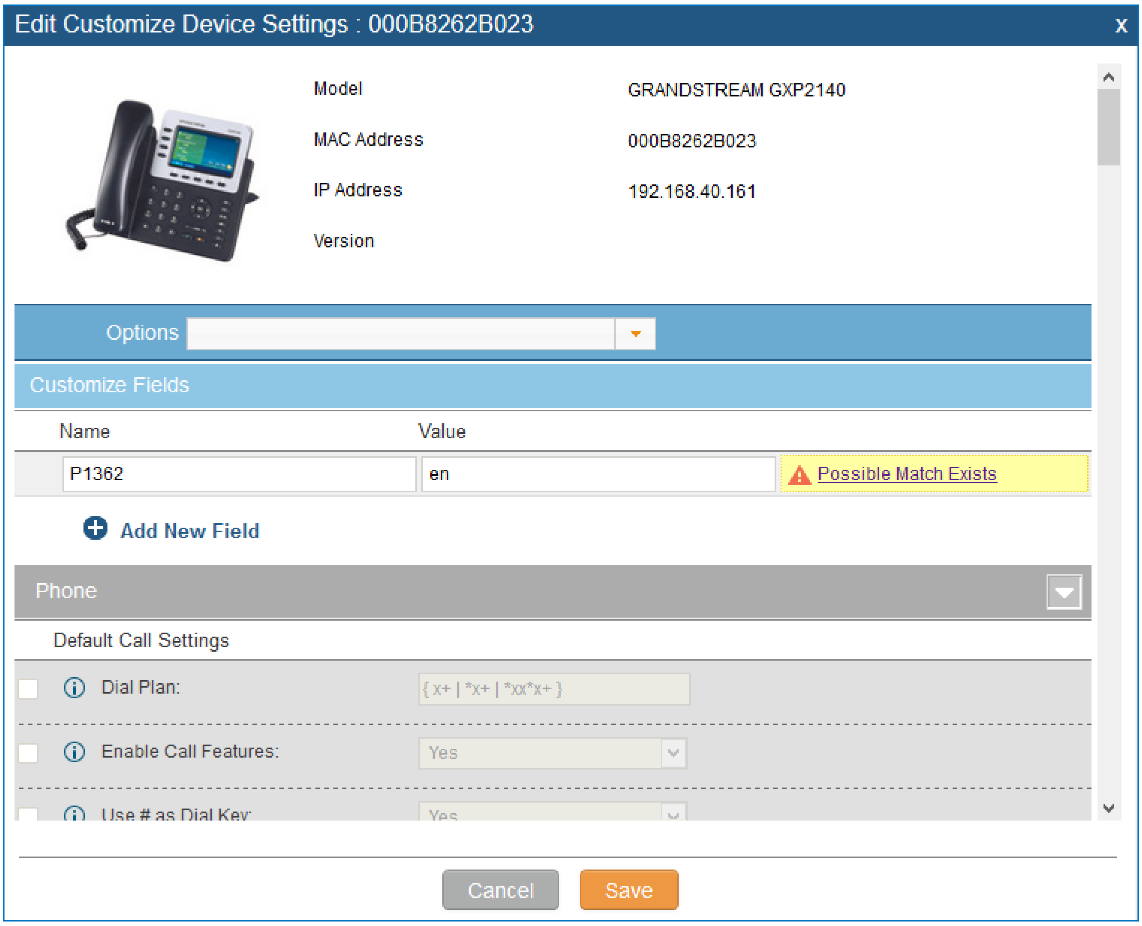


Figure 75: Add P Value in Customize Device Settings

- Select multiple devices that need to be modified and then click on Modify Selected Devices to batch modify devices.

  If selected devices are of the same model, the configuration dialog is like the following figure. Configurations in five levels are all available for users to modify.

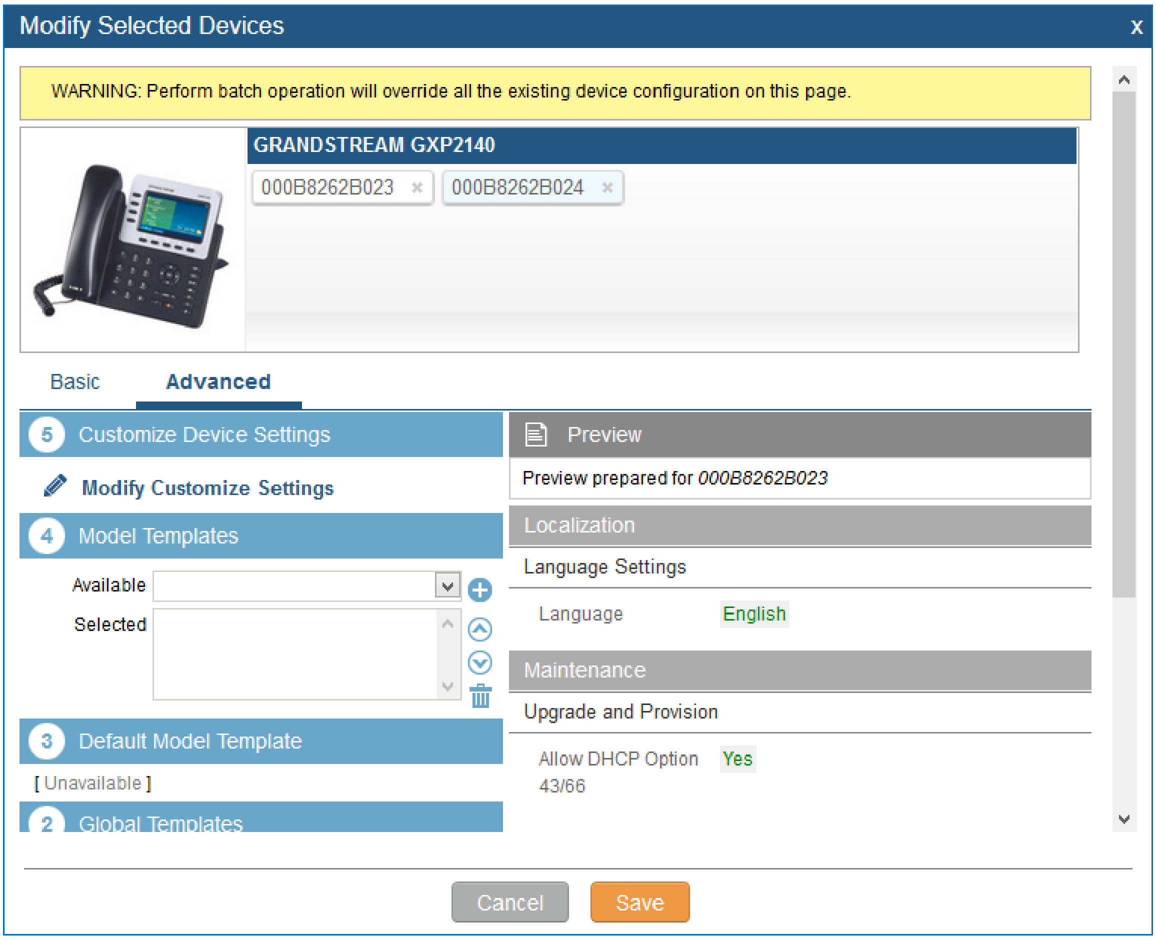


Figure 76: Modify Selected Devices - Same Model

If selected devices are of different models, the configuration dialog is like the following figure. Click on  to view more devices of other models. Users are only allowed to make modifications in Global Templates and Global Policy level.

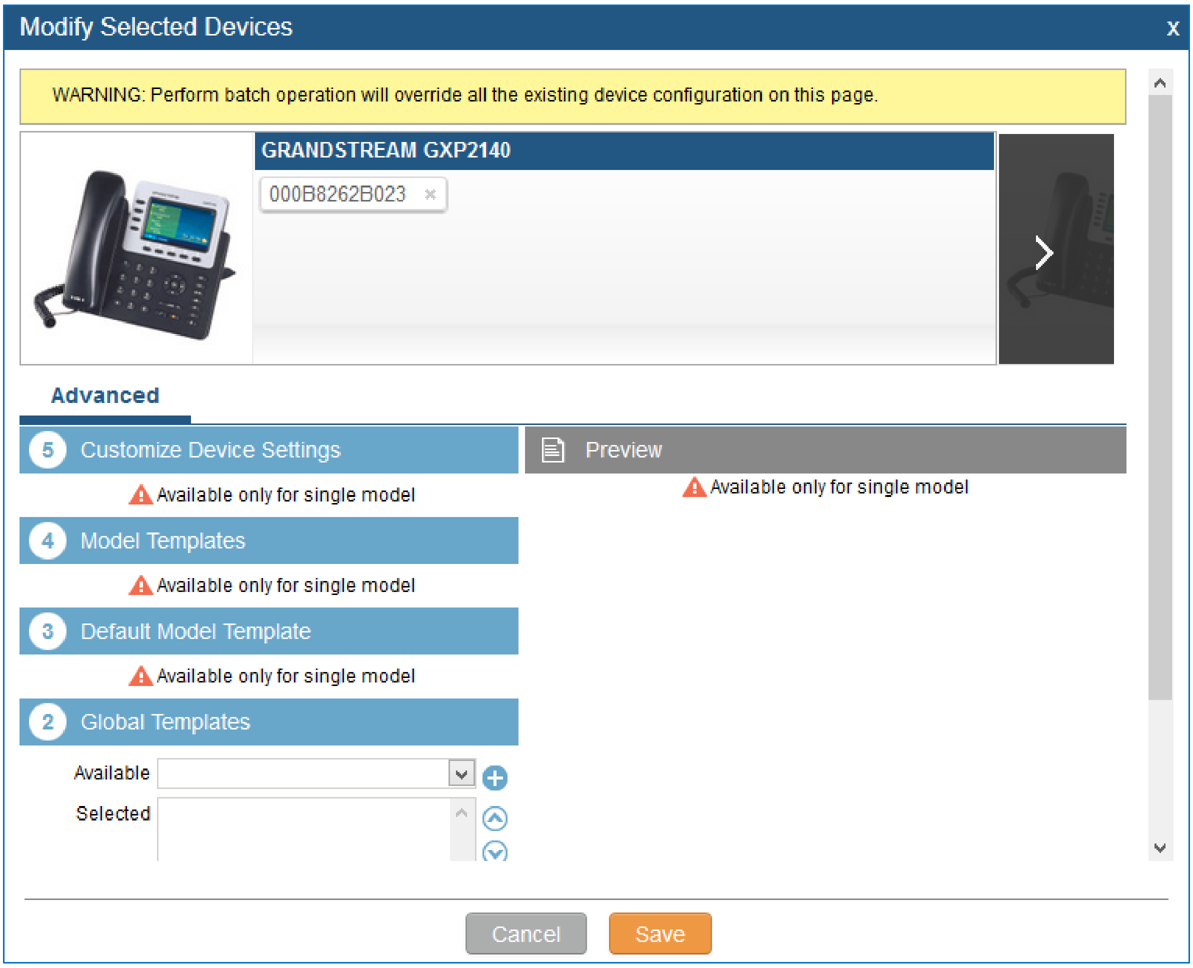


Figure 77: Modify Selected Devices - Different Models

⚠ **Note:**

Performing batch operation will override all the existing device configuration on the page.

After the above configurations, save the changes and go back to web UI->**PBX->Zero Config->Zero Config** page. Users could then click on to send NOTIFY to the SIP end point device and trigger the provisioning process. The device will start downloading the generated configuration file from the URL contained in the NOTIFY message.

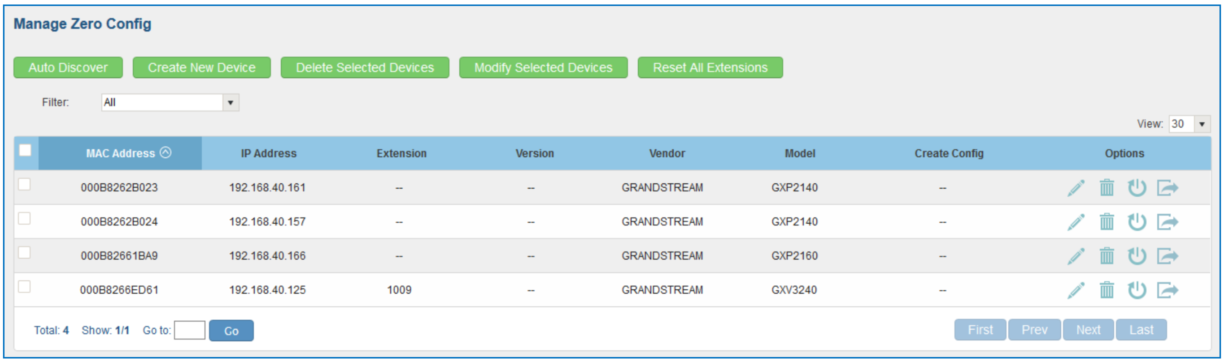


Figure 78: Device List in Zero Config

In this web page, users can also click on “Reset All Extensions” to reset the extensions of all the devices.

SAMPLE APPLICATION

Assuming in a small business office where there are 8 GXP2140 phones used by customer support and 1 GXV3275 phone used by customer support supervisor. 3 of the 8 customer support members speak Spanish and the rest speak English. We could deploy the following configurations to provisioning the office phones for the customer support team.

1. Go to web GUI->**PBX**->**Zero Config**->**Zero Config Settings**, select “Enable Zero Config”.

2. Go to web GUI->**PBX**->**Zero Config**->**Global Policy**, configure Date Format, Time Format and Firmware Source as follows.

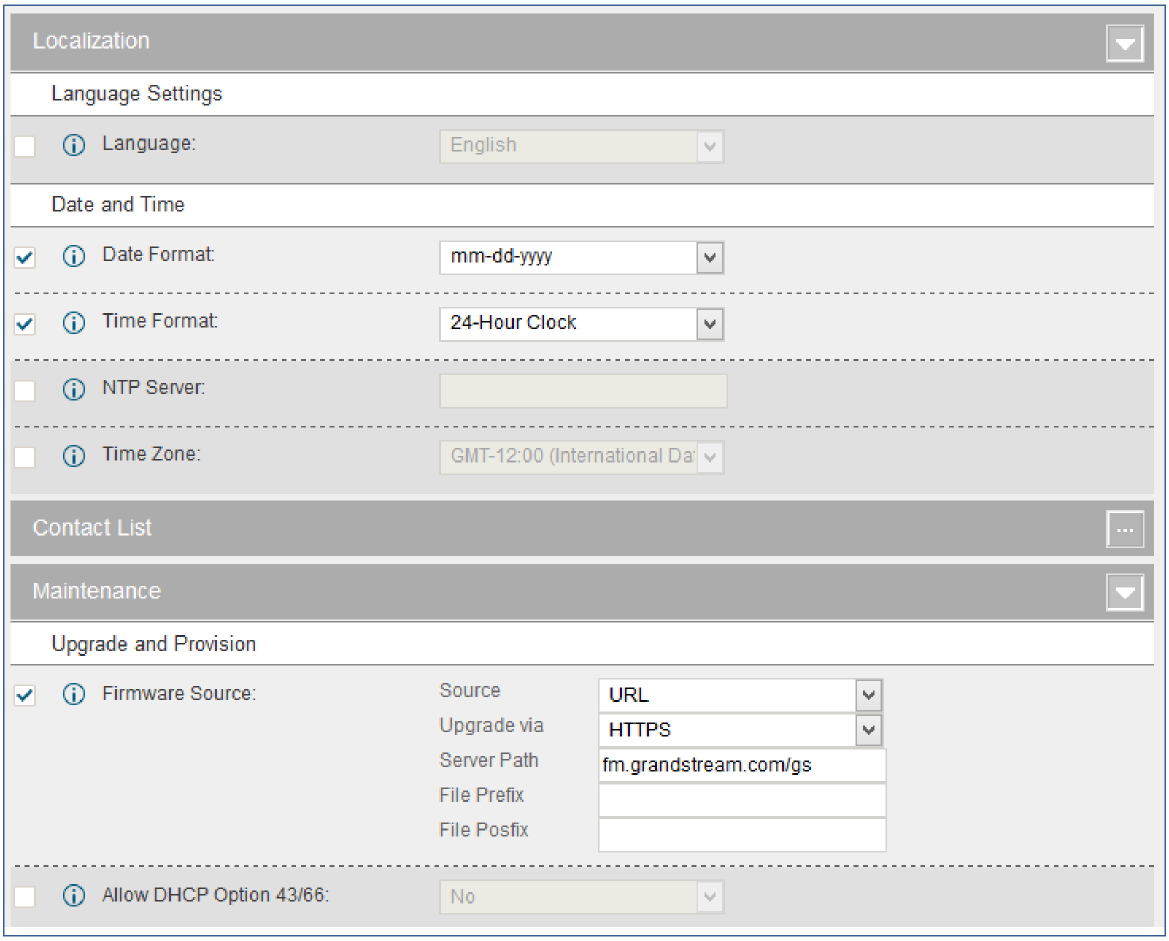


Figure 79: Zero Config Sample - Global Policy

3. Go to web GUI->**PBX**->**Zero Config**->**Model Templates**, create a new model template "English Support Template" for GXP2140. Add option "Language" and set it to "English". Then select the option "Default Model Template" to make it the default model template.

4. Go to web GUI->**PBX**->**Zero Config**->**Model Templates**, create another model template "Spanish Support Template" for GXP2140. Add option "Language" and set it to "Español".

5. After 9 devices are powered up and connected to the LAN network, use "Auto Discover" function or "Create New Device" function to add the devices to the device list on web UI->**PBX**->**Zero Config**->**Zero Config**.

6. On web GUI->**PBX**->**Zero Config**->**Zero Config** page, users could identify the devices by their MAC addresses or IP addresses displayed on the list. Click on to edit the device settings.

7. For each of the 5 phones used by English speaking customer support, in "Basic" settings select an available extension for account 1 and click on "Save". Then click on "Advanced" settings tab to bring up the following dialog. Users will see the English support template is applied since this is the default model template. A preview of the device settings will be listed on the right side.

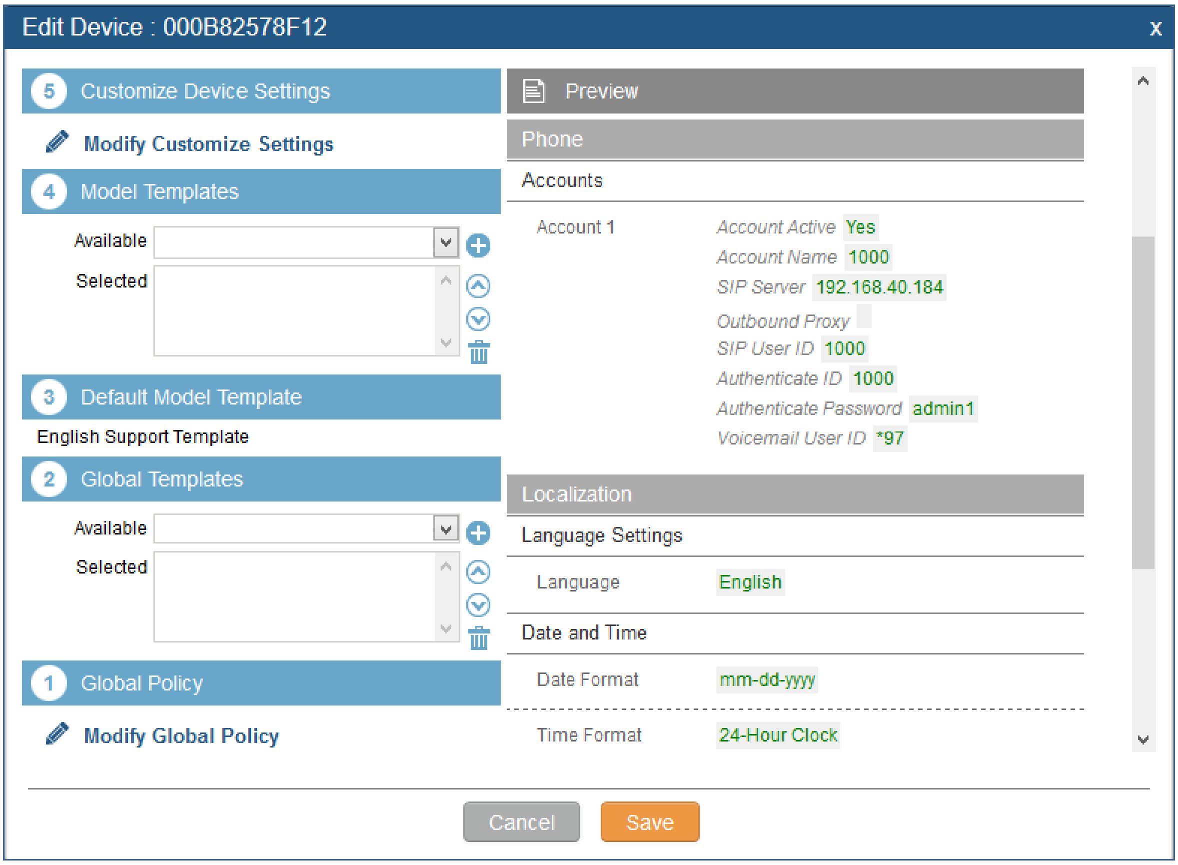


Figure 80: Zero Config Sample - Device Preview 1

8. For the 3 phones used by Spanish support, in "Basic" settings select an available extension for account 1 and click on "Save". Then click on "Advanced" settings tab to bring up the following dialog.

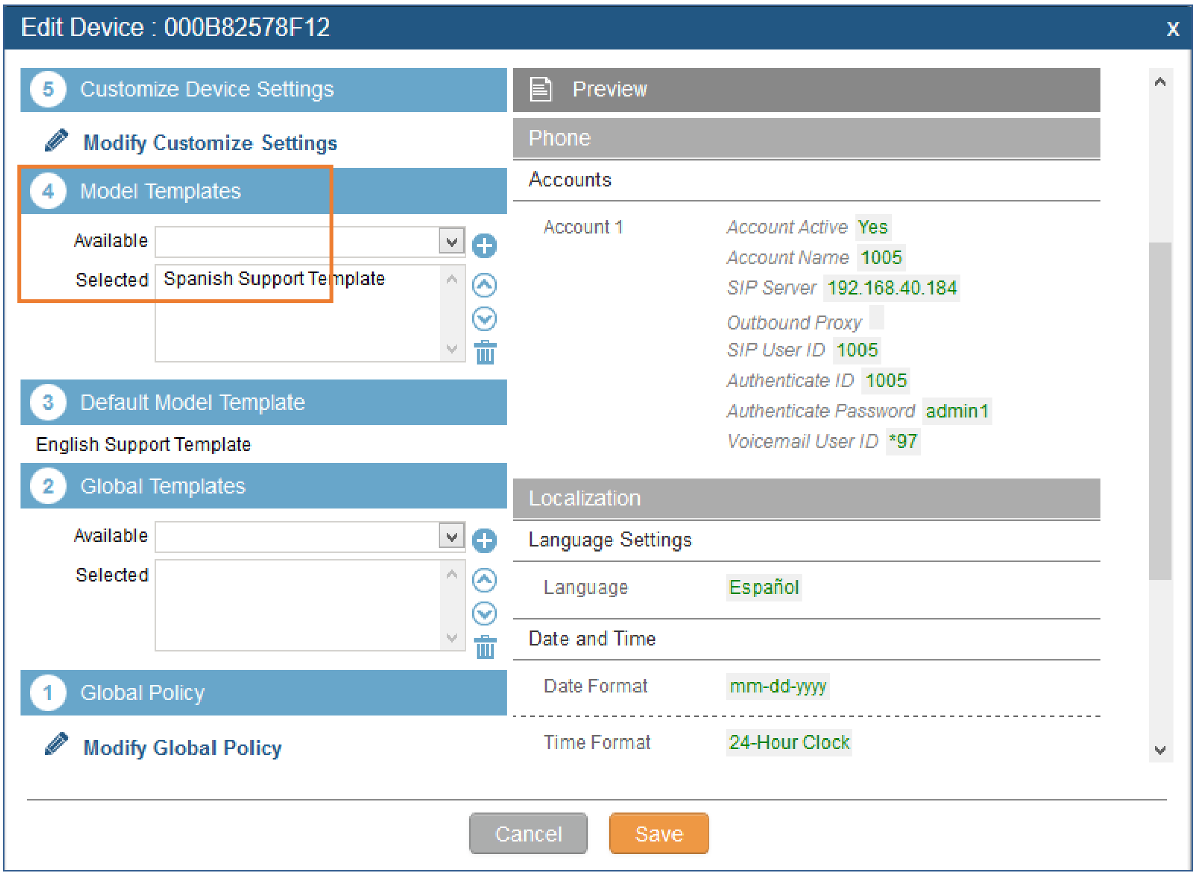


Figure 81: Zero Config Sample - Device Preview 2

Select “Spanish Support Template” in ④ “Model Template”. The preview of the device settings is displayed on the right side and we can see the language is set to “Español” since Model Template has the higher priority for the option “Language”, which overrides the value configured in default model template.

9. For the GXV3275 used by the customer support supervisor, select an available extension for account 1 on “Basic” settings and click on “Save”. Users can see the preview of the device configuration in “Advanced” settings. There is no model template configured for GXV3275.

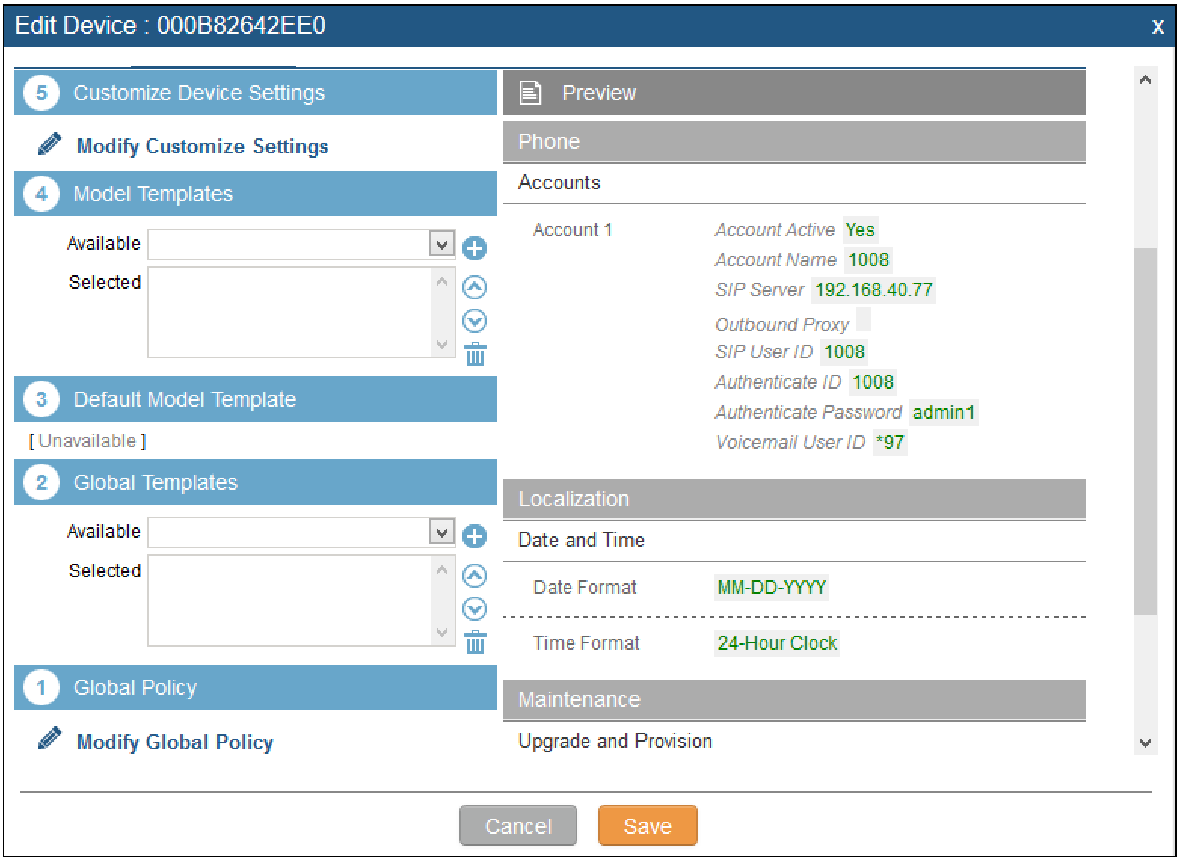


Figure 82: Zero Config Sample - Device Preview 3

10. Click on “Apply Changes” to apply saved changes.

11. On the web UI->**PBX**->**Zero Config**->**Zero Config** page, click on to send NOTIFY to trigger the device to download config file from UCM6200.

Now all the 9 phones in the network will be provisioned with an unique extension registered on the UCM6200. 3 of the phones will be provisioned to display Spanish on LCD and the other 5 will be provisioned to display English on LCD. The GXV3275 used by the supervisor will be provisioned to use the default language on LCD display since it’s not specified in the global policy.

EXTENSIONS

CREATE NEW USER

CREATE NEW SIP EXTENSION

To manually create new SIP user, go to Web GUI->**PBX**->**Basic/Call Routes**->**Extensions**. Click on "Create New User"->"Create New SIP Extension" and a new dialog window will show for users to fill in the extension information.

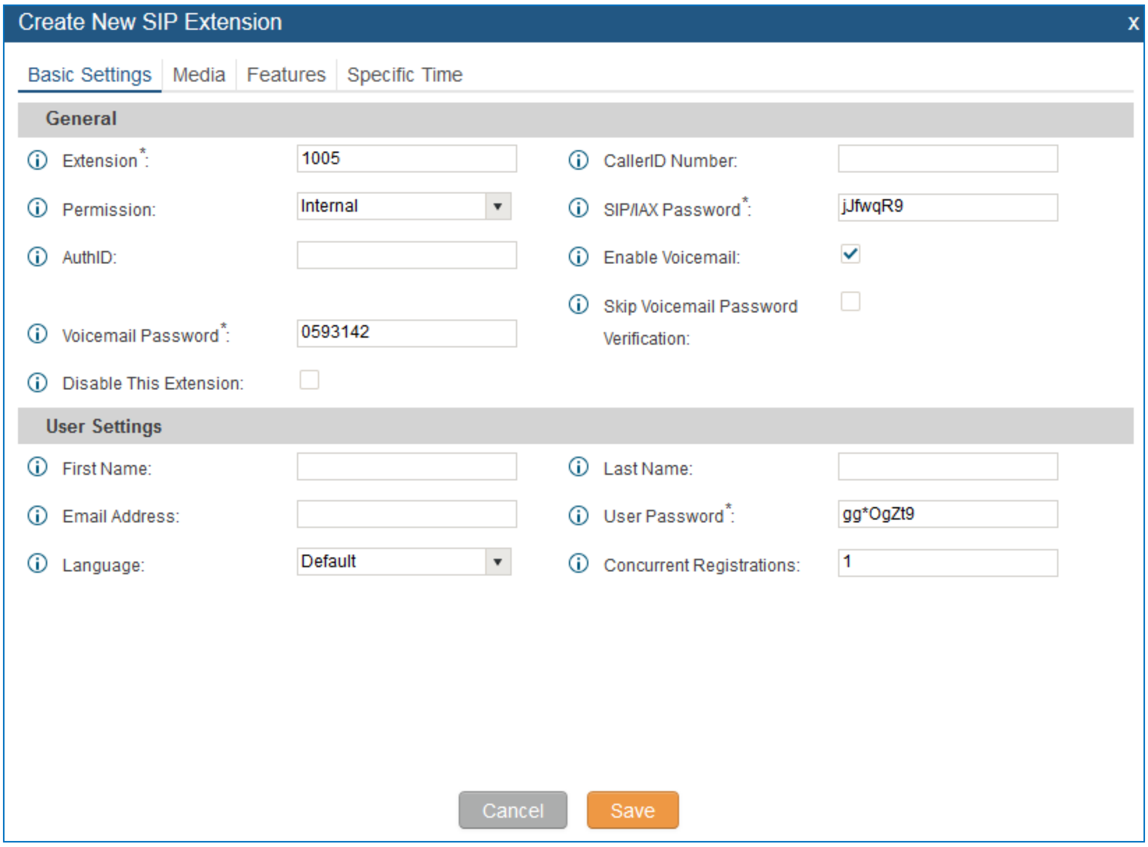


Figure 83: Create New Device

SIP extension options are divided into four categories:

- Basic Settings
- Media
- Features
- Specific Time

Click on the tag to view or edit options belonging to that category.
The configuration parameters are as follows.

Table 33: SIP Extension Configuration Parameters->Basic Settings

General	
Extension	The extension number associated with the user.
CallerID Number	Configure the CallerID Number that would be applied for outbound calls from this user. **Note:** The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
Permission	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". **Note:** Users need to have the same level as or higher level than an outbound rule's privilege in order to make outbound calls using this rule.
SIP/IAX Password	Configure the password for the user. A random secure password will be automatically generated. It is recommended to use this password for security purpose.
Auth ID	Configure the authentication ID for the user. If not configured, the extension number will be used for authentication.
Enable Voicemail	Enable voicemail for the user. The default setting is "Yes".
Voicemail Password	Configure voicemail password (digits only) for the user to access the voicemail box. A random numeric password is automatically generated. It is recommended to use the random generated password for security purpose.
Skip Voicemail Password Verification	When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default this option is disabled.
Disable This Extension	If selected, this extension will be disabled on the UCM6200. **Note:** The disabled extension still exists on the PBX but can’t be used on the end device.
User Settings	
First Name	Configure the first name of the user. The first name can contain characters, letters, digits and _.
Last Name	Configure the last name of the user. The last name can contain

	characters, letters, digits and _.
Email Address	Fill in the Email address for the user. Voicemail will be sent to this Email address.
User Password	Configure the password for user portal access. A random numeric password is automatically generated. It is recommended to use the randomly generated password for security purpose.
Language	Select the voice prompt language to be used for this extension. The default setting is "Default" which is the selected voice prompt language under web GUI->**PBX**->**Internal Options**->**Language**. The dropdown list shows all the current available voice prompt languages on the UCM6200. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under web UI->**PBX**->**Internal Options**->**Language**.
Concurrent Registrations	The maximum endpoints which can be registered into this extension. For security concerns, the default value is 1.

Table 34: SIP Extension Configuration Parameters->Media

SIP Settings	
NAT	Use NAT when the UCM6200 is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is one-way audio issue, usually it's related to NAT configuration or Firewall's support of SIP and RTP ports. The default setting is enabled.
Can Direct Media	By default, the UCM6200 will route the media steams from SIP endpoints through itself. If enabled, the PBX will attempt to negotiate with the endpoints to route the media stream directly. It is not always possible for the UCM6200 to negotiate endpoint-to-endpoint media routing. The default setting is "No".
DTMF Mode	Select DTMF mode for the user to send DTMF. The default setting is "RFC2833". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit PCMU and PCMA are required. When "Auto" is selected, RFC2833 will be used if offered, otherwise "Inband" will be used.
TEL URI	If the phone has an assigned PSTN telephone number, this field should be set to “User=Phone”. “User=Phone” parameter will be attached to the Request-Line and “TO” header in the SIP request to indicate the E.164 number. If set to “Enable”, “Tel” will be used instead of “SIP” in the SIP request.
Enable Keep-alive	If enabled, empty SDP packet will be sent to the SIP server periodically to keep the NAT port open. The default setting is "Yes".

Keep-alive Frequency	Configure the Keep-alive interval (in seconds) to check if the host is up. The default setting is 60 seconds.
Enable T.38 UDPTL	Enable or disable T.38 UDPTL support.
SRTP	Enable SRTP for the call. The default setting is disabled.
Fax Mode	Select Fax mode. The default setting is "None". • None: Disable Fax. • Fax Detect: Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under web UI->PBX->Internal Options->Fax/T.38.
Strategy	This option controls how the extension can be used on devices within different types of network. • Allow All Device in any network can register this extension. • Local Subnet Only Only the user in specific subnet can register this extension. Up to three subnet addresses can be specified. • A Specific IP Address Only the device on the specific IP address can register this extension. The default setting is "Allow All".
Codec Preference	Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G,726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p and VP8.

Table 35: SIP Extension Configuration Parameters->Features

Call Transfer	
Call Forward Unconditional	Configure the Call Forward Unconditional target number. If not configured, the Call Forward Unconditional feature is deactivated. The default setting is deactivated.
CFU Time Condition	Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time condition are "Office Time", "Out of Office Time", "Holiday", "Out of Holiday", "Out of Office Time or Holiday" and "Specific". Note: • "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific

	time. • Office Time and Holiday could be configured on page Settings->Time Settings->Office Time/Holiday page.
Call Forward No Answer	Configure the Call Forward No Answer target number. If not configured, the Call Forward No Answer feature is deactivated. The default setting is deactivated.
CFN Time Condition	Select time condition for Call Forward No Answer. The available time condition are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”. Note: • “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. • Office Time and Holiday could be configured on page Settings->Time Settings->Office Time/Holiday page.
Call Forward Busy	Configure the Call Forward Busy target number. If not configured, the Call Forward Busy feature is deactivated. The default setting is deactivated.
CFB Time Condition	Select time condition for Call Forward Busy. The available time condition are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”. Note: • “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. • Office Time and Holiday could be configured on page Settings->Time Settings->Office Time/Holiday page.
CC Settings	
Enable CC	If enabled, UCM6200 will automatically alert this extension when a called party is available, given that a previous call to that party failed for some reason. By default it’s disabled.
CC Mode	Two modes for Call Completion are supported: • **Normal**: This extension is used as ordinary extension. • **For Trunk**: This extension is registered from a PBX. The default setting is “Normal”.
CC Max Agents	Configure the maximum number of CCSS agents which may be allocated

	for this channel. In other words, this number serves as the maximum number of CC requests this channel is allowed to make. The minimum value is 1.
CC Max Monitors	Configure the maximum number of monitor structures which may be created for this device. In other words, this number tells how many callers may request CC services for a specific device at one time. The minimum value is 1.
Ring Simultaneously	
Ring Simultaneously	Enable this option to have an external number ring simultaneously along with the extension. If a register trunk is used for outbound, the register number will be used to be displayed for the external number as caller ID number.
External Number	Set the external number to be rang simultaneously. '-' is the connection character which will be ignored.
Time Condition for Ring Simultaneously	Ring the external number simultaneously along with the extension on the basis of this time condition.
Other Settings	
Ring Timeout	Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the UCM6200, which can be configured in the global ring timeout setting under web GUI->Internal Options->IVR Prompt: General Preference. The valid range is between 5 seconds and 600 seconds. Note: If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.
Auto Record	Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under web GUI->CDR->Recording Files.
Skip Trunk Auth	• If set to "yes", users can skip entering the password when making outbound calls. • If set to "By Time", users can skip entering the password when making outbound calls during the selected time condition. • If set to "No", users will be asked to enter the password when making outbound calls.
Time Condition for Skip Trunk Auth	If 'Skip Trunk Auth' is set to 'By Time', select a time condition during which users can skip entering password when making outbound calls.
Dial Trunk Password	Configure personal password when making outbound calls via trunk.
Support Hot-Desking Mode	If enabled, SIP Password will accept only alphabet characters and digits.

	Auth ID will be changed to the same as Extension.
Enable LDAP	If enabled, the extension will be added to LDAP Phonebook PBX list.
Enable WebRTC Support	Enable registration and call from WebRTC.
Music On Hold	Specify which Music On Hold class to suggest to the bridged channel when putting them on hold.
Call Duration Limit	The maximum duration of call-blocking.

Table 36: SIP Extension Configuration Parameters->Specific Time

Specific Time	
Time Condition	Click to add Time Condition to configure specific time for this extension.

CREATE NEW IAX EXTENSION

The UCM6200 supports Inter-Asterisk eXchange (IAX) protocol. IAX is used for transporting VoIP telephony sessions between servers and terminal devices. IAX is similar to SIP but also has its own characteristic. For more information, please refer to RFC 5465.

To manually create new IAX user, go to Web GUI->**PBX**->**Basic/Call Routes**->**Extensions**. Click on "Create New User"->"Create New IAX Extension" and a new dialog window will show for users to fill in the extension information. The configuration parameters are as follows.

Table 37: IAX Extension Configuration Parameters->Basic Settings

General	
Extension	The extension number associated with the user.
CallerID Number	Configure the CallerID Number that would be applied for outbound calls from this user. **Note:** The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
Permission	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". **Note:** Users need to have the same level as or higher level than an outbound rule's privilege in order to make outbound calls using this rule.
SIP/IAX Password	Configure the password for the user. A random secure password will be automatically generated. It is recommended to use this password for security purpose.

Enable Voicemail	Enable voicemail for the user. The default setting is "Yes".
Voicemail Password	Configure voicemail password (digits only) for the user to access the voicemail box. A random numeric password is automatically generated. It is recommended to use the random generated password for security purpose.
Skip Voicemail Password Verification	When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default this option is disabled.
Disable This Extension	If selected, this extension will be disabled on the UCM6200. **Note:** The disabled extension still exists on the PBX but can't be used on the end device.
User Settings	
First Name	Configure the first name of the user. The first name can contain characters, letters, digits and _.
Last Name	Configure the last name of the user. The last name can contain characters, letters, digits and _.
Email Address	Fill in the Email address for the user. Voicemail will be sent to this Email address.
User Password	Configure the password for user portal access. A random numeric password is automatically generated. It is recommended to use the randomly generated password for security purpose.
Language	Select the voice prompt language to be used for this extension. The default setting is "Default" which is the selected voice prompt language under web GUI->**PBX**->**Internal Options**->**Language**. The dropdown list shows all the current available voice prompt languages on the UCM6200. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under web UI->**PBX**->**Internal Options**->**Language**.

Table 38: IAX Extension Configuration Parameters->Media

SIP Settings	
Max Number of Calls	Configure the maximum number of calls allowed for each remote IP address.
Require Call Token	Configure to enable/disable requiring call token. If set to "Auto", it might lock out users who depend on backward compatibility when peer authentication credentials are shared between physical endpoints. The default setting is "Yes".

SRTP	Enable SRTP for the call. The default setting is disabled.
Fax Mode	Select Fax Mode. The default setting is "None". • None: Disable Fax. This is the default setting. • Fax Detect: Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under web UI->PBX->Internal Options->Fax/T.38.
Strategy	This option controls how the extension can be used on devices within different types of network. • Allow All Device in any network can register this extension. • Local Subnet Only Only the user in specific subnet can register this extension. Up to three subnet addresses can be specified. • A Specific IP Address Only the device on the specific IP address can register this extension. The default setting is "Allow All".
Codec Preference	Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G,726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p and VP8.

Table 39: IAX Extension Configuration Parameters->Features

Call Transfer	
Call Forward Unconditional	Configure the Call Forward Unconditional target number. If not configured, the Call Forward Unconditional feature is deactivated. The default setting is deactivated.
CFU Time Condition	Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time condition are "Office Time", "Out of Office Time", "Holiday", "Out of Holiday", "Out of Office Time or Holiday" and "Specific". Note: • "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. • Office Time and Holiday could be configured on page Settings->Time

	Settings->Office Time/Holiday page.
Call Forward No Answer	Configure the Call Forward No Answer target number. If not configured, the Call Forward No Answer feature is deactivated. The default setting is deactivated.
CFN Time Condition	Select time condition for Call Forward No Answer. The available time condition are "Office Time", "Out of Office Time", "Holiday", "Out of Holiday", "Out of Office Time or Holiday" and "Specific". Note: • "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. • Office Time and Holiday could be configured on page Settings->Time Settings->Office Time/Holiday page.
Call Forward Busy	Configure the Call Forward Busy target number. If not configured, the Call Forward Busy feature is deactivated. The default setting is deactivated.
CFB Time Condition	Select time condition for Call Forward Busy. The available time condition are "Office Time", "Out of Office Time", "Holiday", "Out of Holiday", "Out of Office Time or Holiday" and "Specific". Note: • "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. • Office Time and Holiday could be configured on page Settings->Time Settings->Office Time/Holiday page.
Ring Simultaneously	
Ring Simultaneously	Enable this option to have an external number ring simultaneously along with the extension. If a register trunk is used for outbound, the register number will be used to be displayed for the external number as caller ID number.
External Number	Set the external number to be rang simultaneously. '-' is the connection character which will be ignored.
Time Condition for Ring Simultaneously	Ring the external number simultaneously along with the extension on the basis of this time condition.
Other Settings	
Ring Timeout	Configure the number of seconds to ring the user before the call is

	forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the UCM6200, which can be configured in the global ring timeout setting under web GUI->Internal Options->IVR Prompt: General Preference. The valid range is between 5 seconds and 600 seconds. Note: If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.
Auto Record	Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under web GUI->CDR->Recording Files.
Skip Trunk Auth	• If set to "Yes", users can skip entering the password when making outbound calls. • If set to "By Time", users can skip entering the password when making outbound calls during the selected time condition. • If set to "No", users will be asked to enter the password when making outbound calls.
Time Condition for Skip Trunk Auth	If "Skip Trunk Auth" is set to "By Time", select a time condition during which users can skip entering password when making outbound calls.
Dial Trunk Password	Configure personal password when making outbound calls via trunk.
Enable LDAP	If enabled, the extension will be added to LDAP Phonebook PBX lists.
Music On Hold	Configure the Music On Hold class to suggest to the bridged channel when putting them on hold.
Call Duration Limit	The maximum duration of call-blocking.

Table 40: IAX Extension Configuration Parameters->Specific Time

Specific Time	
Time Condition	Click to add Time Condition to configure specific time for this extension.

CREATE NEW FXS EXTENSION

The UCM6200 supports Foreign eXchange Subscriber (FXS) interface. FXS is used when user needs to connect analog phone lines or FAX machines to the UCM6200.

To manually create new FXS user, go to Web GUI->**PBX**->**Basic/Call Routes**->**Extensions**. Click on "Create New User"->"Create New FXS Extension" and a new dialog window will show for users to fill in the extension information. The configuration parameters are as follows.

Table 41: FXS Extension Configuration Parameters->Basic Settings

General	
Extension	The extension number associated with the user.
Analog Station	Select the FXS port to be assigned for this extension.
CallerID Number	Configure the CallerID Number that would be applied for outbound calls from this user. **Note:** The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
Permission	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". **Note:** Users need to have the same level as or higher level than an outbound rule's privilege in order to make outbound calls using this rule.
Enable Voicemail	Enable voicemail for the user. The default setting is "Yes".
Voicemail Password	Configure voicemail password (digits only) for the user to access the voicemail box. A random numeric password is automatically generated. It is recommended to use the random generated password for security purpose.
Skip Voicemail Password Verification	When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default this option is disabled.
Disable This Extension	If selected, this extension will be disabled on the UCM6200. **Note:** The disabled extension still exists on the PBX but can't be used on the end device.
User Settings	
First Name	Configure the first name of the user. The first name can contain characters, letters, digits and _.
Last Name	Configure the last name of the user. The last name can contain characters, letters, digits and _.
Email Address	Fill in the Email address for the user. Voicemail will be sent to this Email address.
User Password	Configure the password for user portal access. A random numeric password is automatically generated. It is recommended to use the randomly generated password for security purpose.

Language	Select the voice prompt language to be used for this extension. The default setting is "Default" which is the selected voice prompt language under web GUI->**PBX**->**Internal Options**->**Language**. The dropdown list shows all the current available voice prompt languages on the UCM6200. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under web UI->**PBX**->**Internal Options**->**Language**.

Table 42: FXS Extension Configuration Parameters->Media

Analog Settings	
Call Waiting	Configure to enable/disable call waiting feature. The default setting is "No".
User '#' as SEND	If configured, the # key can be used as SNED key after dialing the number on the analog phone. The default setting is "Yes".
RX Gain	Configure the RX gain for the receiving channel of analog FXS port. The valid range is -30dB to +6dB. The default setting is 0.
TX Gain	Configure the TX gain for the transmitting channel of analog FXS port. The valid range is -30dB to +6dB. The default setting is 0.
MIN RX Flash	Configure the minimum period of time (in milliseconds) that the hook-flash must remain unpressed for the PBX to consider the event as a valid flash event. The valid range is 30ms to 1000ms. The default setting is 200ms.
MAX RX Flash	Configure the maximum period of time (in milliseconds) that the hook-flash must remain unpressed for the PBX to consider the event as a valid flash event. The minimum period of time is 256ms and it can't be modified. The default setting is 1250ms.
Enable Polarity Reversal	If enabled, a polarity reversal will be marked as received when an outgoing call is answered by the remote party. For some countries, a polarity reversal is used for signaling the disconnection of a phone line and the call will be considered as hangup on a polarity reversal. The default setting is "Yes".
Echo Cancellation	Specify "ON", "OFF" or a value (the power of 2) from 32 to 1024 as the number of taps of cancellation. **Note:** When configuring the number of taps, the number 256 is not translated into 256ms of echo cancellation. Instead, 256 taps means 256/8 = 32 ms. The default setting is "ON", which is 128 taps.
3-Way Calling	Configure to enable/disable 3-way calling feature on the user. The default setting is enabled.
Send CallerID After	Configure the number of rings before sending CID. Default setting is 1.

Fax Mode	For FXS extension, there are three options available in Fax Mode. The default setting is "None". • None: Disable Fax. • Fax Detect: Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under web UI->PBX->Internal Options->Fax/T.38. • Fax Gateway: If selected, the UCM6200 can support conversation and processing of Fax data from T.30 to T.38 or T.38 to T.30. This feature is only available for FXS or FXO port.

Table 43: FXS Extension Configuration Parameters->Features

Call Transfer	
Call Forward Unconditional	Configure the Call Forward Unconditional target number. If not configured, the Call Forward Unconditional feature is deactivated. The default setting is deactivated.
CFU Time Condition	Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time condition are "Office Time", "Out of Office Time", "Holiday", "Out of Holiday", "Out of Office Time or Holiday" and "Specific". Note: • "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. • Office Time and Holiday could be configured on page Settings->Time Settings->Office Time/Holiday page.
Call Forward No Answer	Configure the Call Forward No Answer target number. If not configured, the Call Forward No Answer feature is deactivated. The default setting is deactivated.
CFN Time Condition	Select time condition for Call Forward No Answer. The available time condition are "Office Time", "Out of Office Time", "Holiday", "Out of Holiday", "Out of Office Time or Holiday" and "Specific". Note: • "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific

	time. • Office Time and Holiday could be configured on page Settings->Time Settings->Office Time/Holiday page.
Call Forward Busy	Configure the Call Forward Busy target number. If not configured, the Call Forward Busy feature is deactivated. The default setting is deactivated.
CFB Time Condition	Select time condition for Call Forward Busy. The available time condition are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”. Note: • “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. • Office Time and Holiday could be configured on page Settings->Time Settings->Office Time/Holiday page.
CC Settings	
Enable CC	If enabled, UCM6200 will automatically alert this extension when a called party is available, given that a previous call to that party failed for some reason.
Ring Simultaneously	
Ring Simultaneously	Enable this option to have an external number ring simultaneously along with the extension. If a register trunk is used for outbound, the register number will be used to be displayed for the external number as caller ID number.
External Number	Set the external number to be rang simultaneously. ‘-’ is the connection character which will be ignored.
Time Condition for Ring Simultaneously	Ring the external number simultaneously along with the extension on the basis of this time condition.
Hotline	
Enable Hotline	If enabled, hotline dialing plan will be activated, a pre-configured number will be used according to the selected Hotline Type.
Hotline Number	Configure the Hotline Number
Hotline Type	Configure the Hotline Type: • **Immediate Hotline:** When the phone is off-hook, UCM6200 will immediately dial the preset number • **Delay Hotline:** When the phone is off-hook, if there is no dialing within 5 seconds, UCM6200 will dial the preset number.
Other Settings	

Ring Timeout	Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the UCM6200, which can be configured in the global ring timeout setting under web GUI->Internal Options->IVR Prompt: General Preference. The valid range is between 5 seconds and 600 seconds. Note: If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.
Auto Record	Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under web GUI->CDR->Recording Files.
Skip Trunk Auth	• If set to "Yes", users can skip entering the password when making outbound calls. • If set to "By Time", users can skip entering the password when making outbound calls during the selected time condition. • If set to "No", users will be asked to enter the password when making outbound calls.
Time Condition for Skip Trunk Auth	If "Skip Trunk Auth" is set to "By Time", select a time condition during which users can skip entering password when making outbound calls.
Dial Trunk Password	Configure personal password when making outbound calls via trunk.
Enable LDAP	If enabled, this extension will be added to LDAP Phonebook PBX list; if disabled, this extension will be skipped when creating LDAP Phonebook.
Music On Hold	Select which Music On Hold class to suggest to extension when putting the active call on hold.
Call Duration Limit	Configure the maximum duration of call-blocking.

Table 44: FXS Extension Configuration Parameters->Specific Time

Specific Time	
Time Condition	Click to add Time Condition to configure specific time for this extension.

BATCH ADD EXTENSIONS

BATCH ADD SIP EXTENSIONS

In order to add multiple SIP extensions, BATCH add can be used to create standardized SIP extension accounts. However, unique extension user name can't be set using BATCH add.
Under Web GUI->**PBX**->**Basic/Call Routes**->**Extensions**, click on "Batch Add Extensions"->"Batch Add SIP Extensions".

Table 45: Batch Add SIP Extension Parameters

General	
Start Extension	Configure the starting extension number of the batch of extensions to be added.
Create Number	Specify the number of extensions to be added. The default setting is 5.
Permission	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". **Note:** Users need to have the same level as or higher level than an outbound rule's privilege in order to make outbound calls from this rule.
Enable Voicemail	Enable Voicemail for the user. The default setting is "Yes".
SIP/IAX Password	Configure the SIP/IAX password for the users. Three options are available to create password for the batch of extensions. • User Random Password. A random secure password will be automatically generated. It is recommended to use this password for security purpose. • Use Extension as Password. • Enter a password to be used on all the extensions in the batch.
Voicemail Password	Configure Voicemail password (digits only) for the users. • User Random Password. A random password in digits will be automatically generated. It is recommended to use this password for security purpose. • Use Extension as Password. • Enter a password to be used on all the extensions in the batch.
Ring Timeout	Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the UCM6200, which can be configured in the global ring timeout setting under web GUI->**Internal Options**->**IVR Prompt**: General Preference. The valid range is between 5 seconds and 600 seconds. **Note:** If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.
Auto Record	Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under web GUI->**CDR**->**Recording Files**.
Skip Voicemail Password Verification	When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default this

	option is disabled.
Music On Hold	Select which Music On Hold class to suggest to extensions when putting them on hold.
Enable LDAP	If enabled, the batch added extensions will be added to LDAP Phonebook PBX list; if disabled, the batch added extensions will be skipped when creating LDAP Phonebook.
Enable WebRTC Support	If enabled, extensions will be able to login to user portal and use Web RTC features.
Call Duration Limit	Configure the maximum duration of call-blocking.
SIP Settings	
NAT	Use NAT when the PBX is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is one-way audio issue, usually it's related to NAT configuration or Firewall's support of SIP and RTP ports. The default setting is enabled.
Can Direct Media	By default, the PBX will route the media steams from SIP endpoints through itself. If enabled, the PBX will attempt to negotiate with the endpoints to route the media stream directly. It is not always possible for the PBX to negotiate endpoint-to-endpoint media routing. The default setting is "No".
DTMF Mode	Select DTMF mode for the user to send DTMF. The default setting is "RFC2833". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit codec PCMU and PCMA are required. When "Auto" is selected, RFC2833 will be used if offered, otherwise "Inband" will be used.
Enable Keep-alive	If enabled, empty SDP packet will be sent to the SIP server periodically to keep the NAT port open. The default setting is "Yes".
Keep-alive Frequency	Configure the number of seconds for the host to be up for Keep-alive. The default setting is 60 seconds.
TEL URI	If the end device/phone has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled.
Concurrent Registrations	The maximum endpoints which can be registered into this extension. For security concerns, the default value is 1.
Other Settings	
SRTP	Enable SRTP for the call. The default setting is "No".
Fax Mode	Select Fax mode for this user. The default setting is “None”.

	• None: Disable Fax. • Fax Detect: Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under web UI->PBX->Internal Options->Fax/T.38.
Strategy	This option controls how the extension can be used on devices within different types of network. • Allow All Device in any network can register this extension. • Local Subnet Only Only the user in specific subnet can register this extension. Up to three subnet addresses can be specified. • A Specific IP Address. Only the device on the specific IP address can register this extension. The default setting is "Allow All".
Enable T.38 UDPTL	Enable or disable T.38 UDPTL Support.
Skip Trunk Auth	If enable “All”, users do not need to enter password when making an outbound call. If enable “Follow Me”, the user can dial out via follow me without password.
Codec Preference	Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.722, G.729, G.723, Ilbc, ADPCM, LPC10, H.264, H.263, H.263p and VP8.

BATCH ADD IAX EXTENSIONS

Under Web GUI->**PBX**->**Basic/Call Routes**->**Extensions**, click on “Batch Add Extensions”->”Batch Add IAX Extensions”.

Table 46: Batch Add IAX Extension Parameters

General	
Start Extension	Configure the starting extension number of the batch of extensions to be added.
Create Number	Specify the number of extensions to be added. The default setting is 5.
Permission	Assign permission level to the user. The available permissions are “Internal”, “Local”, “National” and “International” from the lowest level to the highest level. The default setting is “Internal”. **Note:**

	Users need to have the same level as or higher level than an outbound rule's privilege in order to make outbound calls from this rule.
Enable Voicemail	Enable Voicemail for the user. The default setting is "Yes".
SIP/IAX Password	Configure the SIP/IAX password for the users. Three options are available to create password for the batch of extensions. • User Random Password. A random secure password will be automatically generated. It is recommended to use this password for security purpose. • Use Extension as Password. • Enter a password to be used on all the extensions in the batch.
Voicemail Password	Configure Voicemail password (digits only) for the users. • User Random Password. A random password in digits will be automatically generated. It is recommended to use this password for security purpose. • Use Extension as Password. • Enter a password to be used on all the extensions in the batch.
Ring Timeout	Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the UCM6200, which can be configured in the global ring timeout setting under web GUI->**Internal Options**->**IVR Prompt**: General Preference. The valid range is between 5 seconds and 600 seconds. **Note:** If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.
Auto Record	Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under web GUI->**CDR**->**Recording Files**.
Skip Voicemail Password Verification	When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default this option is disabled.
Music On Hold	Select which Music On Hold class to suggest to extensions when putting them on hold.
Enable LDAP	If enabled, the batch added extensions will be added to LDAP Phonebook PBX list; if disabled, the batch added extensions will be skipped when creating LDAP Phonebook.
Call Duration Limit	Configure the maximum duration of call-blocking.
IAX Settings	
Max Number of Calls	Configure the maximum number of calls allowed for each remote IP

	address.
Require Call Token	Configure to enable/disable requiring call token. If set to "Auto", it might lock out users who depend on backward compatibility when peer authentication credentials are shared between physical endpoints. The default setting is "Yes".
Other Settings	
SRTP	Enable SRTP for the call. The default setting is "No".
Fax Mode	Select Fax Mode for this user. The default setting is “None”. • None: Disable Fax. • Fax Detect: Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under web UI->**PBX**->**Internal Options**->**Fax/T.38**.
Strategy	This option controls how the extension can be used on devices within different types of network. • Allow All Device in any network can register this extension. • Local Subnet Only Only the user in specific subnet can register this extension. Up to three subnet addresses can be specified. • A Specific IP Address. Only the device on the specific IP address can register this extension. The default setting is "Allow All".
Skip Trunk Auth	If enable “All”, users do not need to enter password when making an outbound call. If enable “Follow Me”, the call can dial out via follow me without password.
Codec Preference	Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.722, G.729, G.723, iLBC, ADPCM, LPC10, H.264, H.263, H.263p and VP8.

SEARCH AND EDIT EXTENSION

All the UCM6200 extensions are listed under Web GUI->**PBX**->**Basic/Call Routes**->**Extensions**, with status, Extension, CallerID Name, Technology (SIP, IAX and FXS), IP and Port. Each extension has a checkbox for users to "Modify Selected Extensions" or "Delete Selected Extensions". Also, options "Edit", "Reboot" and "Delete" are available per extension. User can search an extension by specifying the extension number to find an extension quickly.

Figure 84: Manage Extensions

- **Status**

  Users can see the following icon for each extension to indicate the SIP status.

  Green: Free
  Blue: Ringing
  Yellow: In Use
  Grey: Unavailable (the extension is not registered or disabled on the PBX)

- **Edit single extension**

  Click on to start editing the extension parameters.

- **Reboot the user**

  Click on to send NOTIFY reboot event to the device which has an UCM6200 extension already registered. To successfully reboot the user, "Zero Config" needs to be enabled on the UCM6200 web GUI->**PBX->Zero Config->Auto Provisioning Settings**.

- **Delete single extension**

  Click on to delete the extension. Or select the checkbox of the extension and then click on "Delete Selected Extensions".

- **Modify selected extensions**

  Select the checkbox for the extension(s). Then click on "Modify Selected Extensions" to edit the extensions in a batch.

- **Delete selected extensions**

  Select the checkbox for the extension(s). Then click on "Delete Selected Extensions" to delete the extension(s).

EXPORT EXTENSIONS

The extensions configured on the UCM6200 can be exported to csv format file with selected technology "SIP", "IAX" or "FXS". Click on "Export Extensions" button and select technology in the prompt below.

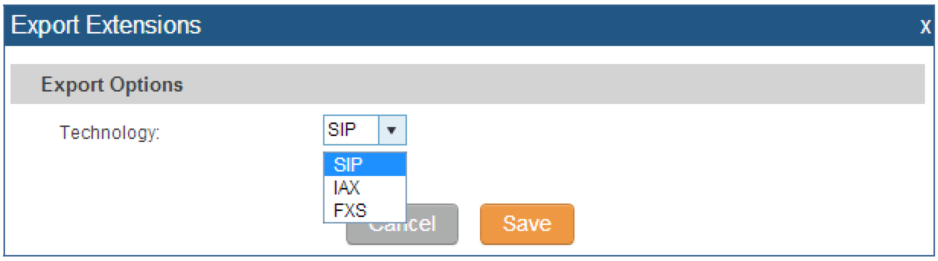


Figure 85: Export Extensions

The exported csv file can be serve as a template for users to fill in desired extension information to be imported to the UCM6200.

IMPORT EXTENSIONS

The capability to import extensions to the UCM6200 provides users flexibility to batch add extensions with similar or different configuration quickly into the PBX system.

1. Export extension csv file from the UCM6200 by clicking on "Export Extensions" button.
2. Fill up the extension information you would like in the exported csv template.
3. Click on "Import Extensions" button. The following dialog will be prompted.

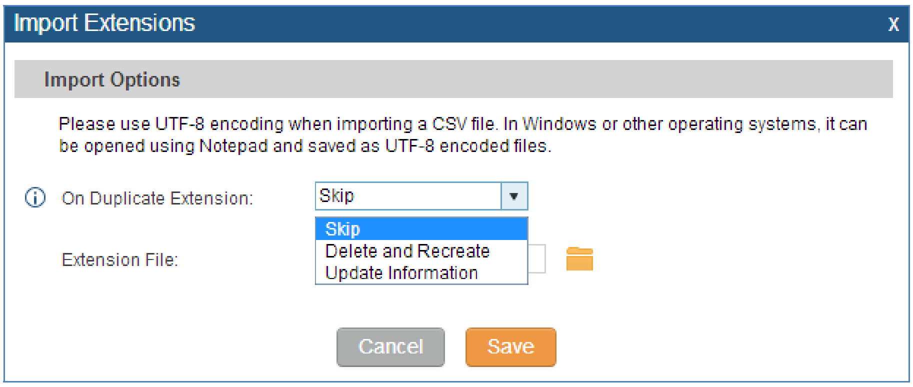


Figure 86: Import Extensions

4. Select the option in "On Duplicate Extension" to define how the duplicate extension(s) in the imported csv file should be treated by the PBX.

- Skip: Duplicate extensions in the csv file will be skipped. The PBX will keep the current extension information as previously configured without change.

- Delete and Recreate: The current extension previously configured will be deleted and the duplicate extension in the csv file will be loaded to the PBX.
- Update Information: The current extension previously configured in the PBX will be kept. However, if the duplicate extension in the csv file has different configuration for any options, it will override the configuration for those options in the extension.

5. Click on to select csv file from local directory in the PC.
6. Click on "Save" to import the csv file.
7. Click on "Apply Changes" to apply the imported file on the UCM6200.

EMAIL TO USER

Once the extensions are created with Email address, the PBX administrator can click on button “Email To User” to send the account registration and configuration information to the user. Please make sure Email setting under web UI->**Settings**->**Email Settings** is properly configured and tested on the UCM6200 before using “Email To User”.

When click on “Email To User” button, the following message will be prompted in the web page. Click on OK to confirm sending the account information to all users’ Email addresses.

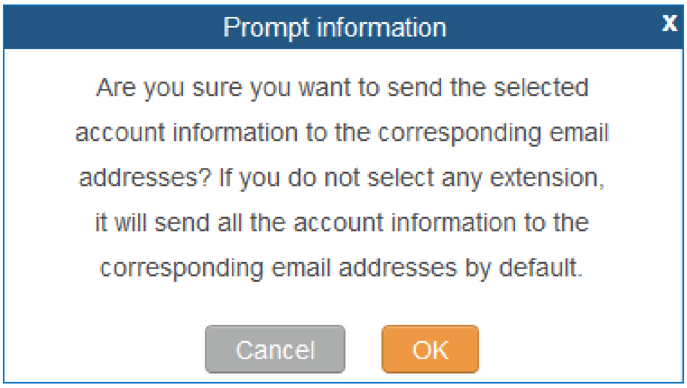


Figure 87: Email To User - Prompt Information

The user will receive Email including account registration information and LDAP configuration. A QR code is also generated for Mobile applications to scan it and get automatically provisioned. QR code provisioning is supported on Grandstream Softphone GS Wave Android™ application and iOS application.

Account Name : 1001
SIP Server : 192.168.2.1
SIP User ID : 1001
Authenticate ID : 1001
Authenticate Password : t*297eoS1h
Name :

This is the QR code of this account.

Figure 88: Account Registration Information and QR Code

Server Address : 192.168.2.1
Port : 389
Base : dc=pbx,dc=com
This is the QR code of this LDAP config.

Figure 89: LDAP Client Information and QR Code

MULTIPLE REGISTRATIONS PER EXTENSION

UCM6200 supports multiple registrations per extension so that users can use the same extension on devices in different locations.

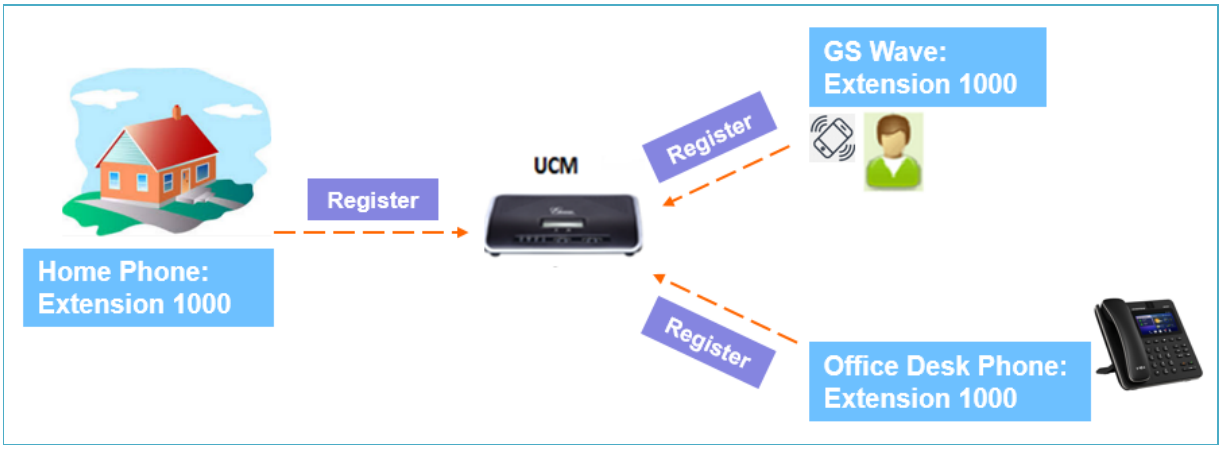


Figure 90: Multiple Registrations per Extension

This feature can be enabled by configuring option “Concurrent Registrations” under web **UI->PBX->Basic/Call Routes->Edit Extension**. The default value is set to 1 for security purpose.

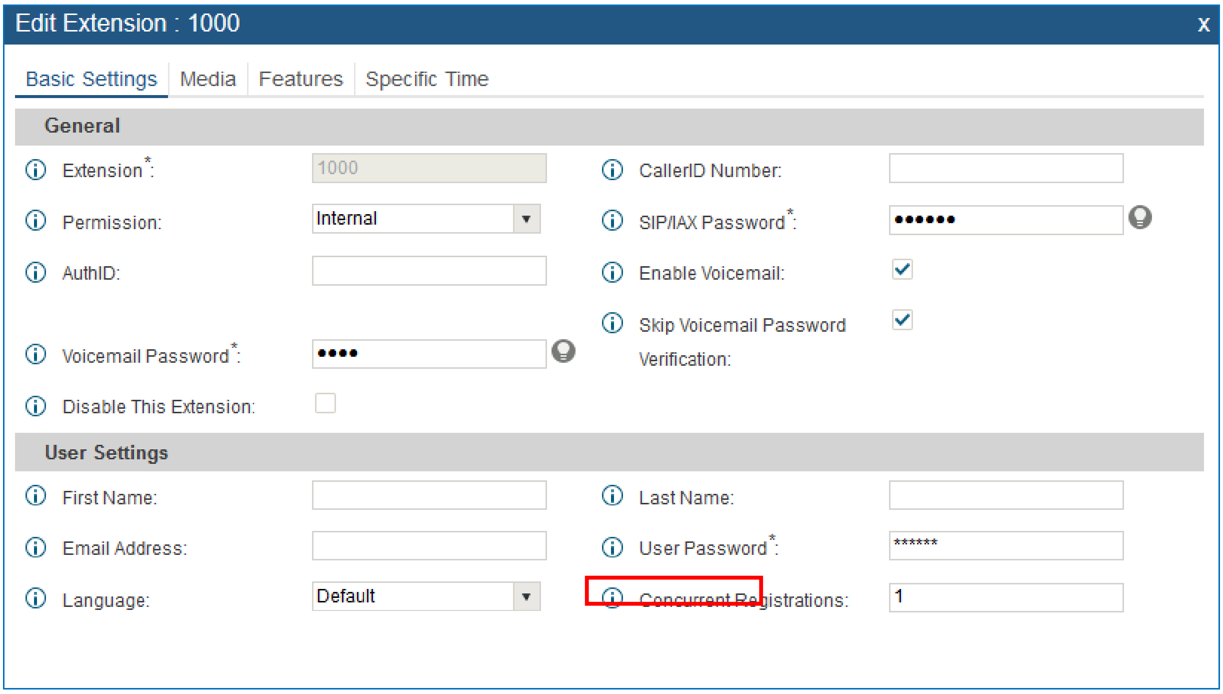


Figure 91: Extension - Concurrent Registration

SMS MESSAGE SUPPORT

The UCM6200 provides built-in SIP SMS message support. For SIP end devices such as Grandstream GXP or GXV phones that supports SIP message, after an UCM6200 account is registered on the end device, the user can send and receive SMS message. Please refer to the end device documentation on how to send and receive SMS message.

SMS Message support is a new feature added since firmware 1.0.10.x.

Figure 92: SMS Message Support

TRUNKS

ANALOG TRUNKS

Go to Web GUI->**PBX**->**Basic/Call Routes**->**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.

ANALOG TRUNK CONFIGURATION

The analog trunk options are listed in the table below.

Table 47: Analog Trunk Configuration Parameters

Channels	Select the channel for the analog trunk. • UCM6202: 2 channels • UCM6204: 4 channels • UCM6208: 8 channels
Trunk Name	Specify a unique label to identify the trunk when listed in outbound rules, incoming rules and etc.
SLA Mode	Enable this option to satisfy two primary use cases, which include emulating a simple key system and creating shared extensions on a PBX. Enable SLA Mode will disable polarity reversal.
Barge Allowed	The barge option specifies whether or not other stations are allowed to join a call in progress on this trunk. If enabled, the other stations can press the line button to join the call. The default setting is Yes.
Hold Access	The hold option specifies hold permissions for this trunk. If set to “Open”, any station can place this trunk on hold and any other station is allowed to retrieve the call. If set to “Private”, only the station that places the call on hold can retrieve the call. The default setting is Yes.
Advanced Options	
Enable Polarity Reversal	If enabled, a polarity reversal will be marked as received when an outgoing call is answered by the remote party. For some countries, a polarity reversal is used for signaling the disconnection of a phone line and the call will be considered as “hangup” on a polarity reversal. The

	default setting is "No".
Polarity on Answer Delay	When FXO port answers the call, FXS may send a Polarity Reversal. If this interval is shorter than the value of "Polarity on Answer Delay", the Polarity Reversal will be ignored. Otherwise, the FXO will onhook to disconnect the call. The default setting is 600ms.
Current Disconnect Threshold (ms)	This is the periodic time (in ms) that the UCM6200 will use to check on a voltage drop in the line. The default setting is 200. The valid range is 50 to 3000.
Ring Timeout	Configure the ring timeout (in ms). Trunk (FXO) devices must have a timeout to determine if there was a hangup before the line is answered. This value can be used to configure how long it takes before the UCM6200 considers a non-ringing line with hangup activity. The default setting is 8000.
RX Gain	Configure the RX gain for the receiving channel of analog FXO port. The valid range is from -13.5 (dB) to + 12.0 (dB). The default setting is 0.
TX Gain	Configure the TX gain for the transmitting channel of analog FXO port. The valid range is from -13.5 (dB) to + 12.0 (dB). The default setting is 0.
Use CallerID	Configure to enable CallerID detection. The default setting is "Yes".
Caller ID Scheme	Select the Caller ID scheme for this trunk. The default setting is "Bellcore/Telcordia".
FXO Dial Delay(ms)	Configure the time interval between off-hook and first dialed digit for outbound calls.
Auto Record	Enable automatic recording for the calls using this trunk. The default setting is disabled. The recording files can be accessed under web GUI->**CDR**->**Recording Files**.
Disable This Trunk	If selected, the trunk will be disabled.
DAHDI Out Line Selection	This is to implement analog trunk outbound line selection strategy. Three options are available: • Ascend When the call goes out from this analog trunk, it will always try to use the first idle FXO port. The port order that the call will use to go out would be port 1->port 2->port 10->port 16. Every time it will start with port 1 (if it's idle). • Poll When the call goes out from this analog trunk, it will use the port that is not used last time. And it will always use the port in the order of port 1->2->10->16->1->2->10->16->1->2->10->16..., following the last port being used. • Descend

	When the call goes out from this analog trunk, it will always try to use the last idle FXO port. The port order that the call will use to go out would be port 16->port 10->port 2->port 1. Every time it will start with port 16 (if it's idle). The default setting is “Ascend” mode.
Tone Settings	
Busy Detection	Busy Detection is used to detect far end hangup or for detecting busy signal. The default setting is "Yes".
Busy Tone Count	If "Busy Detection" is enabled, users can specify the number of busy tones to be played before hanging up. The default setting is 2. Better results might be achieved if set to 4, 6 or even 8. Please note that the higher the number is, the more time is needed to hangup the channel. However, this might lower the probability to get random hangup.
Congestion Detection	Congestion detection is used to detect far end congestion signal. The default setting is "Yes".
Congestion Count	If "Congestion Detection" is enabled, users can specify the number of congestion tones to wait for. The default setting is 2.
Tone Country	Select the country for tone settings. If "Custom" is selected, users could manually configure the values for Busy Tone and Congestion Tone. The default setting is "United States of America (USA)".
Busy Tone	**Syntax:** f1=val[@level][,f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]]; Frequencies are in Hz and cadence on and off are in ms. Frequencies Range: [0, 4000) Busy Level Range: (-300, 0) Cadence Range: [0, 16383]. Select Tone Country "Custom" to manually configure Busy Tone value. **Default value:** f1=480@-50,f2=620@-50,c=500/500
Congestion Tone	**Syntax:** f1=val[@level][,f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]]; Frequencies are in Hz and cadence on and off are in ms. Frequencies Range: [0, 4000) Busy Level Range: (-300, 0) Cadence Range: [0, 16383]. Select Tone Country "Custom" to manually configure Busy Tone value. **Default value:** f1=480@-50,f2=620@-50,c=250/250

PSTN Detection	Click on "Detect" to detect the busy tone, Polarity Reversal and Current Disconnect by PSTN. Before the detecting, please make sure there are more than one channel configured and working properly. If the detection has busy tone, the "Tone Country" option will be set as "Custom".

PSTN DETECTION

The UCM6200 provides PSTN detection function to help users detect the busy tone, Polarity Reversal and Current Disconnect by making a call from the PSTN line to another destination. The detecting call will be answered and up for about 1 minute. Once done, the detecting result will show and can be used for the UCM6200 settings.

1. Go to UCM6200 web GUI->**PBX**->**Basic/Call Routes**->**Analog Trunks** page.
2. Click to edit the analog trunk created for the FXO port.
3. In the dialog window to edit the analog trunk, go to "Tone Settings" section and there are two methods to set the busy tone.

- Tone Country. The default setting is "United States of America (USA)".
- PSTN Detection.

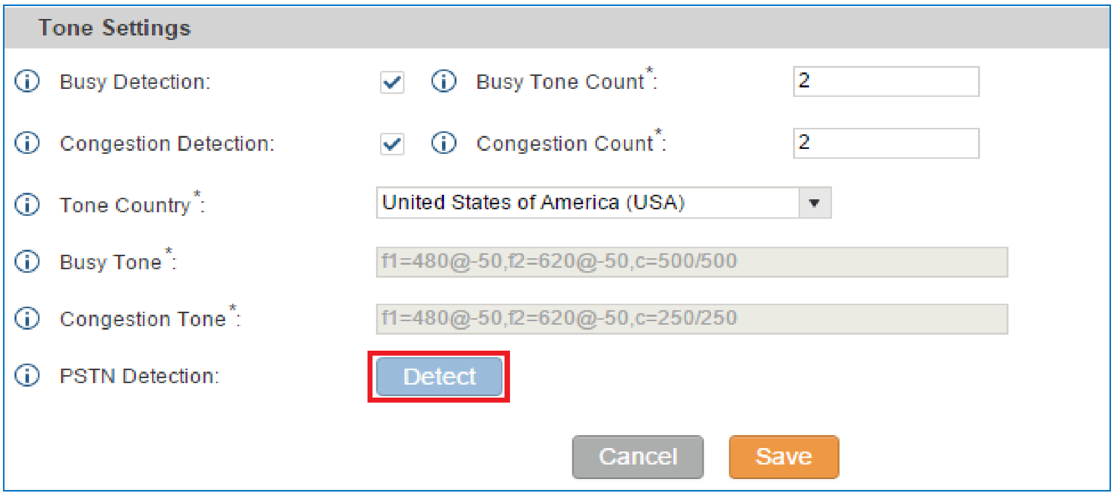


Figure 93: UCM6200 FXO Tone Settings

4. Click on "Detect" to start PSTN detection.

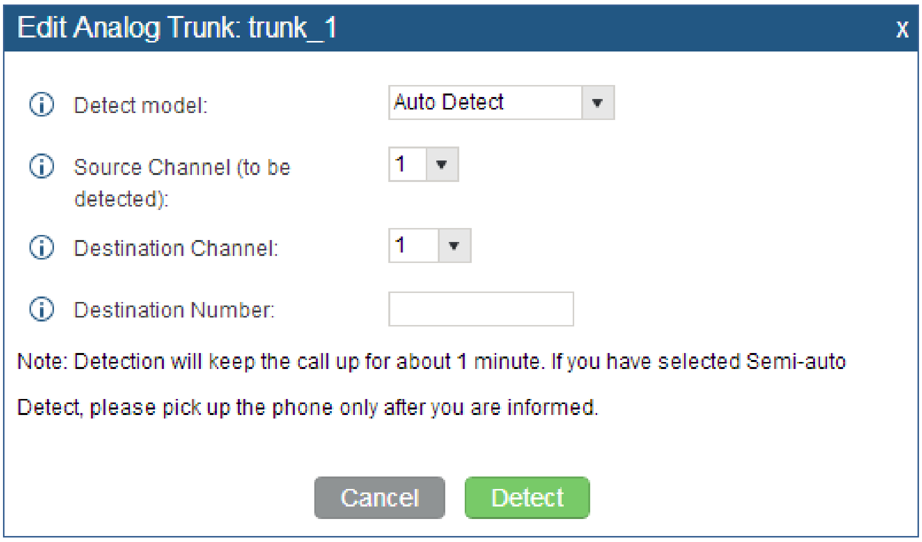


Figure 94: UCM6200 PSTN Detection

- If there are two FXO ports connected to PSTN lines, use the following settings for auto-detection.

  Detect Model: Auto Detect.
  Source Channel: The source channel to be detected.
  Destination Channel: The channel to help detecting. For example, the second FXO port.
  Destination Number: The number to be dialed for detecting. This number must be the actual PSTN number for the FXO port used as the destination channel.

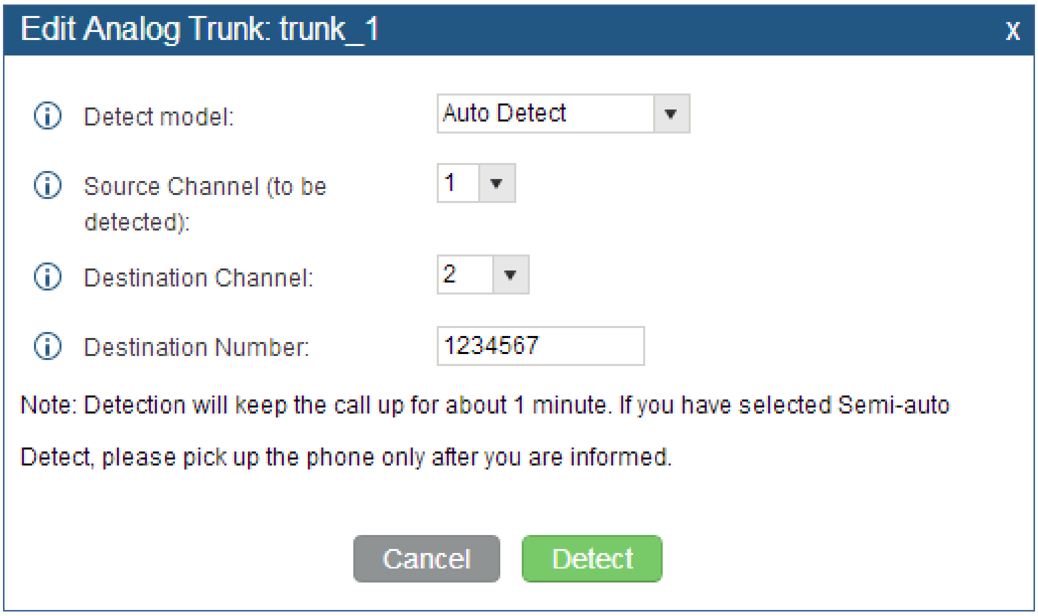


Figure 95: UCM6200 PSTN Detection: Auto Detect

- If there is only one FXO port connected to PSTN line, use the following settings for auto-detection.

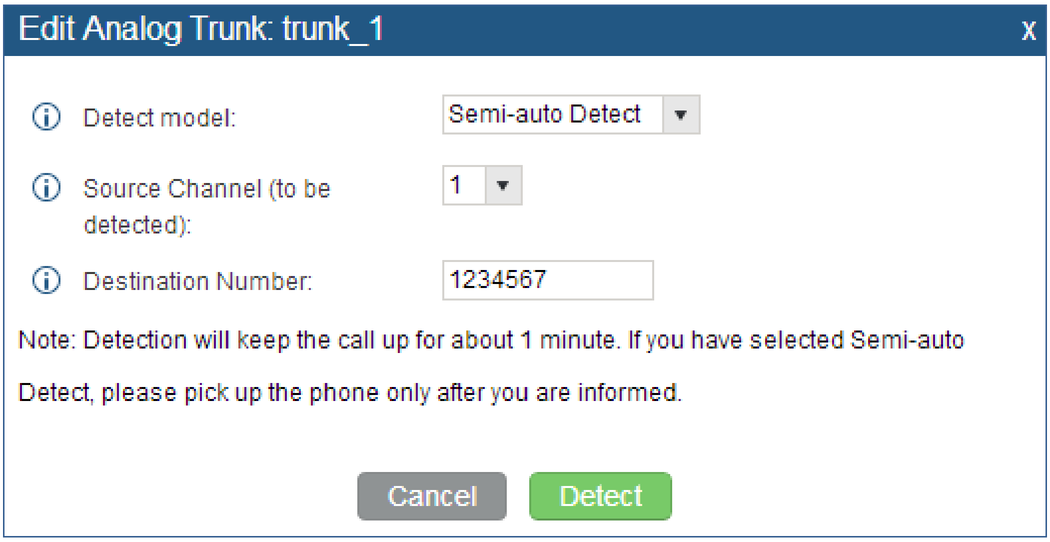


Figure 96: UCM6200 PSTN Detection: Semi-Auto Detect

Detect Model: Semi-auto Detect.
Source Channel: The source channel to be detected.
Destination Number: The number to be dialed for detecting. This number could be a cell phone number or other PSTN number that can be reached from the source channel PSTN number.

5. Click "Detect" to start detecting. The source channel will initiate a call to the destination number. For "Auto Detect", the call will be automatically answered. For "Semi-auto Detect", the UCM6200 web GUI will display prompt to notify the user to answer or hang up the call to finish the detecting process.

6. Once done, the detected result will show. Users could save the detecting result as the current UCM6200 settings.

Table 48: PSTN Detection for Analog Trunk

Detect Model	Select "Auto Detect" or "Semi-auto Detect" for PSTN detection. • Auto Detect Please make sure two or more channels are connected to the UCM6200 and in idle status before starting the detection. During the detection, one channel will be used as caller (Source Channel) and another channel will be used as callee (Destination Channel). The UCM6200 will control the call to be established and hang up between caller and callee to finish the detection. • Semi-auto Detect Semi-auto detection requires answering or hanging up the call

	manually. Please make sure one channel is connected to the UCM6200 and in idle status before starting the detection. During the detection, source channel will be used as caller and send the call to the configured Destination Number. Users will then need follow the prompts in web GUI to help finish the detection. The default setting is "Auto Detect".
Source Channel	Select the channel to be detected.
Destination Channel	Select the channel to help detect when "Auto Detect" is used.
Destination Number	Configure the number to be called to help the detection.

⚠ **Note:**

- The PSTN detection process will keep the call up for about 1 minute.
- If "Semi-auto Detect' is used, please pick up the call only after informed from the web GUI prompt.
- Once the detection is successful, the detected parameters "Busy Tone", "Polarity Reversal" and "Current Disconnect by PSTN" will be filled into the corresponding fields in the analog trunk configuration.

VOIP TRUNKS

VoIP trunks can be configured in UCM6200 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 SIP Trunk.
- Click on to start LDAP Sync.
- Click on to delete the VoIP trunk.

For VoIP trunk example, please refer to the document in the following link:
http://www.grandstream.com/sites/default/files/Resources/ucm_to_ucm_peer_guide.pdf

The VoIP trunk options are listed in the table below.

Table 49: Create New SIP Trunk

Type	Select the VoIP trunk type. • Peer SIP Trunk • Register SIP Trunk
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Keep Original CID	Keep the CID from the inbound call when dialing out. This setting will override "Keep Trunk CID" option. Please make sure that the peer PBX at the other side supports to match user entry using "username" field from authentication line.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
NAT	Turn on this setting when the PBX is using public IP and communicating with devices behind NAT. If there is one-way audio issue, usually it is related to NAT configuration or SIP/RTP port support on the firewall.
Disable This Trunk	If checked, the trunk will be disabled. **Note:** If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
TEL URI	If the trunk has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled.
Caller ID	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored. When making outgoing calls, the following rules are used to determine which CallerID will be used if they exist: • The CallerID configured for the extension will be looked up first.

	• If no CallerID is configured for the extension, the CallerID configured for the trunk will be used. • If the above two are missing, the "Global Outbound CID" defined in Web GUI->**PBX**->**Internal Options**->**General** will be used.
Need Registration	Select whether the trunk needs to register on the external server or not when "Register SIP Trunk" type is selected. The default setting is No.
Username	Enter the username to register to the trunk from the provider when "Register SIP Trunk" type is selected.
Password	Enter the password to register to the trunk from the provider when "Register SIP Trunk" is selected.
Auth ID	Enter the Authentication ID for "Register SIP Trunk" type.
Auto Record	Enable automatic recording for the calls using this trunk (for SIP trunk only). The default setting is disabled. The recording files can be accessed under web GUI->**CDR**->**Recording Files**.

Table 50: SIP Register Trunk Configuration Parameters

Basic Settings	
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Transport	Configure the SIP transport protocol to be used in this trunk. The default setting is "All - UDP Primary". • UDP Only • TCP Only • TLS Only • All - UDP Primary: UDP is the primary transport protocol when all the other SIP transport methods are available too. • All - TCP Primary: TCP is the primary transport protocol when all the other SIP transport methods are available too. • All – TLS Primary: TLS is the primary transport protocol when all the other SIP transport methods are available too.
Keep Original CID	Keep the CID from the inbound call when dialing out. This setting will override "Keep Trunk CID" option. Please make sure that the peer PBX at the other side supports to match user entry using "username" field from authentication line.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
NAT	Turn on this option when the PBX is using public IP and communicating

	with devices behind NAT. If there is one-way audio issue, usually it's related to NAT configuration or SIP/RTP port configuration on the firewall.
Disable This Trunk	If selected, the trunk will be disabled. **Note:** If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
TEL URI	If the trunk has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled.
Need Registration	Select whether the trunk needs to register on the external server or not when "Register SIP Trunk" type is selected. The default setting is No.
Username	Enter the username to register to the trunk from the provider when "Register SIP Trunk" type is selected.
Password	Enter the password to register to the trunk from the provider when "Register SIP Trunk" is selected.
Auth ID	Enter the Authentication ID for "Register SIP Trunk" type.
Auto Record	Enable automatic recording for the calls using this trunk (for SIP trunk only). The default setting is disabled. The recording files can be accessed under web GUI->**CDR**->**Recording Files**.
Advanced Settings	
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p and VP8.
From Domain	Configure the actual domain name where the extension comes from. This can be used to override the From Header. For example, "trunk.UCM6200.provider.com" is the From Domain in From Header: sip:1234567@trunk.UCM6200.provider.com.
From User	Configure the actual user name of the extension. This can be used to override the From Header. There are cases where there is a single ID for registration (single trunk) with multiple DIDs. For example, "1234567" is the From User in From Header: sip:1234567@trunk.UCM6200.provider.com.
Send PPI Header	If enabled, the SIP INVITE message sent to the trunk will contain PPI (P-Preferred-Identity) header. The default setting is "No". **Note:**

	"Send PPI Header" and "Send PAI Header" cannot be enabled at the same time. Only one of the two headers is allowed to be contained in the SIP INVITE message.
Send PAI Header	If enabled, the SIP INVITE message sent to the trunk will contain PAI (P-Asserted-Identity) header. The default setting is "No". **Note:** "Send PPI Header" and "Send PAI Header" cannot be enabled at the same time. Only one of the two headers is allowed to be contained in the SIP INVITE message.
Outbound Proxy Support	Select to enable outbound proxy in this trunk. The default setting is "No".
Outbound Proxy	When outbound proxy support is enabled, enter the IP address or URL of the outbound proxy.
DID Mode	Configure where to get the destination ID of an incoming SIP call, from SIP Request-line or To-header. The default is set to "Request-line".
DTMF Mode	Configure the default DTMF mode when sending DTMF on this trunk. • Default: The global setting of DTMF mode will be used. The global setting for DTMF Mode setting is under web UI->**PBX**->**SIP Settings**->**ToS**. • RFC2833: Send DTMF using RFC2833. • Info: Send DTMF using SIP INFO message. • Inband: Send DTMF using inband audio. This requires 64 bit codec, i.e., PCMU and PCMA. • Auto: Send DTMF using RFC2833 if offered. Otherwise, inband will be used.
Enable Qualify	If enabled, the UCM6200 will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
Qualify Timeout	When "Enable Qualify" option is set to "Yes", configure the timeout (in ms) for the Qualify SIP message. If no response is received within the timeout, the device is considered offline. The default setting is 1000ms.
Qualify Frequency	When "Enable Qualify" option is set to "Yes", configure the interval (in seconds) of the SIP OPTIONS message sent to the device to check if the device is still online. The default setting is 60 seconds.
Maximum Number of Call Lines	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limit.
Fax Mode	Select Fax mode. The default setting is "None". • None: Disable Fax. • Fax Detect: Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address

	configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under web UI->PBX->Internal Options->Fax/T.38.
SRTP	Enable SRTP for the VoIP trunk. The default setting is "No".
CC Settings	
Enable CC	If enabled, the system will automatically alert the user when a called party is available, given that a previous call to that party failed for some reason.
CC Max Agents	Configure the maximum number of CCSS agents which may be allocated for this channel. In other words, this number serves as the maximum number of CC requests this channel is allowed to make. The minimum value is 1.
CC Max Monitors	Configure the maximum number of monitor structures which may be created for this device. In other words, this number tells how many callers may request CC services for a specific device at one time. The minimum value is 1.

Table 51: SIP Peer Trunk Configuration Parameters

Basic Settings	
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Transport	Configure the SIP transport protocol to be used in this trunk. The default setting is "All - UDP Primary". • UDP Only • TCP Only • TLS Only • All - UDP Primary: UDP is the primary transport protocol when all the other SIP transport methods are available too. • All - TCP Primary: TCP is the primary transport protocol when all the other SIP transport methods are available too. • All – TLS Primary: TLS is the primary transport protocol when all the other SIP transport methods are available too.
Keep Original CID	Keep the CID from the inbound call when dialing out, this setting will override "Keep Trunk CID" option. Please make sure that the peer PBX at the other side supports to match user entry using "username" field from authentication line.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".

NAT	Turn on this option when the PBX is using public IP and communicating awith devices behind NAT. If there is one-way audio issue, usually it's related to NAT configuration or SIP/RTP port configuration on the firewall.
Disable This Trunk	If selected, the trunk will be disabled. **Note:** If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
TEL URI	If the trunk has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled.
Caller ID	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored. When making outgoing calls, the following rules are used to determine which CallerID will be used if they exist: • The CallerID configured for the extension will be looked up first. • If no CallerID configured for the extension, the CallerID configured for the trunk will be used. • If the above two are missing, the "Global Outbound CID" defined in Web GUI->**PBX**->**Internal Options**->**General** will be used.
CallerID Name	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
Auto Record	Enable automatic recording for the calls using this trunk (for SIP trunk only). The default setting is disabled. The recording files can be accessed under web GUI->**CDR**->**Recording Files**.
Advanced Settings	
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p and VP8.
DID Mode	Configure where to get the destination ID of an incoming SIP call, from SIP Request-line or To-header. The default is set to "Request-line".
DTMF Mode	Configure the default DTMF mode when sending DTMF on this trunk. • Default: The global setting of DTMF mode will be used. The global setting for DTMF Mode setting is under web UI->**PBX**->**SIP Settings**->**ToS**. • RFC2833: Send DTMF using RFC2833.

	• Info: Send DTMF using SIP INFO message. • Inband: Send DTMF using inband audio. This requires 64 bit codec, i.e., PCMU and PCMA. • Auto: Send DTMF using RFC2833 if offered. Otherwise, inband will be used.
Enable Qualify	If enabled, the UCM6200 will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
Qualify Timeout	When "Enable Qualify" option is set to "Yes", configure the timeout (in ms) for the Qualify SIP message. If no response is received within the timeout, the device is considered offline. The default setting is 1000ms.
Qualify Frequency	When "Enable Qualify" option is set to "Yes", configure the interval (in seconds) of the SIP OPTIONS message sent to the device to check if the device is still online. The default setting is 60 seconds.
Maximum Number of Call Lines	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limite.
Fax Mode	Select Fax mode. The default setting is “None”. • None: Disable Fax. • Fax Detect: Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under web UI->PBX->Internal Options->Fax/T.38.
SRTP	Enable SRTP for the VoIP trunk. The default setting is "No".
Sync LDAP Enable	If enabled, the local UCM6200 will automatically provide and update the local LDAP contacts to the remote UCM6200 SIP peer trunk. In order to ensure successful synchronization, the remote UCM6200 peer also needs to enable this option on the SIP peer trunk. The default setting is "No".
Sync LDAP Password	This is the password used for LDAP contact file encryption and decryption during the LDAP sync process. The password must be the same on both UCM6200 peers o ensure successful synchronization.
Sync LDAP Port	Configure the TCP port used LDAP sync feature between two peer UCM6200.
LDAP Outbound Rule	Specify an outbound rule for LDAP sync feature. The UCM6200 will automatically modify the remote contacts by adding prefix parsed from this rule.
LDAP Dialed Prefix	Specify the prefix for LDAP sync feature. The UCM6200 will automatically modify the remote contacts by adding this prefix.
CC Settings	

Enable CC	If enabled, the system will automatically alert the user when a called party is available, given that a previous call to that party failed for some reason.
CC Max Agents	Configure the maximum number of CCSS agents which may be allocated for this channel. In other words, this number serves as the maximum number of CC requests this channel is allowed to make. The minimum value is 1.
CC Max Monitors	Configure the maximum number of monitor structures which may be created for this device. In other words, this number tells how many callers may request CC services for a specific device at one time. The minimum value is 1.

Table 52: Create New IAX Trunk

Type	Select the VoIP trunk type. • Peer IAX Trunk • Register IAX Trunk
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
Username	Enter the username to register to the trunk from the provider when "Register IAX Trunk" type is selected.
Password	Enter the password to register to the trunk from the provider when "Register IAX Trunk" type is selected.
Disable This Trunk	If selected, the trunk will be disabled.

Table 53: IAX Register Trunk Configuration Parameters

Basic Settings	
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
Disable This Trunk	If selected, the trunk will be disabled.
Caller ID	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible

	to set the CallerID with this option and this option will be ignored. When making outgoing calls, the following rules are used to determine which CallerID will be used if they exist: • The CallerID configured for the extension will be looked up first. • If no CallerID configured for the extension, the CallerID configured for the trunk will be used. • If the above two are missing, the "Global Outbound CID" defined in Web GUI->**PBX**->**Internal Options**->**General** will be used.
CallerID Name	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
Username	Enter the username to register to the trunk from the provider.
Password	Enter the password to register to the trunk from the provider.
Advanced Settings	
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p and VP8.
Enable Qualify	If enabled, the UCM6200 will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
Qualify Timeout	When "Enable Qualify" option is set to "Yes", configure the timeout (in ms) for the Qualify SIP message. If no response is received within the timeout, the device is considered offline. The default setting is 1000ms.
Qualify Frequency	When "Enable Qualify" option is set to "Yes", configure the interval (in seconds) of the SIP OPTIONS message sent to the device to check if the device is still online. The default setting is 60 seconds.
Maximum Number of Call Lines	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limited.
Fax Mode	Select Fax mode. The default setting is “None”. • None: Disable Fax. • Fax Detect: Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under web UI->PBX->Internal Options->Fax/T.38.

Table 54: IAX Peer Trunk Configuration Parameters

Basic Settings	
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.

Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
Disable This Trunk	If selected, the trunk will be disabled.
Caller ID	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored. When making outgoing calls, the following rules are used to determine which CallerID will be used if they exist: • The CallerID configured for the extension will be looked up first. • If no CallerID configured for the extension, the CallerID configured for the trunk will be used. • If the above two are missing, the "Global Outbound CID" defined in Web GUI->**PBX**->**Internal Options**->**General** will be used.
CallerID Name	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
Advanced Settings	
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p and VP8.
Enable Qualify	If enabled, the UCM6200 will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
Qualify Timeout	When "Enable Qualify" option is set to "Yes", configure the timeout (in ms) for the Qualify SIP message. If no response is received within the timeout, the device is considered offline. The default setting is 1000ms.
Qualify Frequency	When "Enable Qualify" option is set to "Yes", configure the interval (in seconds) of the SIP OPTIONS message sent to the device to check if the device is still online. The default setting is 60 seconds.
Maximum Number of Call Lines	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limited.
Fax Mode	Select Fax mode. The default setting is “None”. • None: Disable Fax. • Fax Detect: Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under web UI->PBX->Internal Options->Fax/T.38.

DIRECT OUTWARD DIALING (DOD)

The UCM6200 provides Direct Outward Dialing (DOD) which is a service of a local phone company (or local exchange carrier) that allows subscribers within a company's PBX system to connect to outside lines directly.

Example of how DOD is used:

Company ABC has a SIP trunk. This SIP trunk has 4 DIDs associated to it. The main number of the office is routed to an auto attendant. The other three numbers are direct lines to specific users of the company. At the moment when a user makes an outbound call their caller ID shows up as the main office number. This poses a problem as the CEO would like their calls to come from their direct line. This can be accomplished by configuring DOD for the CEO's extension.

Steps on how to configure DOD on the UCM6200:

1. To setup DOD go to UCM6200 web GUI->**PBX**->**Basic/Call Routes**->**VoIP Trunks** page.
2. Click DOD to access the DOD options for the selected SIP Trunk.
3. Click "Create a new DOD" to begin your DOD setup
4. For "DOD Number" enter one of the numbers (DIDs) from your SIP trunk provider. In the example above Company ABC received 4 DIDs from their provider. ABC will enter in the number for the CEO's direct line.
5. Select an extension from the "Available Extensions" list. Users have the option of selecting more than one extension. In this case, Company ABC would select the CEO's extension. After making the selection, click on the button to move the extension(s) to the "Selected Extensions" list.

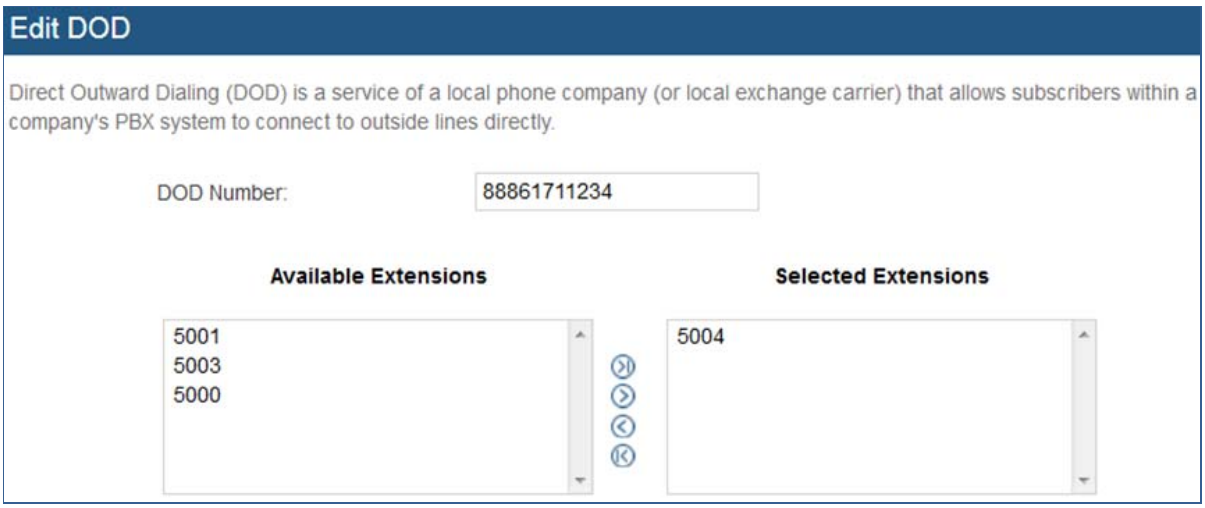


Figure 97: DOD extension selection

6. Click "Save" at the bottom.
   Once completed, the user will return to the EDIT DOD page that shows all the extensions that are associated to a particular DOD.

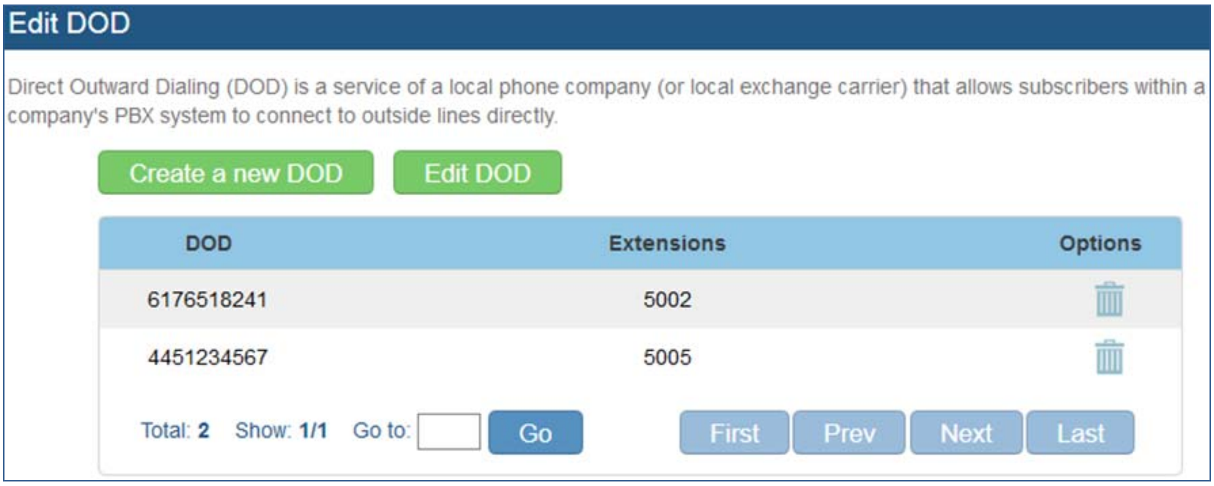


Figure 98: Edit DOD

SLA STATION

The UCM6200 supports SLA that allows mapping the key with LED on a multi-line phone to different external lines. When there is an incoming call and the phone starts to ring, the LED on the key will flash in red and the call can be picked up by pressing this key. This allows users to know if the line is occupied or not. The SLA function on the UCM6200 is similar to BLF but SLA is used to monitor external line i.e., analog trunk on the UCM6200. Users could configure the phone with BLF mode on the MPK to monitor the analog trunk status or press the line key pick up call from the analog trunk on the UCM6200.

CREATE/EDIT SLA STATION

SLA Station can be configured on web GUI->**PBX->Basic/Call Routes->SLA Station.**

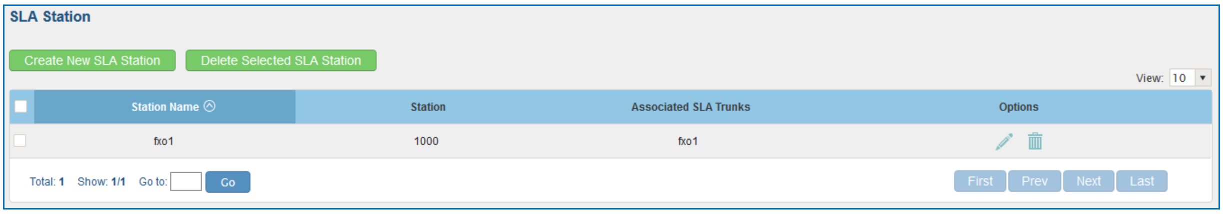


Figure 99: SLA Station

- Click on "Create New SLA Station" to add a SLA Station.
- Click on to edit the SLA Station. The following table shows the SLA Station configuration parameters.
- Click on to delete the SLA Station.

Table 55: SLA Station Configuration Parameters

Station Name	Configure a name to identify the SLA Station.
Station	Specify a SIP extension as a station that will be using SLA.
Available SLA Trunks	Existing Analog Trunks with SLA Mode enabled will be listed here.
Selected SLA Trunks	Select a trunk for this SLA from the Available SLA Trunks list. Click on to arrange the order. If there are multiple trunks selected, when there are calls on those trunks at the same time, pressing the LINE key on the phone will pick up the call on the first trunk here.
SLA Station Options	

Ring Timeout	Configure the time (in seconds) to ring the station before the call is considered unanswered. No timeout is set by default. If set to 0, there will be no timeout.
Ring Delay	Configure the time (in seconds) for delay before ringing the station when a call first coming in on the shared line. No delay is set by default. If set to 0, there will be no delay.
Hold Access	This option defines the competence of the hold action for one particular trunk. If set to "open", any station could hold a call on that trunk or resume one held session; if set to "private", only the station that places the trunk call on hold could resume the session. The default setting is "open".

SAMPLE CONFIGURATION

1. On the UCM6200, go to web UI->**Basic/Call Routes**->**Analog Trunks** page. Create analog trunk or edit the existing analog trunk. Make sure "SLA Mode" is enabled for the analog trunk. Once enabled, this analog trunk will be only available for the SLA stations created under web UI->**Basic/Call Routes**->**SLA Station** page.

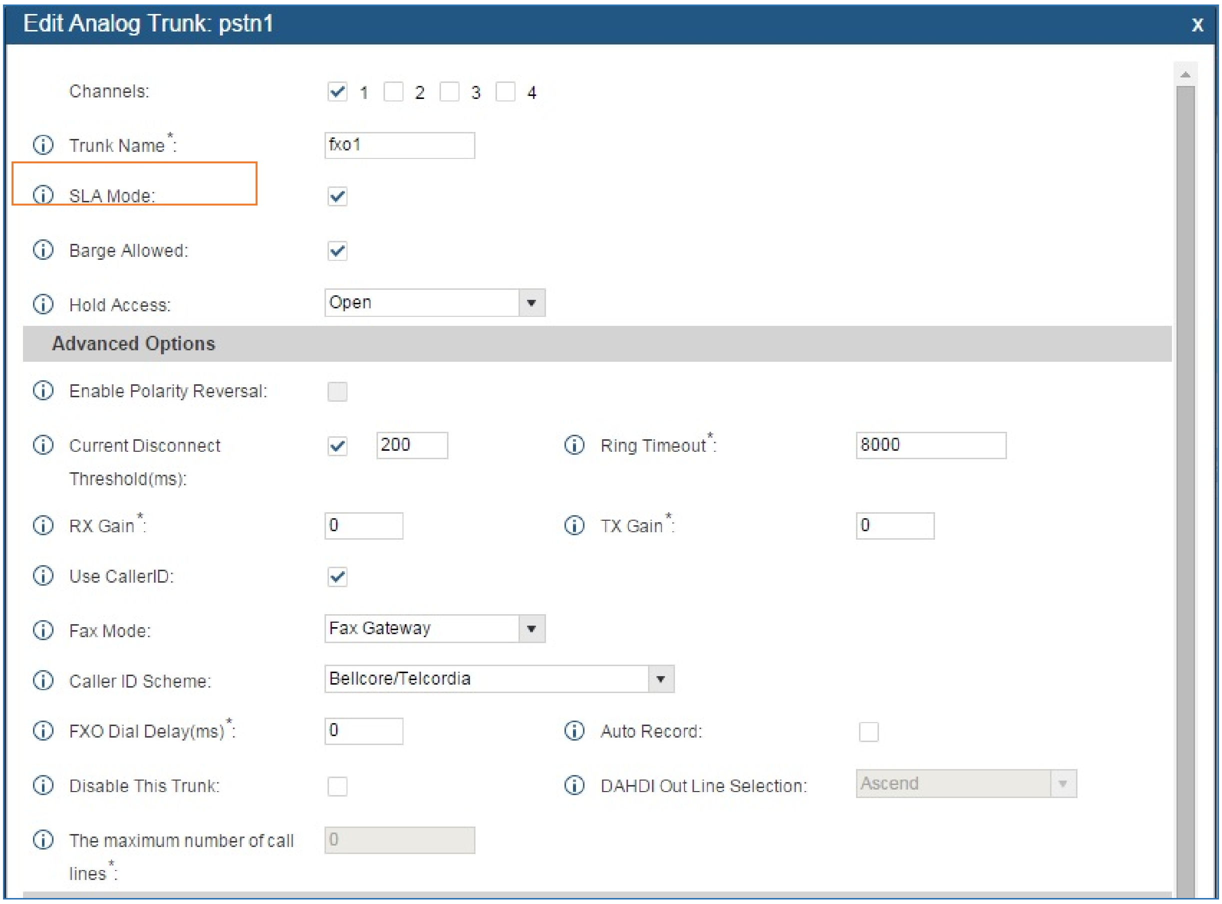


Figure 100: Enable SLA Mode for Analog Trunk

Click on "Save". The analog trunk will be listed with trunk mode "SLA".

Trunks	Trunk Mode	Analog Ports	Options
fxo1	SLA	1	

Figure 101: Analog Trunk with SLA Mode Enabled

2. On the UCM6200, go to web UI->**Basic/Call Routes**->**SLA Station** page, click on "Create New SLA Station". Please refer to section ***[CREATE/EDIT SLA STATION]*** for the configuration parameters. Users can create one or more SLA stations to monitor the analog trunk. The following figure shows two stations, 1002 and 1005, are configured to be associated with SLA trunk "fxo1".

Station Name	Station	Associated SLA Trunks	Options
sla2	1002	fxo1	
testsla	1005	fxo1	

Figure 102: SLA Example - SLA Station

3. On the SIP phone 1, configure to register UCM6200 extension 1002. Configure the MPK as BLF mode

and the value must be set to “extension_trunkname”, which is 1002_fxo1 in this case.

4. On the SIP phone 2, configure to register UCM6200 extension 1005. Configure the MPK as BLF mode and value must be set to “extension_trunkname”, which is 1005_fxo1 in this case.

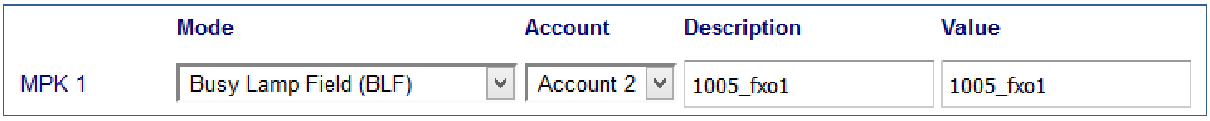


Figure 103: SLA Example - MPK Configuration

Now the SLA station is ready to use. The following functions can be achieved by this configuration.

- Making an outbound call from the station/extension, using LINE key
  When the extension is in idle state, pressing the line key for this extension on the phone to off hook. Then dial the station’s extension number, for example, dial 1002 on phone 1 (or dial 1005 on phone 2), to hear the dial tone. Then the users could dial external number for the outbound call.
- Making an outbound call from the station/extension, using BLF key
  When the extension is in idle state, pressing the MPK and users could dial external numbers directly.
- Answering call using LINE key
  When the station is ringing, pressing the LINE key to answer the incoming call.
- Barging-in active call using BLF key
  When there is an active call between an SLA station and an external number using the SLA trunk, other SLA stations monitoring the same trunk could join the call by pressing the BLF key if “Barge Allowed” is enabled for the analog trunk.
- Hold/Unhold using BLF key
  If the external line is previously put on hold by an SLA station, another station that monitors the same SLA trunk could unhold the call by pressing the BLF key if “Hold Access” is set to “open” on the analog trunk and the SLA station.