

# Grandstream Networks, Inc.

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UCM630xA Series

Enterprise-Grade Unified Communication Solutions

**User Manual**



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## DOCUMENT PURPOSE

The intent of this document is to provide device administrators an overview of the specifications and features of the Grandstream UCM630xA IPPBX system. To learn more about the UCM630xA, please visit <https://www.grandstream.com/support> to download additional guides.

This guide covers following main topics:

- [Product overview](#)
- [Installation](#)
- [Getting started](#)
- [System settings](#)
- [Provisioning](#)
- [Extensions](#)
- [Extension groups](#)
- [Analog Trunks](#)
- [VoIP Trunks](#)
- [SLA station](#)
- [Call routes](#)
- [Meeting room.](#)
- [Meeting schedule](#)
- [IVR](#)
- [Language settings for voice prompt](#)
- [Voicemail](#)
- [Ring group](#)
- [Paging and intercom group](#)
- [Call queue](#)
- [Pickup groups](#)
- [PIN Groups](#)
- [Music on hold](#)
- [Fax Server](#)
- [Busy camp-on](#)
- [Presence](#)
- [Follow me](#)
- [Speed Dial](#)
- [DISA](#)
- [Callback](#)
- [BLF and event list](#)
- [Dial by name](#)
- [Active calls and monitor](#)
- [Call features](#)
- [Call recording](#)
- [CTI Server](#)
- [Asterisk manager interface \(AMI\)](#)
- [CRM integration](#)
- [PMS integration](#)
- [Wakeup service](#)
- [Announcements center](#)
- [Status and reporting](#)
- [CDR \(Call Details Record\)](#)
- [User Portal](#)
- [Upgrading and maintenance](#)
- [Backup/restore](#)
- [Troubleshooting](#)



## CHANGE LOG

This section documents significant changes from previous versions of the UCM630xA 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 1.0.7.9**

- This is the initial version.



## WELCOME

Thank you for purchasing Grandstream UCM630xA series IP PBX appliance. The UCM6300A series allows businesses to build powerful and scalable unified communication and collaboration solutions. This series of IP PBXs provide a platform that unifies all business communication on one centralized network, including voice, video calling, voice meeting, video surveillance, web meetings, data, analytics, mobility, facility access, intercoms and more. The UCM6300A series supports up to 1500 users and includes a built-in web meetings and meeting solution that allows employees to connect from the desktop, mobile, GVC series devices and IP phones. It can be paired with the UCM6300A ecosystem to offer a hybrid platform that combines the control of an on-premises IP PBX with the remote access of a cloud solution. The UCM630xA ecosystem consists of the Wave app for desktop and mobile, which provides a hub for collaborating remotely, and UCM RemoteConnect, a cloud NAT traversal service for ensuring secure remote connections. The UCM6300A series also offers cloud setup and management through GDMS and an API for integration with third-party platforms. By offering a high-end unified communications and collaboration solution packed with a suite of mobility, security, meeting and collaboration tools, the UCM6300A series provides a powerful platform for any organization. The UCM6300A/6302A support 1 x ARM Cortex-A53 Quad-Core CPU and 2GB of memory while UCM6304A/6308A support 2 x ARM Cortex-A53 Quad-Core CPU and 2GB for master and 1GB slave memory.

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 **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 UCM630xA as it may cause damage to the product and void the manufacturer warranty.

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This document is subject to change without notice. The latest electronic version of this user manual is available for download here:

<https://www.grandstream.com/support>

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# PRODUCT OVERVIEW

## Technical Specifications

The following table resumes all the technical specifications including the protocols / standards supported, voice codecs, telephony features, languages, and upgrade/provisioning settings for UCM630xA series.

**Table 1: Technical Specifications**

Interfaces	
<b>Analog Telephone FXS Ports</b>	<ul style="list-style-type: none"> <li>UCM6300A: 0 port</li> <li>UCM6302A: 2 ports with lifeline support</li> <li>UCM6304A: 4 ports with lifeline support</li> <li>UCM6308A: 8 ports with lifeline support</li> </ul> Each port supports 3 REN
<b>PSTN Line FXO Ports</b>	<ul style="list-style-type: none"> <li>UCM6300A: 0 port</li> <li>UCM6302A: 2 ports</li> <li>UCM6304A: 4 ports</li> <li>UCM6308A: 8 ports</li> </ul> All ports have lifeline capability in case of power outage
<b>Network Interfaces</b>	Three self-adaptive Gigabit ports (switched, routed or dual card mode) with PoE+
<b>NAT Router</b>	Yes (supports router mode and switch mode)
<b>Peripheral Ports</b>	<ul style="list-style-type: none"> <li>UCM6300A: USB 3.0, and SD card interface</li> <li>UCM6302A: USB 2.0, USB 3.0, and SD card interface</li> <li>UCM6304A/6308A: 2x USB 3.0 and SD card interface</li> </ul>
<b>LED Indicators</b>	<ul style="list-style-type: none"> <li>UCM6300A/6302A/UCM6304A: None</li> <li>UCM6308A: Power 1/2, FXS, FXO, LAN, WAN, Heartbeat</li> </ul>
<b>LCD Display</b>	<ul style="list-style-type: none"> <li>UCM6300A/6302A/6304A: 320*240 LCD with touch screen for Shortcut Keys and Scroll Bar</li> <li>UCM6308A: 128x32 dot matrix graphic LCD with DOWN and OK buttons</li> </ul>
<b>Reset Switch</b>	Yes, long press for factory reset and short press for reboot
Voice Capabilities	
<b>Voice-over-Packet Capabilities</b>	LEC with NLP Packetized Voice Protocol Unit, 128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711, NetEQ, FEC 2.0, jitter resilience up to 50% audio packet loss
<b>Voice and Fax Codecs</b>	Opus, G.711 A-law/U-law, G.722, G722.1 G722.1C, G.723.1 5.3K/6.3K, G.726-32, G.729A/B, iLBC, GSM; T.38



<b>QoS</b>	Layer 2 QoS (802.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS
<b>Signaling and Control</b>	
<b>DTMF Methods</b>	Inband, RFC4733, and SIP INFO
<b>Provisioning Protocol and Plug-and-Play</b>	Mass provisioning using AES encrypted XML configuration file, auto-discovery & auto-provisioning of Grandstream IP endpoints via ZeroConfig (DHCP Option 66 multicast SIP SUBSCRIBE mDNS), eventlist between local and remote trunk
<b>Network Protocols</b>	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, STUN, SRTP, TLS, LDAP, HDLC, HDLC-ETH, PPP, Frame Relay (pending), IPv6, OpenVPN®
<b>API</b>	Full API available for third-party platform and application integration
<b>Disconnect Methods</b>	Busy/Congestion/Howl Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect
<b>Security</b>	
<b>Media Encryption</b>	SRTP, TLS1.2, HTTPS, SSH, 802.1x
<b>Physical</b>	
<b>Universal Power Supply</b>	<ul style="list-style-type: none"> <li>UCM6300A/6302A/6304A: Input: 100 ~ 240VAC, 50/60Hz; Output: DC+12V, 1.5A</li> <li>UCM6308A: 2x DC 12V Power Jack Input: 100~240VAC,50/60Hz; Output: DC+12V, 2A</li> </ul>
<b>Dimensions</b>	<ul style="list-style-type: none"> <li>UCM6300A/6302A/6304A: 270mm(L) x 175mm(W) x 36mm(H)</li> <li>UCM6308A: 485mm(L) x 187.2mm(W) x 46.2mm(H)</li> </ul>
<b>Environmental</b>	<ul style="list-style-type: none"> <li>Operating: 32 - 113°F / 0 ~ 45°C, Humidity 10 - 90% (non-condensing)</li> <li>Storage: 14 - 140°F / -10 ~ 60°C, Humidity 10 - 90% (non-condensing)</li> </ul>
<b>Mounting</b>	Wall mount (sold separately) & Desktop for UCM6300A/6302A/6304A and Desktop & Rack mount for UCM6308A.
<b>Weight</b>	<ul style="list-style-type: none"> <li>UCM6300A: Unit weight 705g, Package weight 1200g</li> <li>UCM6302A: Unit weight 725g, Package weight 1221g</li> <li>UCM6304A: Unit weight 775g, Package weight 1271g</li> <li>UCM6308A: Unit weight 2540g, Package weight 3465g</li> </ul>
<b>Additional Features</b>	
<b>Multi-language Support</b>	<ul style="list-style-type: none"> <li>Web UI: English, Simplified Chinese, Traditional Chinese, Spanish, French, Portuguese, German, Russian, Italian, Polish, Czech, Turkish</li> <li>Customizable IVR/voice prompts: English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic, Netherlands</li> <li>Customizable language pack to support any other languages</li> </ul>



<b>Caller ID</b>	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT, NTT								
<b>Polarity Reversal/ Wink</b>	Yes, with enable/disable option upon call establishment and termination								
<b>Call Center</b>	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/workload, in-queue announcement								
<b>Customizable Auto Attendant</b>	Up to 5 layers of IVR (Interactive Voice Response) in multiple languages								
<b>Telephony Operating System</b>	Based on Asterisk version 16								
<b>Maximum Call Capacity</b>	<table border="1"> <tr> <td><b>UCM6300A</b></td> <td> <ul style="list-style-type: none"> <li>Users: <b>250</b></li> <li>Concurrent calls (G.711): <b>50</b></li> <li>Max concurrent SRTP calls (G.711): <b>50</b></li> </ul> </td> </tr> <tr> <td><b>UCM6302A</b></td> <td> <ul style="list-style-type: none"> <li>Users: <b>500</b></li> <li>Concurrent calls (G.711): <b>75</b></li> <li>Max concurrent SRTP calls (G.711): <b>75</b></li> </ul> </td> </tr> <tr> <td><b>UCM6304A</b></td> <td> <ul style="list-style-type: none"> <li>Users: <b>1000</b></li> <li>Concurrent calls (G.711): <b>150</b></li> <li>Max concurrent SRTP calls (G.711): <b>120</b></li> </ul> </td> </tr> <tr> <td><b>UCM6308A</b></td> <td> <ul style="list-style-type: none"> <li>Users: <b>1500</b></li> <li>Concurrent calls (G.711): <b>200</b></li> <li>Max concurrent SRTP calls (G.711): <b>150</b></li> </ul> </td> </tr> </table>	<b>UCM6300A</b>	<ul style="list-style-type: none"> <li>Users: <b>250</b></li> <li>Concurrent calls (G.711): <b>50</b></li> <li>Max concurrent SRTP calls (G.711): <b>50</b></li> </ul>	<b>UCM6302A</b>	<ul style="list-style-type: none"> <li>Users: <b>500</b></li> <li>Concurrent calls (G.711): <b>75</b></li> <li>Max concurrent SRTP calls (G.711): <b>75</b></li> </ul>	<b>UCM6304A</b>	<ul style="list-style-type: none"> <li>Users: <b>1000</b></li> <li>Concurrent calls (G.711): <b>150</b></li> <li>Max concurrent SRTP calls (G.711): <b>120</b></li> </ul>	<b>UCM6308A</b>	<ul style="list-style-type: none"> <li>Users: <b>1500</b></li> <li>Concurrent calls (G.711): <b>200</b></li> <li>Max concurrent SRTP calls (G.711): <b>150</b></li> </ul>
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<b>Maximum Attendees of Meeting Bridges</b>	<table border="1"> <tr> <td><b>UCM6300A</b></td> <td> <ul style="list-style-type: none"> <li><b>3</b> Meeting rooms and up to <b>50</b> parties</li> </ul> </td> </tr> <tr> <td><b>UCM6302A</b></td> <td> <ul style="list-style-type: none"> <li><b>5</b> Meeting rooms and up to <b>75</b> parties</li> </ul> </td> </tr> <tr> <td><b>UCM6304A</b></td> <td> <ul style="list-style-type: none"> <li><b>7</b> Meeting rooms and up to <b>120</b> parties</li> </ul> </td> </tr> <tr> <td><b>UCM6308A</b></td> <td> <ul style="list-style-type: none"> <li><b>9</b> Meeting rooms and up to <b>150</b> parties</li> </ul> </td> </tr> </table>	<b>UCM6300A</b>	<ul style="list-style-type: none"> <li><b>3</b> Meeting rooms and up to <b>50</b> parties</li> </ul>	<b>UCM6302A</b>	<ul style="list-style-type: none"> <li><b>5</b> Meeting rooms and up to <b>75</b> parties</li> </ul>	<b>UCM6304A</b>	<ul style="list-style-type: none"> <li><b>7</b> Meeting rooms and up to <b>120</b> parties</li> </ul>	<b>UCM6308A</b>	<ul style="list-style-type: none"> <li><b>9</b> Meeting rooms and up to <b>150</b> parties</li> </ul>
	<b>UCM6300A</b>	<ul style="list-style-type: none"> <li><b>3</b> Meeting rooms and up to <b>50</b> parties</li> </ul>							
	<b>UCM6302A</b>	<ul style="list-style-type: none"> <li><b>5</b> Meeting rooms and up to <b>75</b> parties</li> </ul>							
	<b>UCM6304A</b>	<ul style="list-style-type: none"> <li><b>7</b> Meeting rooms and up to <b>120</b> parties</li> </ul>							
<b>UCM6308A</b>	<ul style="list-style-type: none"> <li><b>9</b> Meeting rooms and up to <b>150</b> parties</li> </ul>								





<b>Call Features</b>	Call park, call forward, call transfer, call waiting, caller ID, call record, call history, ringtone, IVR, music on hold, call routes, DID, DOD, DND, DISA, ring group, ring simultaneously, time schedule, PIN groups, call queue, pickup group, paging/intercom, voicemail, call wakeup, SCA, BLF, voicemail to email, speed dial, call back, dial by name, emergency call, call follow me, blacklist/whitelist, voice meeting, eventlist, feature codes, busy camp-on/ call completion, voice control
<b>Wave Mobile App</b>	Allows Android & iOS users to join UCM-hosted meetings & communicate with other users/solutions registered to the UCM6300A
<b>Firmware Upgrade</b>	Supported by Grandstream Device Management System (GDMS), a zero-touch cloud provisioning and management system, It provides a centralized interface to provision, manage, monitor, and troubleshoot Grandstream products
<b>Compliance</b>	<ul style="list-style-type: none"> <li>• FCC: Part 15 (CFR 47) Class B, Part 68</li> <li>• CE: EN 55032, EN 55035, EN61000-3-2, EN61000-3-3, EN 62368.1, ES 203 021, ITU K.21</li> <li>• IC: ICES-003, CS-03 Part I Issue 9</li> <li>• RCM: AS/NZS CISPR 32, AS/NZS 62368.1, AS/CA S002, AS/CA S003.1/.2</li> <li>• Power adapter: UL 60950-1 or UL 62368-1</li> </ul>

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 **Note:**

UCM630xA FXS ports lifeline functionality:

The UCM630xA 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 UCM630xA series, the device needs to be properly powered up and connected to a network. This section describes detailed information on installation, connection, and warranty policy of the UCM630xA series.

### Equipment Packaging

Table 2: UCM630xA Equipment Packaging

<b>Main Case</b>	1
<b>Power Adaptor</b>	1
<b>Ethernet Cable</b>	1
<b>Quick Installation Guide</b>	1

### UCM6300A front and back view

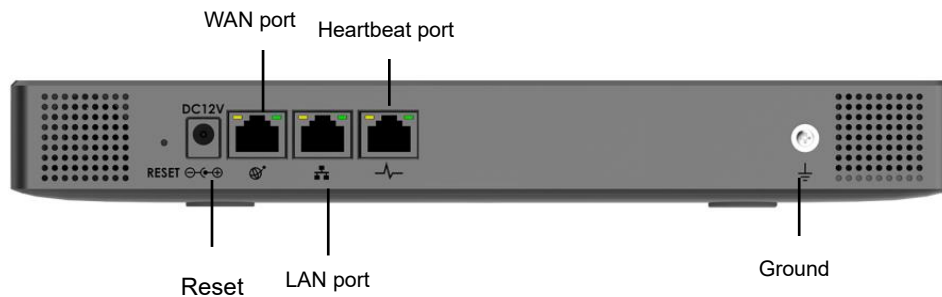


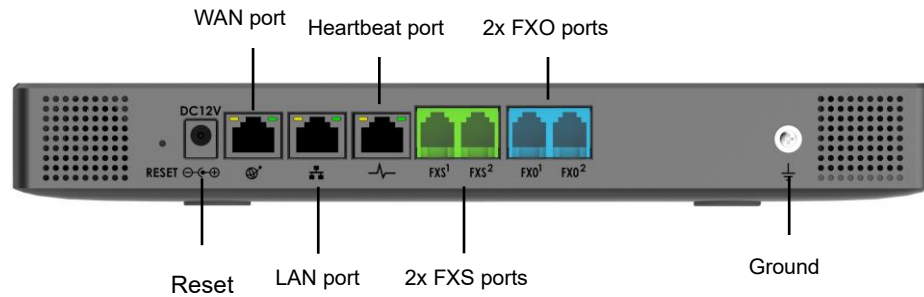
Figure 1: UCM6300A Back View



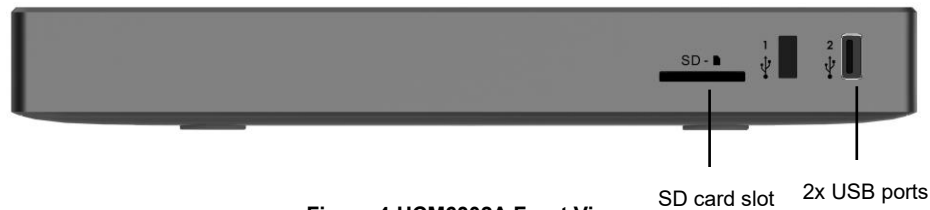
Figure 2: UCM6300A Front View



### UCM6302A front and back view

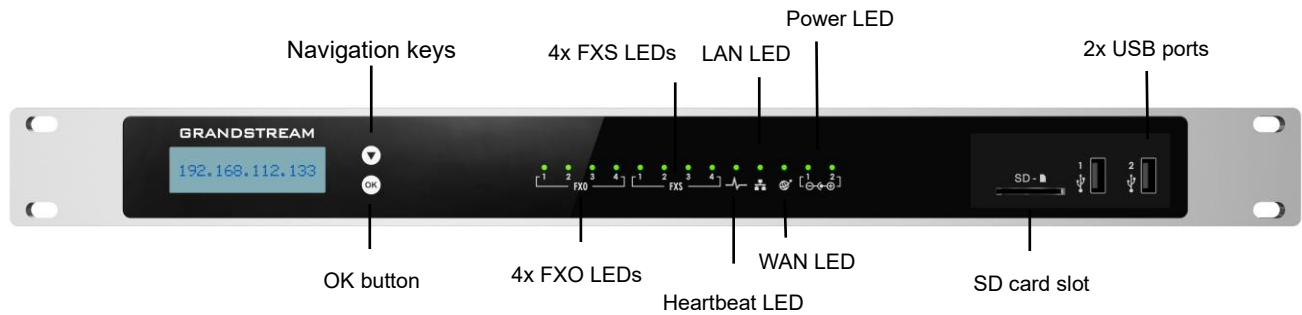


**Figure 3: UCM6302A Back View**

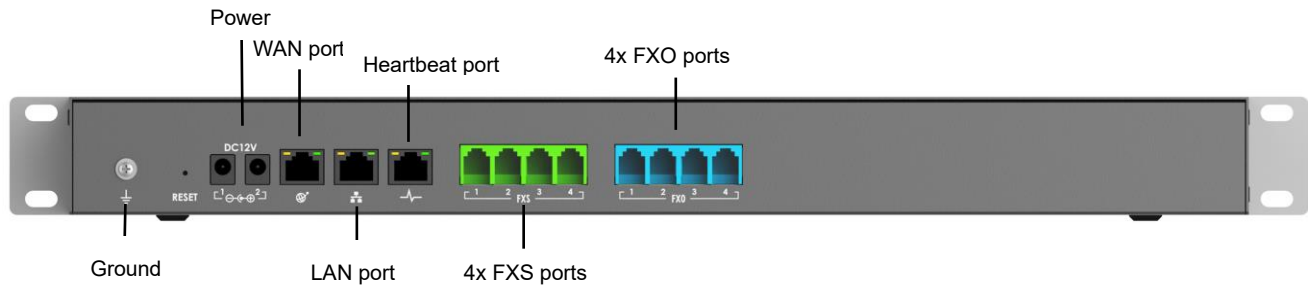


**Figure 4: UCM6302A Front View**

### UCM6304A front and back view



**Figure 5: UCM6304A Front View**



**Figure 6: UCM6304A Back View**



## UCM6308A front and back view

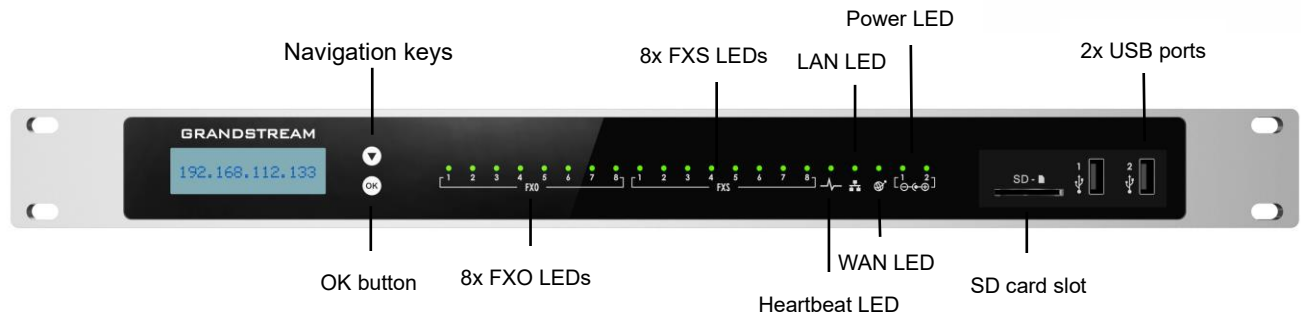


Figure 7: UCM6308A Front View

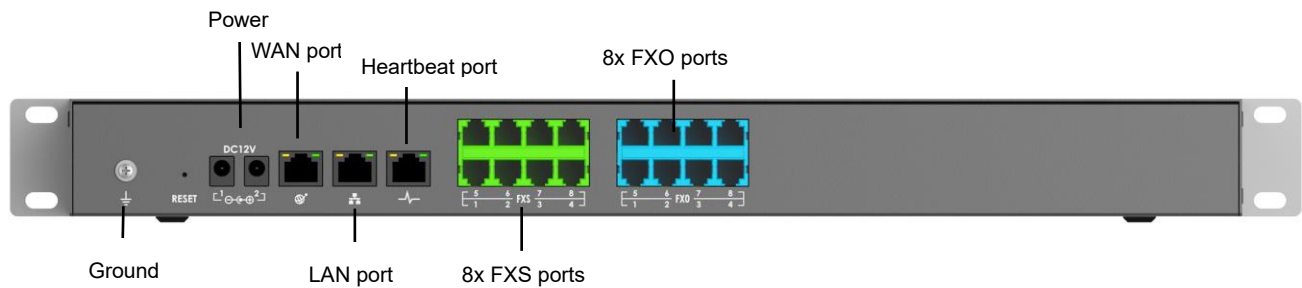


Figure 8: UCM6308A Back View

## Safety Compliances

The UCM630xA series IP PBX complies with FCC/CE and various safety standards. The UCM630xA power adapter is compliant with the UL standard. Use the universal power adapter provided with the UCM630xA package only. The manufacturer's warranty does not cover damages to the device caused by unsupported power adapters.

## Warranty

If the UCM630xA 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 an RMA (Return Materials Authorization) number before the product is returned. Grandstream reserves the right to remedy warranty policy without prior notification.

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### **Warning:**

Use the power adapter provided with the UCM630xA 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.

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## GETTING STARTED

To get started with the UCM630xA setup process, use the following available interfaces: LCD display, and web portal.

- The LCD display shows hardware, software, interface status and network information and can be navigated via the Slide control and Touch keys. From here, users can configure basic network settings, run diagnostic tests, and factory reset.
- The web portal (may also be referred to as web UI in this guide) is the primary method of configuring the UCM.

This section will provide step-by-step instructions on how to use these interfaces to quickly set up the UCM and start making and receiving calls with it.

### Use the LCD Menu

- **Idle Screen**  
Once the device has booted up completely, the LCD will show the UCM model, hardware version and IP address. Upon menu key press timeout (30 seconds), the screen will default back to this information.
- **Menu**  
Pressing the Home button will show the main menu. All available menu options are found in [Table 3: LCD Menu Options].
- **Menu Navigation**  
Scrolling sown using slide control through the menu options. Press the OK button to select an option.
- **Exit**  
Selecting the Back option will return to the previous menu. For the Device Info, Network Info, and Web Info screens that have no Back option, pressing the OK button will return to the previous menu.
- **LCD Backlight**  
The LCD backlight will turn on upon button press and will go off when idle for 30 seconds.

The following table summarizes the layout of the LCD menu of UCM630x.



**Table 3: LCD Menu Options**

<b>View Events</b>	<ul style="list-style-type: none"> <li>• <b>Critical Events</b></li> <li>• <b>Other Events</b></li> </ul>
<b>Device Info</b>	<ul style="list-style-type: none"> <li>• <b>Hardware:</b> Hardware version number</li> <li>• <b>Software:</b> Software version number</li> <li>• <b>P/N:</b> Part number</li> <li>• <b>WAN MAC:</b> WAN side MAC address</li> <li>• <b>LAN MAC:</b> LAN side MAC address</li> <li>• <b>Uptime:</b> System uptime</li> </ul>
<b>Network Info</b>	<ul style="list-style-type: none"> <li>• <b>WAN Mode:</b> DHCP, Static IP, or PPPoE</li> <li>• <b>WAN IP:</b> IP address</li> <li>• <b>WAN Subnet Mask</b></li> <li>• <b>LAN IP:</b> IP address</li> <li>• <b>LAN Subnet Mask</b></li> </ul>
<b>Network Menu</b>	<ul style="list-style-type: none"> <li>• <b>WAN Mode:</b> Select WAN mode as DHCP, Static IP or PPPoE</li> <li>• <b>Static Route Reset:</b> Select this to reset static route settings.</li> </ul>
<b>Factory Menu</b>	<ul style="list-style-type: none"> <li>• <b>Reboot</b></li> <li>• <b>Factory Reset</b></li> <li>• <b>LCD Test Patterns</b> Press DOWN and OK buttons to scroll through and select different LCD patterns to test. Once a test is done, press the OK button to return to the previous menu.</li> <li>• <b>Fan Mode</b> Select Auto or On.</li> <li>• <b>LED Test Patterns</b> All On, All Off, and Blinking are the available options. Selecting Back in the menu will revert the LED indicators back to their actual status.</li> <li>• <b>RTC Test Patterns</b> Select either 2022-02-22 22:22 or 2011-01-11 11:11 to start the RTC (Real-Time Clock) test pattern. Check the system time from either the LCD idle screen or in the web portal System Status→System Information→General page. To revert back to the correct time, manually reboot the device.</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>Hardware Testing</b> Select Test SVIP to verify hardware connections within the device. The result will display on the LCD when the test is complete.</li> </ul>
<b>Web Info</b>	<ul style="list-style-type: none"> <li>• <b>Protocol:</b> Web access protocol (HTTP/ HTTPS). HTTPS is used by default.</li> <li>• <b>Port:</b> Web access port number, which is 8089 by default.</li> </ul>
<b>SSH Switch</b>	<ul style="list-style-type: none"> <li>• <b>Enable SSH</b></li> <li>• <b>Disable SSH</b></li> </ul> <p>SSH access is disabled by default</p>

## Use the LED Indicators

The UCM6300A/6302A has LED indicators on the network port to display connection status and the following picture shows the other ports status.

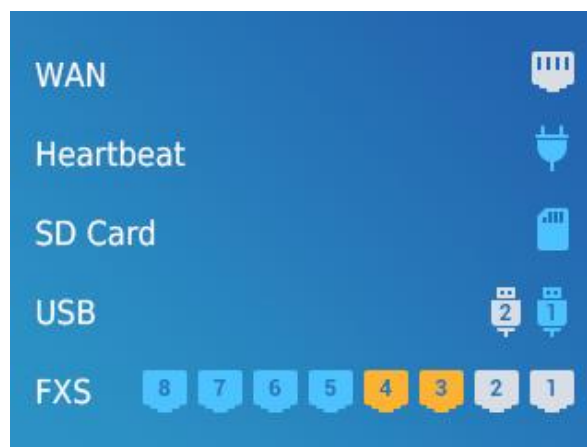


Figure 9: Ports Status

The UCM6304A/6308A has LED indicators in the front to display connection status. The following table shows the status definitions.

Table 4: UCM6304A/6308A LED Indicators

LED Indicator	LED Status
Power 1/Power 2	<ul style="list-style-type: none"> <li>■ <b>Solid:</b> Connected</li> <li>■ <b>Fast Blinking:</b> Data Transferring</li> <li>■ <b>Slow Blinking:</b> Trying to connect</li> <li>■ <b>OFF:</b> Not Connected</li> </ul>
PoE	
LAN	
WAN	
USB	
SD	
FXS ports	
FXO ports	



## Using the Web UI

### Accessing the Web UI

The UCM's 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 (version 8+), Mozilla Firefox, Google Chrome, etc. To access the UCM's web portal, follow the steps below:

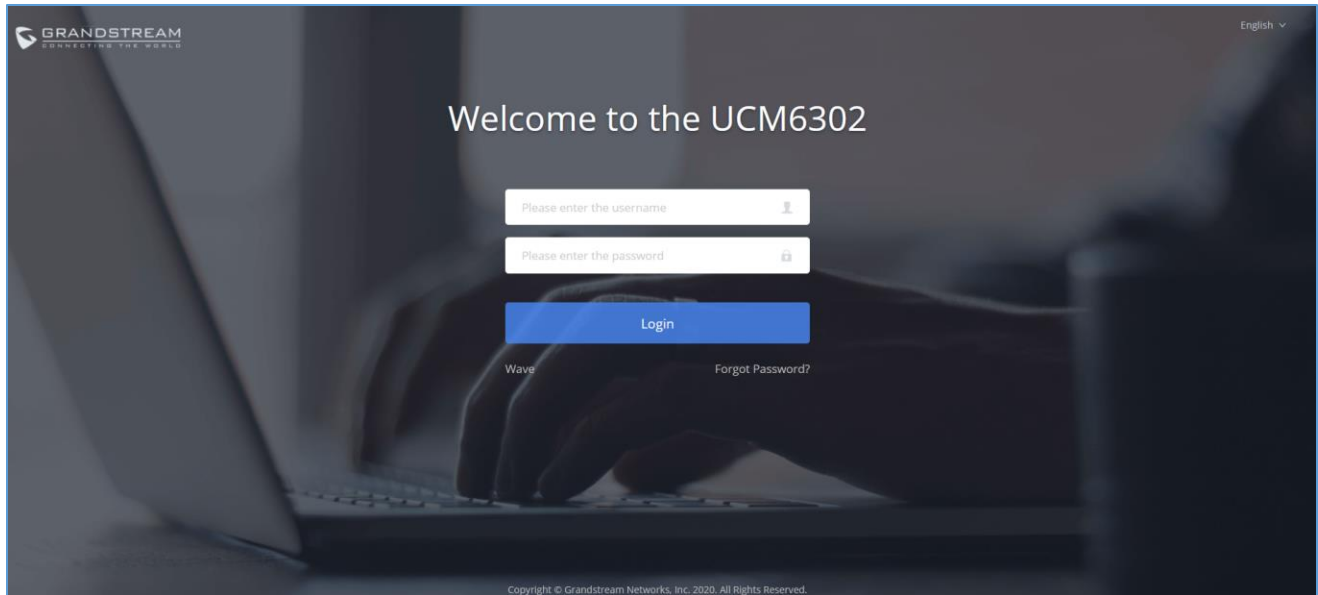


Figure 10: UCM6302A Web GUI Login Page

1. Make sure your computer is on the same network as the UCM.
2. Make sure that the UCM's IP address is displayed on its LCD.
3. Enter the UCM's IP address into a web browsers' address bar. The login page should appear (please see the above image).
4. Enter default administrator username "admin" and password can be found on the sticker at the back of the UCM.

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 **Note:**

By default, the UCM630xA has **Redirect From Port 80** enabled. As such, if users type in the UCM630xA 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, and 192.168.40.167 is entered into the web browser, the web page will be redirected to: <https://192.168.40.167:8089>

The option **Redirect From Port 80** can be found under the UCM630xA Web GUI→**System Settings**→**HTTP Server**.

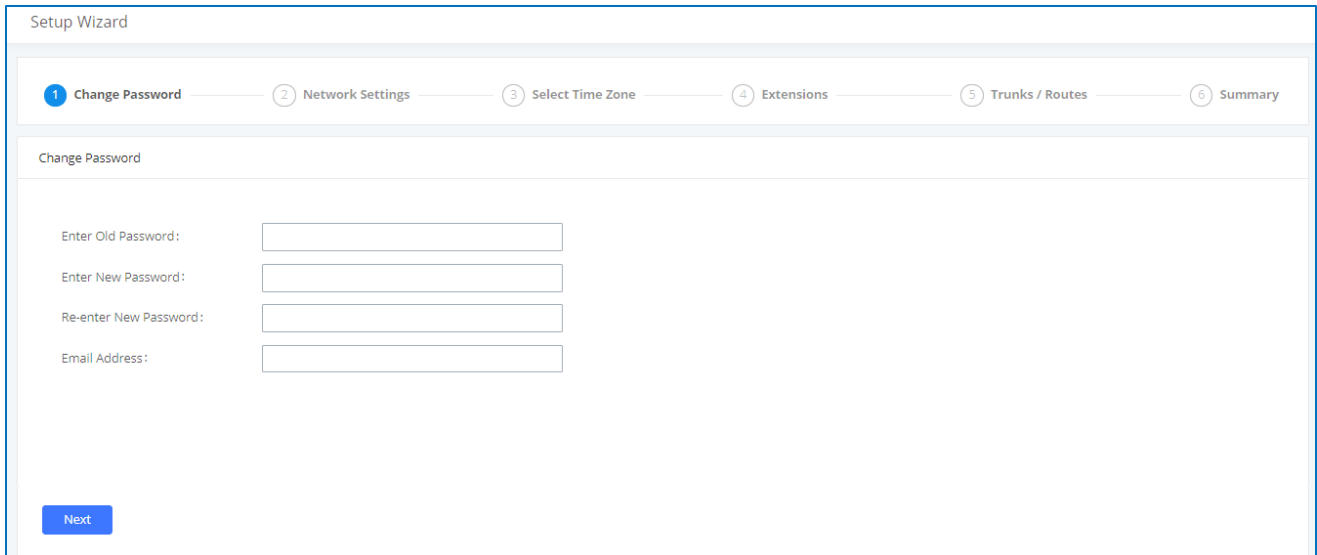
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## Setup Wizard

After logging into the UCM web portal for the first time, the setup wizard will guide the user through basic configurations such as time zone, network settings, trunks, and routing rules.



The screenshot shows the 'Setup Wizard' interface. At the top, a progress bar indicates six steps: 1. Change Password (active), 2. Network Settings, 3. Select Time Zone, 4. Extensions, 5. Trunks / Routes, and 6. Summary. Below the progress bar, the 'Change Password' section contains four input fields: 'Enter Old Password:', 'Enter New Password:', 'Re-enter New Password:', and 'Email Address:'. A blue 'Next' button is located at the bottom left of the form.

**Figure 11: UCM630xA Setup Wizard**

The setup wizard can be closed and reopened at any time. At the end of the wizard, a summary of the pending configuration changes can be reviewed before applying them.

## Main Settings

There are 8 main sections in the web portal to manage various features of the UCM.

- **System Status:** Displays the dashboard, system information, current active calls, and network status.
- **Extensions/Trunks:** Manages extensions, trunks, and routing rules.
- **Call Features:** Manages various features of the UCM such as the IVR and voicemail.
- **PBX Settings:** Manages the settings related to PBX functionality such as SIP settings and interface settings.
- **System Settings:** Manages the settings related to the UCM system itself such as network and security settings.
- **CDR:** Contains the call detail records, statistics, and audio recordings of calls processed by the UCM.
- **Value-Added Features:** Manages the settings of features unrelated to core PBX functionality such as Zero Config provisioning and CRM/PMS integrations.
- **Maintenance:** Manages settings and logs related to system management and maintenance such as user management, activity logs, backup settings, upgrade settings and troubleshooting tools.



## Web GUI Languages

Currently the UCM630xA series Web GUI supports **English, Simplified Chinese, Traditional Chinese, Spanish, French, Portuguese, Russian, Italian, Polish, German etc.**

Users can select the UCM's web UI display language in the top-right corner of the page.

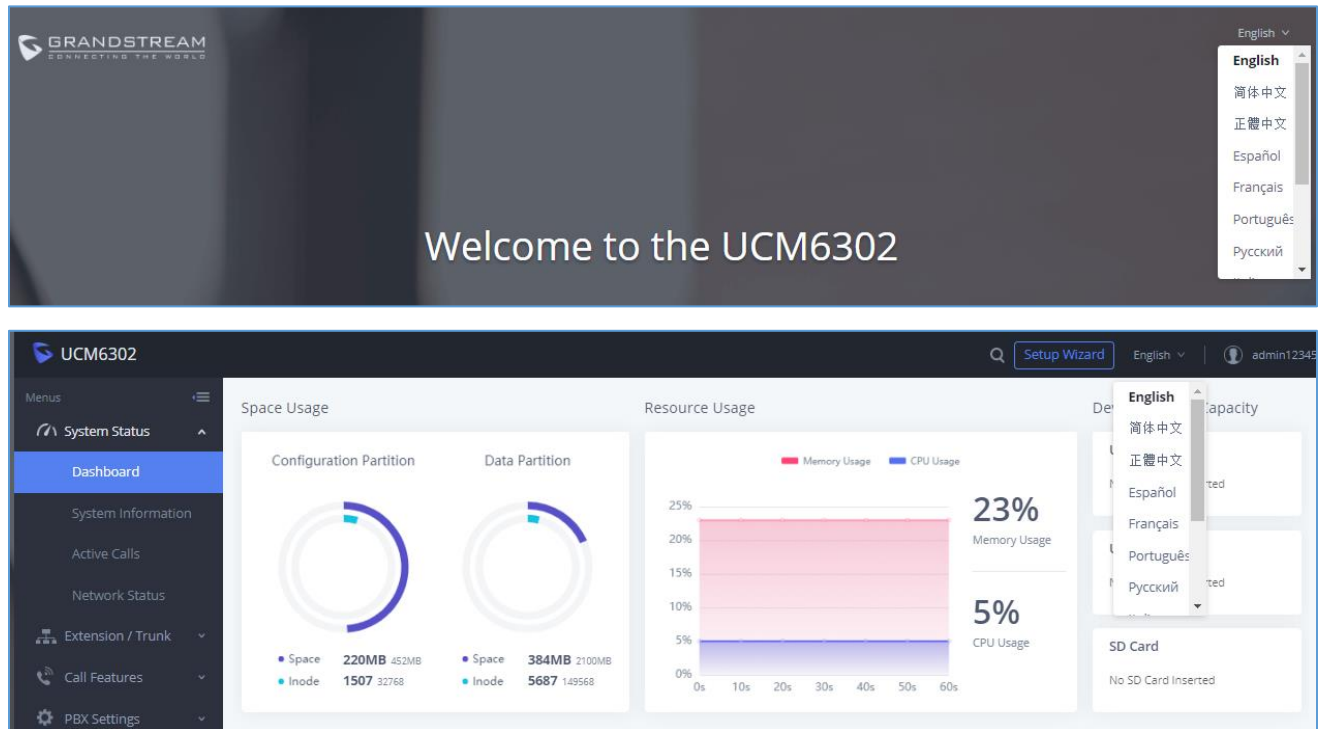
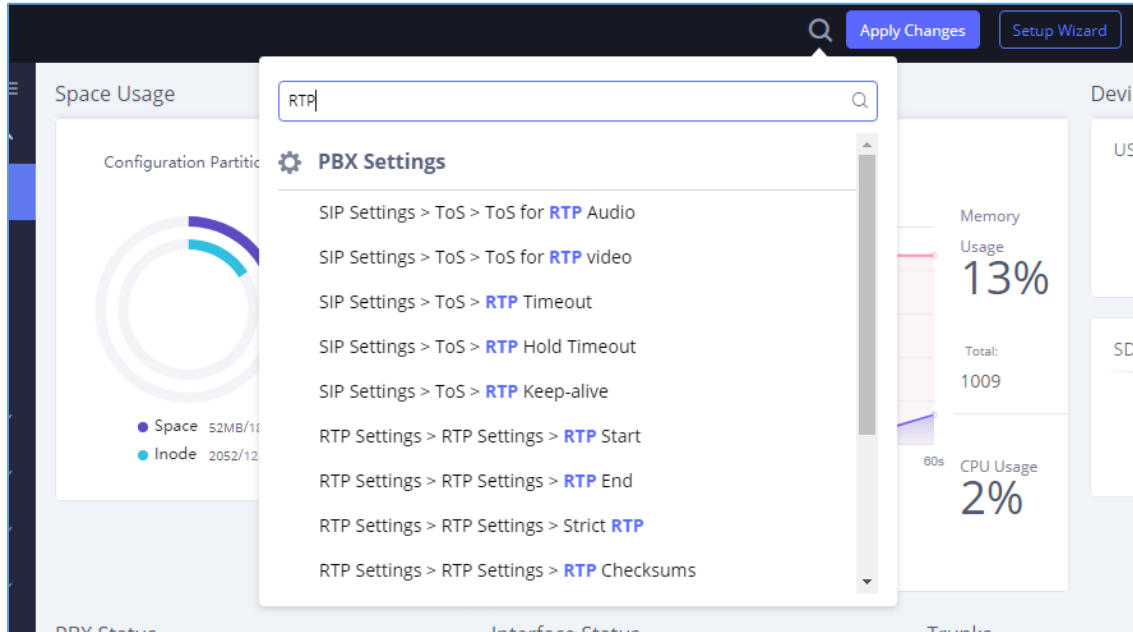


Figure 12: UCM630xA Web GUI Language

## Web GUI Search Bar

Users can search for options in the web portal with the search bar on the top right of the page.





**Figure 13: Web GUI Search Bar**

## Saving and Applying Changes

After making changes to a page, click on the "Save" button to save them and then the "Apply Changes" button that finalizes the changes. If a modification requires a reboot, a prompt will appear asking to reboot the device.

## Setting Up an Extension

Power on the UCM630xA and your SIP endpoint. Connect both devices to the same network and follow the steps below to set up an extension.

1. Log into the UCM web portal and navigate to **Extension/Trunk→Extensions**
2. Click on the "Add" button to start creating a new extension. The Extension and SIP/IAX Password information will be used to register to this extension. To set up voicemail, the Voicemail Password will be required.
3. To register an endpoint to this extension, go into your endpoint's web UI and edit the desired account. Enter the newly created extension's number, SIP user ID, and password into their corresponding fields on the endpoint. Enter the UCM's IP address into the SIP server field. If setting up voicemail, enter \*97 into the Voice Mail Access Number field. This field may be named differently on other devices.
4. To access the extension's voicemail, use the newly registered extension to dial \*97 and access the personal voicemail system. Once prompted, enter the voicemail password. If successful, you will now be prompted with various voicemail options.
5. You have now set up an extension on an endpoint.

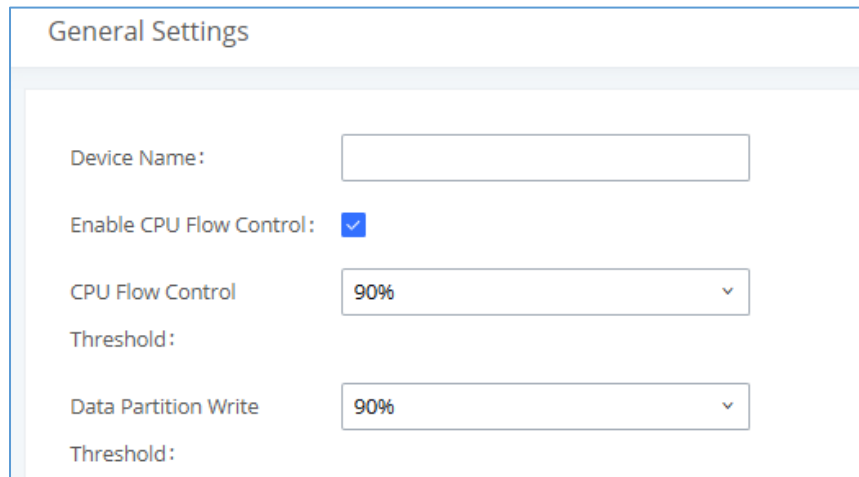


## SYSTEM SETTINGS

This section will explain the available system-wide parameters and configuration options on the UCM630xA series. This includes settings for the following items: General Settings, HTTP server, network Settings, OpenVPN, DDNS Settings, Security Settings, LDAP server, Time settings, Email settings and TR-069.

### General Settings

System administrators can prevent the UCM from making calls and/or writing to the data partition (e.g., CDR, recordings, etc.) once the system reaches a specified threshold of storage usage and CPU usage respectively. These options are located in the System Settings → General Settings page.



The screenshot shows the 'General Settings' interface with the following fields:

- Device Name:
- Enable CPU Flow Control:
- CPU Flow Control Threshold:
- Data Partition Write Threshold:

Figure 14: General Settings Interface

Table 5: General Settings Parameters

General Settings	
<b>Device Name</b>	Configure the name of the UCM.
<b>Enable CPU Flow Control</b>	Enables the CPU flow control.
<b>CPU Flow Control Threshold</b>	Used to set the threshold generated by the CPU Flow Control. When the system CPU reaches the threshold, it will prohibit the new calls. Default value is <b>90%</b> .
<b>Data Partition Write Threshold</b>	Used to set a threshold to stop writing data partition. When the disk data partition reaches the threshold configured, the data partition writing will be stopped. Default value is <b>90%</b> .



## HTTP Server

The UCM630xA's embedded web server responds to HTTPS GET/POST requests and allows users to configure the UCM via web browsers such as Microsoft IE, Mozilla Firefox, and Google Chrome. By default, users can access the UCM by just typing its IP address into a browser address bar. The browser will automatically be redirected to HTTPS using port 8089. For example, typing in "192.168.40.50" into the address bar will redirect the browser to "<https://192.168.40.50:8089>". This behavior can be changed in the **System Settings**→**HTTP Server** page.

Table 6: HTTP Server Settings

UCM Web Settings	
<b>Redirect From Port 80</b>	Toggles automatic redirection to UCM's web portal from port 80. If disabled, users will need to manually add the UCM's configured HTTPS port to the server address when accessing the UCM web portal via browser. Default is "Enabled".
<b>Port</b>	Specifies the port number used to access the UCM HTTP server. Default is "8089".
<b>Enable IP Address Whitelist</b>	If enabled, only the server addresses in whitelist will be able to access the UCM's web portal. It is highly recommended to add the IP address currently used to access the UCM web page before enabling this option. Default is "Disabled".
<b>Permitted IP(s)</b>	List of addresses that can access the UCM web portal. Ex: 192.168.6.233 / 255.255.255.255
<b>External Host</b>	Configure a URL and port (optional) used to access the UCM web portal if the UCM is behind NAT.
GS Wave Settings	
<b>External Host</b>	Configure a URL and port (optional) used to access the UCM web portal if the UCM is behind NAT.
<b>Port</b>	The port to access Wave Web and Wave Mobile. If behind NAT, please make sure to map the external port to this port.
Certificate Settings	
<b>Certificate Options</b>	<p>Selects the method of acquiring SSL certificates for the UCM web server. Two methods are currently available:</p> <ul style="list-style-type: none"> <li>• <b>Upload Certificate:</b> Upload the appropriate files from one's own PC.</li> <li>• <b>Request Certificate:</b> Enter the domain for which to request a certificate for from "Let's Encrypt".</li> </ul>



<b>TLS Private Key</b>	Uploads the private key for the HTTP server. <b>Note:</b> Key file must be under 2MB in file size and in *.pem format. File name will automatically be changed to "private.pem".
<b>TLS Cert</b>	Uploads the certificate for the HTTP server. <b>Note:</b> Certificate must be under 2MB in file size and in *.pem format. This will be used for TLS connections and contains private key for the client and signed certificate for the server.
<b>Domain</b>	Enter the domain to request the certificate for and click on <a href="#">Request Certificate</a> to request the certificate.

If the protocol or port has been changed, the user will be logged out and redirected to the new URL.

## Network Settings

After successfully connecting the UCM630xA to the network for the first time, users could login the Web GUI and go to **System Settings**→**Network Settings** to configure the network parameters for the device.

- UCM630xA 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→**System Settings**→**Network Settings** page to configure LAN settings, WAN settings, 802.1X and Port Forwarding.

## Basic Settings

Please refer to the following tables for basic network configuration parameters on UCM6300A, UCM6302A, UCM6304A and UCM6308A, respectively.

**Table 7: UCM630xA Network Settings→Basic Settings**

<b>Method</b>	<p>Select "Route", "Switch" or "Dual" mode on the network interface of UCM630xA. The default setting is "Route".</p> <ul style="list-style-type: none"> <li>• <b>Route</b> WAN port will be used for uplink connection. LAN port will function similarly to a regular router port.</li> <li>• <b>Switch</b> WAN port will be used for uplink connection. LAN port will be used as a bridge for connections.</li> <li>• <b>Dual</b> Both WAN and LAN ports will be used for uplink connections labeled as LAN2 and</li> </ul>
---------------	--



	LAN1, respectively. The port selected as the Default Interface will need to have a gateway IP address configured if it is using a static IP.
<b>MTU</b>	Specifies the maximum transmission unit value. Default is 1492.
<b>IPv4 Address</b>	
<b>Preferred DNS Server</b>	If configured, this will be used as the Primary DNS server.
<b>WAN (when "Method" is set to "Route")</b>	
<b>IP Method</b>	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
<b>IP Address</b>	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
<b>Subnet Mask</b>	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
<b>Gateway IP</b>	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
<b>DNS Server 1</b>	Enter the DNS server 1 address for static IP settings.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static IP settings.
<b>Username</b>	Enter the username to connect via PPPoE.
<b>Password</b>	Enter the password to connect via PPPoE.
<b>Layer 2 QoS 802.1Q/VLAN Tag</b>	Assign the VLAN tag of the layer 2 QoS packets for WAN port. The default value is 0.
<b>Layer 2 QoS 802.1p Priority Value</b>	Assign the priority value of the layer 2 QoS packets for WAN port. The default value is 0.
<b>LAN (when Method is set to "Route")</b>	
<b>IP Address</b>	Enter the IP address assigned to LAN port. The default setting is 192.168.2.1.
<b>Subnet Mask</b>	Enter the subnet mask. The default setting is 255.255.255.0.
<b>DHCP Server Enable</b>	Enable or disable DHCP server capability. The default setting is "Yes".
<b>DNS Server 1</b>	Enter DNS server address 1. The default setting is 8.8.8.8.
<b>DNS Server 2</b>	Enter DNS server address 2. The default setting is 208.67.222.222.
<b>Allow IP Address From</b>	Enter the DHCP IP Pool starting address. The default setting is 192.168.2.100.
<b>Allow IP Address To</b>	Enter the DHCP IP Pool ending address. The default setting is 192.168.2.254.
<b>Default IP Lease Time</b>	Enter the IP lease time (in seconds). The default setting is 43200.



**LAN (when Method is set to "Switch")**

<b>IP Method</b>	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
<b>IP Address</b>	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
<b>Subnet Mask</b>	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
<b>Gateway IP</b>	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
<b>DNS Server 1</b>	Enter the DNS server 1 address for static IP settings.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static IP settings.
<b>Username</b>	Enter the username to connect via PPPoE.
<b>Password</b>	Enter the password to connect via PPPoE.
<b>Layer 2 QoS 802.1Q/VLAN Tag</b>	Assign the VLAN tag of the layer 2 QoS packets for LAN port. The default value is 0.
<b>Layer 2 QoS 802.1p Priority Value</b>	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")**

<b>Default Interface</b>	If "Dual" is selected as "Method", users will need assign the default interface to be LAN 1 (mapped to UCM6302A WAN port) or LAN 2 (mapped to UCM6302A LAN port) and then configure network settings for LAN 1/LAN 2. The default interface is LAN 2.
<b>IP Method</b>	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
<b>IP Address</b>	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
<b>Subnet Mask</b>	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
<b>Gateway IP</b>	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.
<b>DNS Server 1</b>	Enter the DNS server 1 address for static IP settings.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static IP settings.
<b>Username</b>	Enter the username to connect via PPPoE.
<b>Password</b>	Enter the password to connect via PPPoE.
<b>Layer 2 QoS 802.1Q/VLAN Tag</b>	Assign the VLAN tag of the layer 2 QoS packets for LAN port. The default value is 0.
<b>Layer 2 QoS 802.1p Priority Value</b>	Assign the priority value of the layer 2 QoS packets for LAN port. The default value is 0.





IPv6 Address	
<b>WAN (when "Method" is set to "Route")</b>	
<b>IP Method</b>	Select Auto or Static. The default setting is Auto
<b>IP Address</b>	Enter the IP address for static IP settings.
<b>IP Prefixlen</b>	Enter the Prefix length for static settings. Default is 64
<b>DNS Server 1</b>	Enter the DNS server 1 address for static settings.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static settings.
<b>LAN (when Method is set to "Route")</b>	
<b>DHCP Server</b>	Select Disable, Auto or DHCPv6. <ul style="list-style-type: none"> <li><b>Disable:</b> the DHCPv6 server is disabled.</li> <li><b>Auto:</b> Stateless address auto configuration using NDP protocol.</li> <li><b>DHCPv6:</b> Stateful address auto configuration using DHCPv6 protocol.</li> </ul> The default setting is Disabled.
<b>DHCP Prefix</b>	Enter DHCP prefix. (Default is 2001:db8:2:2::)
<b>DHCP prefixlen</b>	Enter the Prefix length for static settings. Default is 64
<b>DNS Server 1</b>	Enter the DNS server 1 address for static settings. Default is (2001:4860:4860::8888 )
<b>DNS Server 2</b>	Enter the DNS server 2 address for static settings. Default is (2001:4860:4860::8844 )
<b>Allow IP Address From</b>	Configure starting IP address assigned by the DHCP prefix and DHCP prefixlen.
<b>Allow IP Address To</b>	Configure the ending IP address assigned by the DHCP Prefix and DHCP prefixlen.
<b>Default IP Lease Time</b>	Configure the lease time (in second) of the IP address.
<b>LAN (when Method is set to "Switch")</b>	
<b>IP Method</b>	Select Auto or Static. The default setting is Auto
<b>IP Address</b>	Enter the IP address for static IP settings.
<b>IP Prefixlen</b>	Enter the Prefix length for static settings. Default is 64
<b>DNS Server 1</b>	Enter the DNS server 1 address for static settings.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static settings.
<b>LAN 1 / LAN 2 (when Method is set to "Dual")</b>	
<b>Default Interface</b>	Users will need assign the default interface to be LAN 1 (mapped to UCM630xA WAN port) or LAN 2 (mapped to UCM630xA LAN port) and then configure network settings for LAN 1/LAN 2. The default interface is LAN 1.
<b>IP Method</b>	Select Auto or Static. The default setting is Auto
<b>IP Address</b>	Enter the IP address for static IP settings.



<b>IP Prefixlen</b>	Enter the Prefix length for static settings. Default is 64
<b>DNS Server 1</b>	Enter the DNS server 1 address for static settings.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static settings.

- **Method: Route**

When the UCM630xA 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.

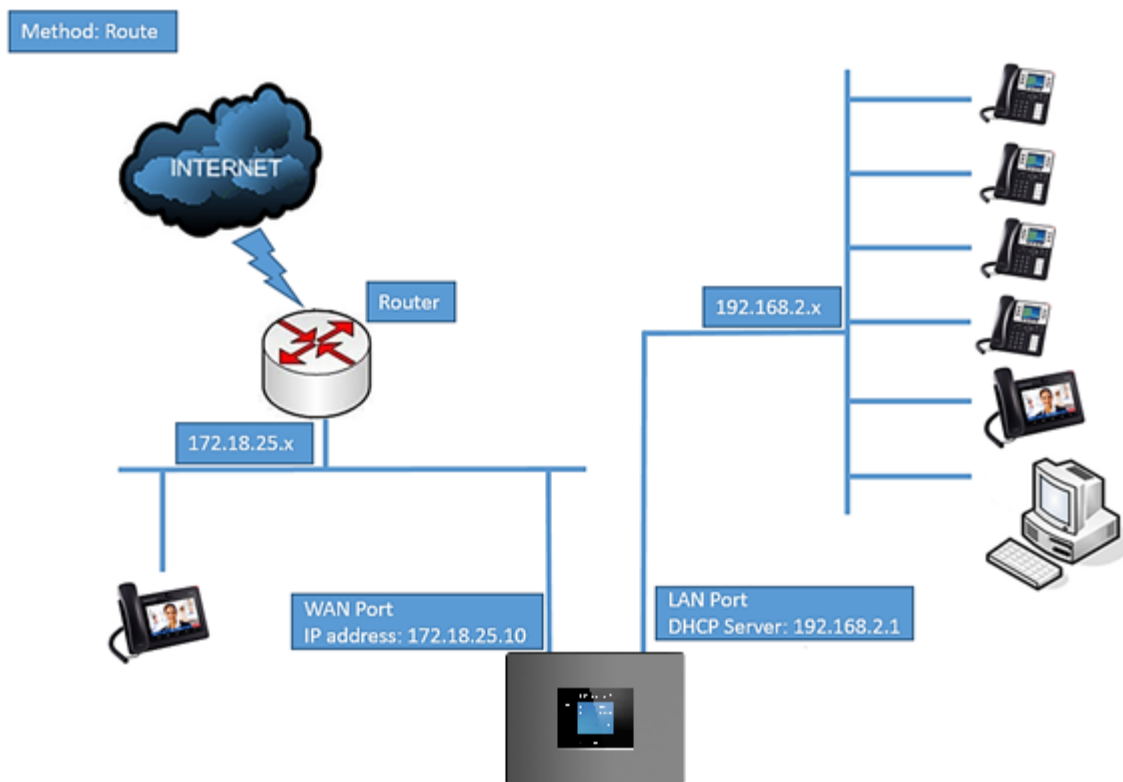


Figure 15: UCM6302A Network Interface Method: Route

- **Method: Switch**

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



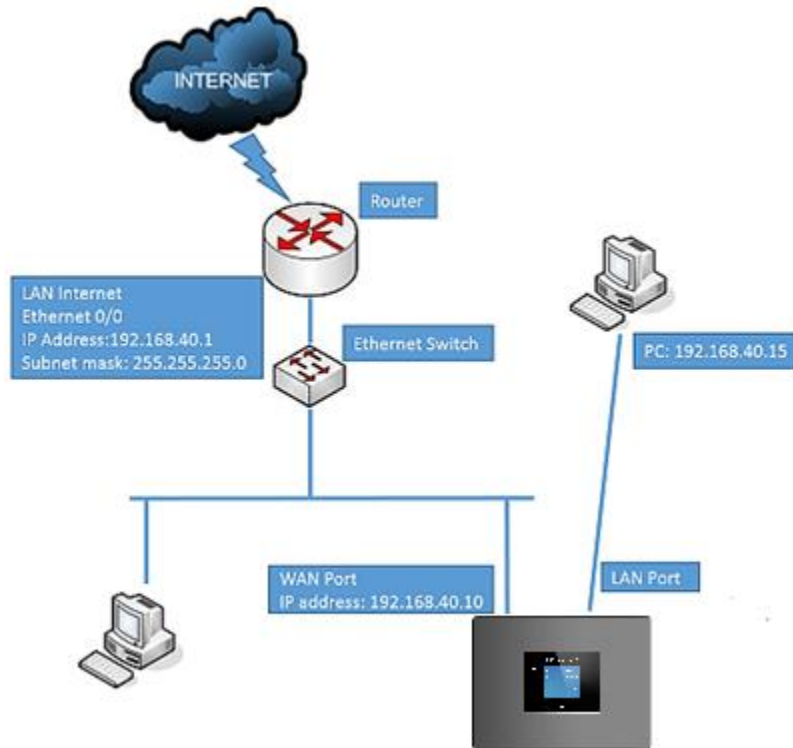


Figure 16: UCM6302A Network Interface Method: Switch

- **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.

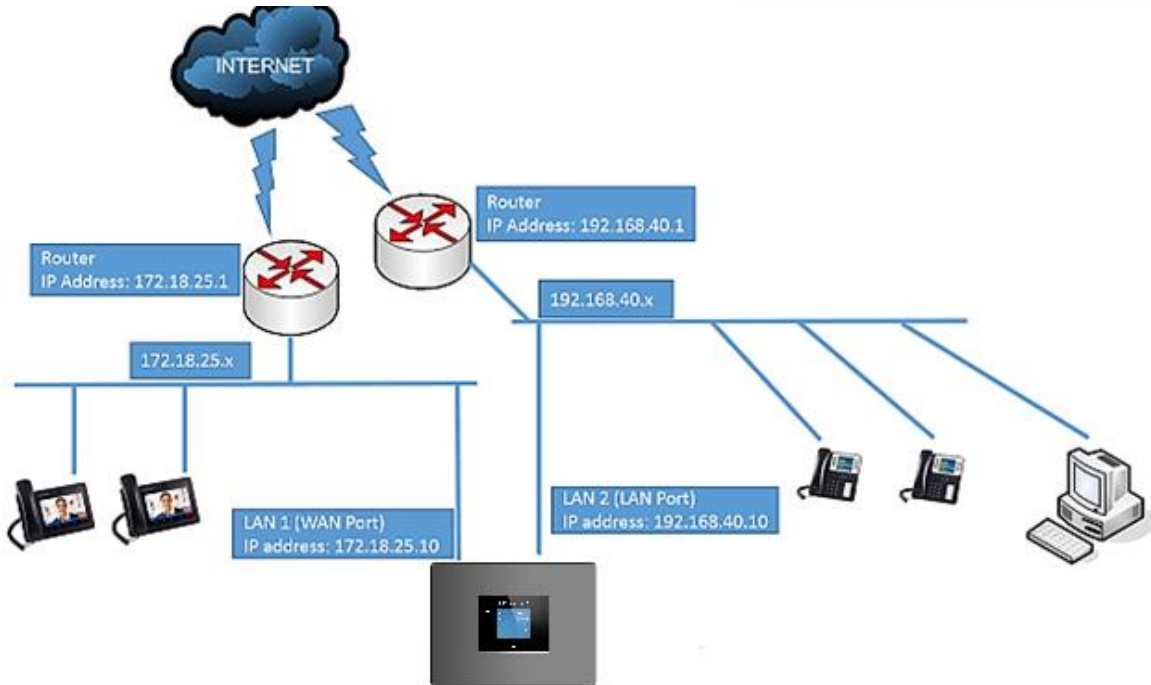


Figure 17: UCM6302A Network Interface Method: Dual

## 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 can access Internet or other LAN resources. The UCM630xA supports 802.1X as a supplicant/client to be authenticated. The following diagram and figure show UCM630xA use 802.1X mode “EAP-MD5” on WAN port as client in the network to access Internet.

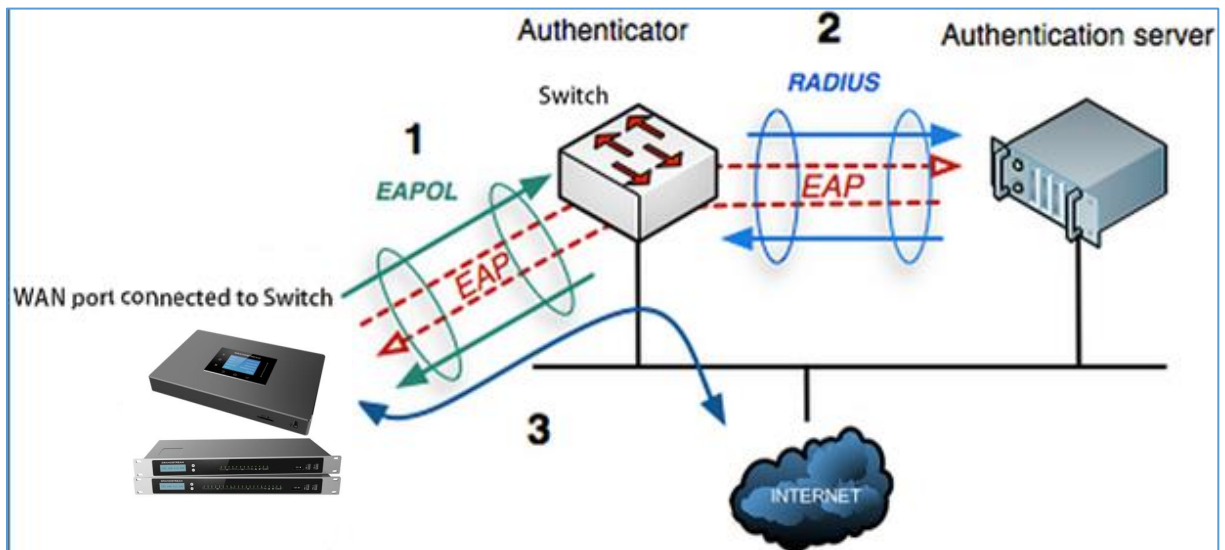


Figure 18: UCM630xA Using 802.1X as Client



### Network Settings

Basic Settings
802.1X Settings
Static Routes
Port Forwarding

802.1X Mode:

\* Identity:

\* MD5 Password:

**Figure 19: UCM630xA Using 802.1X EAP-MD5**

The following table shows the configuration parameters for 802.1X on UCM630xA. 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, users will also need to upload 802.1X CA Certificate and 802.1X Client Certificate, which should be also generated from the RADIUS server.



**Table 8: UCM630xA Network Settings→802.1X**

<b>802.1X Mode</b>	Select 802.1X mode. The default setting is "Disable". The supported 802.1X mode are: <ul style="list-style-type: none"> <li>EAP-MD5</li> <li>EAP-TLS</li> <li>EAP-PEAPv0/MSCHAPv2</li> </ul>
<b>Identity</b>	Enter 802.1X mode Identity information.
<b>MD5 Password</b>	Enter 802.1X mode MD5 password information.
<b>802.1X CA Certificate</b>	Select 802.1X certificate from local PC and then upload.
<b>802.1X Client Certificate</b>	Select 802.1X client certificate from local PC and then upload.

## Static Routes

The UCM630xA 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 UCM630xA Web GUI→**System Settings**→**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 UCM630xA as a failover backup, etc.



- Click on “**Add IPv4 Static Route**” to create a new IPv4 static route or click on “**Add IPv6 Static Route**” to create a new IPv6 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.

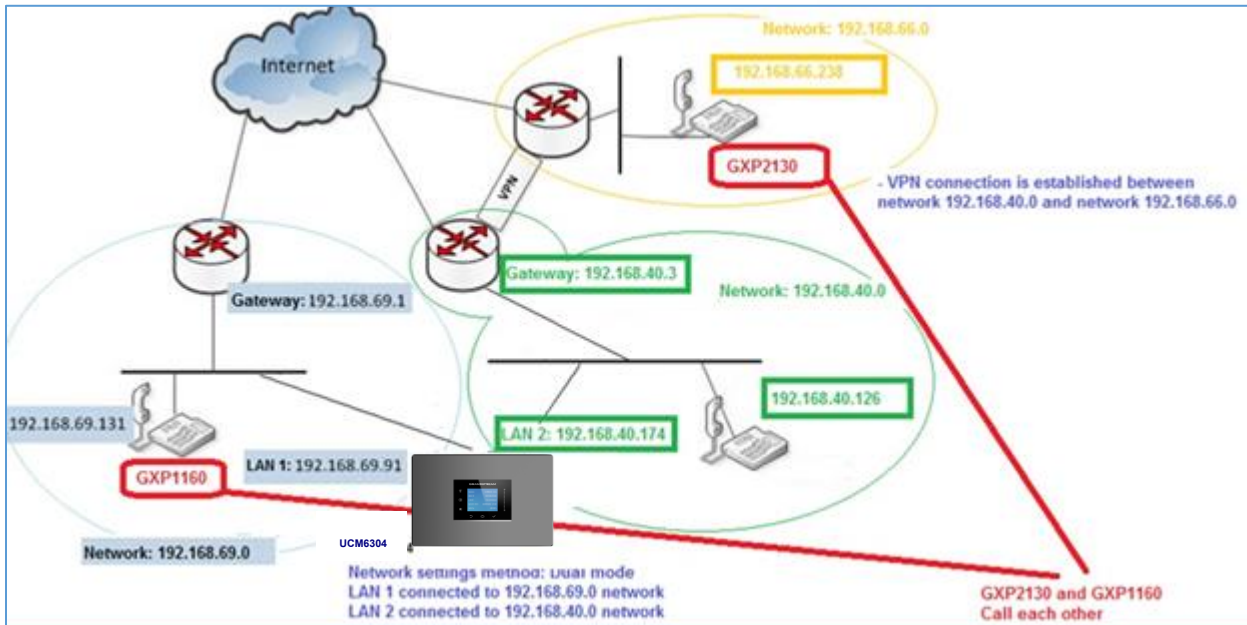
**Table 9: UCM630xA Network Settings→Static Routes**

<b>Destination</b>	Configure the destination IPv4 address or the destination IPv6 subnet for the UCM630xA to reach using the static route. Example: IPv4 address - <b>192.168.66.4</b> IPv6 subnet - <b>2001:740:D::1/64</b>
<b>Subnet Mask</b>	Configure the subnet mask for the above destination address. If left blank, the default value is 255.255.255.255. Example: <b>255.255.255.0</b>
<b>Gateway</b>	Configure the IPv4 or IPv6 gateway address so that the UCM630xA can reach the destination via this gateway. Gateway address is optional. Example: <b>192.168.40.5 or 2001:740:D::1</b>
<b>Interface</b>	Specify the network interface on the UCM630xA 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 UCM6304A.





**Figure 20: UCM6304A Static Route Sample**

The network topology of the above diagram is as below:

- Network 192.168.69.0 has IP phones registered to UCM6304A LAN 1 address
- Network 192.168.40.0 has IP phones registered to UCM6304A LAN 2 address
- Network 192.168.66.0 has IP phones registered to UCM6304A 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 UCM6304A. Therefore, we need configure a static route on the UCM6304A so that the phones in isolated networks can make calls between each other.



### Create New IPV4 Static Route

\* Destination:

Subnet Mask:

Gateway:

\* Protocol Type:  ▾

**Figure 21: UCM6304A Static Route Configuration**

## Port Forwarding

The UCM network interface supports router function which provides users the ability to do port forwarding. If LAN mode is set to "Route" under Web GUI→**System Settings**→**Network Settings**→**Basic Settings** page, port forwarding is available for configuration.

The port forwarding configuration is under Web GUI→**System Settings**→**Network Settings**→**Port Forwarding** page. Please see related settings in the table below.

**Table 10: UCM630xA Network Settings→Port Forwarding**


<b>WAN Port</b>	<p>Specify the WAN port number or a range of WAN ports. Unlimited number of ports can be configured.</p> <p><b>Note:</b> 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.</p>
<b>LAN IP</b>	Specify the LAN IP address.
<b>LAN Port</b>	<p>Specify the LAN port number or a range of LAN ports.</p> <p><b>Note:</b> 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</p>





	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.
<b>Protocol Type</b>	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 GUI access to public side.

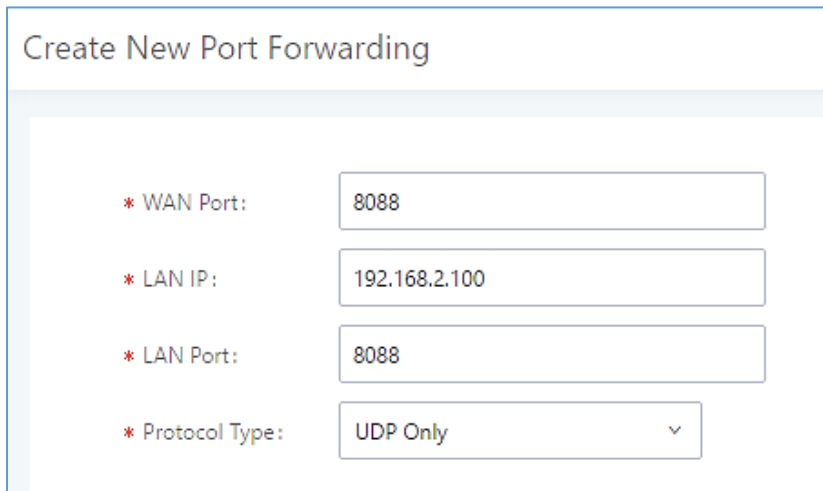
- UCM630xA network mode is set to "Route".
- UCM630xA WAN port is connected to uplink switch, with a public IP address configured, e.g. 1.1.1.1.
- UCM630xA LAN port provides DHCP pool that connects to multiple phone devices in the LAN network 192.168.2.x. The UCM60X is used as a router, with gateway address 192.168.2.1.
- There is a GXP2160 connected under the LAN interface network of the UCM630xA. It obtains IP address 192.168.2.100 from UCM630xA DHCP pool.
- On the UCM630xA Web GUI → **System Settings** → **Network Settings** → **Port Forwarding**, configure a port forwarding entry as the figure shows below.
- Click on 

**WAN Port:** This is the port opened on the WAN side for access purpose.

**LAN IP:** This is the GXP2160 IP address, under the LAN interface network of the UCM630xA.

**LAN Port:** This is the port opened on the GXP2160 side for access purpose.

**Protocol Type:** We select TCP here for Web GUI access using HTTP.

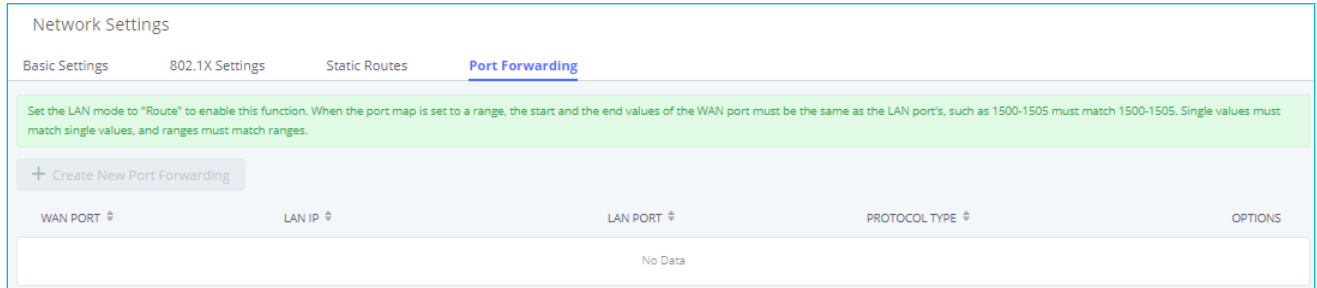


The screenshot shows a web form titled "Create New Port Forwarding". It contains the following fields:

- \* WAN Port:** Input field containing "8088".
- \* LAN IP:** Input field containing "192.168.2.100".
- \* LAN Port:** Input field containing "8088".
- \* Protocol Type:** Dropdown menu with "UDP Only" selected.

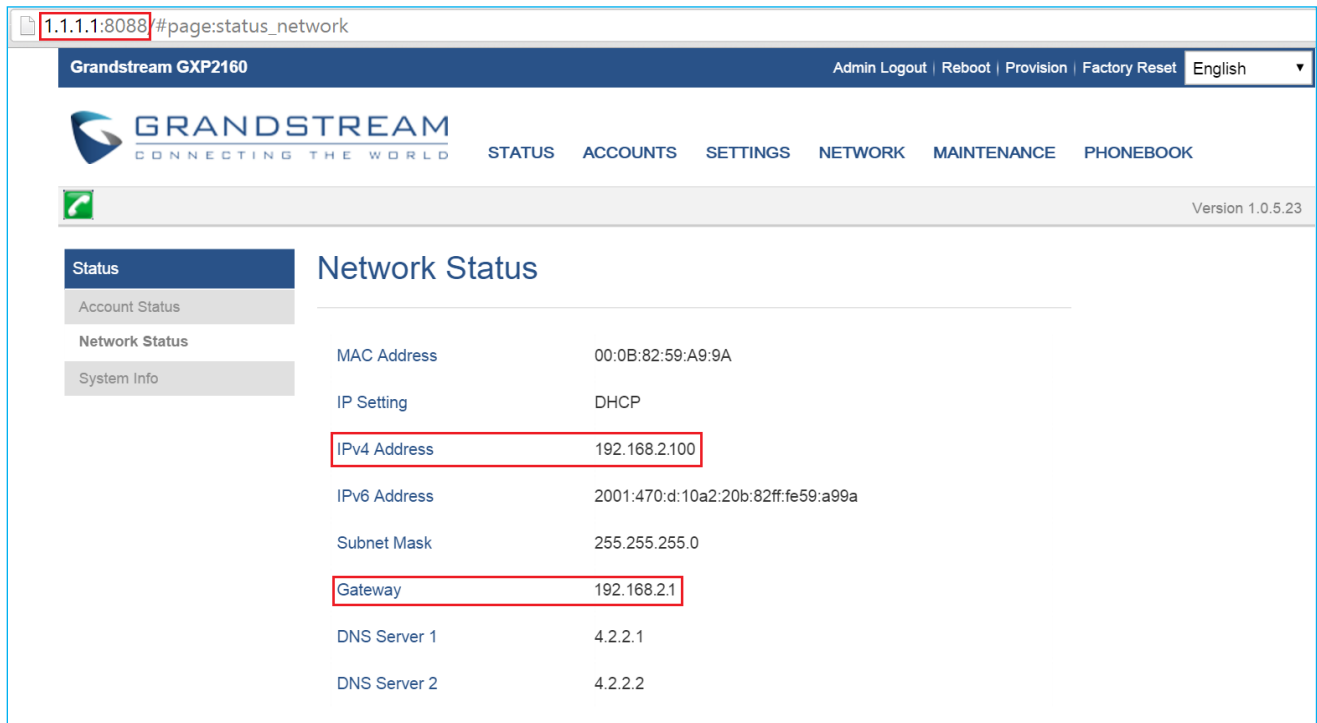
**Figure 22: Create New Port Forwarding**





**Figure 23: UCM630xA Port Forwarding Configuration**

This will allow users to access the GXP2160 Web GUI from public side, by typing in public IP address (example: 1.1.1.1:8088).



**Figure 24: GXP2160 Web Access using UCM6302A Port Forwarding**

## ARP Settings

The ARP settings can be configured under Web GUI→**System Settings**→**Network Settings**→**ARP Settings**



**Table 11: ARP Settings**

<b>ARP GC Threshold 1</b>	Minimum number of entries to keep. Garbage collector will not purge entries if there are fewer than this number. The default value is 128.
<b>ARP GC Threshold 2</b>	Threshold when garbage collector becomes more aggressive about purging entries. Entries older than 5 seconds will be cleared when over this number. The default value is 512.
<b>ARP GC Threshold 3</b>	Maximum number of non-PERMANENT neighbor entries allowed. Increase this when using large numbers of interfaces and when communicating with large numbers of directly connected peers. The default value is 1024.

## OpenVPN®

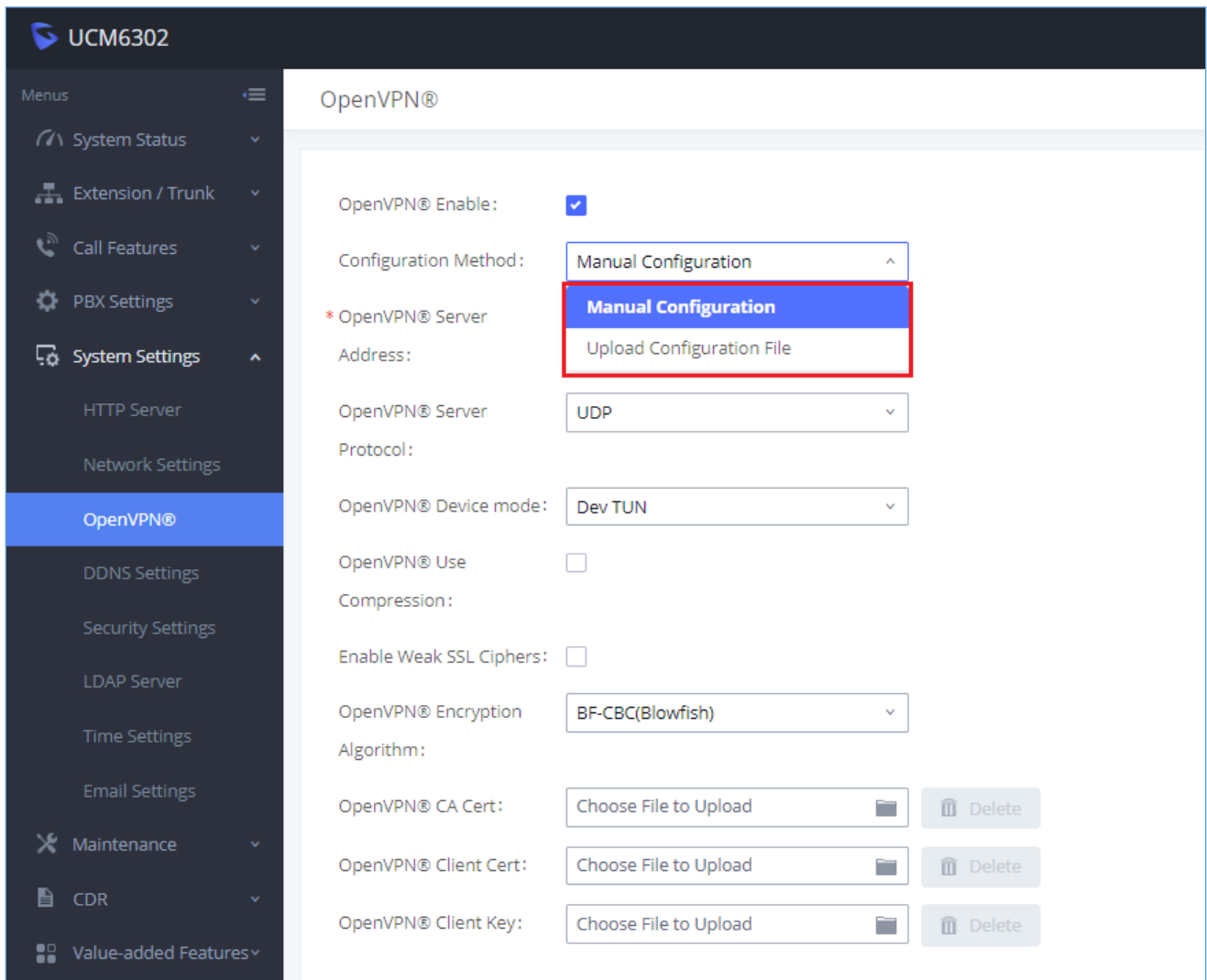
OpenVPN® settings allow the users to configure UCM630xA to use VPN features, the following table gives details about the various options in order to configure the UCM as OpenVPN Client.

**Table 12: UCM630xA System Settings→Network Settings→OpenVPN®**

<b>OpenVPN® Enable</b>	Enable / Disable the OpenVPN® feature.
<b>Configuration Method</b>	Select OpenVPN® configuration method. <b>Manual Configuration:</b> Allows to configure OpenVPN® settings manually. <b>Upload Configuration File:</b> Allows to upload .ovpn and .conf files to the UCM and to automatically configure OpenVPN® settings.
<b>OpenVPN® Server Address</b>	Configures the hostname/IP and port of the server. For example: 192.168.1.2:22
<b>OpenVPN® Server Protocol</b>	Specify the protocol user, user should use the same settings as used on the server
<b>OpenVPN® Device mode</b>	Use the same setting as used on the server. <ul style="list-style-type: none"> <li>• <b>Dev TUN:</b> Create a routed IP tunnel.</li> <li>• <b>Dev TAP:</b> Create an Ethernet tunnel.</li> </ul>
<b>OpenVPN® Use Compression</b>	Compress tunnel packets using the LZO algorithm on the VPN link. Do not enable this unless it is also enabled in the server config file.
<b>Enable Weak SSL Ciphers</b>	Either to enable the Weak SSL ciphers or not.



<b>OpenVPN® Encryption Algorithm</b>	Specify the cryptographic cipher. Users should make sure to use the same setting that they are using on the OpenVPN server.
<b>OpenVPN® CA Cert</b>	Upload as SSL/TLS root certificate. This file will be renamed as 'ca.crt' automatically.
<b>OpenVPN® Client Cert</b>	Upload a client certificate. This file will be renamed as 'cliend.crt' automatically.
<b>OpenVPN® Client Key</b>	Upload a client private key. This file will be renamed as 'client.key' automatically.



**UCM6302**

Menus

- System Status
- Extension / Trunk
- Call Features
- PBX Settings
- System Settings
  - HTTP Server
  - Network Settings
  - OpenVPN®**
  - DDNS Settings
  - Security Settings
  - LDAP Server
  - Time Settings
  - Email Settings
- Maintenance
- CDR
- Value-added Features

**OpenVPN®**

OpenVPN® Enable:

Configuration Method: Manual Configuration

\* OpenVPN® Server: Manual Configuration

Address: Upload Configuration File

OpenVPN® Server: UDP

Protocol: Dev TUN

OpenVPN® Device mode: Dev TUN

OpenVPN® Use:

Compression:

Enable Weak SSL Ciphers:

OpenVPN® Encryption Algorithm: BF-CBC(Blowfish)

OpenVPN® CA Cert: Choose File to Upload [Delete]

OpenVPN® Client Cert: Choose File to Upload [Delete]

OpenVPN® Client Key: Choose File to Upload [Delete]

**Figure 25: Open VPN® Feature on the UCM630xA**



## DDNS Settings

DDNS setting allows user to access UCM630xA 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 UCM630xA needs to have public IP access.








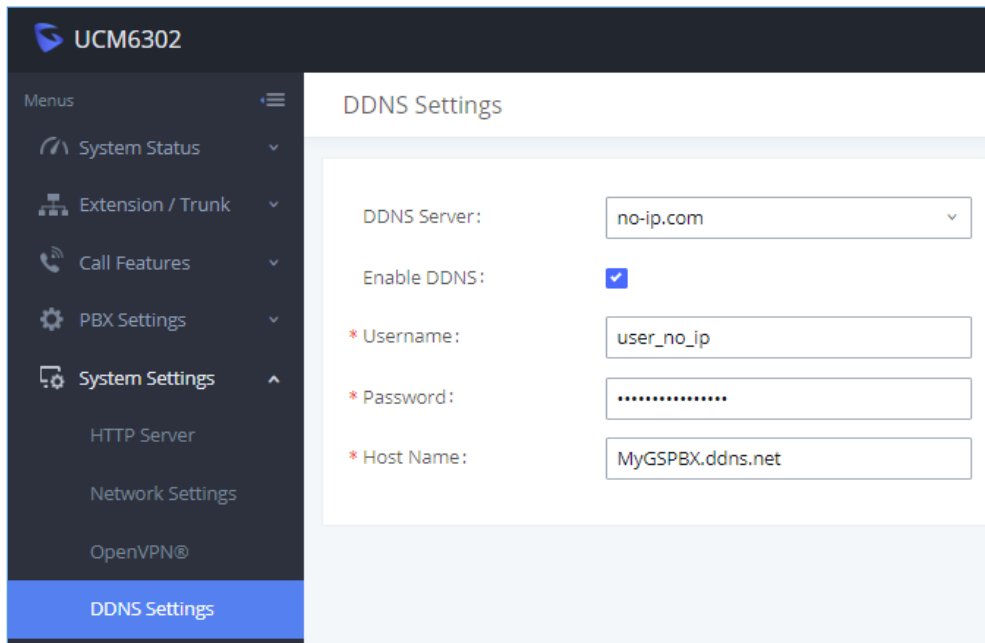
Hostname Information	
Hostname:	haograndstream.ddns.net 
Host Type:	<input checked="" type="radio"/> DNS Host (A) <input type="radio"/> DNS Host (Round Robin) <input type="radio"/> DNS Alias (CNAME)  <input type="radio"/> Port 80 Redirect <input type="radio"/> Web Redirect
IP Address:	<input type="text" value="1.2.3.4"/> Last Update: 2015-01-07 17:29:20 PST 
Assign to Group:	<input type="text" value="- No Group -"/>   <a href="#">Configure Groups</a>
Enable Wildcard:	Wildcards are a Plus / Enhanced feature. <a href="#">Upgrade Now!</a> 
Advanced Records:	TXT, SPF, and SRV records and the use of some special clients are Plus / Enhanced features. <a href="#">Upgrade now</a> to use them. 

Figure 26: Register Domain Name on noip.com

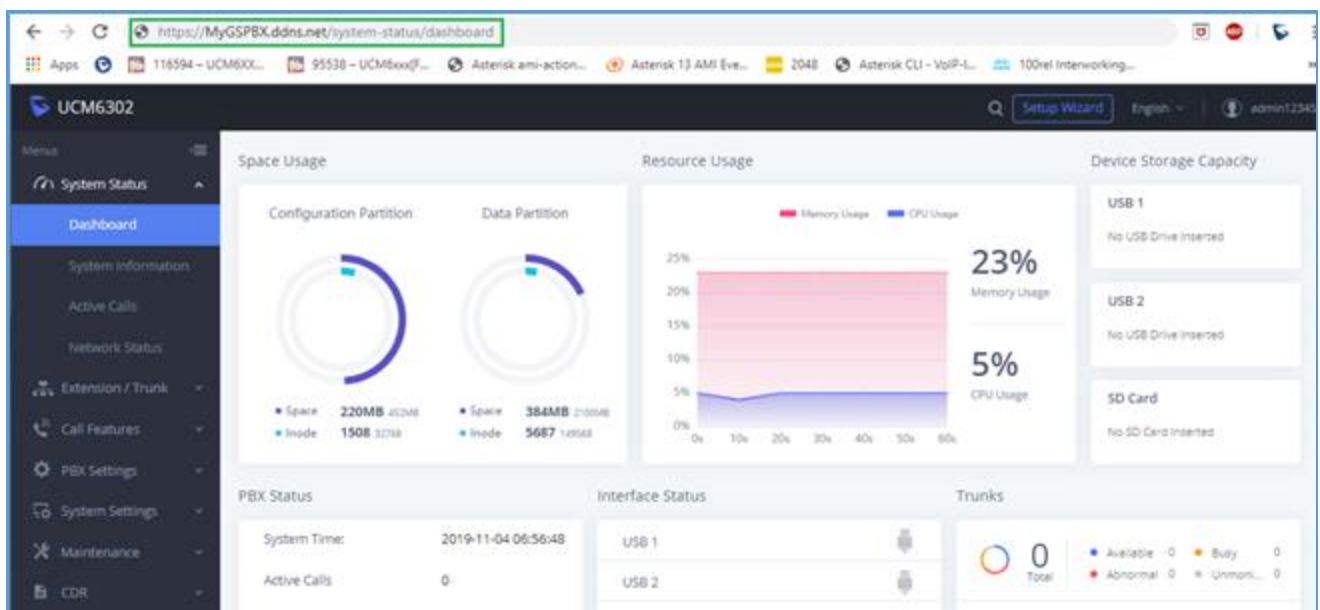
2. On Web GUI→**System Settings**→**Network Settings**→**DDNS Settings**, enable DDNS service and configure username, password, and host name.





**Figure 27: UCM630xA DDNS Setting**

3. Now you can use domain name instead of IP address to connect to the UCM630xA Web GUI.



**Figure 28: Using Domain Name to Connect to UCM630xA**



## Security Settings

The UCM630xA provides users firewall security configurations to prevent certain malicious attack to the UCM630xA system. Users could configure to allow, restrict, or reject specific traffic through the device for security and bandwidth purpose. The UCM630xA also provides Fail2ban feature for authentication errors in SIP REGISTER, INVITE and SUBSCRIBE. To configure firewall settings in the UCM630xA, go to Web GUI→**System Settings**→**Security Settings** page.

### Static Defense

Under Web GUI→**System Settings**→**Security Settings**→**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 UCM630xA.

**Table 13: UCM630xA Firewall→Static Defense→Current Service**

Port	Process	Type	Protocol or Service
7777	Asterisk	TCP/IPv4	SIP
389	Slapd	TCP/IPv4	LDAP
6060	zero_config	UDP/IPv4	UCM630xA zero_config service
5060	Asterisk	UDP/IPv4	SIP
4569	Asterisk	UDP/IPv4	SIP
38563	Asterisk	udp/ipv4	SIP
10000	gs_avs	udp/ipv4	gs_avs
10001	gs_avs	udp/ipv4	gs_avs
10002	gs_avs	udp/ipv4	gs_avs
10003	gs_avs	udp/ipv4	gs_avs
10004	gs_avs	udp/ipv4	gs_avs
10005	gs_avs	udp/ipv4	gs_avs
10006	gs_avs	udp/ipv4	gs_avs
10007	gs_avs	udp/ipv4	gs_avs
10010	gs_avs	udp/ipv4	gs_avs



10012	gs_avs	udp/ipv4	gs_avs
10013	gs_avs	udp/ipv4	gs_avs
10014	gs_avs	udp/ipv4	gs_avs
10015	gs_avs	udp/ipv4	gs_avs
10018	gs_avs	udp/ipv4	gs_avs
10019	gs_avs	udp/ipv4	gs_avs
10020	gs_avs	udp/ipv4	gs_avs
6066	Python	udp/ipv4	python
3306	Mysqld	tcp/ipv4	mysqld
45678	Python	udp/ipv4	python
8439	Lighttpd	tcp/ipv4	HTTP
8088	asterisk	tcp/ipv4	SIP
8888	Pbxmid	tcp/ipv4	pbxmid
25	Master	tcp/ipv4	master
636	Slapd	tcp/ipv4	SLDAP
4569	asterisk	udp/ipv6	SIP
42050	asterisk	udp/ipv6	SIP
7681	Pbxmid	tcp/ipv4	pbxmid

For typical firewall settings, users could configure the following options on the UCM630xA.

**Table 14: Typical Firewall Settings**

<b>Ping Defense Enable</b>	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 (UCM630xA) interface.
<b>SYN-Flood Defense Enable</b>	Allows the UCM630xA to handle excessive amounts of SYN packets from one source and keep the web portal accessible. There are two options available and only one of these options may be enabled at one time. <ul style="list-style-type: none"> <li>eth(0)LAN defends against attacks directed to the LAN IP address of the UCM630xA.</li> <li>eth(1)WAN defends against attacks directed to the WAN IP address of the UCM630xA.</li> </ul>



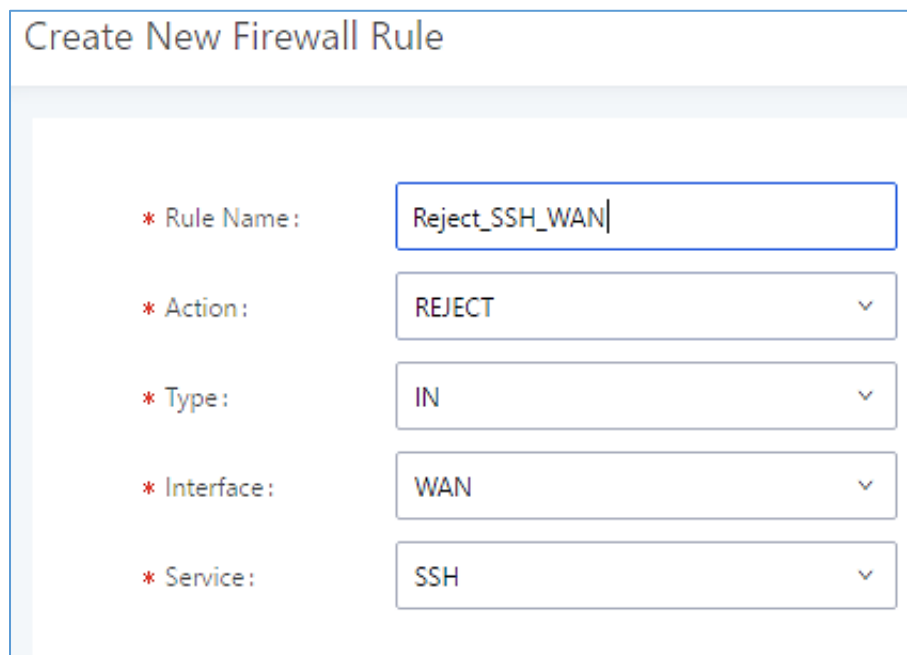


	SYN Flood Defense will limit the amount of SYN packets accepted by the UCM from one source to 10 packets per second. Any excess packets from that source will be discarded.
<b>Ping-of-Death Defense Enable</b>	Enable to prevent Ping-of-Death attack to the device. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (UCM630xA) interface.

Under "Custom Firewall Settings", users could create new rules to accept, reject or drop certain traffic going through the UCM630xA. 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 is 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



**Create New Firewall Rule**

\* Rule Name:

\* Action:

\* Type:

\* Interface:

\* Service:



**Figure 29: Create New Firewall Rule**



**Table 15: Firewall Rule Settings**

<b>Rule Name</b>	Specify the Firewall rule name to identify the firewall rule.
<b>Action</b>	Select the action for the Firewall to perform. <ul style="list-style-type: none"> <li>• ACCEPT</li> <li>• REJECT</li> <li>• DROP</li> </ul>
<b>Type</b>	Select the traffic type. <ul style="list-style-type: none"> <li>• <b>IN</b> If selected, users will need specify the network interface "LAN" or "WAN" (for UCM630xA) for the incoming traffic.</li> <li>• <b>OUT</b></li> </ul>
<b>Interface</b>	Select the interface to use the Firewall rule.
<b>Service</b>	Select the service type. <ul style="list-style-type: none"> <li>• <b>FTP</b></li> <li>• <b>SSH</b></li> <li>• <b>Telnet</b></li> <li>• <b>HTTP</b></li> <li>• <b>LDAP</b></li> <li>• <b>Custom</b> 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".</li> </ul>
<b>Source IP Address and Port</b>	Configure a source subnet and port. If set to "Anywhere" or left empty, traffic from all addresses and ports will be accepted. A single port or a range of ports can be specified (e.g., 10000, 10000-20000).
<b>Destination IP Address and Port</b>	Configure a destination subnet and port. If set to "Anywhere" or left empty, traffic can be sent to all addresses and ports. A single port or a range of ports can be specified (e.g., 10000, 10000-20000).
<b>Protocol</b>	Select the protocol for the rule to be used.

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 UCM630xA series. It can blacklist hosts dynamically when the LAN mode is set to "Route" under Web GUI→**System Settings**→**Network Settings**→**Basic Settings** page. If enabled, the traffic coming into the UCM630xA 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 UCM630xA firewall, which will then be displayed in the web page. Please refer to the following table for dynamic defense options on the UCM630xA.

**Table 16: UCM630xA Firewall Dynamic Defense**

<b>Dynamic Defense Enable</b>	Enable dynamic defense. The default setting is disabled.
<b>Blacklist Update Interval</b>	Configure the blacklist update time interval (in seconds). The default setting is 120.
<b>Connection Threshold</b>	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.
<b>Dynamic Defense Whitelist</b>	Allowed IPs and ports range, multiple IP addresses and port range. For example: <b>192.168.2.100-192.168.2.105, 1000:9999</b>

The following figure shows a configuration example like this:

- If a host at IP address 192.168.5.7 initiates more than 20 TCP connections to the UCM630xA it will be added into UCM630xA blacklist.
- This host 192.168.5.7 will be blocked by the UCM630xA for 500 seconds.
- Since IP range 192.168.5.100-192.168.5.200 is in whitelist, if a host initiates more than 20 TCP connections to the UCM630xA it will not be added into UCM630xA blacklist. It can still establish TCP connection with the UCM630xA.



Static Defense	<u>Dynamic Defense</u>	Fail2Ban	SSH Access
<b>Dynamic Defense</b>			
Dynamic Defense Enable:		<input checked="" type="checkbox"/>	
* Blacklist Update Interval (s):	<input type="text" value="500"/>		
* Connection Threshold:	<input type="text" value="20"/>		
Dynamic Defense Whitelist:	<input type="text" value="192.168.5.100-192.168.5.200 1500:2000"/>		

**Figure 30: Configure Dynamic Defense**

## Fail2ban

Fail2Ban feature on the UCM630xA provides intrusion detection and prevention for authentication errors in SIP REGISTER, INVITE and SUBSCRIBE. Once the entry is detected within "Max Retry Duration", the UCM630xA will act to forbid the host for certain period as defined in "Banned Duration". This feature helps prevent SIP brute force attacks to the PBX system.



### Security Settings

Static Defense
Dynamic Defense
Fail2ban
SSH Access

Global Settings

Enable Fail2Ban:

\* Banned Duration:

\* Max Retry Duration:

\* MaxRetry:

Fail2ban Whitelist:  ⊕

Local Settings

Asterisk Service:

Listening port number:  UDP Port

\* MaxRetry:

Login Attack Defense:

Listening port number:  TCP Port

\* MaxRetry:

**Figure 31: Fail2ban Settings**

**Table 17: Fail2Ban Settings**

Global Settings	
<b>Enable Fail2Ban</b>	Enable Fail2Ban. The default setting is disabled. Please make sure both "Enable Fail2Ban" and "Asterisk Service" are turned on to use Fail2Ban for SIP authentication on the UCM630xA.
<b>Banned Duration</b>	Configure the duration (in seconds) for the detected host to be banned. The default setting is 600. If set to 0, the host will be always banned.



<b>Max Retry Duration</b>	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 600.
<b>MaxRetry</b>	Configure the number of authentication failures during "Max Retry Duration" before the host is banned. The default setting is 5.
<b>Fail2Ban Whitelist</b>	Configure IP address, CIDR mask or DNS host in the whitelist. Fail2Ban will not ban the host with matching address in this list. Up to 20 addresses can be added into the list.
<b>Local Settings</b>	
<b>Asterisk Service</b>	Enable Asterisk service for Fail2Ban. The default setting is disabled. Please make sure both "Enable Fail2Ban" and "Asterisk Service" are turned on to use Fail2Ban for SIP authentication on the UCM630xA.
<b>Listening Port Number</b>	Configure the listening port number for the service. By default, port 5060 will be used for UDP and TCP, and port 5061 will be used for TCP.
<b>MaxRetry</b>	Configure the number of authentication failures during "Max Retry Duration" before the host is banned. The default setting is 5. Please make sure this option is properly configured as it will override the "MaxRetry" value under "Global Settings".
<b>Login Attack Defense</b>	Enables defense against excessive login attacks to the UCM's web GUI. The default setting is disabled.
<b>Listening Port Number</b>	This is the Web GUI listening port number which is configured under <b>System Settings</b> → <b>HTTP Server</b> → <b>Port</b> . The default is 8089.
<b>MaxRetry</b>	When the number of failed login attempts from an IP address exceeds the MaxRetry number, that IP address will be banned from accessing the Web GUI.
<b>Blacklist</b>	
<b>Blacklist</b>	Users will be able to view the IPs that have been blocked by UCM.

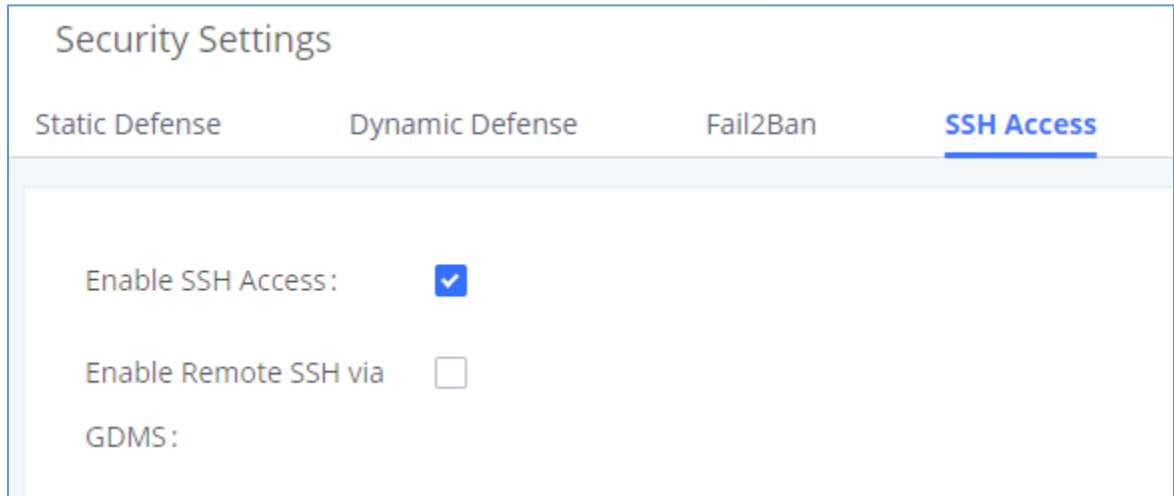
## SSH Access

SSH switch now is available via Web GUI and LCD. User can enable or disable SSH access directly from Web GUI or LCD screen. For web SSH access, please log in UCM630xA web interface and go to Web GUI→**System Settings**→**Security Settings**→**SSH Access**.

The "Enable SSH access" option is for system debugging. If you enable this option, the system will allow SSH access. The SSH connection also requires the username and password of the super administrator. This option is turned off by default. It is recommended to turn off this option when debugging is not required.

Enable the "Enable remote SSH (via GDMS)" option, the system will allow remote SSH access via the GDMS platform. This option is turned off by default, and it is strongly recommended to turn off this option when remote troubleshooting is not required.





**Figure 32: SSH Access**

**Table 18: SSH Access**

<b>Enable SSH Access</b>	This option is used for system debugging. Once enabled, UCM will allow SSH access. The SSH connection requires super administrator's username and password. The default setting is "No". It is recommended to set it to "No" if there is no need for debugging.
<b>Enable Remote SSH via GDMS</b>	If this option is enabled, remote SSH access will be allowed through the GDMS platform. It is strongly recommended to keep this disabled unless remote troubleshooting is necessary.

## LDAP Server

The UCM630xA has an embedded LDAP/LDAPS 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 UCM630xA 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 UCM630xA 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 UCM630xA, the LDAP directory will be set up on the phone and can be used right away for users to access all phonebooks.

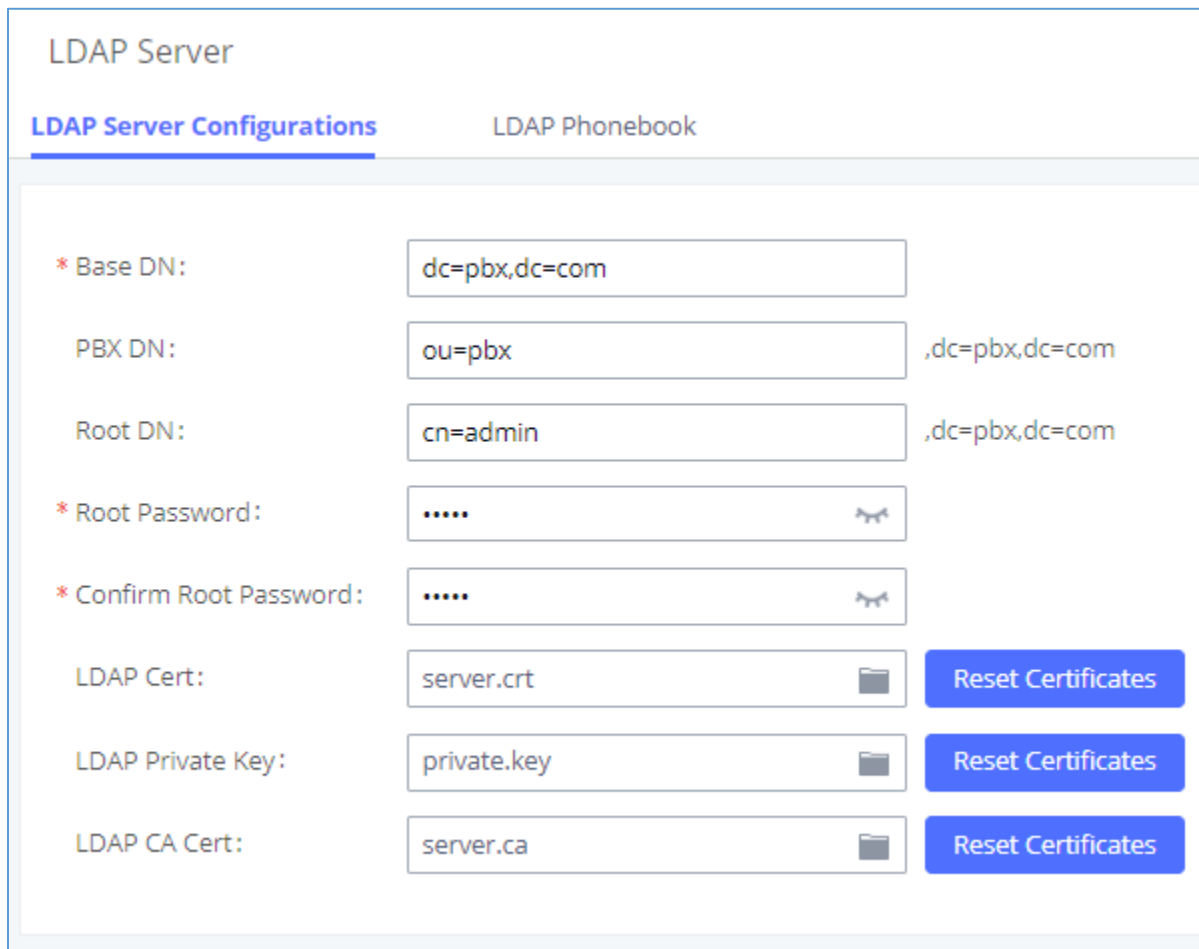
Additionally, users could manually configure the LDAP client settings to manipulate the built-in LDAP server on the UCM630xA. If the UCM630xA has multiple LDAP phonebooks created, in the LDAP client configuration, users could use "dc=pbx,dc=com" as Base DN to have access to all phonebooks on the UCM630xA 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 another 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 UCM630xA.




LDAP Server	
LDAP Server Configurations	LDAP Phonebook
* Base DN:	dc=pbx,dc=com
PBX DN:	ou=pbx ,dc=pbx,dc=com
Root DN:	cn=admin ,dc=pbx,dc=com
* Root Password:	.....
* Confirm Root Password:	.....
LDAP Cert:	server.crt <span>Reset Certificates</span>
LDAP Private Key:	private.key <span>Reset Certificates</span>
LDAP CA Cert:	server.ca <span>Reset Certificates</span>

Figure 33: LDAP Server Configurations

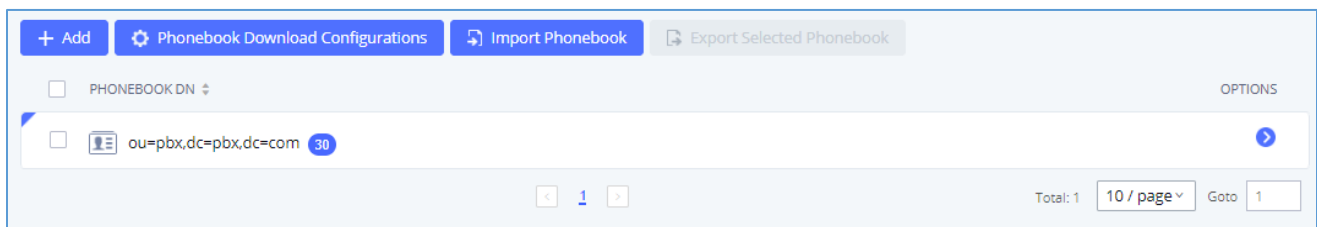




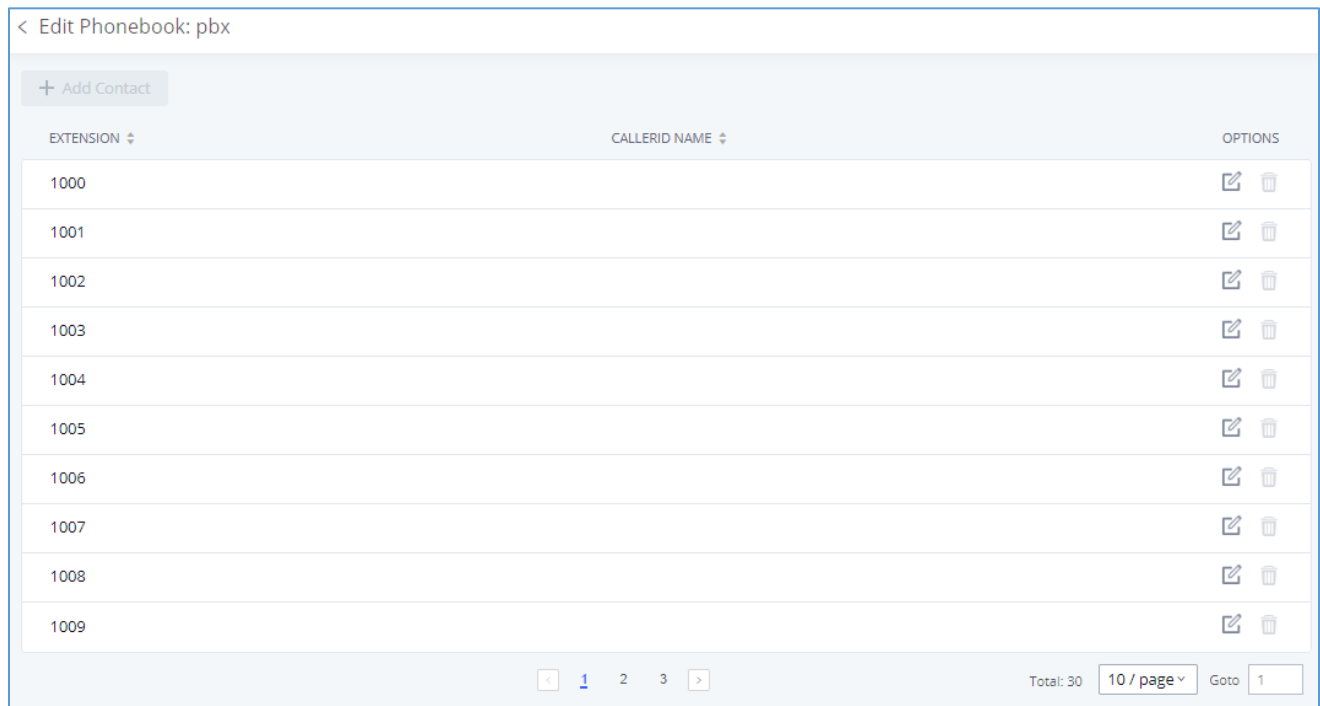
The UCM630xA LDAP server supports anonymous access (read-only) by default. Therefore, the LDAP client does not have to configure username and password to access the phonebook directory. The "Root DN" and "Root Password" here are for LDAP management and configuration where users will need provide for 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.

The UCM630xA support secure LDAP (LDAPS) where the communication is encrypted and secure.



**Figure 34: Default LDAP Phonebook DN**



**Figure 35: 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→**Extension/Trunk**→**Extensions** first. The default LDAP phonebook will then be updated automatically.

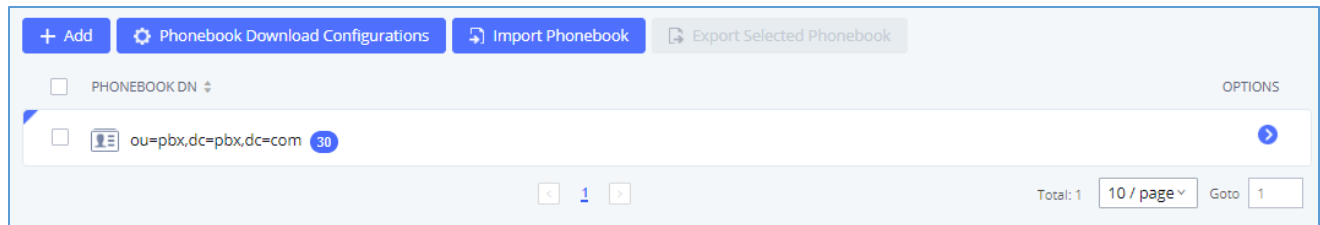


Figure 36: LDAP Server→LDAP Phonebook

- **Add new phonebook**

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

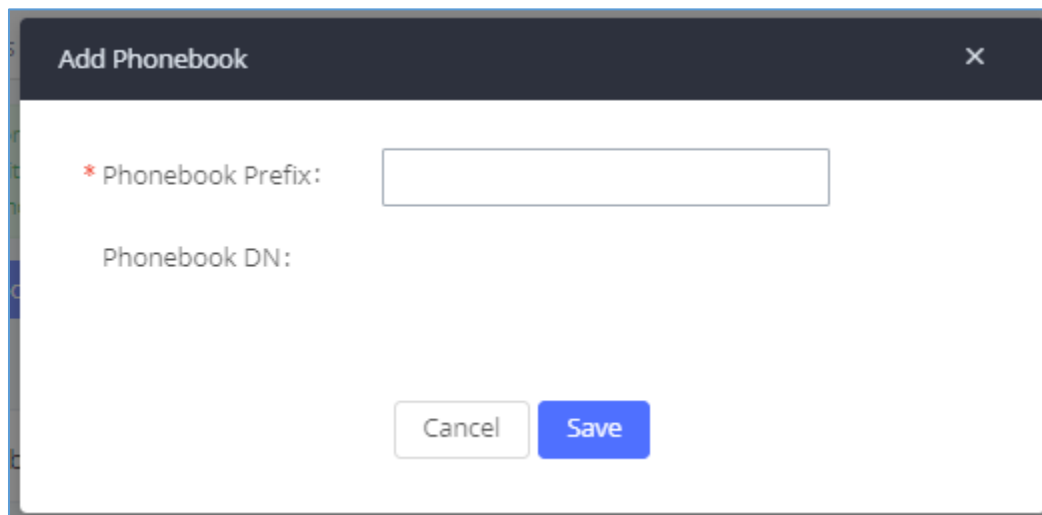


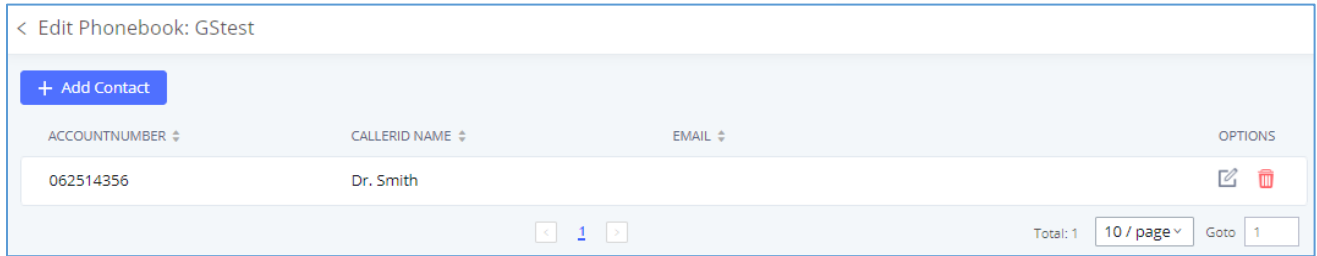


Figure 37: Add LDAP Phonebook

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.

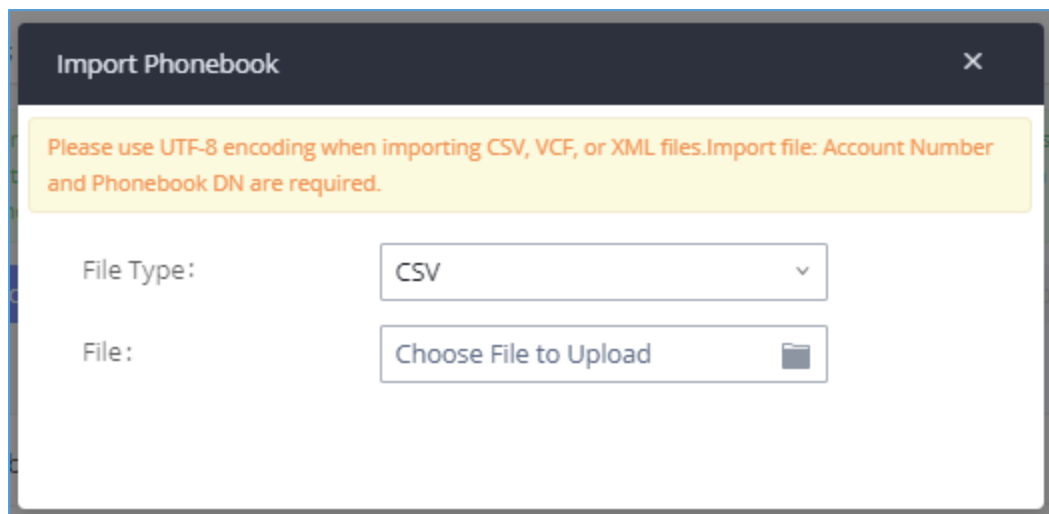




**Figure 38: Edit LDAP Phonebook**

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

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



**Figure 39: Import Phonebook**

The file to be imported must be a CSV, VCF or XML file with UTF-8 encoding. Users can open the 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 UCM630xA 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 Name	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

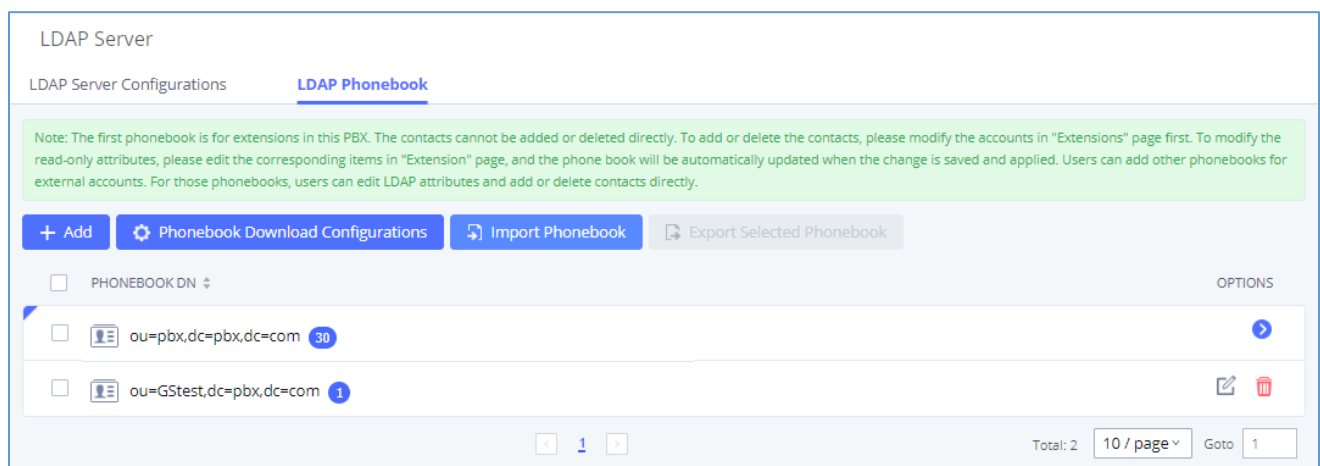
**Figure 40: Phonebook CSV File Format**



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 UCM630xA 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 UCM630xA LDAP Phonebook, the contacts in the CSV file will be added into the existing phonebook. If the phonebook DN does not exist on the UCM630xA 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 UCM630xA.



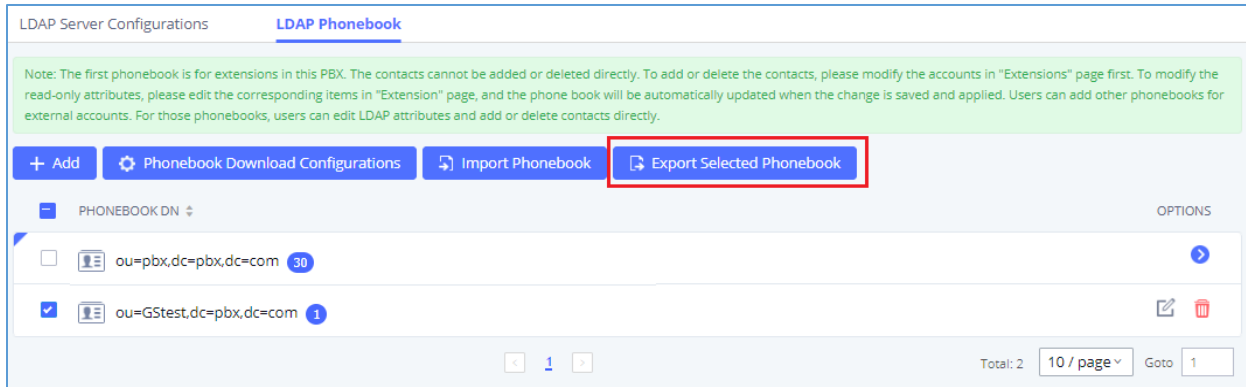
**Figure 41: LDAP Phonebook After Import**

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 UCM630xA 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, VFC or XML file for the users to add more contacts in it and import to the UCM630xA again.





**Figure 42: Export Selected LDAP Phonebook**

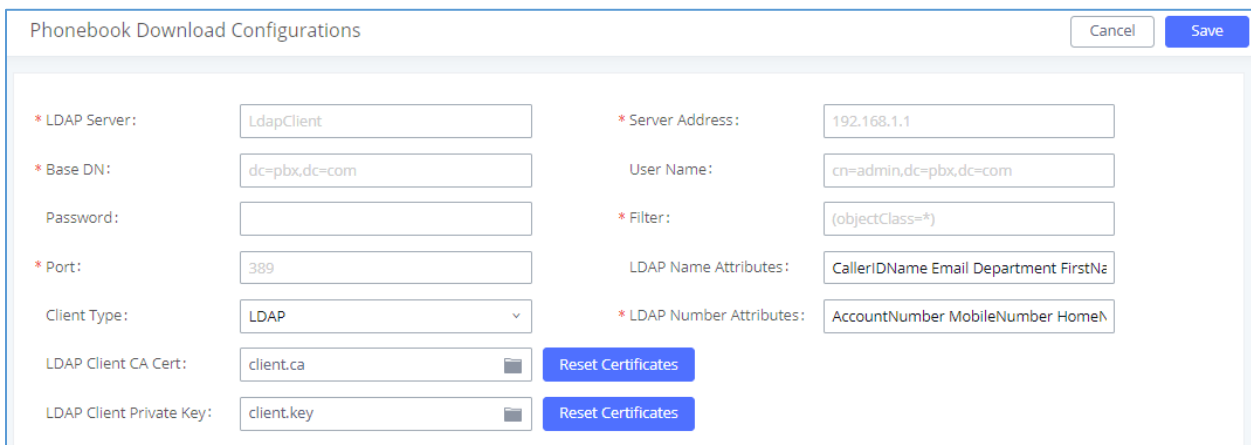
## LDAP Client Configurations

The configuration on LDAP client is useful when you use other LDAP servers. Here we provide an example on how to configure the LDAP client on the UCM.

Assuming the remote server base dn is “**dc=pbx,dc=com**”, configure the LDAP client as follows:

- **LDAP Server** : Enter a name for the remote LDAP server
- **Server Address** : Enter the IP address or domain name for remote LDAP server.
- **Base DN**: dc=pbx,dc=com
- **Username**: Enter username if authentication is required
- **Password**: Enter password if authentication is required
- **Filter**: Enter the filter. Ex: ((CallerIDName=%)(AccountNumber=%))
- **Port**: Enter the port number. Ex:389
- **LDAP Name Attributes**: Enter the name attributes for remote server
- **LDAP Number Attributes**: Enter the number attributes for remote server

The following figure gives a sample configuration for UCM acting as a LDAP client.



**Figure 43: LDAP Client Configurations**



To configure Grandstream IP phones as the LDAP clients for UCM, please refer to the following example:

- **Server Address:** The IP address or domain name of the UCM
- **Base DN:** dc=pbx,dc=com
- **Username:** 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 GXP2170 to successfully use the LDAP server as configured in **[Figure 33: LDAP Server Configurations]**.



### LDAP

LDAP protocol	<input type="text" value="LDAP"/>
Server Address	<input type="text" value="192.168.40.134"/>
Port	<input type="text" value="389"/>
Base	<input type="text" value="dc=pbx,dc=com"/>
User Name	<input type="text"/>
Password	<input type="password"/>
LDAP Number Filter	<input type="text" value="(AccountNumber=%)"/>
LDAP Name Filter	<input type="text" value="(CallerIDName=%)"/>
LDAP Version	<input type="radio"/> Version 2 <input checked="" type="radio"/> Version 3
LDAP Name Attributes	<input type="text" value="CallerIDName"/>
LDAP Number Attributes	<input type="text" value="AccountNumber"/>
LDAP Display Name	<input type="text" value="AccountNumber CallerIDName"/>
Max. Hits	<input type="text" value="50"/>
Search Timeout	<input type="text" value="30"/>
Sort Results	<input checked="" type="radio"/> No <input type="radio"/> Yes
LDAP Lookup	<input checked="" type="checkbox"/> Incoming Calls <input checked="" type="checkbox"/> Outgoing Calls
Lookup Display Name	<input type="text"/>

**Figure 44: GXP2170 LDAP Phonebook Configuration**

## Time Settings

### Automatic Date and Time

The current system time on the UCM630xA can be found under Web GUI→**System Status**→**Dashboard**→**PBX Status**.

To configure the UCM630xA to update time automatically, go to Web GUI→**System Settings**→**Time Settings**→**Automatic date and Time**.

---

 **Note:**

The configurations under Web GUI→**Settings**→**Time Settings**→ **Automatic date and Time** page require reboot to take effect. Please consider configuring auto time updating related changes when setting up the UCM630xA for the first time to avoid service interrupt after installation and deployment in production.

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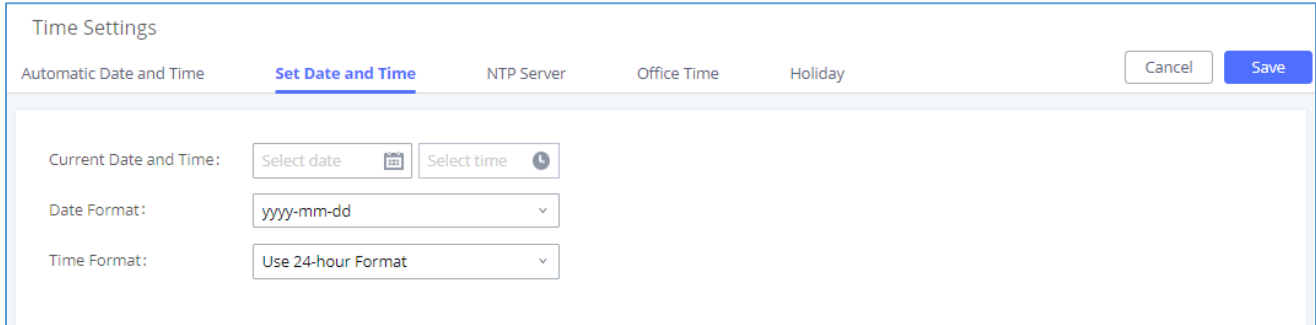


**Table 19: Time Auto Updating**

<b>Remote NTP Server</b>	Specify the URL or IP address of the NTP server for the UCM630xA to synchronize the date and time. The default NTP server is pool.ntp.org.
<b>Enable DHCP Option 2</b>	If set to "Yes", the UCM630xA can get provisioned for Time Zone from DHCP Option 2 in the local server automatically. The default setting is "Yes".
<b>Enable DHCP Option 42</b>	If set to "Yes", the UCM630xA can 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".
<b>Time Zone</b>	Select the proper time zone option so the UCM630xA can display correct time accordingly.

## Set Date and Time

To manually set the time on the UCM630xA, go to Web GUI→**System Settings**→**Time Settings**→**Set Date and Time**. The format is YYYY-MM-DD HH:MM:SS.


**Figure 45: Set Time Manually**

### Note:

Manually setup time will take effect immediately after saving and applying change in the Web GUI. If users would like to reboot the UCM630xA 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**→**Auto Time Updating** page are unchecked or set to empty. Otherwise, time auto updating settings in this page will take effect after reboot.



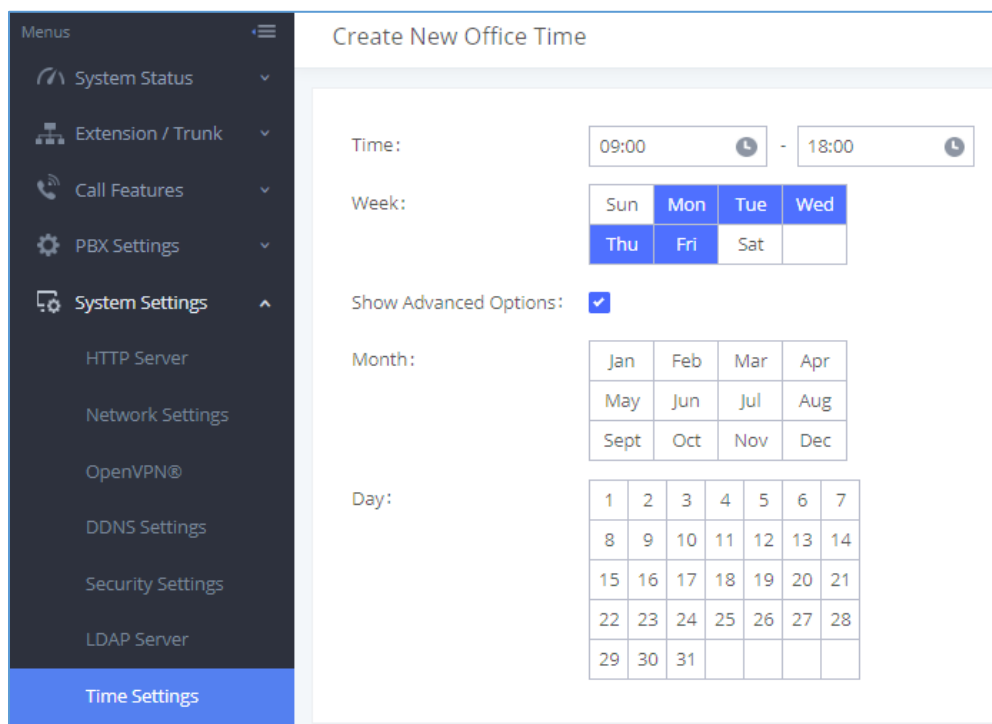


## NTP Server

The UCM630xA can be used as an NTP server for the NTP clients to synchronize their time with. To configure the UCM630xA as the NTP server, set "Enable NTP server" to "Yes" under Web GUI→**System Settings**→**Time Settings**→**NTP Server**. On the client side, point the NTP server address to the UCM630xA IP address or host name to use the UCM630xA as the NTP server.

## Office Time

On the UCM630xA, 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 GUI→**System Settings**→**Time Settings**→**Office Time**. Click on "Add" to create an office time.





**Figure 46: Create New Office Time**

**Table 20: Create New Office Time**



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

Time Settings						
Automatic Date and Time		Set Date and Time		NTP Server	<b>Office Time</b>	Holiday
+ Add		Delete				
<input type="checkbox"/>	INDEX	TIME	WEEK	MONTH	DAY	OPTIONS
<input type="checkbox"/>	1	09:00-18:00	Mon Tue Wed Thu Fri	Default	Default	 

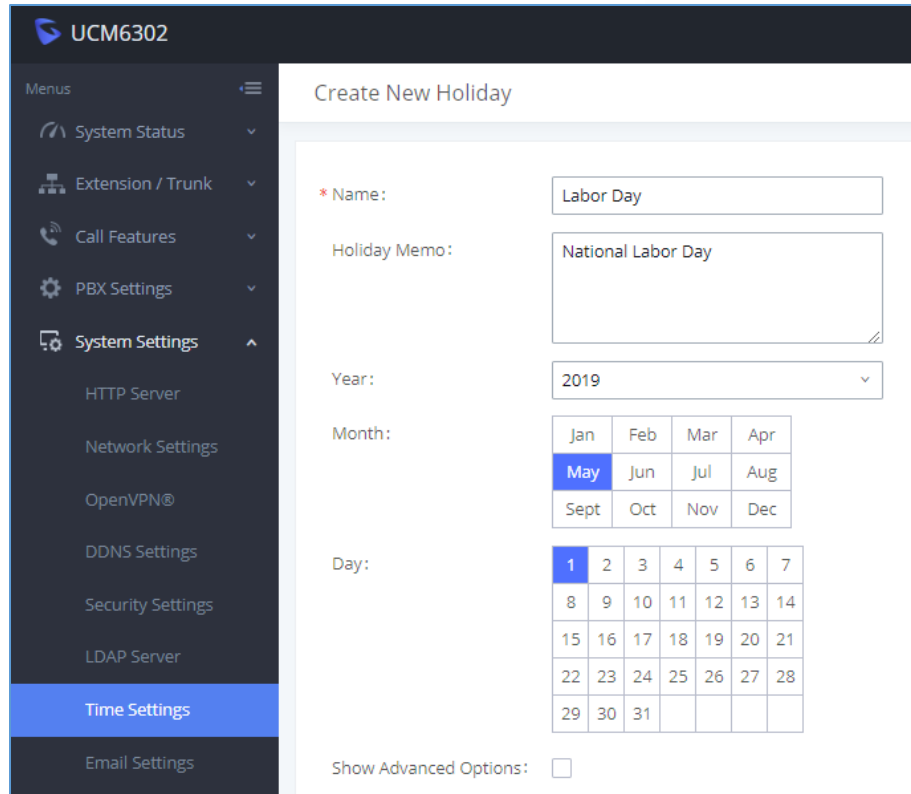
**Figure 47: Settings→Time Settings→Office Time**

- Click on  to edit the office time.
- Click on  to delete the office time.
- Click on **"Delete"** to delete multiple selected office times at once.

## Holiday

On the UCM630xA, 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 GUI→**System Settings**→**Time Settings**→**Holiday**. Click on **"Add"** to create holiday time.





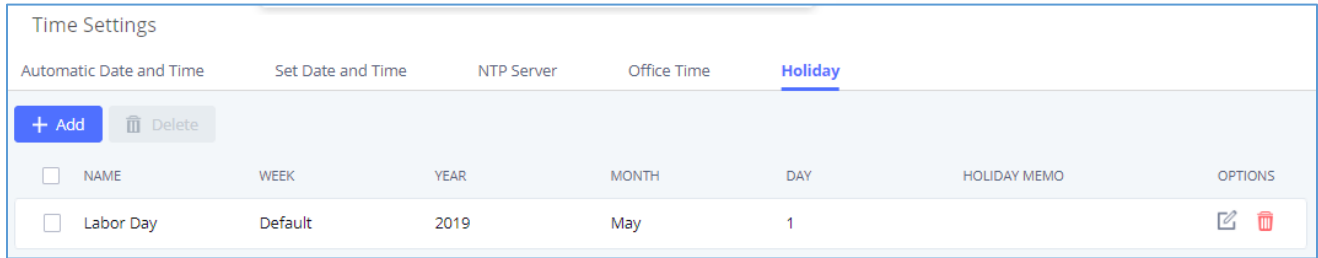
**Figure 48: Create New Holiday**

**Table 21: Create New Holiday**



<b>Name</b>	Specify the holiday name to identify this holiday.
<b>Holiday Memo</b>	Create a note for the holiday.
<b>Month</b>	Select the month for the holiday.
<b>Day</b>	Select the day for the holiday.
<b>Show Advanced Options</b>	Check this option to show advanced options. If selected, please specify the days as holiday in one week below.
<b>Week</b>	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 49: Settings→Time Settings→Holiday**

- Click on  to edit the holiday.
- Click on  to delete the holiday.
- Click on **"Delete"** 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:

[https://www.grandstream.com/hubfs/Product\\_Documentation/office\\_time\\_and\\_holiday\\_on\\_ucm6xxx.pdf](https://www.grandstream.com/hubfs/Product_Documentation/office_time_and_holiday_on_ucm6xxx.pdf)

## Email Settings

### Email settings

The Email application on the UCM630xA can be used to send out alert event Emails, Voicemail (Voicemail-To-Email) etc. The configuration parameters can be accessed via Web GUI→**System Settings→Email Settings→Email Settings**.

**Table 22: Email Settings**

<b>TLS Enable</b>	Enable or disable TLS during transferring/submitted your Email to another SMTP server. The default setting is "Yes".
<b>Type</b>	Select Email type. <ul style="list-style-type: none"> <li>• <b>MTA:</b> 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.</li> <li>• <b>Client:</b> Submit Emails to the SMTP server. A SMTP server is required, and users need login with correct credentials.</li> </ul>
<b>Domain</b>	Specify the domain name to be used in the Email when using type "MTA".



<b>SMTP Server</b>	Specify the SMTP server when using type "Client".
<b>Enable SASL Authentication</b>	Enable SASL Authentication. When disabled, UCM will not try to use the username and password for mail client login authentication. Most of the mail server requires login authentication while some others private mail servers allow anonymous login which requires disabling this option to send Email as normal. For Exchange Server, please disable this option.
<b>Username</b>	Username is required when using type "Client". Normally it is the Email address.
<b>Password</b>	Password to login for the above Username (Email address) is required when using type "Client".
<b>POP/POP3 Server Address</b>	Configure the POP/POP3 server address for the configured username Example: pop.gmail.com
<b>POP/POP3 Server Port</b>	Configure the POP/POP3 server port for the configured username Example: 995
<b>Display Name</b>	Specify the display name in the FROM header in the Email.
<b>Sender</b>	Specify the sender's Email address. For example: pbx@example.mycompany.com.

The following figure shows a sample Email setting on the UCM630xA, assuming the Email is using 192.168.6.202 as the SMTP server.



### Email Settings

Email Settings
Email Template
Email Send Log

TLS Enable:	<input checked="" type="checkbox"/>
Type:	<input type="text" value="Client"/>
* SMTP Server:	<input type="text" value="192.168.6.202:25"/>
* Enable SASL Authentication:	<input checked="" type="checkbox"/>
* Username:	<input type="text" value="GStest@gmail.com"/>
* Password:	<input type="password" value="....."/>
POP/POP3 Server Address:	<input type="text"/>
POP/POP3 Server Port:	<input type="text"/>
* Display Name:	<input type="text" value="PBX"/>
* Sender:	<input type="text" value="GStest@gmail.com"/>

















Figure 50: UCM630xA 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 UCM630xA.

### Email Templates

The Email templates on the UCM630xA can be used for email notification, the configuration parameters can be accessed via Web GUI→**Settings**→**Email Settings**→**Email Templates**.



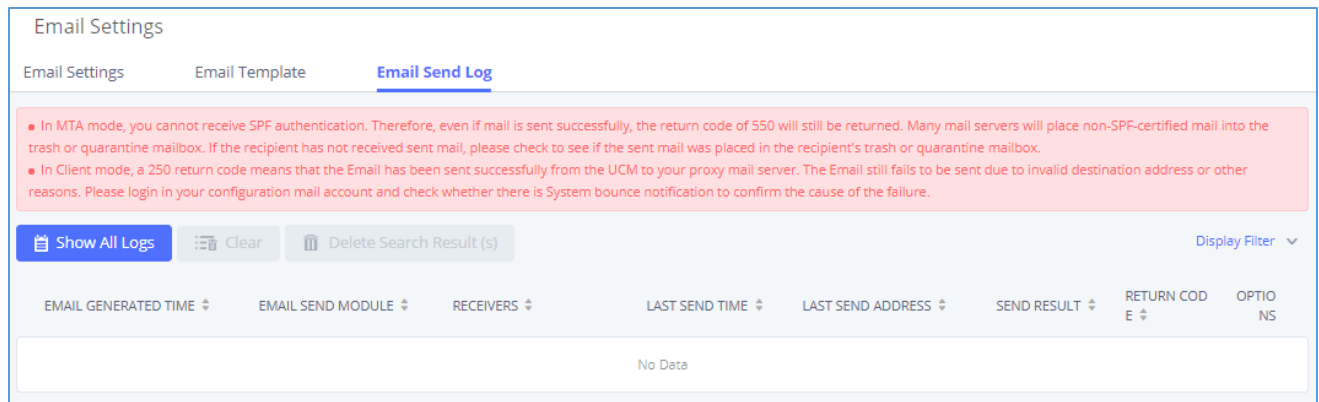
Email Settings				
Email Settings		<a href="#">Email Template</a>	Email Send Log	
TYPE	NAME	TIME	OPTIONS	
Scheduled Conference Report	conferencescheduledreport_template.html	2021-01-14 13:35:53 UTC+01:00		
Call Queue Statistics	callqueuestatistics_template.html	2020-12-30 12:53:28 UTC+01:00		
Emergency Calls	emergency_template.html	2020-12-30 12:53:28 UTC+01:00		
User Password	password_template.html	2020-12-30 12:53:28 UTC+01:00		
Conference Report	conferencereport_template.html	2020-12-30 12:53:28 UTC+01:00		
Audio Conference Schedule	conference_template.html	2020-12-30 12:53:28 UTC+01:00		
PMS	pms_template.html	2020-12-30 12:53:28 UTC+01:00		
Voicemail	voicemail_template.html	2020-12-30 12:53:28 UTC+01:00		
Fax Sending	sendfax_template.html	2020-12-30 12:53:28 UTC+01:00		
Fax	fax_template.html	2020-12-30 12:53:28 UTC+01:00		
Video Conference Schedule	mcm_template.html	2020-12-30 12:53:28 UTC+01:00		
CDR	cdr_template.html	2020-12-30 12:53:28 UTC+01:00		
Extension	account_template.html	2020-12-30 12:53:28 UTC+01:00		
Alert Events	alert_template.html	2020-12-30 12:53:28 UTC+01:00		
Reset Password	resetpassword_template.html	2020-12-30 12:53:28 UTC+01:00		
Missed Calls	missedcall_template.html	2020-12-30 12:53:28 UTC+01:00		

**Figure 51: Email Template**

## Email Send Log

Under UCM Web GUI→**System Settings**→**Email Settings**→**Email Send Log**, users could search, filter and check whether the Email is sent out successfully or not. This page will also display the corresponding error message if the Email is not sent out successfully.





**Figure 52: Email Send Log**

**Table 23: Email Log – Display Filter**

Field	Description
Start Time	Enter the start time for filter
End Time	Enter the end time for filter
Receivers	Enter the email recipient, while searching for multiple recipients, please separate then with comma and no spaces.
Send Result	Enter the status of the send result to filter with
Return Code	Enter the email code to filter with
Email Send Module	Select the email module to filter with from the drop-down list, which contains: All Modules Extension Voicemail Meeting Schedule User Password Alert Events CDR Test

Email logs will be shown on bottom of the “Email Send Log” page, as shown on the following figure.





EMAIL GENERATED TIME	EMAIL SEND MODULE	RECEIVERS	LAST SEND TIME	LAST SEND ADDRESS	SEND RESULT	RETURN CODE	OPTIONS
2020-12-22 18:00:03	Video Conference Schedule	m.g@gmail.com	12-22 18:00:05	m.g@gmail.com	sent	250	ⓘ
2020-12-21 18:00:04	Video Conference Schedule	m.g@gmail.com	12-21 18:00:06	m.g@gmail.com	sent	250	ⓘ
2020-12-20 18:00:04	Video Conference Schedule	m.g@gmail.com	12-20 18:00:07	m.g@gmail.com	sent	250	ⓘ
2020-12-19 18:00:03	Video Conference Schedule	m.g@gmail.com	12-19 18:00:06	m.g@gmail.com	sent	250	ⓘ
2020-12-18 18:00:03	Video Conference Schedule	m.g@gmail.com	12-18 18:00:06	m.g@gmail.com	sent	250	ⓘ
2020-12-17 18:00:03	Video Conference Schedule	m.g@gmail.com	12-17 18:00:06	m.g@gmail.com	sent	250	ⓘ

Total: 32    10 / page    Goto 1

**Figure 53: Email Logs**

Below are the codes returned when sending emails and their description:

**Table 24: Email Codes**

Code	Description
<b>250</b>	Mail sent successfully
<b>501</b>	Address format parsing error, 501 will be returned when there are unacceptable characters in the recipient's email address in MTA mode. Please check if the recipient's email address format is correct. The "sender" configured on the client is your mail account.
<b>535</b>	The user name and password verification in the client mode is incorrect. Please check whether the user name and password are configured correctly.
<b>550</b>	Possible reasons: 1. The recipient's mailbox user name does not exist or is in a banned state, please check whether the email recipient is the correct email address. 2. The number of destination addresses sent by the sender exceeds the maximum limit per day and is temporarily blacklisted. Please reduce the sending frequency or try again the next day. 3. The sender's IP does not pass the SPF permission test of the sending domain. Emails sent in MTA mode may return this error code even if they are sent.
<b>552</b>	The sent email is too large or the email attachment type is prohibited
<b>553</b>	The sender and the email account are inconsistent, please configure the sender as your email account correctly.
<b>554</b>	The email was identified as spam. Please reduce the sending frequency or try again the next day



<b>none</b>	<p>Indicates that there is no return code.</p> <p>If the sending result is "deferred", the general reason is that the mail service area is configured incorrectly. Please check whether the server configuration is correct.</p> <p>If the sending result is "bounced", the general reason is that the receiving email address domain name is wrong, please check whether the email recipient is the correct email address. If it is in MTA mode, please check whether the "domain" is configured to be in the same domain name as the "recipient".</p>
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## TR-069

To configure TR-069 on Grandstream devices, set following parameters:

Parameter	Description
<b>Enable TR-069</b>	Toggle it on to enable TR-069. It is enabled by default
<b>ACS URL</b>	URL for TR-069 Auto Configuration Servers (ACS), e.g., <a href="http://myacs.grandstream.com">http://myacs.grandstream.com</a>
<b>TR-069 Username</b>	ACS username for TR-069, must be the same as in the ACS configuration.
<b>TR-069 Password</b>	ACS password for TR-069, must be the same as in the ACS configuration.
<b>Periodic Inform Enable</b>	Enables periodic inform. If set to 'Yes', device will send inform packets to the ACS.
<b>Periodic Inform Interval</b>	Periodic time when UCM630xA will send inform packets to TR-069 ACS server. This option is specified in seconds.
<b>ACS Connection Request Username</b>	The username for the ACS to connect to UCM.
<b>ACS Connection Request Password</b>	The password for the ACS to connect to UCM.
<b>Connection Request Port</b>	Port for incoming connection requests. The default value is <b>7547</b> .
<b>CPE Cert File</b>	The Cert file for UCM to connect to the ACS via SSL.
<b>CPE Cert Key</b>	The Cert key for UCM to connect to the ACS via SSL.



# 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 UCM630xA 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 UCM630xA. The settings for this feature can be accessed via Web GUI→**Value-added Features**→**Zero Config**.

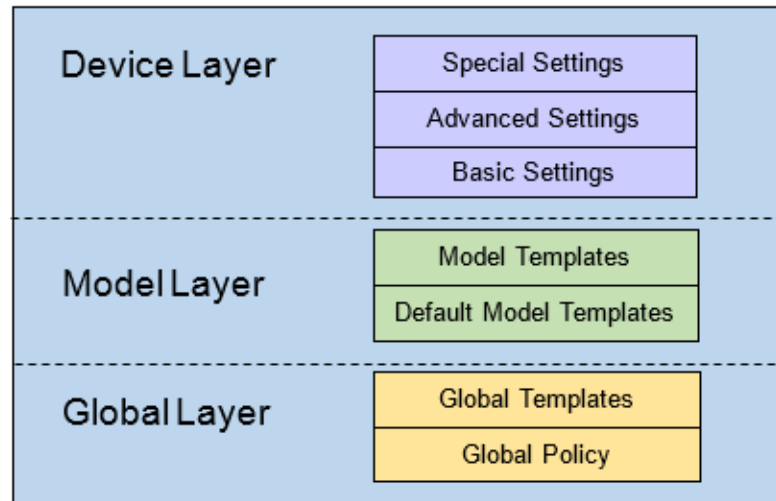
## Configuration Architecture for End Point Device

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 54: Zero Config Configuration Architecture for End Point Device**

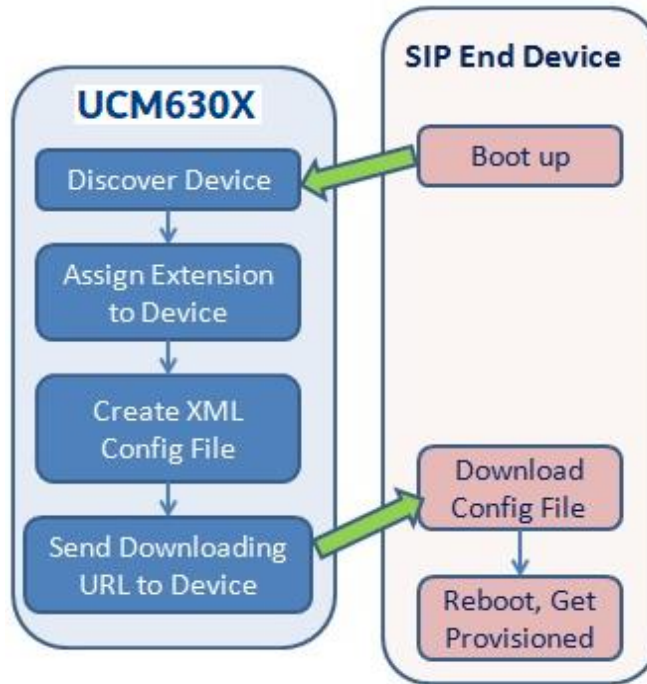
The configuration options in model layer and device layer have all the option in global layers already, i.e., the options in global layer is a subset of the options in model layer and device layer. If an option is set in all three layers with different values, the highest layer value will override the 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 UCM630xA 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 UCM630xA for auto provisioning. Three methods of auto provisioning are used.





**Figure 55: UCM630xA Zero Config**

- **SIP SUBSCRIBE**

When the phone boots up, it sends out SUBSCRIBE to a multicast IP address in the LAN. The UCM630xA 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 UCM630xA and take the new configuration.

- **DHCP OPTION 66**

Route mode needs to be set to use this feature. When the phone restarts (by default DHCP Option 66 is turned on), it will send out a DHCP DISCOVER request. The UCM630xA 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 UCM630xA.

- **mDNS**

When the phone boots up, it sends out mDNS query to get the TFTP server address. The UCM630xA will respond with its own address. The phone will then send TFTP request to download the XML config file from the UCM630xA.

To start the auto provisioning process, under Web GUI→**Value-added Features**→**Zero Config**→**Zero Config Settings**, fill in the auto provision information.



Zero Config

Zero Config    Global Policy    Global Templates    Model Templates    Model Update    Zero Config Settings

---

**Basic Settings**

Enable Zero Config:

Enable Automatic Configuration:

Assignment:

**Extension Assignment**

Auto provision automatically provides an extension to the device.  
 There are two methods of auto provision: SIP SUBSCRIBE and DHCP Option 66.  
 For example, when the device boots up, it will send SIP SUBSCRIBE multicast in the LAN. The PBX will find it, create an account and return a URL of the config file for the device to download.

Auto Assign Extension:

Zero Config Extension Segment: 5000 - 6299 [Zero Config Extension Segment](#)

Enable Pick Extension:

Pick Extension Segment: 4000 - 4999 [Pick Extension Segment](#)

Pick Extension Period (hour):

**Network Settings**

Subnet Whitelist:  -

[Add Subnet Whitelist](#) +

Save

**Figure 56: Auto Provision Settings**

**Table 25: Auto Provision Settings**

<b>Enable Zero Config</b>	Enable or disable the zero-config feature on the PBX. The default setting is enabled.
<b>Enable Automatic Configuration Assignment</b>	<p>By default, this is disabled. If disabled, when SIP device boots up, the UCM630xA 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 UCM630xA.</p> <p><b>Note:</b>          When disabled, SIP devices can still be provisioned by manually sending NOTIFY from the UCM630xA which will include the XML config file URL for the SIP device to download.</p>
<b>Auto Assign Extension</b>	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.



<b>Zero Config Extension Segment</b>	Click on the link "Zero Config Extension Segment" to specify the 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 GUI→ <b>PBX Settings</b> → <b>General Settings</b> → <b>General page</b> →Extension Preference section: "Auto Provision Extensions".
<b>Enable Pick Extension</b>	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.
<b>Pick Extension Segment</b>	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 GUI → <b>PBX Settings</b> → <b>General Settings</b> → <b>General page</b> → Extension Preference section: "Pick Extensions".
<b>Pick Extension Period (hour)</b>	Specify the number of minutes to allow the phones being provisioned to pick extensions.
<b>Subnet Whitelist</b>	This feature allows the UCM to provision devices in different subnets other than UCM network. Enter subnets IP addresses to allow devices within these subnets to be provisioned. The syntax is <b>&lt;IP&gt;/&lt;CIDR&gt;</b> . <u>Examples:</u> 10.0.0.1/8 192.168.6.0/24 <b>Note:</b> Only private IP ranges (10.0.0.0   172.16.0.0   192.168.0.0) are supported.

Please make sure an extension is manually assigned to the phone or "Automatically Assign Extension" is enabled during provisioning. After the configuration on the UCM630xA Web GUI, click on "Save" and "Apply Changes". Once the phone boots up and picks up the config file from the UCM630xA, it will take the configuration right away.

## Discovery

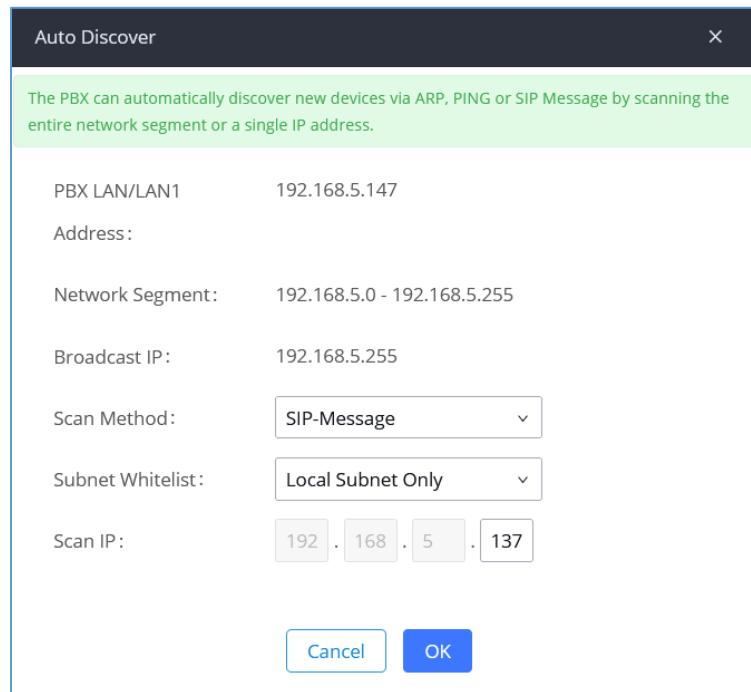
Grandstream endpoints are automatically discovered after bootup. Users could also manually discover device by specifying the IP address or scanning the entire LAN network. Three methods are supported to scan the devices.

- PING
- ARP



- SIP Message (NOTIFY)

Click on "Auto Discover" under Web GUI→Value-added Features→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 UCM630xA. 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 UCM630xA Web GUI→Value-added Features→Zero Config→Auto Provisioning Settings.



The PBX can automatically discover new devices via ARP, PING or SIP Message by scanning the entire network segment or a single IP address.

PBX LAN/LAN1: 192.168.5.147

Address:

Network Segment: 192.168.5.0 - 192.168.5.255

Broadcast IP: 192.168.5.255

Scan Method: SIP-Message


























Subnet Whitelist: Local Subnet Only

Scan IP: 192 . 168 . 5 . 137

Buttons: Cancel, OK

**Figure 57: 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 /Reboot /Access Device WebGUI) are displayed in the list.

<input type="checkbox"/>	MAC ADDRESS	IP ADDRESS	EXTENSION	VERSION	VENDOR	MODEL	CREATE CONFIG	OPTIONS
<input type="checkbox"/>	000B82000001	<a href="#">192.168.2.111</a>	1000	unknown	GRANDSTREAM	GXV3275	--	    
<input type="checkbox"/>	000B8227FB15	<a href="#">192.168.2.108</a>	--	1.0.3.208	GRANDSTREAM	GXV3275	--	    
<input type="checkbox"/>	000B82A46ACE	<a href="#">192.168.2.106</a>	--	1.0.0.36	GRANDSTREAM	--	--	    
<input type="checkbox"/>	000B82D33AC4	<a href="#">192.168.2.105</a>	--	20.19.10.30	GRANDSTREAM	--	--	    
<input type="checkbox"/>	000B82F66470	<a href="#">192.168.2.107</a>	--	10.19.9.26	GRANDSTREAM	--	--	    

**Figure 58: Discovered Devices**





## Uploading Devices List

Besides the built-in discovery method on the UCM, users could prepare a list of devices on .CSV file and upload it by clicking on the button **"Import"**, after which a success message prompt should be displayed.

Users need to make sure that the CSV file respects the format as shown on the following figure and that the entered information is correct (valid IP address, valid MAC address, device model and an existing account), otherwise the UCM will reject the file and the operation will fail:

	A	B	C	D	E	F	G	H	I	J	K	L	M	N
1	===== Device Start =====													
2	config_name	vendor	state	ip	account_secre	file_url	url_paramete	last_access	mac	version	ad_state	model	hot_desking	port
3		Grandstream		192.168.5.172		https://192.168.5.147:806	#####	000B8249E	1.0.7.10		0	GXW4248	no	5060
4														
5	===== Device Start =====													
6	config_name	vendor	state	ip	account_secre	file_url	url_paramete	last_access	mac	version	ad_state	model	hot_desking	port
7		Grandstream		192.168.5.114		https://192.168.5.147:806	#####	000B826B	1.0.3.227		0	GXV3240	no	5060
8														
9	===== Device Start =====													
10	config_name	vendor	state	ip	account_secre	file_url	url_paramete	last_access	mac	version	ad_state	model	hot_desking	port
11		Grandstream		192.168.5.201		https://192.168.5.147:806	#####	000B826F	1.0.1.106		0	--	no	5080

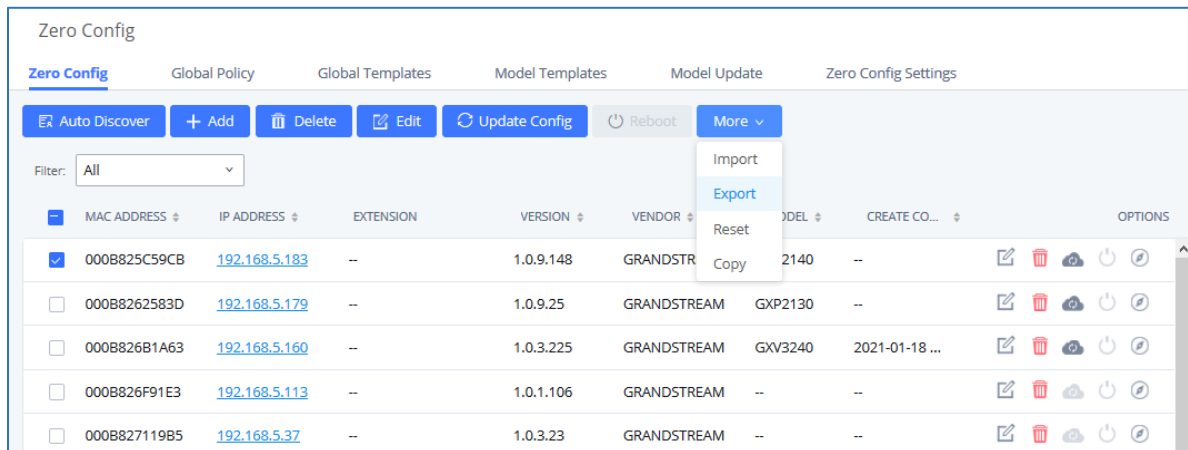
Figure 59: Device List - CSV file Sample

## Managing Discovered Devices

- **Sorting:** Press ▲ or ▼ to sort per MAC Address, IP Address, Version, Vendor, Model or Create Config columns from lower to higher or higher to lower, respectively.

Filter:

- **Filter:** Select a filter to display corresponding results.
  - **All:** Display all discovered devices.
  - **Scan Results:** Display only manually discovered devices. [Discovery]
  - **IP Address:** Enter device IP and press **Search** button.
  - **MAC Address:** Enter device MAC and press **Search** button.
  - **Model:** Enter a model name and press Search button. Example: GXP2130.
  - **Extension:** Enter the extension number and press Search button.

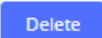


MAC ADDRESS	IP ADDRESS	EXTENSION	VERSION	VENDOR	MODEL	CREATE CO...	OPTIONS
<input checked="" type="checkbox"/> 000B825C59CB	<a href="#">192.168.5.183</a>	--	1.0.9.148	GRANDSTR	2140	--	
<input type="checkbox"/> 000B8262583D	<a href="#">192.168.5.179</a>	--	1.0.9.25	GRANDSTREAM	GXP2130	--	
<input type="checkbox"/> 000B826B1A63	<a href="#">192.168.5.160</a>	--	1.0.3.225	GRANDSTREAM	GXV3240	2021-01-18 ...	
<input type="checkbox"/> 000B826F91E3	<a href="#">192.168.5.113</a>	--	1.0.1.106	GRANDSTREAM	--	--	
<input type="checkbox"/> 000B827119B5	<a href="#">192.168.5.37</a>	--	1.0.3.23	GRANDSTREAM	--	--	

Figure 60: Managing Discovered Devices



From the main menu of zero config, users can perform the following operations:

- Click on  in order to access to the discovery menu as shown on [Discovery] section.
- Click on  to add a new device to zero config database using its MAC address.
- Click on  to delete selected devices from the zero-config database.
- Click on  to modify selected devices.
- Click on  to batch update a list of devices, the UCM on this case will send SIP NOTIFY message to all selected devices in order to update them at once.
- Click on  to reboot selected devices (the selected devices, should have been provisioned with extensions since the phone will authenticate the server which is trying to send it reboot command).
- Click on  to clear all devices configurations.
- Click on  to upload CSV file containing list of devices.
- Click on  to export CSV file containing list of devices. This file can be imported to another UCM to quickly set it up with the original UCM's devices.
- Click on  to copy configuration from one device to another. This can be useful for easily replace devices and note that this feature works only between devices of same model.

All these operations will be detailed on the next sections.

## Global Configuration

### Global Policy

Global configuration will apply to all the connected Grandstream SIP end point devices in the same LAN with the UCM630xA no matter what the Grandstream device model it is. It is divided into two levels:

- **Global Policy**
- **Global Templates**

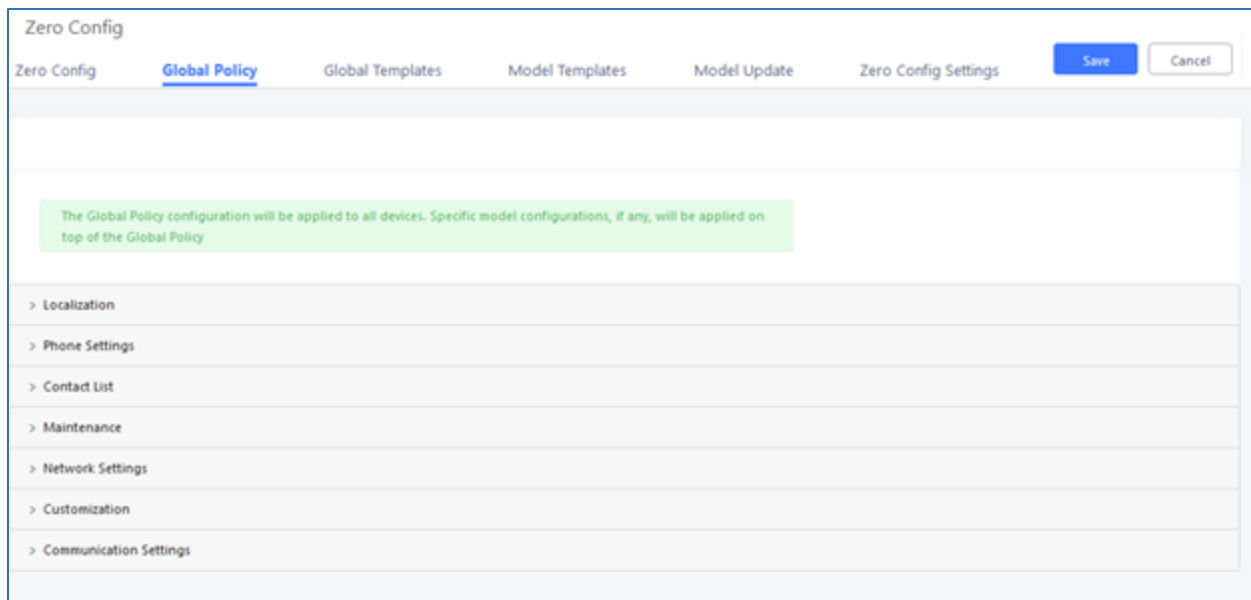
**Note:** **Global Templates** configuration has higher priority to **Global Policy** configuration.



Global Policy can be accessed in Web GUI→**Value-added Features**→**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.
- **Communication Settings:** configure Email and FTP settings

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



**Figure 61: Global Policy Categories**

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

**Table 26: Global Policy Parameters – Localization**

Language settings	
<b>Language</b>	Select the LCD display language on the SIP end device.
Date and Time	
<b>Date Format</b>	Configure the date display format on the SIP end device’s LCD.



<b>Time Format</b>	Configure the time display in 12-hour or 24-hour format on the SIP end device's LCD.
<b>Enable NTP</b>	To enable the NTP service.
<b>NTP Server</b>	Configure the URL or IP address of the NTP server. The SIP end device may obtain the date and time from the server.
<b>NTP Update Interval</b>	Configure the NTP update interval.
<b>Time Zone</b>	Configure the time zone used on the SIP end device.
<b>Enable Daylight Saving Time</b>	Select either to enable or disable the DST.

**Table 27: Global Policy Parameters – Phone Settings**

<b>Default Call Settings</b>	
<b>Dial Plan</b>	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.
<b>Enable Call Features</b>	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.
<b>Use # as Dial Key</b>	If set to “Yes”, pressing the number key “#” will immediately dial out the input digits.
<b>Auto Answer by Call-info</b>	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.
<b>NAT Traversal</b>	Configure if NAT traversal mechanism is activated.
<b>User Random Port</b>	If set to “Yes”, this parameter will force random generation of both the local SIP and RTP ports.
<b>General Settings</b>	
<b>Call Progress Tones</b>	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]]]; <ul style="list-style-type: none"> <li>Frequencies are in Hz and cadence on and off are in 10ms).</li> <li>“on” is the period (in ms) of ringing while “off” is the period of silence. Up to three cadences are supported.</li> <li>Please refer to user manual of the SIP devices to be provisioned for more details</li> </ul>
<b>HEADSET Key Mode</b>	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 28: Global Policy Parameters – Contact List**

LDAP Phonebook	
<b>Source</b>	Select “Manual” or “PBX” as the LDAP configuration source. <ul style="list-style-type: none"> <li>• If “Manual” is selected, the LDAP configuration below will be applied to the SIP end device.</li> <li>• If “PBX” is selected, the LDAP configuration built-in from UCM630xA Web GUI→<b>System Settings</b>→<b>LDAP Server</b> will be applied.</li> </ul>
<b>Address</b>	Configure the IP address or DNS name of the LDAP server.
<b>Port</b>	Configure the LDAP server port. The default value is 389.
<b>Base DN</b>	This is the location in the directory where the search is requested to begin. Example: <ul style="list-style-type: none"> <li>• dc=grandstream, dc=com</li> <li>• ou=Boston, dc=grandstream, dc=com</li> </ul>
<b>Username</b>	Configure the bind “Username” for querying LDAP servers. The field can be left blank if the LDAP server allows anonymous binds.
<b>Password</b>	Configure the bind “Password” for querying LDAP servers. The field can be left blank if the LDAP server allows anonymous binds.
<b>Number Filter</b>	Configure the filter used for number lookups. Please refer to user manual for more details.
<b>Name Filter</b>	Configure the filter used for name lookups. Please refer to user manual for more details.
<b>Version</b>	Select the protocol version for the phone to send the bind requests. The default value is 3.
<b>Name Attribute</b>	Specify the “name” attributes of each record which are returned in the LDAP search result. Example: gn cn sn description
<b>Number Attribute</b>	Specify the “number” attributes of each record which are returned in the LDAP search result. Example: telephoneNumber telephoneNumber Mobile
<b>Display Name</b>	Configure the entry information to be shown on phone’s LCD. Up to 3 fields can be displayed. Example: %cn %sn %telephoneNumber



<b>Max Hits</b>	Specify the maximum number of results to be returned by the LDAP server. Valid range is 1 to 3000. The default value is 50.
<b>Search Timeout</b>	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. Default value is 30.
<b>Sort Results</b>	Specify whether the searching result is sorted or not. Default setting is No.
<b>Incoming Calls</b>	Configure to enable LDAP number searching when receiving calls. The default setting is No.
<b>Outgoing Calls</b>	Configure to enable LDAP number searching when making calls. The default setting is No.
<b>Lookup Display Name</b>	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.
<b>XML Phonebook</b>	
<b>Phonebook XML Server</b>	Select the source of the phonebook XML server. <ul style="list-style-type: none"> <li>• <b>Disable</b> Disable phonebook XML downloading.</li> <li>• <b>Manual</b> 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.</li> <li>• <b>Local UCM Server</b> 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.</li> </ul>
<b>Phonebook Download Interval</b>	Configure the phonebook download interval (in Minute). If set to 0, automatic download will be disabled. Valid range is 5 to 720.
<b>Remove manually edited entries on download</b>	If set to “Yes”, when XML phonebook is downloaded, the entries added manually will be automatically removed.

Table 29: Global Policy Parameters – Maintenance

### Upgrade and Provision

<b>Firmware Source</b>	Firmware source via ZeroConfig provisioning could a URL for external server address, local UCM directory or USB media if plugged in to the UCM630xA. Select a source to get the firmware file:
------------------------	--



	<ul style="list-style-type: none"> <li>• <b>URL</b> If select to use URL to upgrade, complete the configuration for the following four parameters: “Upgrade Via”, “Server Path”, “File Prefix” and “File Postfix”.</li> <li>• <b>Local UCM Server</b> Firmware can be uploaded to the UCM630xA 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.</li> <li>• <b>Local USB Media</b> If selected, the USB storage device needs to be plugged into the UCM630xA and the firmware file must be put under a folder named “ZC_firmware” in the USB storage root directory.</li> <li>• <b>Local SD Card Media</b> If selected, an SD card needs to be plugged into the UCM630xA and the firmware file must be put under a folder named “ZC_firmware” in the USB storage root directory.</li> </ul>
<b>Upgrade via</b>	When URL is selected as firmware source, configure upgrade via TFTP, HTTP or HTTPS.
<b>Server Path</b>	When URL is selected as firmware source, configure the firmware upgrading server path.
<b>File Prefix</b>	Configure the Config Server Path.
<b>Config Server Path</b>	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.
<b>Allow DHCP Option 43/66</b>	If DHCP option 43 or 66 is enabled on the LAN side, the TFTP server can be redirected.
<b>Automatic Upgrade</b>	<p>If enabled, the end point device will automatically upgrade if a new firmware is detected. Users can select automatic upgrading by day, by week or by minute.</p> <ul style="list-style-type: none"> <li>• <b>By week</b> Once selected, specify the day of the week to check HTTP/TFTP server for firmware upgrades or configuration files changes.</li> <li>• <b>By day</b> Once selected, specify the hour of the day to check the HTTP/TFTP server for firmware upgrades or configuration files changes.</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>By minute</b> Once selected, specify the interval <b>X</b> that the SIP end device will request for new firmware every <b>X</b> minutes.</li> </ul>
<b>Firmware Upgrade Rule</b>	Specify how firmware upgrading and provisioning request to be sent.
<b>Zero Config</b>	Select either to enable or disable zero config.
<b>Web Access</b>	
<b>Admin Password</b>	Configure the administrator password for admin level login.
<b>End-User Password</b>	Configure the end-user password for the end user level login.
<b>Web Access Mode</b>	Select HTTP or HTTPS as the web access protocol.
<b>Web Server Port</b>	Configure the port for web access. The valid range is 1 to 65535.
<b>RTSP Port</b>	Configure the RTSP Port.
<b>Enable UPnP Discovery</b>	Select either to enable or disable Enable UPnP Discovery
<b>Login Settings</b>	Configure the login settings.
<b>User Login Timeout</b>	Configure User Login Timeout.
<b>Maximum Consecutive Failed Login Attempts</b>	Configure Maximum Consecutive Failed Login Attempts.
<b>Login Error Lock Time</b>	Configure Login Error Lock Time.
<b>Security</b>	
<b>Disable Telnet/SSH</b>	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.
<b>Syslog</b>	
<b>Syslog Server</b>	Configure the URL/IP address for the syslog server.
<b>Syslog Level</b>	Select the level of logging for syslog.
<b>Send SIP Log</b>	Configure whether the SIP log will be included in the syslog message.

Table 30: Global Policy Parameters – Network Settings

<b>Basic Settings</b>	
<b>IP Address</b>	Configure how the SIP end device shall obtain the IP address. DHCP or PPPoE can be selected. <ul style="list-style-type: none"> <li>• <b>DHCP</b> 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</li> </ul>





	<p>the client and server to exchange vendor class ID information.</p> <ul style="list-style-type: none"> <li>• <b>PPPoE</b> Once selected, users need specify the Account ID, Password and Service Name for PPPoE.</li> </ul>
<b>Host Name</b>	Specifies the name of the client. This field is optional but may be required by Internet Service Providers.
<b>Vendor Class ID</b>	Used by clients and servers to exchange vendor class ID.
<b>Account ID</b>	Enter the PPPoE account ID.
<b>Password</b>	Enter the PPPoE Password.
<b>Service Name</b>	Enter the PPPoE Service Name.
<b>Advanced Setting</b>	
<b>Layer 3 QoS</b>	Define the Layer 3 QoS parameter. This value is used for IP Precedence, Diff-Serv or MPLS. Valid range is 0-63.
<b>Layer 3 QoS For RTP</b>	Assign the priority value of the Layer 3 QoS for RTP packets. Valid range is 0 -63.
<b>Layer 3 QoS For SIP</b>	Assign the priority value of the Layer 3 QoS for SIP packets. Valid range is 0 -63.
<b>Layer 2 QoS Tag</b>	Assign the VLAN Tag of the Layer 2 QoS packets. Valid range is 0 -4095.
<b>Layer 2 QoS Priority Value</b>	Assign the priority value of the Layer 2 QoS packets. Valid range is 0-7.
<b>STUN Server</b>	Configure the IP address or Domain name of the STUN server. Only non-symmetric NAT routers work with STUN.
<b>Keep Alive</b>	Select either to enable or disable Keep Alive.
<b>Keep Alive Interval</b>	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.
<b>Register Expiration</b>	Specify the Register Expiration.
<b>Local SIP Port</b>	Configure Local SIP Port.
<b>Local RTP Port</b>	Configure Local RTP Port.
<b>Auto On-Hook Timer(s)</b>	Configure Auto On-Hook Timer(s).
<b>Ring Timeout</b>	Configure Ring Timeout.
<b>SIP Transport</b>	Select either UDP, TCP or TLS/TCP as SIP transport protocol.
<b>Direct IP Call</b>	Select either to disable or enable Direct IP Call support.
<b>SIP Proxy Compatibility Mode</b>	Select either to disable or enable SIP Proxy Compatibility Mode.



<b>Unregister On Reboot</b>	Select either to disable or enable Unregister On Reboot.
<b>Whitelist</b>	
<b>Whitelist</b>	Select either to enable or disable Whitelist
<b>SIP Phone Number Whitelist</b>	Configure the SIP Phone Number Whitelist.

**Table 31: Global Policy Parameters – Customization**

<b>Wallpaper</b>	
<b>Screen Resolution 1024 x 600</b>	<p>Check this option if the SIP end device shall use 1024 x 600 resolution for the LCD screen wallpaper.</p> <ul style="list-style-type: none"> <li>• <b>Source</b> Configure the location where wallpapers are stored.</li> <li>• <b>File</b> 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 UCM630xA.</li> </ul>
<b>Screen Resolution 800 x 400</b>	<p>Check this option if the SIP end device shall use 800 x 400 resolution for the LCD screen wallpaper.</p> <ul style="list-style-type: none"> <li>• <b>Source</b> Configure the location where wallpapers are stored.</li> <li>• <b>File</b> 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 UCM630xA.</li> </ul>
<b>Screen Resolution 480 x 272</b>	<p>Check this option if the SIP end device shall use 480 x 272 resolution for the LCD screen wallpaper.</p> <ul style="list-style-type: none"> <li>• <b>Source</b> Configure the location where wallpapers are stored.</li> <li>• <b>File</b> 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 UCM630xA.</li> </ul>
<b>Screen Resolution 320 x 240</b>	<p>Check this option if the SIP end device supports 320 x 240 resolution for the LCD screen wallpaper.</p> <ul style="list-style-type: none"> <li>• <b>Source</b> Configure the location where wallpapers are stored.</li> <li>• <b>File</b></li> </ul>



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 UCM630xA.

**Table 32: Global Policy Parameters – Communication Settings**

Email Settings	
<b>SMTP Settings</b>	<p>Check this option to configure the email settings that will be sent to the provisioned phones:</p> <ul style="list-style-type: none"> <li>• <b>Server</b> IP address of the SMTP server</li> <li>• <b>Port</b> SMTP server port</li> <li>• <b>From E-Mail address</b> Email address</li> <li>• <b>Sender Username</b> Username of the sender</li> <li>• <b>Password Recovery Email</b> Email where recovered password will be sent</li> <li>• <b>Alarm receive Email 1</b> Email address where alarms notifications will be sent</li> <li>• <b>Alarm receive Email 1</b> Email address where alarms notifications will be sent</li> <li>• <b>Enable SSL</b> Enable SSL protocol for SMTP</li> </ul>
FTP	
<b>FTP</b>	<p>Check this option to configure the FTP settings that will be sent to the provisioned phones:</p> <ul style="list-style-type: none"> <li>• <b>Storage Server Type</b> Either FTP or Central Storage</li> <li>• <b>Server</b> FTP server address</li> </ul>



- **Port**  
FTP port to be used
- **Username**  
FTP username
- **Path**  
FTP Directory path

## Global Templates

Global Templates can be accessed in Web GUI→**Value-added Features**→**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 templates 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 "Add" to add a global template. Users will see the following configurations.

**Table 33: Create New Template**

<b>Template Name</b>	Create a name to identify this global template.
<b>Description</b>	Provide a description for the global template. This is optional.
<b>Active</b>	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 under the global template.



### Edit Global Templates: test

---

\* Template Name:

Description:

Active:

Options: Phone Settings

▼ Phone Settings

#### Default Call Settings

	Dial Plan:		<input style="width: 100%;" type="text" value="{x+   *x+   *xx*x+}"/>
	Enable Call Features:		<span style="border: 1px solid #ccc; padding: 2px 5px;">Yes</span> ▼
	Use # as Dial Key:		<span style="border: 1px solid #ccc; padding: 2px 5px;">Yes</span> ▼
	Auto Answer by Call-		<span style="border: 1px solid #ccc; padding: 2px 5px;">No</span> ▼

Info:

**Figure 62: 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 delete 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 GUI→Value-added Features→Zero Config→Global Templates page. Users can click on to delete the global template or 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 **Value-added Features→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 templates 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 UCM630xA.

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 “Add” to add a model template.



**Table 34: Create New Model Template**

<b>Model</b>	Select a model to apply this template. The supported Grandstream models are listed in the dropdown list for selection.
<b>Template Name</b>	Create a name for the model template.
<b>Description</b>	Enter a description for the model template. This is optional.
<b>Default Model Template</b>	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.
<b>Active</b>	Check this option to enable the model template.

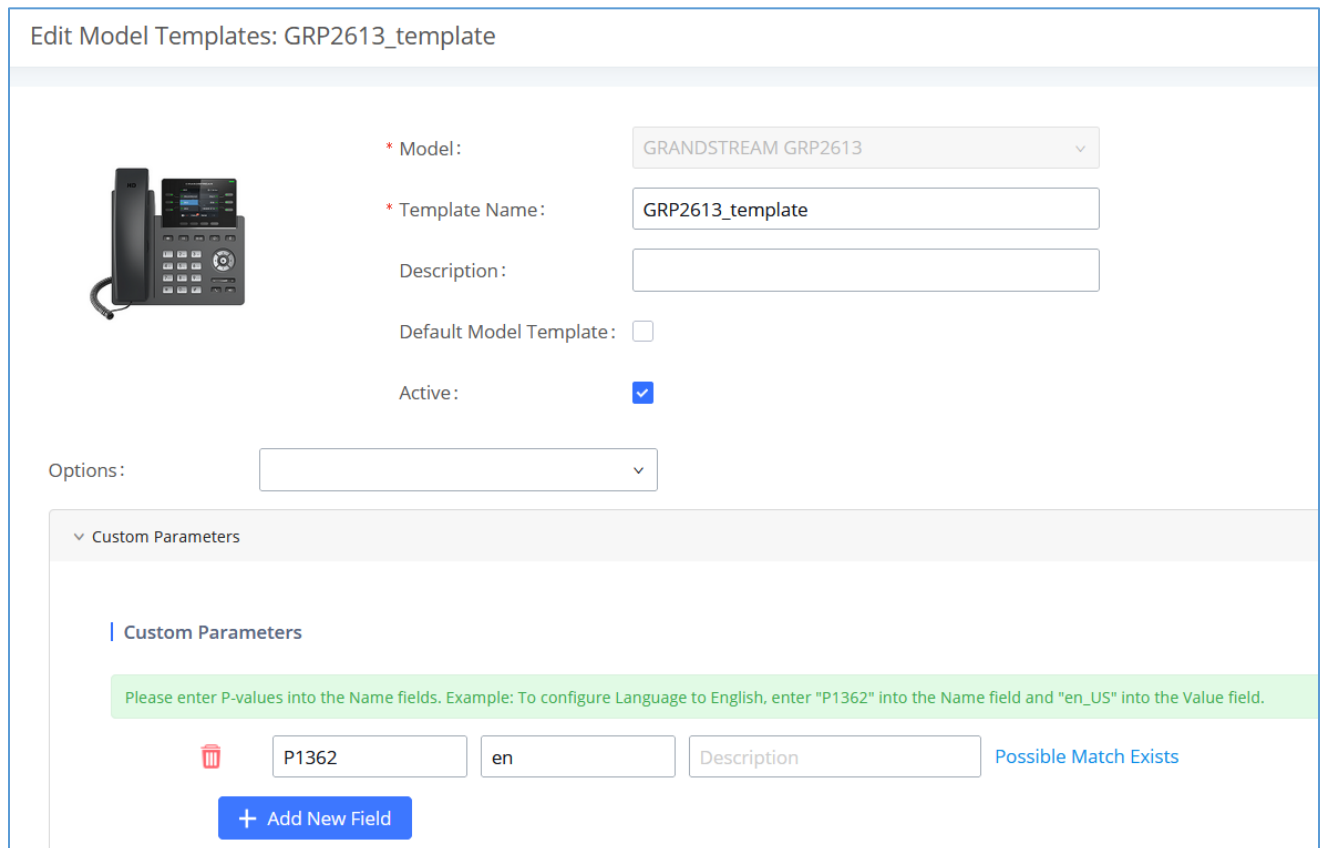
- Click on  to edit the model template.

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 to be modified under 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 <https://www.grandstream.com/support/tools>



**Figure 63: Edit Model Template**

- Click on Save when done. The model template will be displayed on Web GUI → **Value-added Features** → **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

UCM630xA zero config feature supports provisioning all models of Grandstream SIP end devices including OEM device models.

### OEM Models

Users can associate OEM device models with their original Grandstream-branded models, allowing these OEM devices to be provisioned appropriately.

- Click on  button.
- In the *Source Model* field, select the Grandstream device that the OEM model is based on from the dropdown list.
- For the *Destination Model* and *Destination Vendor* field, enter the custom OEM model name and vendor name.
- The newly added OEM model should now be selectable as an option in *Model* fields.

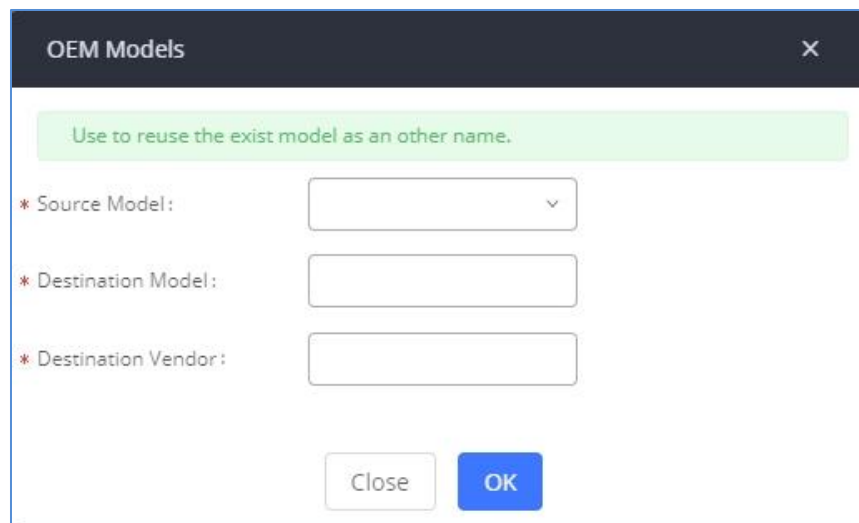




Figure 64: OEM Models

### Model Template Package List

Templates for most of the Grandstream models are built in with the UCM630xA already. Templates for GS Wave and Grandstream surveillance products require users to download and install under Web GUI→**Value-added Features**→**Zero Config**→**Model Update** first before they are available in the UCM630xA for selection. After downloading and installing the model template to the UCM630xA, 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 UCM630xA.

Model Template Package List				
VENDOR	MODEL	VERSION (REMOTE / LOCAL)	SIZE	OPTIONS
Grandstream	DP750	1.0/-	271K	
Grandstream	DP752	1.2/-	58K	
Grandstream	GAC2500	1.0/-	25K	
Grandstream	GDS3705	1.3/-	56K	
Grandstream	GDS3710	1.3/-	97K	
Grandstream	GRP2612	1.0/-	495K	
Grandstream	GRP2612P	1.0/-	495K	
Grandstream	GRP2612W	1.0/-	495K	
Grandstream	GRP2613	1.0/-	67K	
Grandstream	GRP2614	1.3/-	52K	

Total: 37    10 / page

**Figure 65: Template Management**

## Upload Model Template Package

In case the UCM630xA 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 GUI. Please contact Grandstream customer support if the model package is needed for manual uploading.

**Upload Model Template Package**

Choose Model Package to

Upload:

**Figure 66: Upload Model Template Manually**

## Device Configuration

On Web GUI, page **Value-added Features**→**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).

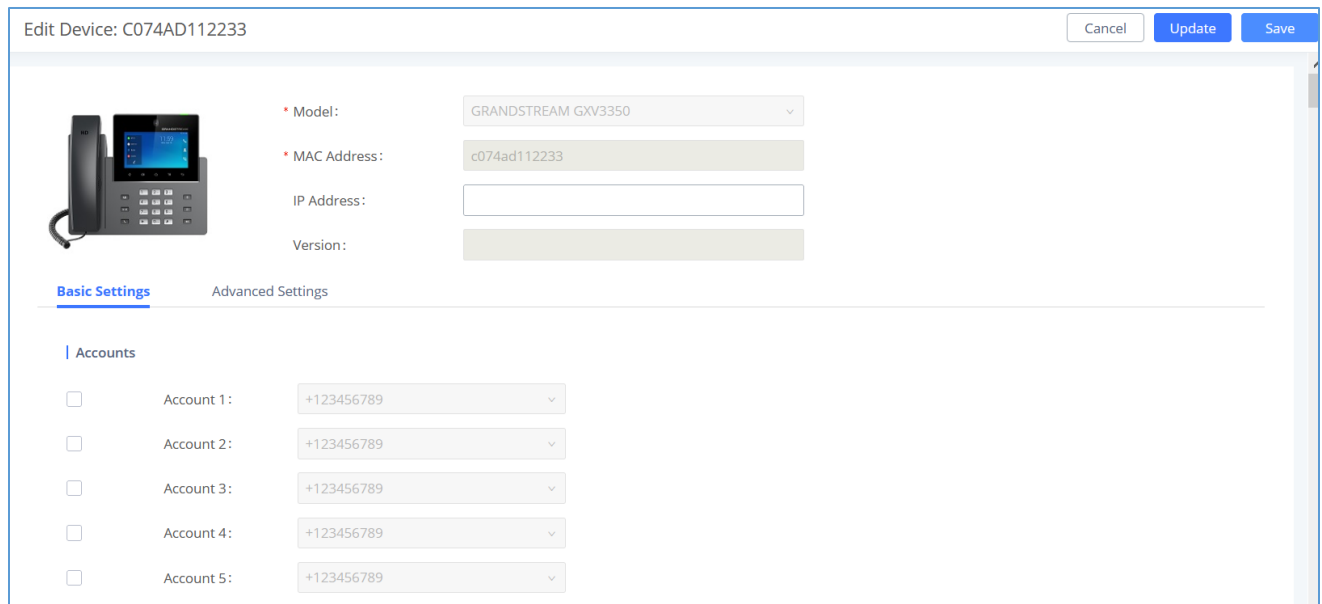


## Create New Device


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 UCM630xA. 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 "Add" 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 "save" to save the configuration for this device.



Edit Device: C074AD112233 Cancel Update Save

 \* Model: GRANDSTREAM GXV3350  
 \* MAC Address: c074ad112233  
 IP Address:   
 Version:

**Basic Settings** | Advanced Settings

**Accounts**


























<input type="checkbox"/>	Account 1:	+123456789	▼
<input type="checkbox"/>	Account 2:	+123456789	▼
<input type="checkbox"/>	Account 3:	+123456789	▼
<input type="checkbox"/>	Account 4:	+123456789	▼
<input type="checkbox"/>	Account 5:	+123456789	▼

Figure 67: Create New Device



## Manage Devices

The device manually created or discovered from Auto Discover will be listed in the Web GUI → **Value-added Features** → **Zero Config** → **Zero Config** page. Users can see the devices with their MAC address, IP address, vendor, model etc.




<input type="checkbox"/>	MAC ADDRESS ↕	IP ADDRESS ↕	EXTENSION	VERSION ↕	VENDOR ↕	MODEL ↕	CREATE CONFIG ↕	OPTIONS
<input type="checkbox"/>	000B82000001	<a href="#">192.168.2.111</a>	1000	unknown	GRANDSTREAM	GXV3275	--	    
<input type="checkbox"/>	000B8227FB15	<a href="#">192.168.2.108</a>	--	1.0.3.208	GRANDSTREAM	GXV3275	--	    
<input type="checkbox"/>	000B82A46ACE	<a href="#">192.168.2.106</a>	--	1.0.0.36	GRANDSTREAM	--	--	    
<input type="checkbox"/>	000B82D33AC4	<a href="#">192.168.2.105</a>	--	20.19.10.30	GRANDSTREAM	--	--	    
<input type="checkbox"/>	000B82F66470	<a href="#">192.168.2.107</a>	--	10.19.9.26	GRANDSTREAM	--	--	    

**Figure 68: Manage Devices**

- Click on  to access the Web GUI 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.

Edit Device: 000B8273C559



\* Model:

\* MAC Address:

IP Address:

Version:

Basic Settings
Advanced Settings

---

**5 Custom Device Settings**

[Modify Custom Settings](#)

**4 Model Templates**

0 item Idle

None

0 Selected

None

**Preview**

**3 Default Model Template**

**Figure 69: 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 GUI→**Value-added Features**→**Zero Config**→**Global Policy** will be applied here. Clicking on “Modify Global Policy” to redirect to page **Value-added Features**→**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 GUI→**Value-added Features**→**Zero Config**→**Model Templates** page. Please see default model template option in [\[Table 34: Create New Model Template\]](#).

(4) Model Templates


Select a model template to be used for the device and click on  to add. Multiple model 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.



Edit Device: 000B82A95F94



Model:	Grandstream GXP2170
MAC Address:	000B82A95F94
IP Address:	192.168.5.115
Version:	1.0.9.132

▼ Custom Fields

Custom Fields

Please use P-values for the Name fields. <br />Example: To configure Language, enter "P1362" into the Language Value field.

✖

P1362

en

Description

Possible Match Exists

+ Add New Field

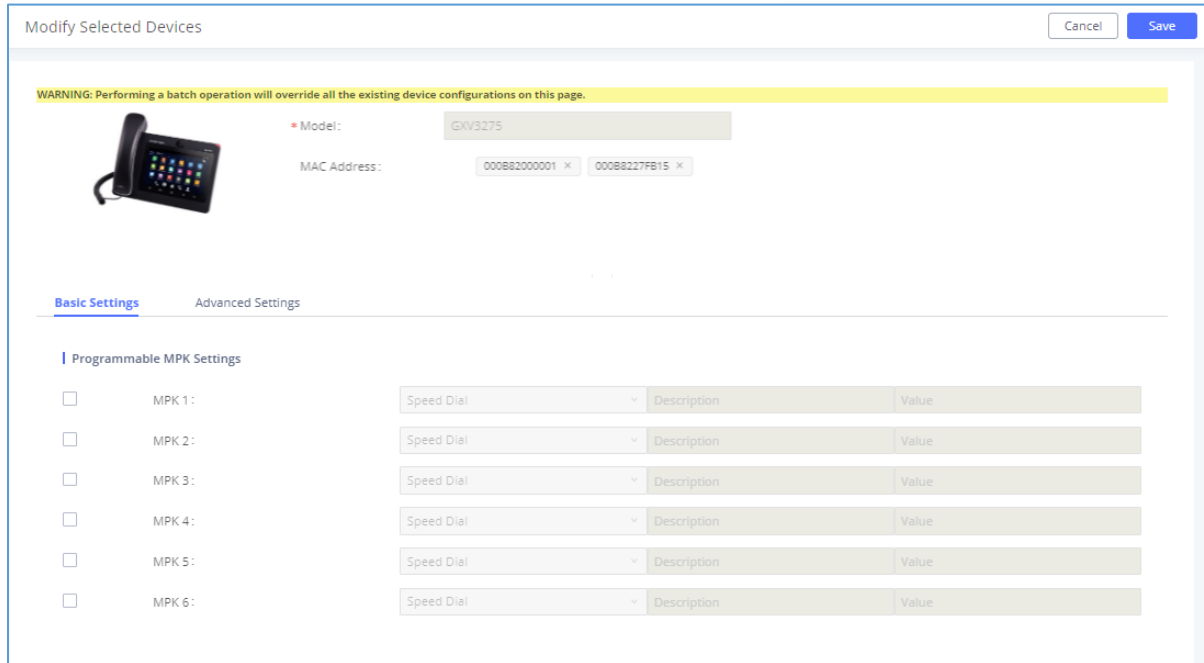
**Figure 70: Edit Customize Device Settings**

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 above 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 [https://content.grandstream.com/hubfs/Grandstream\\_Feb\\_2021/Zip%20File/config-template.zip?hsLang=en](https://content.grandstream.com/hubfs/Grandstream_Feb_2021/Zip%20File/config-template.zip?hsLang=en)


- Select multiple devices that need to be modified and then click on “Update Config” to batch modify devices.

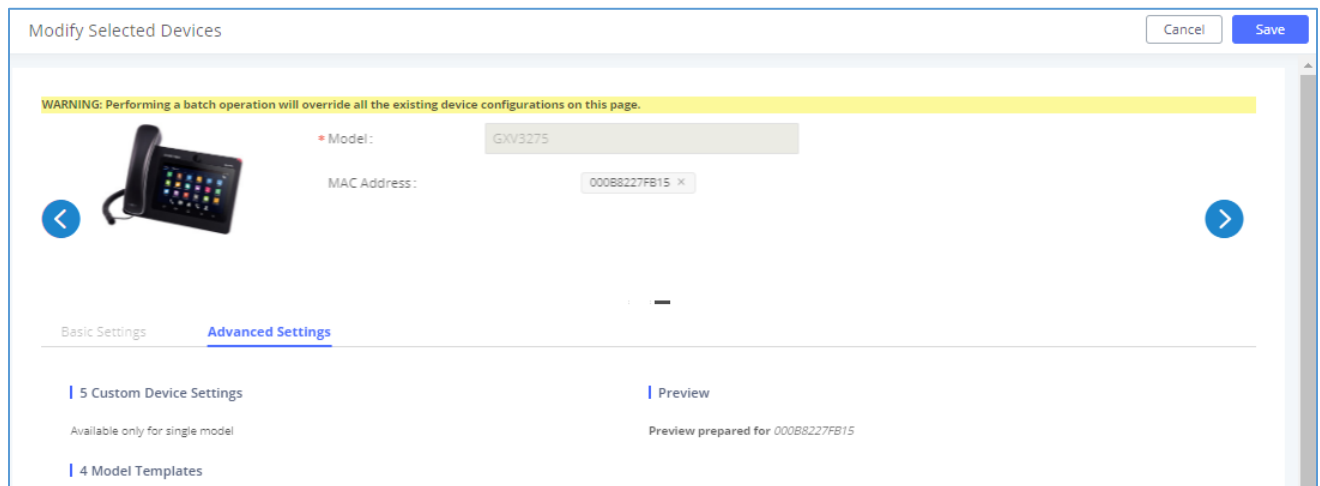
If selected devices are of the same model, the configuration dialog is like the following figure. Configurations in five levels are all available for users to modify.





**Figure 71: 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 72: Modify Selected Devices - Different Models**

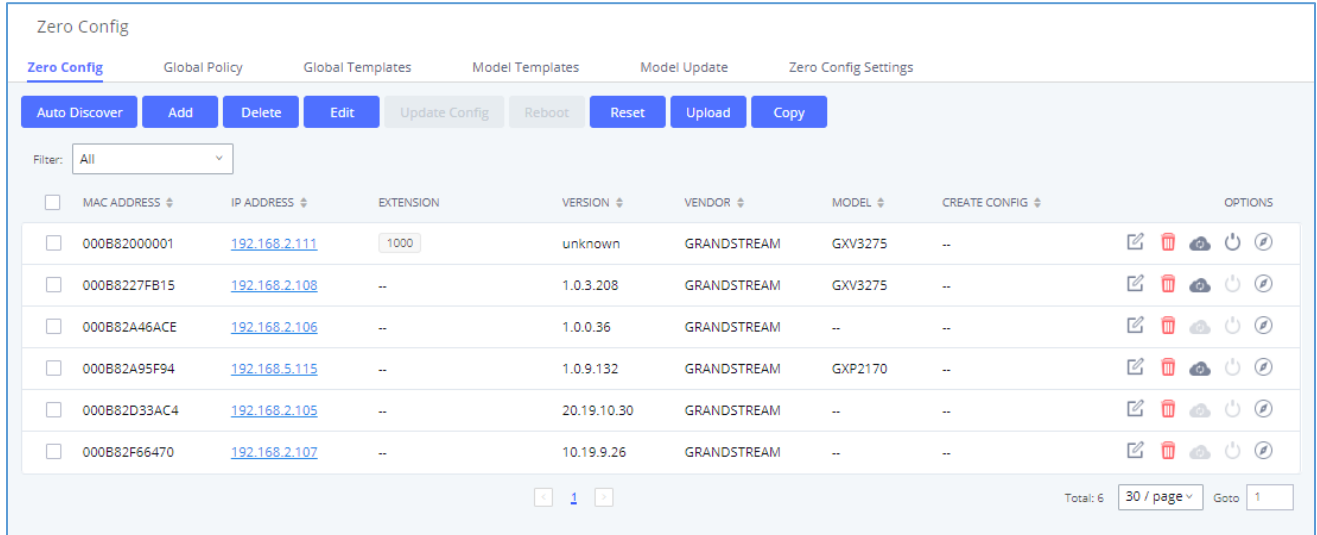


**Note:**

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 GUI→**Value-added Features**→**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.



The screenshot shows the 'Zero Config' web interface. At the top, there are navigation tabs: 'Zero Config', 'Global Policy', 'Global Templates', 'Model Templates', 'Model Update', and 'Zero Config Settings'. Below the tabs are several action buttons: 'Auto Discover', 'Add', 'Delete', 'Edit', 'Update Config', 'Reboot', 'Reset', 'Upload', and 'Copy'. A 'Filter' dropdown is set to 'All'. The main area contains a table with the following columns: 'MAC ADDRESS', 'IP ADDRESS', 'EXTENSION', 'VERSION', 'VENDOR', 'MODEL', 'CREATE CONFIG', and 'OPTIONS'. The table lists six devices, all from the vendor 'GRANDSTREAM'. The 'OPTIONS' column for each row contains icons for edit, delete, notify, power, and refresh.

MAC ADDRESS	IP ADDRESS	EXTENSION	VERSION	VENDOR	MODEL	CREATE CONFIG	OPTIONS
000B82000001	<a href="http://192.168.2.111">192.168.2.111</a>	1000	unknown	GRANDSTREAM	GXV3275	--	
000B8227FB15	<a href="http://192.168.2.108">192.168.2.108</a>	--	1.0.3.208	GRANDSTREAM	GXV3275	--	
000B82A46ACE	<a href="http://192.168.2.106">192.168.2.106</a>	--	1.0.0.36	GRANDSTREAM	--	--	
000B82A95F94	<a href="http://192.168.5.115">192.168.5.115</a>	--	1.0.9.132	GRANDSTREAM	GXP2170	--	
000B82D33AC4	<a href="http://192.168.2.105">192.168.2.105</a>	--	20.19.10.30	GRANDSTREAM	--	--	
000B82F66470	<a href="http://192.168.2.107">192.168.2.107</a>	--	10.19.9.26	GRANDSTREAM	--	--	

At the bottom of the table, there is a pagination control showing 'Total: 6', '30 / page', and 'Goto 1'.

**Figure 73: 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→**Value-added Features**→**Zero Config**→**Zero Config Settings**, select “Enable Zero Config”.
2. Go to Web GUI→**Value-added Features**→**Zero Config**→**Global Policy**, configure Date Format, Time Format and Firmware Source as follows.



v Localization

**| Language Settings**

\* Language: English

**| Date and Time**

Date Format: yyyy-mm-dd

Time Format: 12-hour Clock

Enable NTP: Disabled

NTP Server:

NTP Update Interval: 1440

Time Zone: GMT+08:00 (Beijing, Taipei, Kuala Lu...

Enable Daylight Saving Time: Disabled

> Phone Settings

> Contact List

v Maintenance

**| Upgrade and Provision**

Firmware Source:


Source:	<span style="float: right;">URL</span>
Upgrade via:	<span style="float: right;">TFTP</span>
Server Path:	<span style="float: right;">fm.grandstream.com/gs</span>
File Prefix:	<span style="float: right;"></span>
File Postfix:	<span style="float: right;"></span>

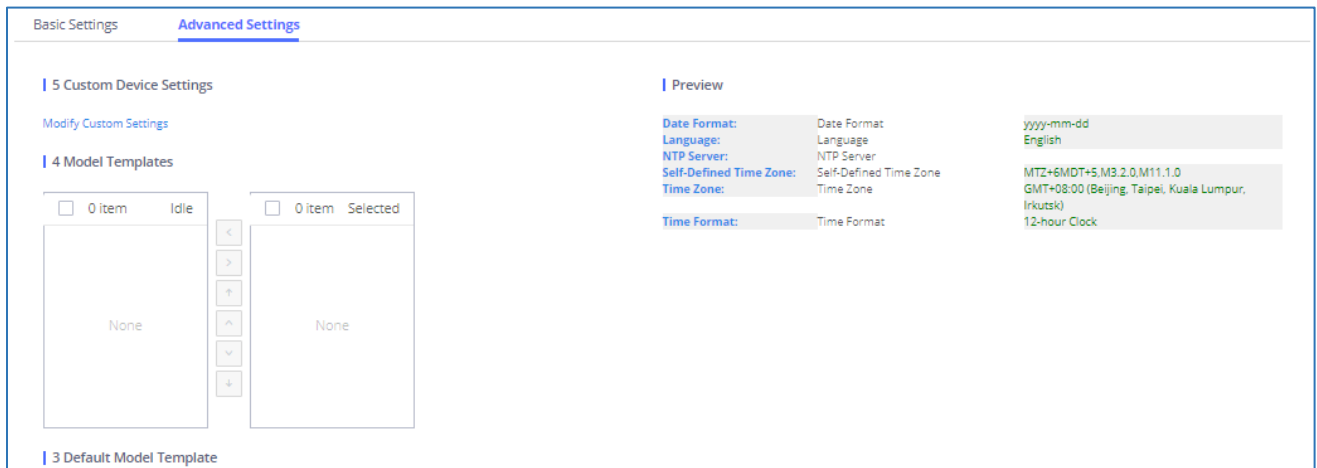
**Figure 74: Zero Config Sample - Global Policy**

3. Go to Web GUI→**Value-added Features**→**Zero Config**→**Model Templates**, create a new model template “English Support Template” for GXP2170. 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→**Value-added Features**→**Zero Config**→**Model Templates**, create another model template “Spanish Support Template” for GXP2170. 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 GUI→**Value-added Features**→**Zero Config**→**Zero Config**.



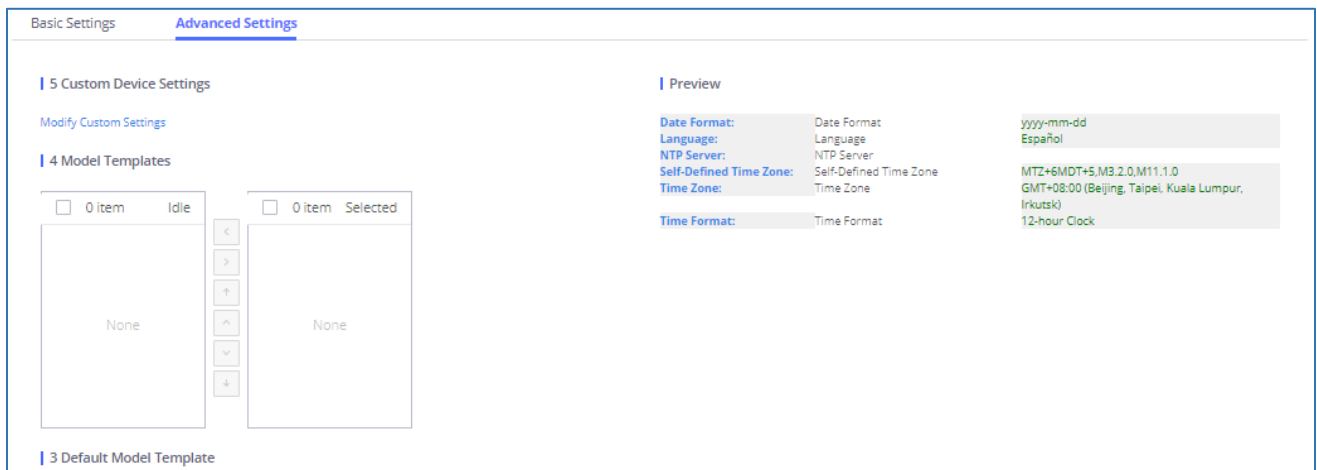


6. On Web GUI→Value-added Features→Zero Config→Zero Config page, users could identify the devices by their MAC addresses or IP addresses displayed on the list. Click on  to edit the device settings.
7. For each of the 5 phones used by English speaking customer support, in “Basic settings” select an available 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 75: 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.

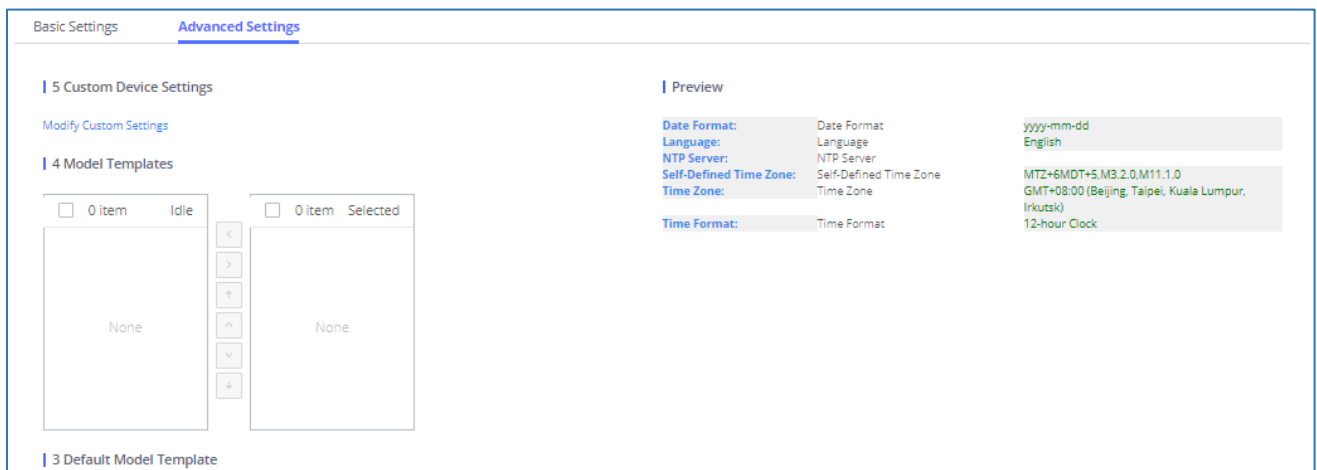


**Figure 76: Zero Config Sample - Device Preview 2**




Select “Spanish Support Template” in “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.

- 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 77: Zero Config Sample - Device Preview 3**

- Click on “Apply Changes” to apply saved changes.
- On the Web GUI → **Value-added Features** → **Zero Config** → **Zero Config** page, click on  to send NOTIFY to trigger the device to download config file from UCM630xA.

Now all the 9 phones in the network will be provisioned with a unique extension registered on the UCM630xA. 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 is not specified in the global policy.

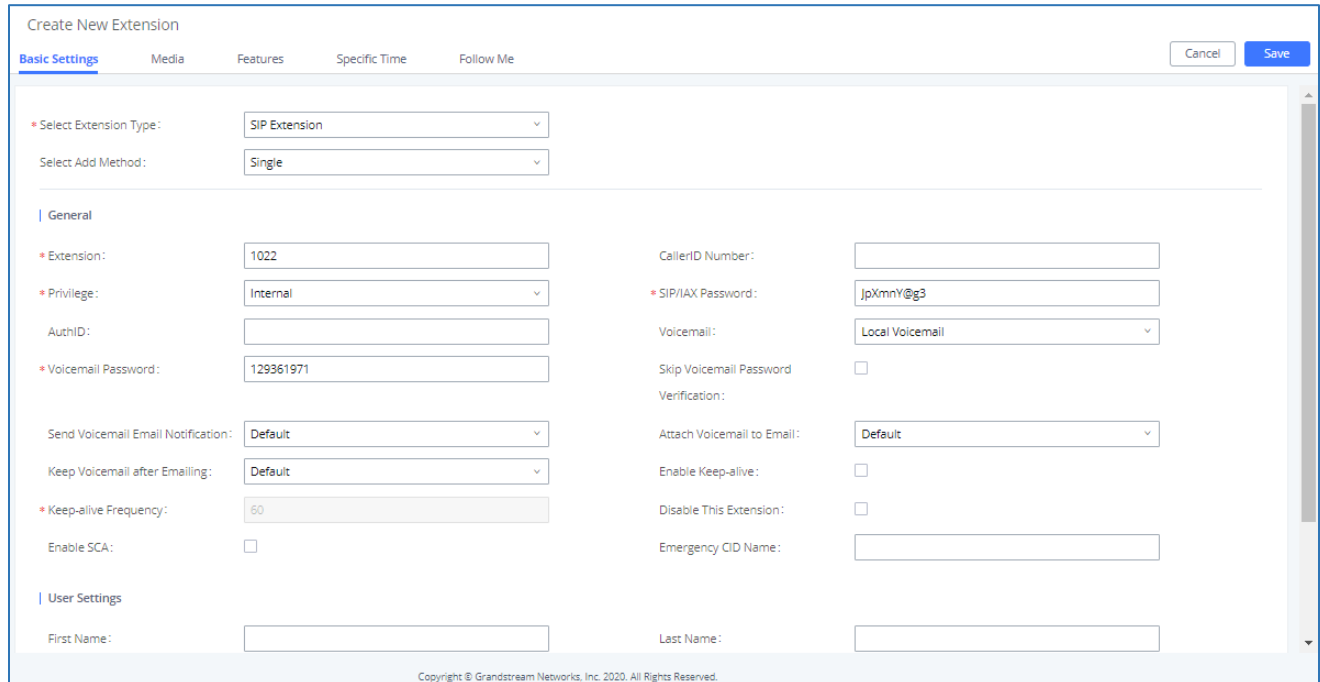


# EXTENSIONS

## Create New User

### Create New SIP Extension

To manually create new SIP user, go to Web GUI→**Extension/Trunk**→**Extensions**. Click on "Add" and a new window will show for users to fill in the extension information.



**Figure 78: Create New Device**

Extension options are divided into four categories:

- Basic Settings
- Media
- Features
- Specific Time
- Follow me

Select first which type of extension: SIP Extension, IAX Extension or FXS Extension. The configuration parameters are as follows.



**Table 35: SIP Extension Configuration Parameters → Basic Settings**

General	
<b>Extension</b>	The extension number associated with the user.
<b>CallerID Number</b>	<p>Configure the CallerID Number that would be applied for outbound calls from this user.</p> <p><b>Note:</b> The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.</p>
<b>Privilege</b>	<p>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".</p> <p><b>Note:</b> Users need to have the same level as or higher level than an outbound rule's privilege to make outbound calls using this rule.</p>
<b>SIP/IAX Password</b>	Configure the password for the user. A random secure password will be automatically generated. It is recommended to use this password for security purpose.
<b>Auth ID</b>	Configure the authentication ID for the user. If not configured, the extension number will be used for authentication.
<b>Voicemail</b>	<p>Configure Voicemail. There are three valid options, and the default option is "Enable Local Voicemail".</p> <ul style="list-style-type: none"> <li>• <b>Disable Voicemail:</b> Disable Voicemail.</li> <li>• <b>Enable Local Voicemail:</b> Enable voicemail for the user.</li> <li>• <b>Enable Remote Voicemail:</b> Forward the notify message from remote voicemail system for the user, and the local voicemail will be disabled. <b>Note:</b> Remote voicemail feature is used only for Infomatec (Brazil).</li> </ul>
<b>Voicemail Password</b>	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.
<b>Skip Voicemail Password Verification</b>	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.
<b>Send Voicemail Email Notification</b>	Configures whether or not to send emails to the extension's email address to notify of new voicemail.
<b>Attach Voicemail to Email</b>	<p>Configures whether or not to attach voicemail audio file to the voicemail notification emails.</p> <p><b>Note:</b> When set to "Default", the global settings in <b>Call Features → Voicemail → Voicemail Email Settings</b> will be used.</p>



<b>Keep Voicemail after Emailing</b>	<p>Whether to keep the local voicemail recording after sending them. If set to "Default", the global settings will be used.</p> <p><b>Note:</b> When set to "Default", the global settings in <b>Call Features</b> → <b>Voicemail</b> → <b>Voicemail Email Settings</b> will be used.</p>
<b>Enable Keep-alive</b>	<p>If enabled, empty SDP packet will be sent to the SIP server periodically to keep the NAT port open. The default setting is "No".</p>
<b>Keep-alive Frequency</b>	<p>Configure the Keep-alive interval (in seconds) to check if the host is up. The default setting is 60 seconds.</p>
<b>Enable SCA</b>	<p>If enabled, (1) Call Forward, Call Waiting and Do Not Disturb settings will not work, (2) Concurrent Registrations can be set only to 1, and (3) Private numbers can be added in Call Features-&gt;SCA page.</p>
<b>Emergency CID Name</b>	<p>CallerID name that will be used for emergency calls and callbacks.</p>
<b>Disable This Extension</b>	<p>If selected, this extension will be disabled on the UCM630xA.</p> <p><b>Note:</b> The disabled extension still exists on the PBX but cannot be used on the end device.</p>
<b>User Settings</b>	
<b>First Name</b>	<p>Configure the first name of the user.</p> <p>The first name can contain characters, letters, digits and _.</p>
<b>Last Name</b>	<p>Configure the last name of the user.</p> <p>The last name can contain characters, letters, digits and _.</p>
<b>Email Address</b>	<p>Fill in the Email address for the user. Voicemail will be sent to this Email address.</p>
<b>User Password</b>	<p>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.</p>
<b>Language</b>	<p>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→<b>PBX Settings</b>→<b>Voice Prompt</b>→<b>Language Settings</b>. The dropdown list shows all the current available voice prompt languages on the UCM630xA. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under Web GUI→<b>PBX Settings</b>→<b>Voice Prompt</b>→<b>Language Settings</b>.</p>
<b>Concurrent Registrations</b>	<p>The maximum endpoints which can be registered into this extension. For security concerns, the default value is 1.</p>
<b>Mobile Phone Number</b>	<p>Configure the phone number for the extension, user can type the related star code for phone number followed by the extension number to directly call this number.</p> <p>Example: user can type *881000 to call the mobile number associated with extension 1000.</p>



**Table 36: SIP Extension Configuration Parameters→Media**

SIP Settings	
<b>NAT</b>	Use NAT when the UCM630xA is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is one-way audio issue, usually it is related to NAT configuration or Firewall's support of SIP and RTP ports. The default setting is enabled.
<b>Enable Direct Media</b>	By default, the UCM630xA will route the media streams 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 UCM630xA to negotiate endpoint-to-endpoint media routing. The default setting is "No".
<b>DTMF Mode</b>	Select DTMF mode for the user to send DTMF. The default setting is "RFC4733". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, a-law or u-law are required. When "Auto" is selected, RFC4733 will be used if offered, otherwise "Inband" will be used.
<b>TEL URI</b>	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.
<b>Alert-Info</b>	When present in an INVITE request, the alert-Info header field specifies and alternative ring tone to the UAS.
<b>Enable T.38 UDPTL</b>	Enable or disable T.38 UDPTL support.
<b>SRTP</b>	Enable SRTP for the call. The default setting is disabled.
<b>Jitter Buffer</b>	Select jitter buffer method. <ul style="list-style-type: none"> <li>• <b>Disable:</b> Jitter buffer will not be used.</li> <li>• <b>Fixed:</b> Jitter buffer with a fixed size (equal to the value of "jitter buffer size")</li> <li>• <b>Adaptive:</b> Jitter buffer with an adaptive size (no more than the value of "max jitter buffer").</li> <li>• <b>NetEQ:</b> Dynamic jitter buffer via NetEQ.</li> </ul>
<b>Packet Loss Retransmission</b>	Configure to enable Packet Loss Retransmission. <ul style="list-style-type: none"> <li>• <b>NACK</b></li> <li>• <b>NACK+RTX(SSRC-GROUP)</b></li> <li>• <b>OFF</b></li> </ul>
<b>Video FEC</b>	Check to enable Forward Error Correction (FEC) for Video.
<b>Audio FEC</b>	Check to enable Forward Error Correction (FEC) for Audio.
<b>FECC</b>	Configure to enable Remote Camera Management.



<b>ACL Policy</b>	Access Control List manages the IP addresses that can register to this extension. <ul style="list-style-type: none"> <li>• <b>Allow All:</b> Any IP address can register to this extension.</li> <li>• <b>Local Network:</b> Only IP addresses in the configured network segments can register to this extension. Press “Add Local Network Address” to add more IP segments.</li> </ul>
<b>Codec Preference</b>	Select 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.265, H.263, H.263p, RTX and VP8.

Table 37: SIP Extension Configuration Parameters→Features

<b>Call Transfer</b>	
<b>Presence Status</b>	Select which presence status to set for the extension and configure call forward conditions for each status. Six possible options are possible: “ <b>Available</b> ”, “ <b>Away</b> ”, “ <b>Chat</b> ”, “ <b>Custom</b> ”, “ <b>DND</b> ” and “ <b>Unavailable</b> ”. More details at [PRESENCE].
<b>Call Forward Unconditional</b>	Enable and configure the Call Forward Unconditional target number. Available options for target number are: <ul style="list-style-type: none"> <li>• “<b>None</b>”: Call forward deactivated.</li> <li>• “<b>Extension</b>”: Select an extension from dropdown list as CFU target.</li> <li>• “<b>Custom Number</b>”: Enter a customer number as target. For example: *97.</li> <li>• “<b>Voicemail</b>”: Select an extension from dropdown list. Incoming calls will be forwarded to voicemail of selected extension.</li> <li>• “<b>Ring Group</b>”: Select a ring group from dropdown list as CFU target.</li> <li>• “<b>Queues</b>”: Select a queue from dropdown list as CFU target.</li> <li>• “<b>Voicemail Group</b>”: Select a voicemail group from dropdown list as CFU target.</li> </ul> The default setting is “None”.
<b>CFU Time Condition</b>	Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”. <p>Note:</p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured under Specific Time section. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</li> </ul>



<b>Call Forward No Answer</b>	<p>Configure the Call Forward No Answer target number. Available options for target number are:</p> <ul style="list-style-type: none"> <li>• <b>“None”</b>: Call forward deactivated.</li> <li>• <b>“Extension”</b>: Select an extension from dropdown list as CFN target.</li> <li>• <b>“Custom Number”</b>: Enter a customer number as target. For example: *97.</li> <li>• <b>“Voicemail”</b>: Select an extension from dropdown list. Incoming calls will be forwarded to voicemail of selected extension.</li> <li>• <b>“Ring Group”</b>: Select a ring group from dropdown list as CFN target.</li> <li>• <b>“Queues”</b>: Select a queue from dropdown list as CFN target.</li> <li>• <b>“Voicemail Group”</b>: Select a voicemail group from dropdown list as CFN target.</li> </ul> <p>The default setting is “None”.</p>
<b>CFN Time Condition</b>	<p>Select time condition for Call Forward No Answer. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured under Specific Time section. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</li> </ul>
<b>Call Forward Busy</b>	<p>Configure the Call Forward Busy target number. Available options for target number are:</p> <ul style="list-style-type: none"> <li>• <b>“None”</b>: Call forward deactivated.</li> <li>• <b>“Extension”</b>: Select an extension from dropdown list as CFB target.</li> <li>• <b>“Custom Number”</b>: Enter a customer number as target. For example: *97.</li> <li>• <b>“Voicemail”</b>: Select an extension from dropdown list. Incoming calls will be forwarded to voicemail of selected extension.</li> <li>• <b>“Ring Group”</b>: Select a ring group from dropdown list as CFB target.</li> <li>• <b>“Queues”</b>: Select a queue from dropdown list as CFB target.</li> <li>• <b>“Voicemail Group”</b>: Select a voicemail group from dropdown list as CFB target.</li> </ul> <p>The default setting is “None”.</p>





<b>CFB Time Condition</b>	<p>Select time condition for Call Forward Busy. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured under Specific Time section. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</li> </ul>
<b>Do Not Disturb</b>	<p>If Do Not Disturb is enabled, all incoming calls will be dropped. All call forward settings will be ignored.</p>
<b>DND Time Condition</b>	<p>Select time condition for Do Not Disturb. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured under Specific Time section. Scroll down the add Time Condition for specific time.</li> </ul> <p>Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</p>
<b>DND Whitelist</b>	<p>If DND is enabled, calls from the whitelisted numbers will not be rejected. Multiple numbers are supported and must be separated by new lines. Pattern matching is supported.</p> <ul style="list-style-type: none"> <li>• <b>Z</b> match any digit from 1-9.</li> <li>• <b>N</b> match any digit from 2-9.</li> <li>• <b>X</b> match any digit from 0-9.</li> </ul>
<b>FWD Whitelist</b>	<p>Calls from users in the forward whitelist will not be forwarded. Pattern matching is supported.</p> <ul style="list-style-type: none"> <li>• <b>Z</b> match any digit from 1-9,</li> <li>• <b>N</b> match any digit from 2-9,</li> <li>• <b>X</b> match any digit from 0-9.</li> </ul>



## CC Settings

<b>Enable CC</b>	If enabled, UCM630xA 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 is disabled.
<b>CC Mode</b>	Two modes for Call Completion are supported: <ul style="list-style-type: none"> <li>• <b>Normal:</b> This extension is used as ordinary extension.</li> <li>• <b>For Trunk:</b> This extension is registered from a PBX.</li> </ul> The default setting is “Normal”.
<b>CC Max Agents</b>	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 can make. The minimum value is 1.
<b>CC Max Monitors</b>	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

<b>Ring Simultaneously</b>	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.
<b>External Number</b>	Set the external number to be rang simultaneously. '-' is the connection character which will be ignored. This field accepts only letters, numbers, and special characters + = * #.
<b>Time Condition for Ring Simultaneously</b>	Ring the external number simultaneously along with the extension on the basis of this time condition.
<b>Use callee DOD on FWD or RS</b>	Use the DOD number when calls are being diverted/forwarded to external destinations or when ring simultaneous is configured.

## Monitor privilege control

<b>Allowed to call-barging</b>	Add members from “Available Extensions” to “Selected Extensions” so that the selected extensions can spy on the used extension using feature code.
--------------------------------	--



### Seamless transfer privilege control

<b>Allowed to seamless transfer</b>	Any extensions on the UCM can perform seamless transfer. When using Pickup Incall feature, only extensions available on the “Selected Extensions” list can perform seamless transfer to the edited extension.
-------------------------------------	---

### PMS Remote Wakeup Whitelist

<b>Select the extensions that can set wakeup service for other extensions</b>	Selected extensions can set a PMS wakeup service for this extension via feature code.
---	---

### Other Settings

<b>Ring Timeout</b>	<p>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 UCM630xA. The valid range is between 5 seconds and 600 seconds.</p> <p><b>Note:</b> If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
<b>Auto Record</b>	Enable automatic recording for the calls using this extension. The default setting is disabled. The recordings can be accessed under Web GUI→ <b>CDR</b> → <b>Recording Files</b> .
<b>Skip Trunk Auth</b>	<ul style="list-style-type: none"> <li>• If set to “yes”, users can skip entering the password when making outbound calls.</li> <li>• If set to “By Time”, users can skip entering the password when making outbound calls during the selected time condition.</li> <li>• If set to “No”, users will be asked to enter the password when making outbound calls.</li> </ul>
<b>Time Condition for Skip Trunk Auth</b>	If ‘Skip Trunk Auth’ is set to ‘By Time’, select a time condition during which users can skip entering password when making outbound calls.
<b>Dial Trunk Password</b>	Configure personal password when making outbound calls via trunk.
<b>Support Hot-Desking Mode</b>	Check to enable Hot-Desking Mode on the extension. Hot-Desking allows to use the same endpoint device and login using extension/password combination. This feature is used in scenarios where different users need to use the same endpoint device during different time of a day for instance. If enabled, SIP Password will accept only alphabet characters and digits. Auth ID will be changed to the same as Extension.



<b>Enable LDAP</b>	If enabled, the extension will be added to LDAP Phonebook PBX list. Default is enabled.
<b>Use MOH as IVR ringback tone</b>	If enabled, when the call to the extension is made through the IVR, the caller will hear MOH as ringback tone instead of the regular ringback tone.
<b>Music On Hold</b>	Specify which Music On Hold class to suggest to the bridged channel when putting them on hold.
<b>Enable WebRTC Support</b>	Enable registration and call from WebRTC. For more details about WebRTC feature refer to [ <b>Wave WebRTC Video Calling</b> ]
<b>Bind PMS Room</b>	If enabled, the system will create a room whose room number, by default, will equal the extension number in PMS module. Note: If this room already exists, the configuration of the existing room will be overwritten.
<b>Call Duration Limit</b>	Check to enable and set the call limit the duration.
<b>Maximum Call Duration (s)</b>	The maximum call duration (in seconds). The default value 0 means no limit. Max value is 86400 seconds
<b>The Maximum Number of Call Lines</b>	The maximum number of simultaneous calls that the extension can have. 0 indicates no limit.
<b>Enable Auto-Answer Support</b>	If enabled, the extension will support auto-answer when indicated by Call-info/Alert-info headers.
<b>Email Missed Calls</b>	Send a log of missed calls to the extension's email address.
<b>Missed Call Type</b>	If <b>Email Missed Calls</b> enabled, users can select the type of missed calls to be sent via email, the available types are: <ul style="list-style-type: none"> <li>• <b>Default:</b> All missed calls will be sent in email notifications.</li> <li>• <b>Missed Internal Call:</b> Only missed local extension-to-extension calls will be sent in email notifications.</li> <li>• <b>Missed External Call:</b> Only missed calls from trunks will be sent in email notifications.</li> </ul>
<b>Call Waiting</b>	Allows calls to the extension even when it is already in a call. This only works if the caller is directly dialing the extension. If disabled, the CC service will take effect only for unanswered and timeout calls.



**Table 38: SIP Extension Configuration Parameters→Specific Time**

Specific Time	
<b>Time Condition</b>	Click to add Time Condition to configure specific time for this extension.

**Table 39: Table 34: SIP Extension Configuration Parameters→Follow Me**

Follow Me	
<b>Enable</b>	Configure to enable or disable Follow Me for this user.
<b>Skip Trunk Auth</b>	If the outbound calls need to check the password, we should enable this option or enable the option "Skip Trunk Auth" of the Extension. Otherwise this Follow Me cannot call out.
<b>Music On Hold Class</b>	Configure the Music On Hold class that the caller would hear while tracking the user.
<b>Enable Destination</b>	Configure to enable destination.
<b>Default Destination</b>	The call will be routed to this destination if no one in the Follow Me answers the call.
<b>Confirm When Answering</b>	If enabled, call will need to be confirmed after answering.
<b>Use Callee DOD for Follow Me</b>	Use the callee DOD number as CID if configured Follow Me numbers are external numbers.
<b>New Follow Me Number</b>	Add a new Follow Me number which could be a "Local Extension" or an "External Number". The selected dial plan should have permissions to dial the defined external number.
<b>Dialing Order</b>	This is the order in which the Follow Me destinations will be dialed to reach the user.

### Create New IAX Extension

The UCM630xA supports Inter-Asterisk eXchange (IAX) protocol. IAX is used for transporting VoIP telephony sessions between servers and terminal devices. IAX is like 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→**Extension/Trunk**→**Extensions**. Click on "Add" and a new dialog window will show for users which need to make sure first to select the extension type to be IAX Extension before proceeding to fill in the extension information. The configuration parameters are as follows.



**Table 40: IAX Extension Configuration Parameters→Basic Settings**

General	
<b>Extension</b>	The extension number associated with the user.
<b>CallerID Number</b>	Configure the CallerID Number that would be applied for outbound calls from this user. <b>Note:</b> The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
<b>Privilege</b>	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". <b>Note:</b> Users need to have the same level as or higher level than an outbound rule's privilege to make outbound calls using this rule.
<b>SIP/IAX Password</b>	Configure the password for the user. A random secure password will be automatically generated. It is recommended to use this password for security purpose.
<b>Voicemail</b>	Configure Voicemail. There are three valid options and the default option is "Enable Local Voicemail". <ul style="list-style-type: none"> <li>• <b>Disable Voicemail:</b> Disable Voicemail.</li> <li>• <b>Enable Local Voicemail:</b> Enable voicemail for the user.</li> </ul>
<b>Voicemail Password</b>	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.
<b>Skip Voicemail Password Verification</b>	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.
<b>Send Voicemail Email Notification</b>	Configures whether or not to send emails to the extension's email address to notify of new voicemail.
<b>Attach Voicemail to Email</b>	Configures whether or not to attach voicemail audio file to the voicemail notification emails.
<b>Keep Voicemail after Emailing</b>	Only applies if extension-level or global Send Voicemail to Email is enabled.
<b>Disable This Extension</b>	If selected, this extension will be disabled on the UCM630xA. <b>Note:</b> The disabled extension still exists on the PBX but cannot be used on the end device.



User Settings	
<b>First Name</b>	Configure the first name of the user. The first name can contain characters, letters, digits and _.
<b>Last Name</b>	Configure the last name of the user. The last name can contain characters, letters, digits and _.
<b>Email Address</b>	Fill in the Email address for the user. Voicemail will be sent to this Email address.
<b>User Password</b>	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.
<b>Language</b>	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→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> . The dropdown list shows all the current available voice prompt languages on the UCM630xA. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under Web GUI→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> .
<b>Mobile Phone Number</b>	Configure the Mobile number of the user.

Table 41: IAX Extension Configuration Parameters→Media

IAX Settings	
<b>Max Number of Calls</b>	Configure the maximum number of calls allowed for each remote IP address.
<b>Require Call Token</b>	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".
<b>SRTP</b>	Enable SRTP for the call. The default setting is disabled.
<b>ACL Policy</b>	Access Control List manages the IP addresses that can register to this extension. <ul style="list-style-type: none"> <li>• <b>Allow All:</b> Any IP address can register to this extension.</li> <li>• <b>Local Network:</b> Only IP addresses in the configured network segments can register to this extension.</li> </ul>
<b>Codec Preference</b>	Select 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.265, H.263, H.263p, RTX and VP8.



**Table 42: IAX Extension Configuration Parameters→Features**

Call Transfer	
<b>Call Forward Unconditional</b>	Configure the Call Forward Unconditional target number. If not configured, the Call Forward Unconditional feature is deactivated. The default setting is deactivated.
<b>CFU Time Condition</b>	<p>Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p>Note:</p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday</b> page.</li> </ul>
<b>Call Forward No Answer</b>	Configure the Call Forward No Answer target number. If not configured, the Call Forward No Answer feature is deactivated. The default setting is deactivated.
<b>CFN Time Condition</b>	<p>Select time condition for Call Forward No Answer. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday</b> page.</li> </ul>
<b>Call Forward Busy</b>	Configure the Call Forward Busy target number. If not configured, the Call Forward Busy feature is deactivated. The default setting is deactivated.
<b>CFB Time Condition</b>	<p>Select time condition for Call Forward Busy. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time</li> </ul>





	<p>period.</p> <ul style="list-style-type: none"> <li>Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> </ul> <p>Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday page</b>.</p>
<b>Do Not Disturb</b>	If Do Not Disturb is enabled, all incoming calls will be dropped. All call forward settings will be ignored.
<b>DND Time Condition</b>	The time condition of DND. The DND will take effect while the time condition is satisfied.
<b>DND Whitelist</b>	<p>If DND is enabled, calls from the whitelisted numbers will not be rejected. Multiple numbers are supported and must be separated by new lines. Pattern matching is supported.</p> <ul style="list-style-type: none"> <li><b>Z</b> match any digit from 1-9.</li> <li><b>N</b> match any digit from 2-9.</li> <li><b>X</b> match any digit from 0-9.</li> </ul>
<b>FWD Whitelist</b>	<p>Calls from users in the forward whitelist will not be forwarded. Pattern matching is supported.</p> <ul style="list-style-type: none"> <li><b>Z</b> match any digit from 1-9.</li> <li><b>N</b> match any digit from 2-9.</li> <li><b>X</b> match any digit from 0-9.</li> </ul>
<b>Ring Simultaneously</b>	
<b>Ring Simultaneously</b>	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.
<b>External Number</b>	Set the external number to be rang simultaneously. '-' is the connection character which will be ignored.
<b>Time Condition for Ring Simultaneously</b>	Ring the external number simultaneously along with the extension on the basis of this time condition.
<b>Use callee DOD on FWD or RS</b>	Use the callee's DOD number as CallerID on Outgoing Forwarding or Ring Simultaneously calls.
<b>Monitor privilege control</b>	
<b>Allow call-barging</b>	Members of the list can spy on this extension via feature codes.



## Seamless transfer privilege control

### Allowed to seamless transfer

Members of the list can seamlessly transfer via feature code.

## Other Settings

<b>Ring Timeout</b>	<p>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 UCM630xA, which can be configured in the global ring timeout setting under Web GUI→PBX Settings→Voice Prompt→Custom Prompt: General Preference. The valid range is between 5 seconds and 600 seconds.</p> <p><b>Note:</b> If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
<b>Auto Record</b>	<p>Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under Web GUI→<b>CDR</b>→<b>Recording Files</b>.</p>
<b>Skip Trunk Auth</b>	<ul style="list-style-type: none"> <li>• If set to “Yes”, users can skip entering the password when making outbound calls.</li> <li>• If set to “By Time”, users can skip entering the password when making outbound calls during the selected time condition.</li> <li>• If set to “No”, users will be asked to enter the password when making outbound calls.</li> </ul>
<b>Time Condition for Skip Trunk Auth</b>	<p>If “Skip Trunk Auth” is set to “By Time”, select a time condition during which users can skip entering password when making outbound calls.</p>
<b>Dial Trunk Password</b>	<p>Configure personal password when making outbound calls via trunk.</p>
<b>Enable LDAP</b>	<p>If enabled, the extension will be added to LDAP Phonebook PBX lists.</p>
<b>Music On Hold</b>	<p>Configure the Music On Hold class to suggest to the bridged channel when putting them on hold.</p>
<b>Use MOH as IVR ringback tone</b>	<p>If enabled, when the call to the extension is made through the IVR, the caller will hear MOH as ringback tone instead of the regular ringback tone.</p>
<b>Call Duration Limit</b>	<p>Check to enable and set the call limit the duration.</p>
<b>Maximum Call Duration (s)</b>	<p>The maximum call duration (in seconds). The default value 0 means no limit. Max value is 86400 seconds</p>



<b>Email Missed Calls</b>	Send a log of missed calls to the extension's email address.
<b>Missed Call Type</b>	<p>If <b>Email Missed Calls</b> enabled, users can select the type of missed calls to be sent via email, the available types are:</p> <ul style="list-style-type: none"> <li>• <b>Default:</b> All missed calls will be sent in email notifications.</li> <li>• <b>Missed Internal Call:</b> Only missed local extension-to-extension calls will be sent in email notifications.</li> <li>• <b>Missed External Call:</b> Only missed calls from trunks will be sent in email notifications.</li> </ul>

**Table 43: IAX Extension Configuration Parameters→Specific Time**

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

**Table 44: IAX Extension Configuration Parameters→Follow Me**

Follow Me	
<b>Enable</b>	Configure to enable or disable Follow Me for this user.
<b>Skip Trunk Auth</b>	If the outbound calls need to check the password, we should enable this option or enable the option "Skip Trunk Auth" of the Extension. Otherwise this Follow Me cannot call out.
<b>Music On Hold Class</b>	Configure the Music On Hold class that the caller would hear while tracking the user.
<b>Enable Destination</b>	Configure to enable destination.
<b>Default Destination</b>	The call will be routed to this destination if no one in the Follow Me answers the call.
<b>Confirm When Answering</b>	If enabled, call will need to be confirmed after answering.
<b>Use Callee DOD for Follow Me</b>	Use the callee DOD number as CID if configured Follow Me numbers are external numbers.
<b>New Follow Me Number</b>	Add a new Follow Me number which could be a "Local Extension" or an "External Number". The selected dial plan should have permissions to dial the defined external number.
<b>Dialing Order</b>	This is the order in which the Follow Me destinations will be dialed to reach the user.



## Create New FXS Extension

The UCM630xA supports Foreign eXchange Subscriber (FXS) interface. FXS is used when user needs to connect analog phone lines or FAX machines to the UCM630xA.

To manually create new FXS user, go to Web GUI→**Extension/Trunk**→**Extensions**. Click on "Add" and a new dialog window will show for users which need to make sure first to select the extension type to be FXS Extension before proceeding to fill in the extension information. The configuration parameters are as follows.

**Table 45: FXS Extension Configuration Parameters→Basic Settings**

General	
<b>Extension</b>	The extension number associated with the user.
<b>Analog Station</b>	Select the FXS port to be assigned for this extension.
<b>Caller ID Number</b>	Configure the CallerID Number that would be applied for outbound calls from this user. <b>Note:</b> The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
<b>Privilege</b>	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". <b>Note:</b> Users need to have the same level as or higher level than an outbound rule's privilege to make outbound calls using this rule.
<b>Voicemail</b>	Configure Voicemail. There are three valid options and the default option is "Enable Local Voicemail". <ul style="list-style-type: none"> <li>• <b>Disable Voicemail:</b> Disable Voicemail.</li> <li>• <b>Enable Local Voicemail:</b> Enable voicemail for the user.</li> </ul>
<b>Voicemail Password</b>	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.
<b>Skip Voicemail Password Verification</b>	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.
<b>Send Voicemail Email Notification</b>	Configures whether or not to send emails to the extension's email address to notify of new voicemail.
<b>Attach Voicemail to Email</b>	Configures whether or not to attach voicemail audio file to the voicemail notification emails.



<b>Keep Voicemail after Emailing</b>	Only applies if extension-level or global Send Voicemail to Email is enabled.
<b>Emergency CID Name</b>	CallerID name that will be used for emergency calls and callbacks.
<b>Disable This Extension</b>	If selected, this extension will be disabled on the UCM630xA. <b>Note:</b> The disabled extension still exists on the PBX but cannot be used on the end device.
<b>User Settings</b>	
<b>First Name</b>	Configure the first name of the user. The first name can contain characters, letters, digits and _.
<b>Last Name</b>	Configure the last name of the user. The last name can contain characters, letters, digits and _.
<b>Email Address</b>	Fill in the Email address for the user. Voicemail will be sent to this Email address.
<b>User Password</b>	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.
<b>Mobile Phone Number</b>	Configure the Mobile number of the user.
<b>Language</b>	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→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> . The dropdown list shows all the current available voice prompt languages on the UCM630xA. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under Web GUI→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> .

Table 46: FXS Extension Configuration Parameters→Media

<b>Analog Settings</b>	
<b>Call Waiting</b>	Configure to enable/disable call waiting feature. The default setting is "No".
<b>User '#' as SEND</b>	If configured, the # key can be used as SNED key after dialing the number on the analog phone. The default setting is "Yes".
<b>RX Gain</b>	Configure the RX gain for the receiving channel of analog FXS port. The valid range is -30dB to +6dB. The default setting is 0.
<b>TX Gain</b>	Configure the TX gain for the transmitting channel of analog FXS port. The valid range is -30dB to +6dB. The default setting is 0.
<b>Min RX Flash</b>	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.



<b>Max RX Flash</b>	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 cannot be modified. The default setting is 1250ms.
<b>Enable Polarity Reversal</b>	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".
<b>Echo Cancellation</b>	Specify "ON", "OFF" or a value (the power of 2) from 32 to 1024 as the number of taps of cancellation. <b>Note:</b> When configuring the number of taps, the number 256 is not translated into 256ms of echo cancellation. Instead, 256 taps mean $256/8 = 32$ ms. The default setting is "ON", which is 128 taps.
<b>3-Way Calling</b>	Configure to enable/disable 3-way calling feature on the user. The default setting is enabled.
<b>Send CallerID After</b>	Configure the number of rings before sending CID. Default setting is 1.
<b>Fax Mode</b>	For FXS extension, there are three options available in Fax Mode. The default setting is "None". <ul style="list-style-type: none"> <li>• <b>None:</b> Disable Fax.</li> <li>• <b>Fax Gateway:</b> If selected, the UCM630xA can support conversation and processing of Fax data from T.30 to T.38 or T.38 to T.30. only for FXS ports.</li> <li>• <b>Fax Detection:</b> During a call, the fax signal from the user/trunk will be detected, and the received fax will be sent to the email address configured for the user. If an email address has been configured for the user, the fax will be sent to the <b>Default Email Address</b> configured in <b>Fax/T.38-&gt;Fax Settings</b> .</li> </ul>

**Table 47: FXS Extension Configuration Parameters→Features**

Call Transfer	
<b>Call Forward Unconditional</b>	Configure the Call Forward Unconditional target number. If not configured, the Call Forward Unconditional feature is deactivated. The default setting is deactivated.
<b>CFU Time Condition</b>	Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time conditions are "Office Time", "Out of Office Time", "Holiday", "Out of Holiday", "Out of Office Time or Holiday" and "Specific". Note: <ul style="list-style-type: none"> <li>• "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> </ul>



	<ul style="list-style-type: none"> <li>Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday</b> page.</li> </ul>
<b>Call Forward No Answer</b>	Configure the Call Forward No Answer target number. If not configured, the Call Forward No Answer feature is deactivated. The default setting is deactivated.
<b>CFN Time Condition</b>	<p>Select time condition for Call Forward No Answer. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>“Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday</b> page.</li> </ul>
<b>Call Forward Busy</b>	Configure the Call Forward Busy target number. If not configured, the Call Forward Busy feature is deactivated. The default setting is deactivated.
<b>CFB Time Condition</b>	<p>Select time condition for Call Forward Busy. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>“Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday</b> page.</li> </ul>
<b>Do Not Disturb</b>	<p>If Do Not Disturb is enabled, all incoming calls will be dropped.</p> <p>All call forward settings will be ignored.</p>
<b>DND Time Condition</b>	The time condition of DND. The DND will take effect while the time condition is satisfied.
<b>DND Whitelist</b>	<p>If DND is enabled, calls from the whitelisted numbers will not be rejected. Multiple numbers are supported and must be separated by new lines. Pattern matching is supported.</p> <ul style="list-style-type: none"> <li><b>Z</b> match any digit from 1-9.</li> <li><b>N</b> match any digit from 2-9.</li> <li><b>X</b> match any digit from 0-9..</li> </ul>



<b>FWD Whitelist</b>	<p>Calls from users in the forward whitelist will not be forwarded.</p> <p>Pattern matching is supported.</p> <ul style="list-style-type: none"> <li>• <b>Z</b> match any digit from 1-9.</li> <li>• <b>N</b> match any digit from 2-9.</li> <li>• <b>X</b> match any digit from 0-9.</li> </ul>
<b>CC Settings</b>	
<b>Enable CC</b>	If enabled, UCM630xA will automatically alert this extension when a called party is available, given that a previous call to that party failed for some reason.
<b>Ring Simultaneously</b>	
<b>Ring Simultaneously</b>	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.
<b>External Number</b>	Set the external number to be rang simultaneously. '-' is the connection character which will be ignored.
<b>Time Condition for Ring Simultaneously</b>	Ring the external number simultaneously along with the extension on the basis of this time condition.
<b>Use callee DOD on FWD or RS</b>	Use the callee's DOD number as CallerID on Outgoing Forwarding or Ring Simultaneously calls.
<b>Hotline</b>	
<b>Enable Hotline</b>	If enabled, hotline dialing plan will be activated, a pre-configured number will be used according to the selected Hotline Type.
<b>Hotline Number</b>	Configure the Hotline Number
<b>Hotline Type</b>	<p>Configure the Hotline Type:</p> <ul style="list-style-type: none"> <li>• <b>Immediate Hotline:</b> When the phone is off-hook, UCM630xA will immediately dial the preset number</li> <li>• <b>Delay Hotline:</b> When the phone is off hook, if there is no dialing within 5 seconds, UCM630xA will dial the preset number.</li> </ul>
<b>Monitor privilege control</b>	
<b>Allow call-barging</b>	Members of the list can spy on this extension via feature codes.





## Seamless transfer privilege control

### Allowed to seamless transfer

Members of the list can seamlessly transfer via feature code.

## 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 UCM630xA, which can be configured in the global ring timeout setting under Web GUI→PBX Settings→Voice Prompt→Custom 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.

### Use MOH as IVR ringback tone

If enabled, when the call to the extension is made through the IVR, the caller will hear MOH as ringback tone instead of the regular ringback tone.

### Music On Hold

Select which Music On Hold class to suggest to extension when putting the active call on hold.

### Call Duration Limit

Check to enable and set the call limit the duration.

### Maximum Call Duration (s)

The maximum call duration (in seconds). The default value 0 means no limit. Max value is 86400 seconds

### Email Missed Calls

Send a log of missed calls to the extension's email address.

### Missed Call Type

If **Email Missed Calls** enabled, users can select the type of missed calls to be sent via email, the available types are:



	<ul style="list-style-type: none"> <li>• <b>Default:</b> All missed calls will be sent in email notifications.</li> <li>• <b>Missed Internal Call:</b> Only missed local extension-to-extension calls will be sent in email notifications.</li> <li>• <b>Missed External Call:</b> Only missed calls from trunks will be sent in email notifications.</li> </ul>
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**Table 48: FXS Extension Configuration Parameters→Specific Time**

Specific Time	
<b>Time Condition</b>	Click to add Time Condition to configure specific time for this extension.

**Table 49: FXS Extension Configuration Parameters→Follow Me**

Follow Me	
<b>Enable</b>	Configure to enable or disable Follow Me for this user.
<b>Skip Trunk Auth</b>	If the outbound calls need to check the password, we should enable this option or enable the option "Skip Trunk Auth" of the Extension. Otherwise this Follow Me cannot call out.
<b>Music On Hold Class</b>	Configure the Music On Hold class that the caller would hear while tracking the user.
<b>Enable Destination</b>	Configure to enable destination.
<b>Default Destination</b>	The call will be routed to this destination if no one in the Follow Me answers the call.
<b>Confirm When Answering</b>	If enabled, call will need to be confirmed after answering.
<b>Use Callee DOD for Follow Me</b>	Use the callee DOD number as CID if configured Follow Me numbers are external numbers.
<b>New Follow Me Number</b>	Add a new Follow Me number which could be a "Local Extension" or an "External Number". The selected dial plan should have permissions to dial the defined external number.
<b>Dialing Order</b>	This is the order in which the Follow Me destinations will be dialed to reach the user.



## Batch Add Extensions

### Batch Add SIP Extensions

To add multiple SIP extensions, BATCH add can be used to create standardized SIP extension accounts. However, unique extension username cannot be set using BATCH add.

Under Web GUI→**Extension/Trunk**→**Extensions**, click on "Add" and select extension type as SIP extension, and "Select Add Method" as Batch.

**Table 50: Batch Add SIP Extension Parameters**

General	
<b>Create Number</b>	Specify the number of extensions to be added. The default setting is 5.
<b>Extension Incrementation</b>	Select how much to increment successive extensions. For example, if the value is 2, the extensions will be 1000,1002,1004,..... <b>Note:</b> Up to 3 characters.
<b>Extension</b>	Configure the starting extension number of the batch of extensions to be added.
<b>Permission</b>	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". <b>Note:</b> Users need to have the same level as or higher level than an outbound rule's privilege to make outbound calls from this rule.
<b>Voicemail</b>	Configure Voicemail. There are three valid options and the default option is "Enable Local Voicemail". <ul style="list-style-type: none"> <li>• <b>Disable Voicemail:</b> Disable Voicemail.</li> <li>• <b>Enable Local Voicemail:</b> Enable voicemail for the user.</li> <li>• <b>Enable Remote Voicemail:</b> Forward the notify message from remote voicemail system for the user, and the local voicemail will be disabled. <b>Note:</b> Remote voicemail feature is used only for Infomatec (Brazil).</li> </ul>
<b>SIP/IAX Password</b>	Configure the SIP/IAX password for the users. Three options are available to create password for the batch of extensions. <ul style="list-style-type: none"> <li>• User Random Password. A random secure password will be automatically generated. It is recommended to use this password for security purpose.</li> <li>• Use Extension as Password.</li> <li>• Enter a password to be used on all the extensions in the batch.</li> </ul>
<b>Voicemail Password</b>	Configure Voicemail password (digits only) for the users. <ul style="list-style-type: none"> <li>• User Random Password. A random password in digits will be automatically generated. It is recommended to use this password for security purpose.</li> <li>• Use Extension as Password.</li> </ul>



	Enter a password to be used on all the extensions in the batch.
<b>Send Voicemail to Email</b>	Send voicemail messages to the configured email address. If set to "Default", the global setting will be used. Global settings can be found in Voicemail->Voicemail Email Settings.
<b>Keep Voicemail after Emailing</b>	Only applies if extension-level or global Send Voicemail to Email is enabled.
<b>CallerID Number</b>	<p>Configure CallerID Number when adding Batch Extensions.</p> <ul style="list-style-type: none"> <li>Use Extension as Number Users can choose to use the extension number as CallerID</li> <li>Use as Number Users can choose to set a specific number instead of using the extension number.</li> </ul>
<b>Skip Voicemail Password Verification</b>	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.
<b>Enable Keep-alive</b>	If enabled, the PBX will regularly send SIP OPTIONS to check if host device is online.
<b>Keep-alive Frequency</b>	Configure the keep-alive interval (in seconds) to check if the host is up.
<b>Disable This Extension</b>	Check this box to disable this extension.
<b>Enable SCA</b>	If enabled, (1) Call Forward, Call Waiting and Do Not Disturb settings will not work, (2) Concurrent Registrations can be set only to 1, and (3) Private numbers can be added in Call Features->SCA page.
<b>Emergency Calls CID</b>	CallerID number that will be used when calling out and receiving direct callbacks.
<b>Language</b>	Select voice prompt language for this extension. If set to "Default", the global setting for voice prompt language will be used.
<b>Media</b>	
<b>NAT</b>	<p>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 is related to NAT configuration or Firewall's support of SIP and RTP ports.</p> <p>The default setting is enabled.</p>
<b>Enable Direct Media</b>	By default, the PBX will route the media streams 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".



<b>DTMF Mode</b>	Select DTMF mode for the user to send DTMF. The default setting is "RFC4733". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, a-law or u-law are required. When "Auto" is selected, RFC4733 will be used if offered, otherwise "Inband" will be used.
<b>Alert-info</b>	When present in an INVITE request, the Alert-info header field specifies an alternative ring tone to the UAS.
<b>SRTP</b>	Enable/disable SRTP for RTP stream encryption.
<b>Packet Loss Retransmission</b>	Configure to enable Packet Loss Retransmission. <ul style="list-style-type: none"> <li>• <b>NACK</b></li> <li>• <b>NACK+RTX(SSRC-GROUP)</b></li> <li>• <b>OFF</b></li> </ul>
<b>Video FEC</b>	Check to enable Forward Error Correction (FEC) for Video.
<b>FECC</b>	Configure to enable FECC
<b>Audio FEC</b>	Check to enable Forward Error Correction (FEC) for Audio.
<b>ACL Policy</b>	Access Control List manages the IP addresses that can register to this extension. <ul style="list-style-type: none"> <li>• <b>Allow All:</b> Any IP address can register to this extension.</li> <li>• <b>Local Network:</b> Only IP addresses in the configured network segments can register to this extension. Press "Add Local Network Address" to add more IP segments.</li> </ul>
<b>Jitter Buffer</b>	Select jitter buffer method. <ul style="list-style-type: none"> <li>• <b>Disable:</b> Jitter buffer will not be used.</li> <li>• <b>Fixed:</b> Jitter buffer with a fixed size (equal to the value of "jitter buffer size")</li> <li>• <b>Adaptive:</b> Jitter buffer with an adaptive size (no more than the value of "max jitter buffer").</li> <li>• <b>NetEQ:</b> Dynamic jitter buffer via NetEQ.</li> </ul>
<b>Codec Preference</b>	Configure the codecs to be used.
<b>Call Transfer</b>	
<b>Presence Status</b>	Select which presence status to set for the extension and configure call forward conditions for each status. Six possible options are possible: <b>"Available"</b> , <b>"Away"</b> , <b>"Chat"</b> , <b>"Custom"</b> , <b>"DND"</b> and <b>"Unavailable"</b> . More details at [PRESENCE].
<b>Call Forward Unconditional</b>	Enable and configure the Call Forward Unconditional target number. Available options for target number are: <ul style="list-style-type: none"> <li>• <b>"None"</b>: Call forward deactivated.</li> <li>• <b>"Extension"</b>: Select an extension from dropdown list as CFU target.</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>“Custom Number”</b>: Enter a customer number as target. For example: *97.</li> <li>• <b>“Voicemail”</b>: Select an extension from dropdown list. Incoming calls will be forwarded to voicemail of selected extension.</li> <li>• <b>“Ring Group”</b>: Select a ring group from dropdown list as CFU target.</li> <li>• <b>“Queues”</b>: Select a queue from dropdown list as CFU target.</li> <li>• <b>“Voicemail Group”</b>: Select a voicemail group from dropdown list as CFU target.</li> </ul> <p>The default setting is “None”.</p>
<b>CFU Time Condition</b>	<p>Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p>Note:</p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</li> </ul>
<b>Call Forward No Answer</b>	<p>Configure the Call Forward No Answer target number. Available options for target number are:</p> <ul style="list-style-type: none"> <li>• <b>“None”</b>: Call forward deactivated.</li> <li>• <b>“Extension”</b>: Select an extension from dropdown list as CFN target.</li> <li>• <b>“Custom Number”</b>: Enter a customer number as target. For example: *97.</li> <li>• <b>“Voicemail”</b>: Select an extension from dropdown list. Incoming calls will be forwarded to voicemail of selected extension.</li> <li>• <b>“Ring Group”</b>: Select a ring group from dropdown list as CFN target.</li> <li>• <b>“Queues”</b>: Select a queue from dropdown list as CFN target.</li> <li>• <b>“Voicemail Group”</b>: Select a voicemail group from dropdown list as CFN target.</li> </ul> <p>The default setting is “None”.</p>
<b>CFN Time Condition</b>	<p>Select time condition for Call Forward No Answer. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time</li> </ul>



	<p>period.</p> <ul style="list-style-type: none"> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</li> </ul>
<b>Call Forward Busy</b>	<p>Configure the Call Forward Busy target number. Available options for target number are:</p> <ul style="list-style-type: none"> <li>• <b>“None”</b>: Call forward deactivated.</li> <li>• <b>“Extension”</b>: Select an extension from dropdown list as CFB target.</li> <li>• <b>“Custom Number”</b>: Enter a customer number as target. For example: *97.</li> <li>• <b>“Voicemail”</b>: Select an extension from dropdown list. Incoming calls will be forwarded to voicemail of selected extension.</li> <li>• <b>“Ring Group”</b>: Select a ring group from dropdown list as CFB target.</li> <li>• <b>“Queues”</b>: Select a queue from dropdown list as CFB target.</li> <li>• <b>“Voicemail Group”</b>: Select a voicemail group from dropdown list as CFB target.</li> </ul> <p>The default setting is “None”.</p>
<b>CFB Time Condition</b>	<p>Select time condition for Call Forward Busy. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</li> </ul>
<b>Do Not Disturb</b>	<p>If Do Not Disturb is enabled, all incoming calls will be dropped.</p> <p>All call forward settings will be ignored.</p>
<b>DND Time Condition</b>	<p>Select time condition for Do Not Disturb. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time</li> </ul>



	<p>period.</p> <ul style="list-style-type: none"> <li>Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> </ul> <p>Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</p>
<b>DND Whitelist</b>	<p>If DND is enabled, calls from the whitelisted numbers will not be rejected. Multiple numbers are supported and must be separated by new lines. Pattern matching is supported.</p> <p>Z match any digit from 1-9, N match any digit from 2-9, X match any digit from 0-9.</p>
<b>FWD Whitelist</b>	<p>Calls from users in the forward whitelist will not be forwarded. Pattern matching is supported.</p> <p>Z match any digit from 1-9, N match any digit from 2-9, X match any digit from 0-9.</p>
<b>CC Settings</b>	
<b>Enable CC</b>	<p>If enabled, UCM630xA 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 is disabled.</p>
<b>CC Mode</b>	<p>Two modes for Call Completion are supported:</p> <ul style="list-style-type: none"> <li><b>Normal:</b> This extension is used as ordinary extension.</li> <li><b>For Trunk:</b> This extension is registered from a PBX.</li> </ul> <p>The default setting is “Normal”.</p>
<b>CC Max Agents</b>	<p>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 can make. The minimum value is 1.</p>
<b>CC Max Monitors</b>	<p>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.</p>
<b>Ring Simultaneously</b>	
<b>Ring Simultaneously</b>	<p>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.</p>
<b>External Number</b>	<p>Set the external number to be rang simultaneously. '-' is the connection character which will be ignored.</p>





	This field accepts only letters, numbers, and special characters + = * #.
<b>Time Condition for Ring Simultaneously</b>	Ring the external number simultaneously along with the extension on the basis of this time condition.
<b>Use callee DOD on FWD or RS</b>	Use the DOD number when calls are being diverted/forwarded to external destinations or when ring simultaneous is configured.
<b>Monitor privilege control</b>	
<b>Allowed to call-barging</b>	Add members from “Available Extensions” to “Selected Extensions” so that the selected extensions can spy on the used extension using feature code.
<b>Seamless transfer privilege control</b>	
<b>Allowed to seamless transfer</b>	Any extensions on the UCM can perform seamless transfer. When using Pickup Incall feature, only extensions available on the “Selected Extensions” list can perform seamless transfer to the edited extension.
<b>Other Settings</b>	
<b>Ring Timeout</b>	<p>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 UCM630xA, which can be configured in the global ring timeout setting under Web GUI→PBX Settings→Voice Prompt→Custom Prompt: General Preference. The valid range is between 3 seconds and 600 seconds.</p> <p><b>Note:</b> If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
<b>Auto Record</b>	Enable automatic recording for the calls using this extension. The default setting is disabled. The recordings can be accessed under Web GUI→ <b>CDR</b> → <b>Recording Files</b> .
<b>Skip Trunk Auth</b>	<ul style="list-style-type: none"> <li>• If set to “yes”, users can skip entering the password when making outbound calls.</li> <li>• If set to “By Time”, users can skip entering the password when making outbound calls during the selected time condition.</li> <li>• If set to “No”, users will be asked to enter the password when making outbound calls.</li> </ul>
<b>Time Condition for Skip Trunk Auth</b>	If ‘Skip Trunk Auth’ is set to ‘By Time’, select a time condition during which users can skip entering password when making outbound calls.
<b>Dial Trunk Password</b>	Configure personal password when making outbound calls via trunk.
<b>Enable LDAP</b>	If enabled, the extension will be added to LDAP Phonebook PBX list.



<b>Enable WebRTC Support</b>	Enable registration and call from WebRTC.
<b>Bind PMS Room</b>	If enabled, the system will create a room whose room number, by default, will equal the extension number in PMS module. Note: If this room already exists, the configuration of the existing room will be overwritten.
<b>Music On Hold</b>	Specify which Music On Hold class to suggest to the bridged channel when putting them on hold.
<b>Call Duration Limit</b>	The maximum duration of call-blocking.
<b>Maximum Call Duration</b>	The maximum call duration (in seconds). The default value 0 means no limit.
<b>Call Waiting</b>	<p>If disabled, UCM will not invite the extension when it is already in a call and will do the same work as the user is busy.</p> <p><b>Note:</b> the option only works when the caller dials the extension directly.</p>

## Batch Add IAX Extensions

Under Web GUI → **Extension/Trunk** → **Extensions**, click on “Add”, then select extension type as IAX Extension and the add method to be Batch.

**Table 51: Batch Add IAX Extension Parameters**

General	
<b>Create Number</b>	Specify the number of extensions to be added. The default setting is 5.
<b>Extension Incrementation</b>	Select how much to increment successive extensions. For example, if the value is 2, the extensions will be 1000,1002,1004,.....
<b>Extension</b>	The extension number associated with this particular user/phone.
<b>Permission</b>	<p>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”.</p> <p><b>Note:</b> 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.</p>
<b>CallerID Number</b>	Configure the Caller ID number displayed when dialing calls from this user. Note: The Caller ID usage might be limited by your VoIP provider. In Batch Add Method, “e” means to use the extension as the number.
<b>Voicemail</b>	<p>Configure Voicemail. There are three valid options and the default option is “Enable Local Voicemail”.</p> <ul style="list-style-type: none"> <li>• <b>Disable Voicemail:</b> Disable Voicemail.</li> <li>• <b>Enable Local Voicemail:</b> Enable voicemail for the user.</li> </ul>



<b>SIP/IAX Password</b>	<p>Configure the SIP/IAX password for the users. Three options are available to create password for the batch of extensions.</p> <ul style="list-style-type: none"> <li>• User Random Password. A random secure password will be automatically generated. It is recommended to use this password for security purpose.</li> <li>• Use Extension as Password.</li> <li>• Enter a password to be used on all the extensions in the batch.</li> </ul>
<b>Voicemail Password</b>	<p>Configure Voicemail password (digits only) for the users.</p> <ul style="list-style-type: none"> <li>• User Random Password. A random password in digits will be automatically generated. It is recommended to use this password for security purpose.</li> <li>• Use Extension as Password.</li> <li>• Enter a password to be used on all the extensions in the batch.</li> </ul>
<b>Send Voicemail to Email</b>	<p>Send voicemail messages to the configured email address. If set to "Default", the global setting will be used. Global settings can be found in Voicemail-&gt;Voicemail Email Settings.</p>
<b>Keep Voicemail after Emailing</b>	<p>Only applies if extension-level or global Send Voicemail to Email is enabled.</p>
<b>Auto Record</b>	<p>Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under Web GUI→<b>CDR</b>→<b>Recording Files</b>.</p>
<b>Skip Voicemail Password Verification</b>	<p>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.</p>
<b>Disable This Extension</b>	<p>Check this box to disable this extension.</p>
<b>Language</b>	<p>Select voice prompt language for this extension. If set to "Default", the global setting for voice prompt language will be used.</p>
<b>IAX Settings</b>	
<b>Max Number of Calls</b>	<p>Configure the maximum number of calls allowed for each remote IP address.</p>
<b>Require Call Token</b>	<p>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".</p>
<b>SRTP</b>	<p>Enable/disable SRTP for RTP stream encryption.</p>
<b>ACL Policy</b>	<p>Access Control List manages the IP addresses that can register to this extension.</p> <ul style="list-style-type: none"> <li>• <b>Allow All:</b> Any IP address can register to this extension.</li> <li>• <b>Local Network:</b> Only IP addresses in the configured network segments can</li> </ul>



	register to this extension.
<b>Codec Preference</b>	Configure the codecs to be used.

Call Transfer	
<b>Call Forward Unconditional</b>	Enable and configure the Call Forward Unconditional target number.
<b>CFU Time Condition</b>	<p>Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p>Note:</p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</li> </ul>
<b>Call Forward No Answer</b>	Configure the Call Forward No Answer target number.
<b>CFN Time Condition</b>	<p>Select time condition for Call Forward No Answer. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p>Notes:</p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</li> </ul>
<b>Call Forward Busy</b>	Configure the Call Forward Busy target number.
<b>CFB Time Condition</b>	<p>Select time condition for Call Forward Busy. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p>Notes:</p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> </ul>



	<ul style="list-style-type: none"> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</li> </ul>
<b>Do Not Disturb</b>	<p>If Do Not Disturb is enabled, all incoming calls will be dropped.</p> <p>All call forward settings will be ignored.</p>
<b>DND Time Condition</b>	<p>Select time condition for Do Not Disturb. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</li> </ul>
<b>DND Whitelist</b>	<p>If DND is enabled, calls from the whitelisted numbers will not be rejected. Multiple numbers are supported and must be separated by new lines. Pattern matching is supported.</p> <ul style="list-style-type: none"> <li>• <b>Z</b> match any digit from 1-9,</li> <li>• <b>N</b> match any digit from 2-9,</li> <li>• <b>X</b> match any digit from 0-9.</li> </ul>
<b>FWD Whitelist</b>	<p>Calls from users in the forward whitelist will not be forwarded. Pattern matching is supported.</p> <ul style="list-style-type: none"> <li>• <b>Z</b> match any digit from 1-9,</li> <li>• <b>N</b> match any digit from 2-9,</li> <li>• <b>X</b> match any digit from 0-9.</li> </ul>
<b>Ring Simultaneously</b>	
<b>Ring Simultaneously</b>	<p>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.</p>
<b>External Number</b>	<p>Set the external number to be rang simultaneously. ‘-’ is the connection character which will be ignored.</p> <p>This field accepts only letters, numbers, and special characters + = * #.</p>




<b>Time Condition for Ring Simultaneously</b>	Ring the external number simultaneously along with the extension on the basis of this time condition.
<b>Use callee DOD on FWD or RS</b>	Use the DOD number when calls are being diverted/forwarded to external destinations or when ring simultaneous is configured.
<b>Monitor privilege control</b>	
<b>Allowed to call-barging</b>	Add members from “Available Extensions” to “Selected Extensions” so that the selected extensions can spy on the used extension using feature code.
<b>Seamless transfer privilege control</b>	
<b>Allowed to seamless transfer</b>	Any extensions on the UCM can perform seamless transfer. When using Pickup Incall feature, only extensions available on the “Selected Extensions” list can perform seamless transfer to the edited extension.
<b>Other Settings</b>	
<b>Ring Timeout</b>	<p>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 UCM630xA, which can be configured in the global ring timeout setting under Web GUI→PBX Settings→Voice Prompt→Custom Prompt: General Preference. The valid range is between 5 seconds and 600 seconds.</p> <p><b>Note:</b> If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
<b>Auto Record</b>	Enable automatic recording for the calls using this extension. The default setting is disabled. The recordings can be accessed under Web GUI→CDR→Recording Files.
<b>Skip Trunk Auth</b>	<ul style="list-style-type: none"> <li>• If set to “yes”, users can skip entering the password when making outbound calls.</li> <li>• If set to “By Time”, users can skip entering the password when making outbound calls during the selected time condition.</li> <li>• If set to “No”, users will be asked to enter the password when making outbound calls.</li> </ul>
<b>Time Condition for Skip Trunk Auth</b>	If ‘Skip Trunk Auth’ is set to ‘By Time’, select a time condition during which users can skip entering password when making outbound calls.
<b>Dial Trunk Password</b>	Configure personal password when making outbound calls via trunk.
<b>Enable LDAP</b>	If enabled, the extension will be added to LDAP Phonebook PBX list.






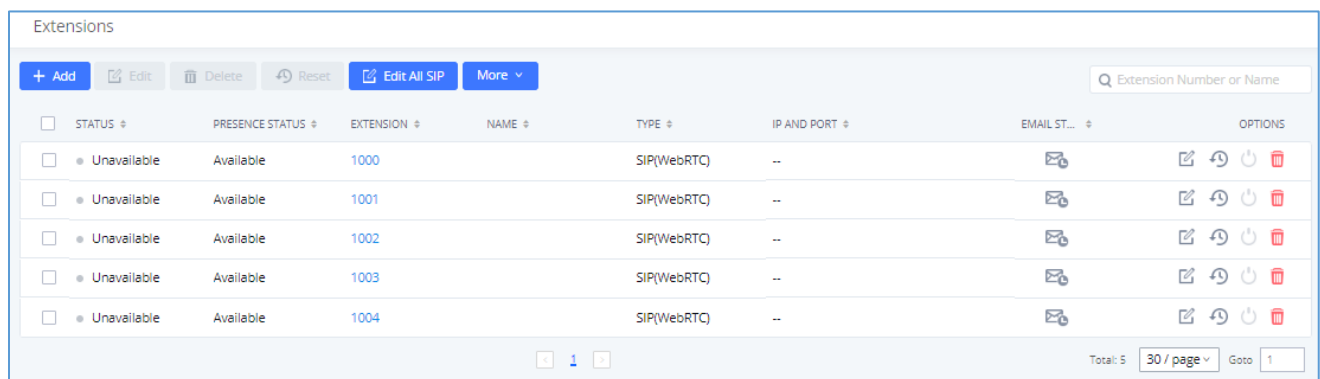
<b>Music On Hold</b>	Specify which Music On Hold class to suggest to the bridged channel when putting them on hold.
<b>Call Duration Limit</b>	Check to enable and set the call limit the duration.
<b>Maximum Call Duration (s)</b>	The maximum call duration (in seconds). The default value 0 means no limit. Max value is 86400 seconds

## Batch Extension Resetting Functionality

Users can select multiple extensions and reset their settings to default by pressing the reset button  and confirm the reset functionality. Once done, all settings in Basic Settings page will be restored to default values with the exception of Concurrent Registrations. User voicemail password will be reset to Random Password. User voicemail prompts and recordings will be deleted. User Management settings will also be restored to default with the exception of usernames and custom privileges

## Search and Edit Extension

All the UCM630xA extensions are listed under Web GUI → **Extension/Trunk** → **Extensions**, with status, Extension, CallerID Name, Technology (SIP, IAX and FXS), IP and Port. Each extension has a checkbox for users to "Edit" or "Delete". 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.



STATUS	PRESENCE STATUS	EXTENSION	NAME	TYPE	IP AND PORT	EMAIL ST...	OPTIONS
Unavailable	Available	1000		SIP(WebRTC)	--		
Unavailable	Available	1001		SIP(WebRTC)	--		
Unavailable	Available	1002		SIP(WebRTC)	--		
Unavailable	Available	1003		SIP(WebRTC)	--		
Unavailable	Available	1004		SIP(WebRTC)	--		

Figure 79: Manage Extensions


- **Status**

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


- Green: Idle
- 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.

- **Reset single extension**


Click on  to reset the extension parameters to default (except concurrent registration).

Other settings will be restored to default in **Maintenance**→**User Management**→**User Information** except username and permissions and delete the user voicemail prompt and voice messages.

- **Reboot the user**

Click on  to send NOTIFY reboot event to the device which has an UCM630xA extension already registered. To successfully reboot the user, "Zero Config" needs to be enabled on the UCM630xA Web GUI→**Value-added Features**→**Zero Config**→**Zero Config 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**

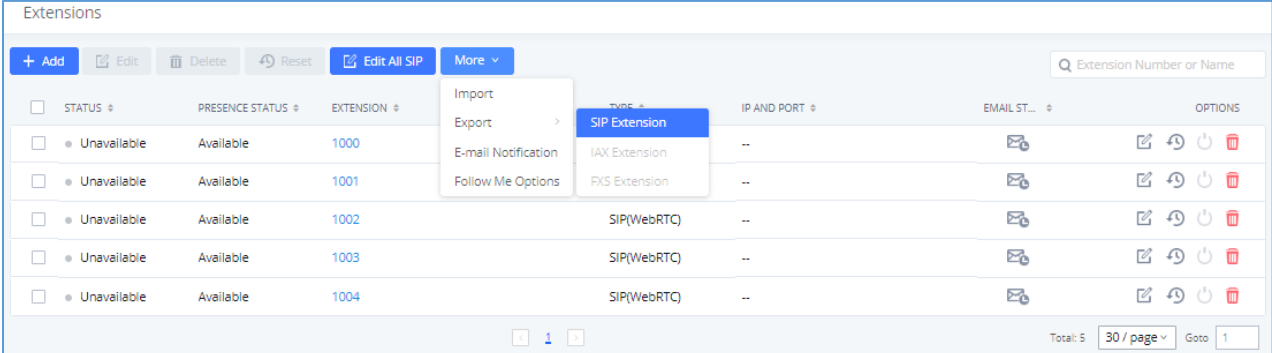
Select the checkbox for the extension(s). Then click on "Edit" to edit the extensions in a batch.

- **Delete selected extensions**

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

## Export Extensions

The extensions configured on the UCM630xA 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.



The screenshot shows the 'Extensions' management interface. At the top, there are buttons for '+ Add', 'Edit', 'Delete', 'Reset', 'Edit All SIP', and 'More'. A search bar is on the right. Below the buttons is a table with columns: STATUS, PRESENCE STATUS, EXTENSION, TYPE, IP AND PORT, EMAIL ST..., and OPTIONS. The table lists five extensions (1000-1004) with status 'Unavailable' and presence 'Available'. The 'More' menu is open, showing options: Import, Export, E-mail Notification, and Follow Me Options. The 'Export' option is selected, and a sub-menu is visible with 'SIP Extension', 'IAX Extension', and 'FXS Extension' options.

STATUS	PRESENCE STATUS	EXTENSION	TYPE	IP AND PORT	EMAIL ST...	OPTIONS
<input type="checkbox"/>	Unavailable	Available	1000	--		
<input type="checkbox"/>	Unavailable	Available	1001	--		
<input type="checkbox"/>	Unavailable	Available	1002	SIP(WebRTC)		
<input type="checkbox"/>	Unavailable	Available	1003	SIP(WebRTC)		
<input type="checkbox"/>	Unavailable	Available	1004	SIP(WebRTC)		

Total: 5 | 30 / page | Goto 1

Figure 80: Export Extensions





The exported csv file can serve as a template for users to fill in desired extension information to be imported to the UCM630xA.

## Import Extensions

The capability to import extensions to the UCM630xA provides users flexibility to batch add extensions with similar or different configuration quickly into the PBX system.

1. Export extension csv file from the UCM630xA 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.

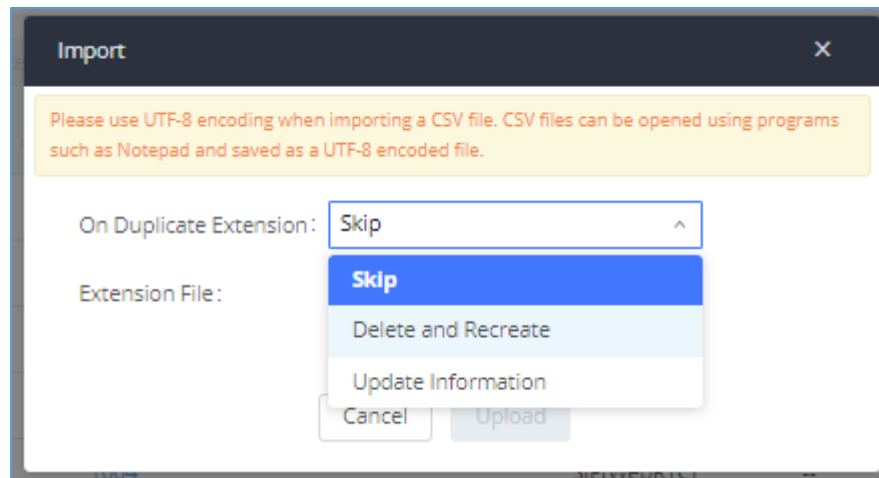


Figure 81: Import Extensions

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 "Choose file to upload" to select csv file from local directory in the PC.
6. Click on "Apply Changes" to apply the imported file on the UCM630xA.

Example of file to import:



A	B	C	D	E	F	G	H	I	J	K	L	M	N
Extension	Technology	Enable Voicemail	CallerID	SIP/IAX Password	Voicema	Skip Voicemail Password Verification	Ring Timeout	Auto Record	SRTP	Fax Mode	Strategy	Local Subnet 1	Local Sub
1000	SIP	yes	1000	admin123	61783	no		no	no	None	Allow All		
1001	SIP	yes	1001	admin123	955921	no		no	no	None	Allow All		
1002	SIP	yes	1002	admin123	269824	no		no	no	None	Allow All		
1003	SIP	yes	1003	admin123	363196	no		no	no	None	Allow All		
1004	SIP	yes	1004	admin123	12860	no		no	no	None	Allow All		

Figure 82: Import File

Table 52: SIP extensions Imported File Example

Field	Supported values
Extension	Digits
Technology	SIP/SIP(WebRTC)
Enable Voicemail	yes/no/remote
CallerID Number	Digits
SIP/IAX Password	Alphanumeric characters
Voicemail Password	Digits
Skip Voicemail Password Verification	yes/no
Ring Timeout	Empty/ 3 to 600 (in second)
SRTP	yes/no
Strategy	Allow All/Local Subnet Only/A Specific IP Address
Local Subnet 1	IP address/Mask
Local Subnet 2	IP address/Mask
Local Subnet 3	IP address/Mask
Local Subnet 4	IP address/Mask
Local Subnet 5	IP address/Mask
Local Subnet 6	IP address/Mask
Local Subnet 7	IP address/Mask
Local Subnet 8	IP address/Mask
Local Subnet 9	IP address/Mask
Local Subnet 10	IP address/Mask
Specific IP Address	IP address
Skip Trunk Auth	yes/no/bytime
Codec Preference	PCMU,PCMA,GSM,G.726,G.722,G.729,H.264, H.265,ILBC,AAL2-G.726- 32,ADPCM,G.723,H.263,H.263p,vp8,opus
Permission	Internal/Local/National/International
NAT	yes/no
DTMF Mode	RFC4733/info/inband/auto
Insecure	Port



<b>Enable Keep-alive</b>	Yes/no
<b>Keep-alive Frequency</b>	Value from 1-3600
<b>AuthID</b>	Alphanumeric value without special characters
<b>TEL URI</b>	Disabled/user=phone/enabled
<b>Call Forward Busy</b>	Digits
<b>Call Forward No Answer</b>	Digits
<b>Call Forward Unconditional</b>	Digits
<b>Support Hot-Desking Mode</b>	Yes/no
<b>Dial Trunk Password</b>	Digits
<b>Disable This Extension</b>	Yes/no
<b>CFU Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFN Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFB Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Music On Hold</b>	Default/ringbacktone_default
<b>CC Agent Policy</b>	If CC is disabled use: never If CC is set to normal use: generic If CC is set to trunk use: native
<b>CC Monitor Policy</b>	Generic/never
<b>CCBS Available Timer</b>	3600/4800
<b>CCNR Available Timer</b>	3600/7200
<b>CC Offer Timer</b>	60/120
<b>CC Max Agents</b>	Value from 1-999
<b>CC Max Monitors</b>	Value from 1-999
<b>Ring simultaneously</b>	Yes/no
<b>External Number</b>	Digits
<b>Time Condition for Ring Simultaneously</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Time Condition for Skip Trunk Auth</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Enable LDAP</b>	Yes/no
<b>Enable T.38 UDPTL</b>	Yes/no
<b>Max Contacts</b>	Values from 1-10
<b>Enable WebRTC</b>	Yes/no



<b>Alert-Info</b>	None/Ring 1/Ring2/Ring3/Ring 4/Ring 5/Ring 6/Ring 7/ Ring 8/Ring 9/Ring 10/bellcore-dr1/bellcore-dr2/ bellcore-dr3/ bellcore-dr4/ bellcore-dr5/custom
<b>Do Not Disturb</b>	Yes/no
<b>DND Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Custom Auto answer</b>	Yes/no
<b>Do Not Disturb Whitelist</b>	Empty/digits
<b>User Password</b>	Alphanumeric characters.
<b>First Name</b>	Alphanumeric without special characters.
<b>Last Name</b>	Alphanumeric without special characters.
<b>Email Address</b>	Email address
<b>Language</b>	Default/en/zh
<b>Phone Number</b>	Digits
<b>Call-Barging Monitor</b>	Extensions allowed to call barging
<b>Seamless Transfer Members</b>	Extensions allowed to seamless transfer

**Table 53: IAX extensions Imported File Example**

<b>Field</b>	<b>Supported values</b>
<b>Extension</b>	Digits
<b>Technology</b>	IAX
<b>Enable Voicemail</b>	yes/no
<b>CallerID Number</b>	Digits
<b>SIP/IAX Password</b>	Alphanumeric characters
<b>Voicemail Password</b>	Digits
<b>Skip Voicemail Password Verification</b>	yes/no
<b>Ring Timeout</b>	Empty/ 3 to 600 (in second)
<b>SRTP</b>	yes/no
<b>Strategy</b>	Allow All/Local Subnet Only/A Specific IP Address
<b>Local Subnet 1</b>	IP address/Mask
<b>Local Subnet 2</b>	IP address/Mask
<b>Local Subnet 3</b>	IP address/Mask
<b>Local Subnet 4</b>	IP address/Mask
<b>Local Subnet 5</b>	IP address/Mask
<b>Local Subnet 6</b>	IP address/Mask



<b>Local Subnet 7</b>	IP address/Mask
<b>Local Subnet 8</b>	IP address/Mask
<b>Local Subnet 9</b>	IP address/Mask
<b>Local Subnet 10</b>	IP address/Mask
<b>Specific IP Address</b>	IP address
<b>Skip Trunk Auth</b>	yes/no/bytime
<b>Codec Preference</b>	PCMU,PCMA,GSM,G.726,G.722,G.729,H.264, H.265,ILBC,AAL2-G.726- 32,ADPCM,G.723,H.263,H.263p,vp8,opus
<b>Permission</b>	Internal/Local/National/International
<b>NAT</b>	yes/no
<b>Call Forward Busy</b>	Digits
<b>Call Forward No Answer</b>	Digits
<b>Call Forward Unconditional</b>	Digits
<b>Require Call Token</b>	Yes/no/auto
<b>Max Number of Calls</b>	Values from 1-512
<b>Dial Trunk Password</b>	Digits
<b>Disable This Extension</b>	Yes/no
<b>CFU Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFN Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFB Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Music On Hold</b>	Default/ringbacktone_default
<b>Ring simultaneously</b>	Yes/no
<b>External Number</b>	Digits
<b>Time Condition for Ring Simultaneously</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Time Condition for Skip Trunk Auth</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Enable LDAP</b>	Yes/no
<b>Limit Max time (s)</b>	empty
<b>Do Not Disturb</b>	Yes/no
<b>DND Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Do Not Disturb Whitelist</b>	Empty/digits



<b>User Password</b>	Alphanumeric characters.
<b>First Name</b>	Alphanumeric without special characters.
<b>Last Name</b>	Alphanumeric without special characters.
<b>Email Address</b>	Email address
<b>Language</b>	Default/en/zh
<b>Phone Number</b>	Digits
<b>Call-Barging Monitor</b>	Extensions allowed to call barging
<b>Seamless Transfer Members</b>	Extensions allowed to seamless transfer

**Table 54: FXS Extensions Imported File Example**

<b>Field</b>	<b>Supported values</b>
<b>Extension</b>	Digits
<b>Technology</b>	FXS
<b>Analog Station</b>	FXS1/FXS2
<b>Enable Voicemail</b>	yes/no
<b>CallerID Number</b>	Digits
<b>Voicemail Password</b>	Digits
<b>Skip Voicemail Password Verification</b>	yes/no
<b>Ring Timeout</b>	Empty/ 3 to 600 (in second)
<b>Auto Record</b>	yes/no
<b>Fax Mode</b>	None/Fax Gateway/Fax Detection
<b>Skip Trunk Auth</b>	Yes/no/bytime
<b>Permission</b>	Internal/Local/National/International
<b>Call Forward Busy</b>	Digits
<b>Call Forward No Answer</b>	Digits
<b>Call Forward Unconditional</b>	Digits
<b>Call Waiting</b>	Yes/no
<b>Use # as SEND</b>	Yes/no
<b>RX Gain</b>	Values from -30→6
<b>TX Gain</b>	Values from -30→6
<b>MIN RX Flash</b>	Values from: 30 – 1000
<b>MAX RX Flash</b>	Values from: 40 – 2000
<b>Enable Polarity Reversal</b>	Yes/no
<b>Echo Cancellation</b>	On/Off/32/64/128/256/512/1024



<b>3-Way Calling</b>	Yes/no
<b>Send CallerID After</b>	1/2
<b>Dial Trunk Password</b>	digits
<b>Disable This Extension</b>	Yes/no
<b>CFU Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFN Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFB Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Music On Hold</b>	Default/ringbacktone_default
<b>CC Agent Policy</b>	If CC is disabled use: never If CC is set to normal use: generic If CC is set to trunk use: native
<b>CC Monitor Policy</b>	Generic/never
<b>CCBS Available Timer</b>	3600/4800
<b>CCNR Available Timer</b>	3600/7200
<b>CC Offer Timer</b>	60/120
<b>CC Max Agents</b>	Value from 1-999
<b>CC Max Monitors</b>	Value from 1-999
<b>Ring simultaneously</b>	Yes/no
<b>External Number</b>	Digits
<b>Time Condition for Ring Simultaneously</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Time Condition for Skip Trunk Auth</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Enable LDAP</b>	Yes/no
<b>Enable Hotline</b>	Yes/no
<b>Hotline Type</b>	Immediate hotline/delay hotline
<b>Hotline Number</b>	digits
<b>Limit Max time (s)</b>	empty
<b>Do Not Disturb</b>	Yes/no
<b>DND Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Do Not Disturb Whitelist</b>	Empty/digits
<b>User Password</b>	Alphanumeric characters.
<b>First Name</b>	Alphanumeric without special characters.



<b>Last Name</b>	Alphanumeric without special characters.
<b>Email Address</b>	Email address
<b>Language</b>	Default/en/zh
<b>Phone Number</b>	Digits
<b>Call-Barging Monitor</b>	Extensions allowed to call barging
<b>Seamless Transfer Members</b>	Extensions allowed to seamless transfer

The CSV file should contain all the above fields, if one of them is missing or empty, the UCM630xA will display the following error message for missing fields.

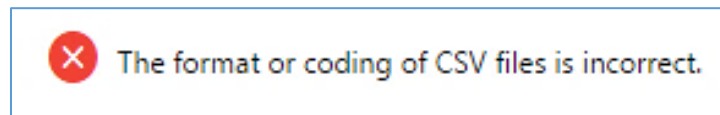


Figure 83: Import Error

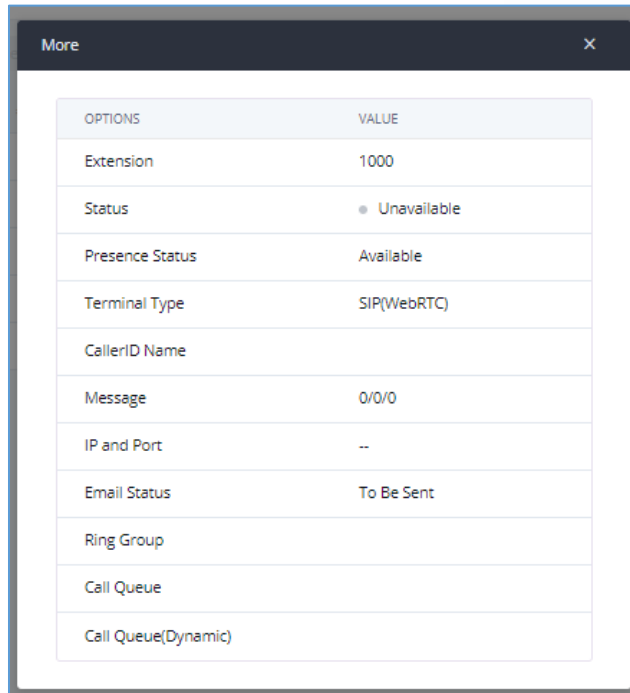
## Extension Details

Users can click on an extension number in the *Extensions* list page and quickly view information about it such as:

- **Extension:** Shows the Extension number.
- **Status:** Shows the status of the extension.
- **Presence status:** Indicates the Presence Status of this extension.
- **Terminal Type:** Shows the Type of the terminal using this extension (SIP, FXS...etc.).
- **Caller ID Name:** Reveals the Caller ID Name configured on the extension.
- **Messages:** Shows the messages stats.
- **IP and Port:** The IP address and the ports of the device using the extension.
- **Email status:** Show the Email status (sent, to be sent...etc.).
- **Ring Group:** Indicates the ring groups that this extension belongs to.
- **Call Queue:** Indicates the Cal Queues that this extension belongs to.
- **Call Queue (Dynamic):** Indicates the Call Queues that this extension belongs to as a dynamic agent.







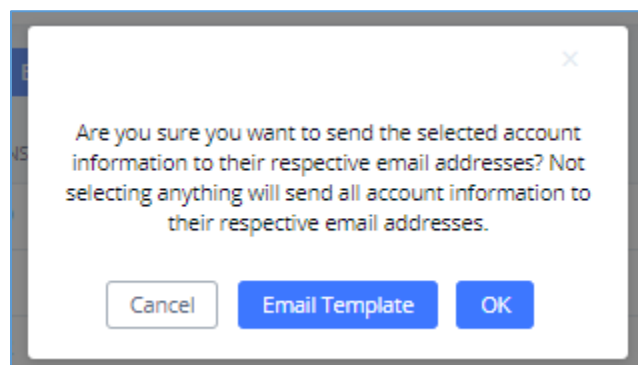
OPTIONS	VALUE
Extension	1000
Status	<input checked="" type="radio"/> Unavailable
Presence Status	Available
Terminal Type	SIP(WebRTC)
CallerID Name	
Message	0/0/0
IP and Port	--
Email Status	To Be Sent
Ring Group	
Call Queue	
Call Queue(Dynamic)	

**Figure 84: Extension Details**

## E-mail Notification

Once the extensions are created with Email addresses, the PBX administrator can click on button “E-mail Notification” to send the account registration and configuration information to the user. Please make sure Email setting under Web GUI→**System Settings**→**Email Settings** is properly configured and tested on the UCM630xA before using “E-mail Notification”.

When click on ”More” > “E-mail Notification” 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 85: E-mail Notification - Prompt Information**



The user will receive Email including account registration information as well as the GSWave Settings with the QR code:

General Settings	
Server Address	<a href="https://192.168.5.147:5060">192.168.5.147:5060</a>
Account Name	Mia
SIP User ID	1000
Authenticate ID	1000
Authenticate Password	pas1

Figure 86: Account Registration Information

GSWave Settings	
Login URL	<a href="https://192.168.5.147:8090/#/">https://192.168.5.147:8090/#/</a>
Login URL for Public	<a href="https://c074ad0a8c94-10671.b.gdms.cloud/#/">https://c074ad0a8c94-10671.b.gdms.cloud/#/</a>
Login Name	1000
Login Password	pas1



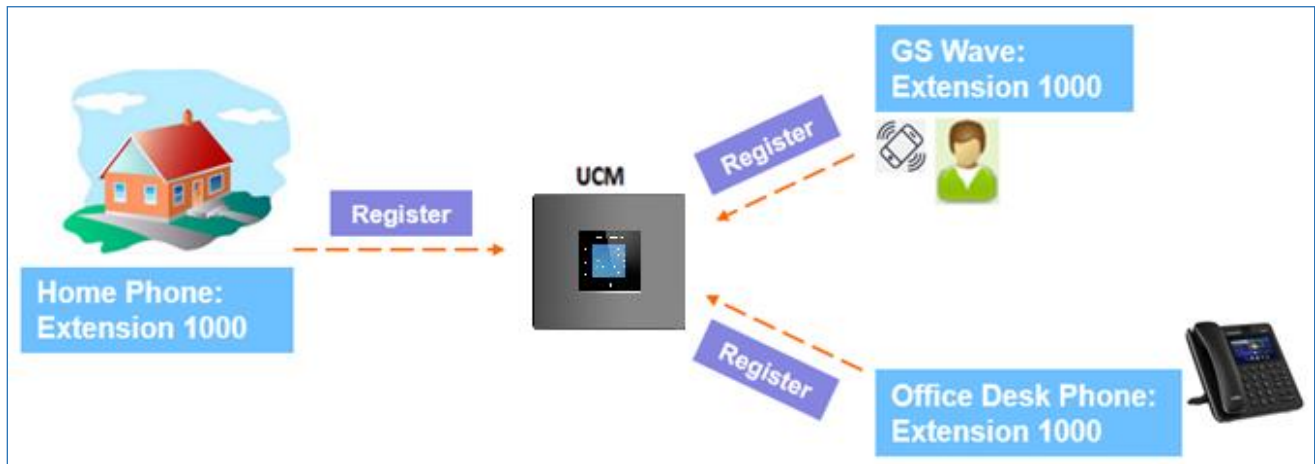
Use Web App to scan qr code and log in

Figure 87: GSWave Settings and QR Code



## Multiple Registrations per Extension

UCM630xA supports multiple registrations per extension so that users can use the same extension on devices in different locations.



**Figure 88: Multiple Registrations per Extension**

This feature can be enabled by configuring option “Concurrent Registrations” under Web GUI → **Extension/Trunk** → **Edit Extension**. The default value is set to 1 for security purpose. Maximum is 10.

The screenshot shows the 'Edit Extension: 1000' web interface. The 'Basic Settings' tab is active. Under the 'User Settings' section, the 'Concurrent Registrations' field is set to 3. Other visible fields include Extension (1000), Permission (Internal), CallerID Number (1000), SIP/IAX Password (masked), Voicemail (Local Voicemail), Skip Voicemail Password (unchecked), Verification (unchecked), Keep Voicemail after Emailing (Default), Keep-alive Frequency (60), Enable SCA (unchecked), First Name, Last Name, User Password (masked), Language (Default), and Mobile Phone Number.

**Figure 89: Extension - Concurrent Registration**



## SMS Message Support

The UCM630xA provides built-in SIP SMS message support. For SIP end devices such as Grandstream GXP or GXV phones that supports SIP message, after an UCM630xA 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.



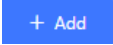


Figure 90: SMS Message Support

## EXTENSION GROUPS

The UCM630xA extension group feature allows users to assign and categorize extensions in different groups to better manage the configurations on the UCM630xA. For example, when configuring "Enable Filter on Source Caller ID", users could select a group instead of each person's extension to assign. This feature simplifies the configuration process and helps manage and categorize the extensions for business environment.

### Configure Extension Groups

Extension group can be configured via Web GUI → **Extension/Trunk** → **Extension Groups**.

- Click on  to create a new extension group.
- Click on  to edit the extension group.
- Click on  to delete the extension group.

Select extensions from the list on the left side to the right side.

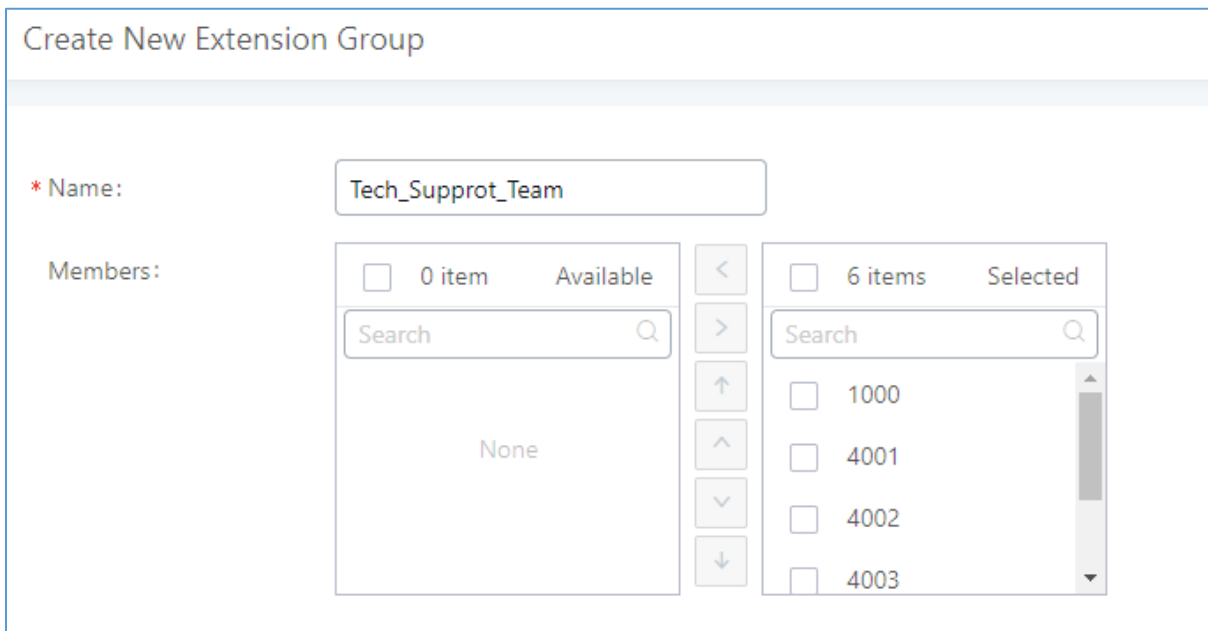


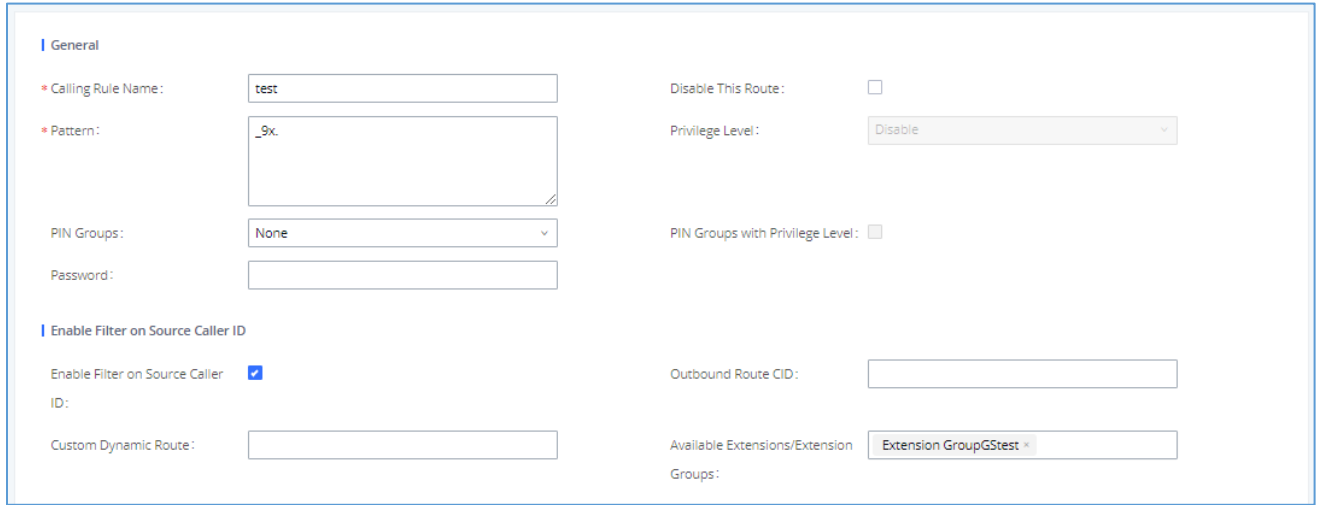
Figure 91: Edit Extension Group



Click on     in order to change the ringing priority of the members selected on the group.

## Using Extension Groups

Here is an example where the extension group can be used. Go to Web GUI→**Extension/Trunk**→**Outbound Routes** and select "Enable Filter on Source Caller ID". Both single extensions and extension groups will show up for users to select.



The screenshot shows the configuration interface for an Outbound Route. It is divided into two main sections: 'General' and 'Enable Filter on Source Caller ID'.



- General Section:**
  - Calling Rule Name:
  - Pattern:
  - PIN Groups:
  - Password:
  - Disable This Route:
  - Privilege Level:
  - PIN Groups with Privilege Level:
- Enable Filter on Source Caller ID Section:**
  - Enable Filter on Source Caller ID:
  - ID:
  - Custom Dynamic Route:
  - Outbound Route CID:
  - Available Extensions/Extension Groups:

**Figure 92: Select Extension Group in Outbound Route**



## ANALOG TRUNKS

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

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

### Analog Trunk Configuration

The analog trunk options are listed in the table below.

**Table 55: Analog Trunk Configuration Parameters**

<b>FXO Port</b>	Select the channel for the analog trunk. <ul style="list-style-type: none"> <li>• UCM6300A: 1 channel</li> <li>• UCM6302A: 2 channels</li> <li>• UCM6304A: 4 channels</li> <li>• UCM6308A: 8 channels</li> </ul>
<b>Trunk Name</b>	Specify a unique label to identify the trunk when listed in outbound rules, incoming rules and etc.
<b>Advanced Options</b>	
<b>SLA Mode</b>	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.
<b>Barge Allowed</b>	The barge option specifies whether other stations can 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.
<b>Hold Access</b>	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.
<b>Enable Polarity Reversal</b>	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".



<b>Polarity on Answer Delay</b>	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.
<b>Current Disconnect Threshold (ms)</b>	This is the periodic time (in ms) that the UCM630xA will use to check on a voltage drop in the line. The default setting is 200. The valid range is 50 to 3000.
<b>Ring Timeout</b>	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 UCM630xA considers a non-ringing line with Hangup activity. The default setting is 8000.
<b>RX Gain</b>	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.
<b>TX Gain</b>	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.
<b>Use CallerID</b>	Configure to enable CallerID detection. The default setting is “Yes”.
<b>Caller ID Scheme</b>	<p>Select the Caller ID scheme for this trunk.</p> <ul style="list-style-type: none"> <li>• Bellcore/Telcordia.</li> <li>• ETSI-FSK During Ringing</li> <li>• ETSI-FSK Prior to Ringing with DTAS</li> <li>• ETSI-FSK Prior to Ringing with LR</li> <li>• ETSI-FSK Prior to Ringing with RP</li> <li>• ETSI-DTMF During Ringing</li> <li>• ETSI-DTMF Prior to Ringing with DTAS</li> <li>• ETSI-DTMF Prior to Ringing with LR</li> <li>• ETSI-DTMF Prior to Ringing with RP</li> <li>• SIN 227-BT</li> <li>• NTT Japan</li> <li>• Auto Detect</li> </ul> <p>If you are not sure which scheme to choose, please select “Auto Detect”. The default setting is “Bellcore/Telcordia”.</p>
<b>Fax Mode</b>	<p>Configures how faxes to this extension will be handled.</p> <ul style="list-style-type: none"> <li>• <b>None:</b> Faxes will not be processed.</li> <li>• <b>Fax Gateway:</b> Faxes to this extension will be processed and converted from T.30 to T.38 or vice-versa. FXS/FXO ports only.</li> </ul> <p>The default setting is None.</p>





<b>FXO Dial Delay (ms)</b>	Configure the time interval between off-hook and first dialed digit for outbound calls.
<b>Auto Record</b>	Enable automatic recording for the calls using this trunk. The default setting is disabled. The recording files can be accessed under Web GUI→ <b>CDR</b> → <b>Recording Files</b> .
<b>Disable This Trunk</b>	If selected, the trunk will be disabled and incoming/Outgoing calls via this trunk will not be possible.
<b>DAHDI Out Line Selection</b>	<p>This is to implement analog trunk outbound line selection strategy. Three options are available:</p> <ul style="list-style-type: none"> <li>• <b>Ascend</b> 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 if UCM6302A is used would be port 1→port 2→. Every time it will start with port 1 (if it is idle).</li> <li>• <b>Poll</b> 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→1→2→1→2→..., following the last port being used in case UCM6302A is used.</li> <li>• <b>Descend</b> 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 if UCM6302A is used would be port 2→port 1. Every time it will start with port 2 (if it is idle).</li> </ul> <p>The default setting is “Ascend” mode.</p>
<b>Echo Cancellation Mode</b>	<p>The Non-Linear Processing (NLP) in echo cancellation helps to remove/suppress residual echo components that could not be removed by the LEC (Line Echo Canceller). Following modes are supported:</p> <ul style="list-style-type: none"> <li>• <b>Default:</b> The NLP limits the signal level to the background noise level when active, and the background noise level adjustment is low.</li> <li>• <b>High Noise Level Adjustment:</b> The NLP limits the signal level to the background noise level when active, and the background noise level adjustment is high.</li> <li>• <b>Noise Masking:</b> The NLP sends sign noise when active, and the background noise level adjustment is high.</li> <li>• <b>White Noise Injection:</b> The NLP injects white noise when active. The level corresponds to the background noise level at Sin, and the background noise level adjustment is high.</li> </ul>



<b>Direct Callback</b>	<p>Allows external numbers the option to get directed to the extension that last called them.</p> <p>For Example: User 2002 has dialed external number 061234575 but they were not reachable, once they have received missed call notification on their phone, they would mostly call back the number, if the option "Direct Callback" is enabled then they will be directly bridged to user 2002 regardless of the configured inbound destination.</p>
<b>Tone Settings</b>	
<b>Busy Detection</b>	<p>Busy Detection is used to detect far end Hangup or for detecting busy signal. The default setting is "Yes".</p>
<b>Busy Tone Count</b>	<p>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.</p>
<b>Congestion Detection</b>	<p>Congestion detection is used to detect far end congestion signal. The default setting is "Yes".</p>
<b>Congestion Count</b>	<p>If "Congestion Detection" is enabled, users can specify the number of congestion tones to wait for. The default setting is 2.</p>
<b>Tone Country</b>	<p>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)".</p>
<b>Busy Tone</b>	<p><b>Syntax:</b>  <code>f1=val[@level],[f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]];</code>          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.  <b>Default value:</b>  <code>f1=480@-50,f2=620@-50,c=500/500</code></p>
<b>Congestion Tone</b>	<p><b>Syntax:</b>  <code>f1=val[@level],[f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]];</code>          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].</p>

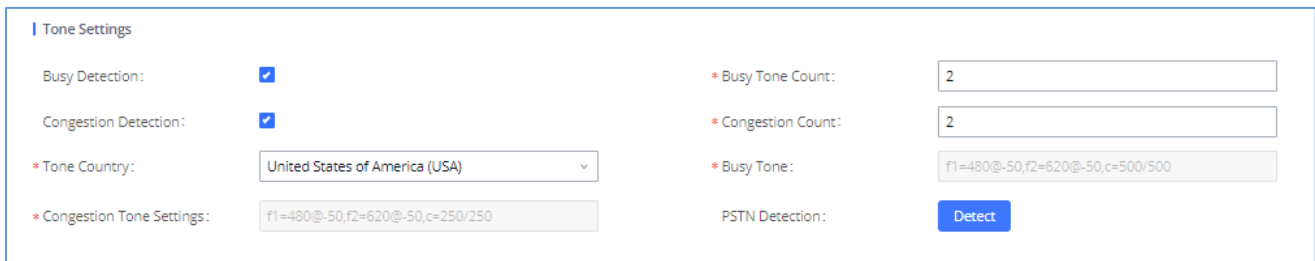


	Select Tone Country "Custom" to manually configure Busy Tone value. <b>Default value:</b> f1=480@-50,f2=620@-50,c=250/250
<b>PSTN Detection</b>	Click on "Detect" to detect the busy tone, Polarity Reversal and Current Disconnect by PSTN. Before the detecting, please make sure there are more than one channel configured and working properly. If the detection has busy tone, the "Tone Country" option will be set as "Custom".

## PSTN Detection

The UCM630xA 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 UCM630xA settings.

1. Go to UCM630xA Web GUI → **Extension/Trunk** → **Analog Trunks** page.
2. Click to edit the analog trunk created for the FXO port.
3. In the 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.



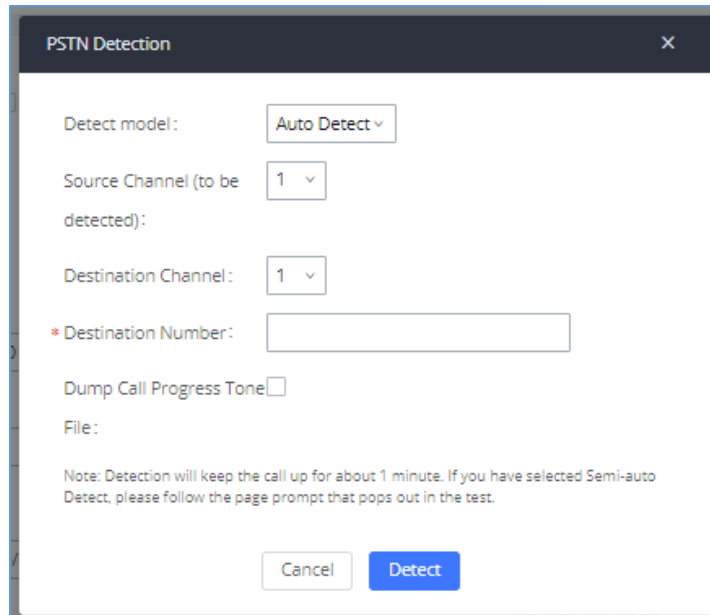
The screenshot shows the 'Tone Settings' configuration page. It includes the following fields and controls:

- Busy Detection:**
- Congestion Detection:**
- \* Tone Country:** United States of America (USA) (dropdown menu)
- \* Congestion Tone Settings:** f1=480@-50,f2=620@-50,c=250/250
- \* Busy Tone Count:** 2
- \* Congestion Count:** 2
- \* Busy Tone:** f1=480@-50,f2=620@-50,c=500/500
- PSTN Detection:**

**Figure 93: UCM630xA FXO Tone Settings**

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

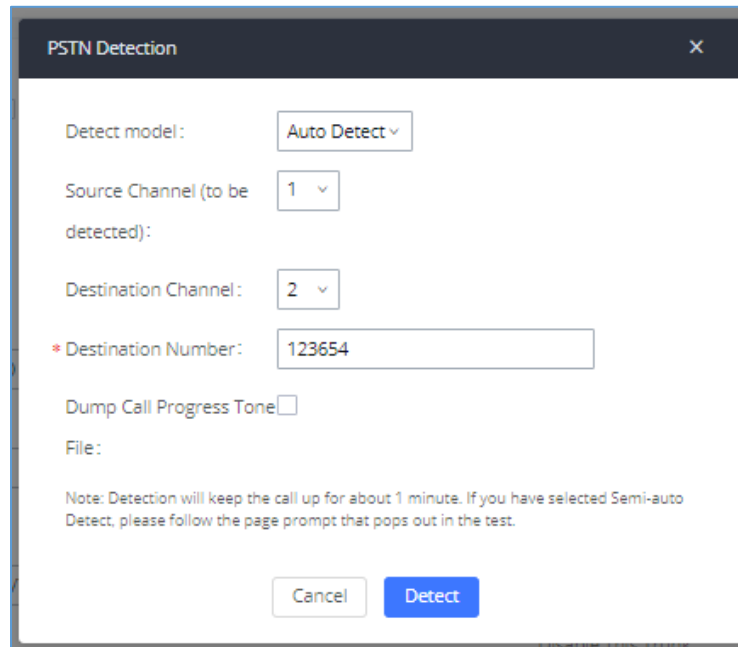




**Figure 94: UCM630xA PSTN Detection**

- 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.





**PSTN Detection** [X]

Detect model: Auto Detect ▾

Source Channel (to be detected): 1 ▾

Destination Channel: 2 ▾

\* Destination Number: 123654

Dump Call Progress Tone

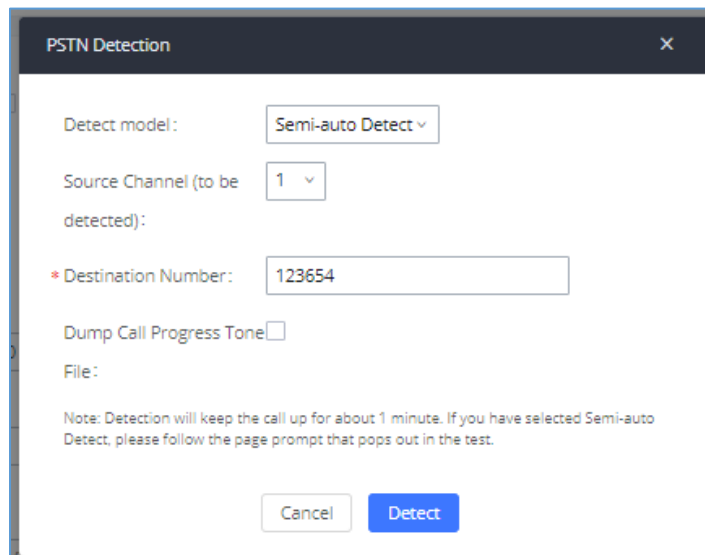
File:

Note: Detection will keep the call up for about 1 minute. If you have selected Semi-auto Detect, please follow the page prompt that pops out in the test.

Cancel Detect

**Figure 95: UCM630xA PSTN Detection: Auto Detect**

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



**PSTN Detection** [X]

Detect model: Semi-auto Detect ▾

Source Channel (to be detected): 1 ▾

\* Destination Number: 123654

Dump Call Progress Tone

File:

Note: Detection will keep the call up for about 1 minute. If you have selected Semi-auto Detect, please follow the page prompt that pops out in the test.

Cancel Detect

**Figure 96: UCM630xA 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 UCM630xA 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 UCM630xA settings.

**Table 56: PSTN Detection for Analog Trunk**

<b>Detect Model</b>	<p>Select "Auto Detect" or "Semi-auto Detect" for PSTN detection.</p> <ul style="list-style-type: none"> <li>• <b>Auto Detect</b> Please make sure two or more channels are connected to the UCM630xA 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 UCM630xA will control the call to be established and hang up between caller and callee to finish the detection.</li> <li>• <b>Semi-auto Detect</b> Semi-auto detection requires answering or hanging up the call manually. Please make sure one channel is connected to the UCM630xA 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.</li> </ul> <p>The default setting is "Auto Detect".</p>
<b>Source Channel</b>	Select the channel to be detected.
<b>Destination Channel</b>	Select the channel to help detect when "Auto Detect" is used.
<b>Destination Number</b>	Configure the number to be called to help the detection.
<b>Dump Call Progress Tone File</b>	Choose whether to save the calling tone file, it is not checked by default.

---

 **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 Trunk Configuration

VoIP trunks can be configured in UCM630xA under Web GUI→**Extension/Trunk**→**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 "Add SIP Trunk" or "Add 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:

[https://www.grandstream.com/hubfs/Product\\_Documentation/ucm6xxx\\_sip\\_trunk\\_guide.pdf](https://www.grandstream.com/hubfs/Product_Documentation/ucm6xxx_sip_trunk_guide.pdf)

The VoIP trunk options are listed in the table below.

**Table 57: Create New SIP Trunk**

<b>Type</b>	Select the VoIP trunk type. <ul style="list-style-type: none"> <li>• Peer SIP Trunk</li> <li>• Register SIP Trunk</li> </ul>
<b>Provider Name</b>	Configure a unique label (up to 64 character) to identify this trunk when listed in outbound rules, inbound rules etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider's server of the trunk.
<b>Keep Original CID</b>	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.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".



<b>NAT</b>	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.
<b>Disable This Trunk</b>	If checked, the trunk will be disabled. <b>Note:</b> If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
<b>TEL URI</b>	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.
<b>Caller ID Number</b>	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.  <b>Important Note:</b> When making outgoing calls, the following priority order rule will be used to determine which CallerID will be set before sending out the call:  From user (Register Trunk Only) → CID from inbound call ( <b>Keep Original CID Enabled</b> ) → Trunk Username/CallerID ( <b>Keep Trunk CID Enabled</b> ) → DOD → Extension CallerID Number → Trunk Username/CallerID ( <b>Keep Trunk CID Disabled</b> ) → Global Outbound CID.
<b>CallerID Name</b>	Configure the new name of the caller when the extension has no CallerID Name configured.
<b>Need Registration</b>	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.
<b>Allow outgoing calls if registration fails</b>	Uncheck to block outgoing calls if registration fails. If "Need Registration" option is unchecked, this setting will be ignored.
<b>Username</b>	Enter the username to register to the trunk from the provider when "Register SIP Trunk" type is selected.
<b>Password</b>	Enter the password to register to the trunk from the provider when "Register SIP Trunk" is selected.
<b>Auth ID</b>	Enter the Authentication ID for "Register SIP Trunk" type.
<b>Auto Record</b>	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→ <b>CDR</b> → <b>Recording Files</b> .





<b>Direct Callback</b>	<p>Allows external numbers the option to get directed to the extension that last called them.</p> <p>For Example: User 2002 has dialed external number 061234575 but they were not reachable, once they have received missed call notification on their phone, they would mostly call back the number, if the option “Direct Callback” is enabled then they will be directly bridged to user 2002 regardless of the configured inbound destination.</p>
<b>RemoteConnect Mode</b>	<p>If enabled, the RemoteConnect related parameters will be set synchronously. Please make sure the trunk host is allocated by GDMS or it supports TLS.</p>

**Table 58: SIP Register Trunk Configuration Parameters**

Basic Settings	
<b>Provider Name</b>	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider’s server of the trunk.
<b>Transport</b>	<p>Configure the SIP transport protocol to be used in this trunk. The default setting is "UDP".</p> <ul style="list-style-type: none"> <li>• UDP</li> <li>• TCP</li> <li>• TLS</li> </ul>
<b>SIP URI Scheme When Using TLS</b>	When TLS is selected as Transport for register trunk, users can select between SIP and SIPS URI scheme
<b>Keep Original CID</b>	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.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension’s CID when the extension has CID configured. The default setting is “No”.
<b>NAT</b>	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 is related to NAT configuration or SIP/RTP port configuration on the firewall.
<b>Disable This Trunk</b>	<p>If selected, the trunk will be disabled.</p> <p><b>Note:</b> If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.</p>
<b>TEL URI</b>	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.



<b>Need Registration</b>	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.
<b>Allow outgoing calls if registration failure</b>	If enabled outgoing calls even if the registration to this trunk fail will still be able to go through. Note that if we uncheck "Need Registration" option, this option will be ignored.
<b>CallerID Name</b>	Configure the new name of the caller when the extension has no CallerID Name configured.
<b>From Domain</b>	Configure the actual domain name where the extension comes from. This can be used to override the "From" Header. For example, "trunk.UCM630xA.provider.com" is the From Domain in From Header: sip:1234567@trunk.UCM630xA.provider.com.
<b>From User</b>	Configure the actual username 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.UCM630xA.provider.com.
<b>Username</b>	Enter the username to register to the trunk from the provider when "Register SIP Trunk" type is selected.
<b>Password</b>	Enter the password to register to the trunk when "Register SIP Trunk" is selected.
<b>Auth ID</b>	Enter the Authentication ID for "Register SIP Trunk" type.
<b>Auth Trunk</b>	If enabled, the UCM will send 401 response to the incoming call to authenticate the trunk.
<b>Auto Record</b>	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→ <b>CDR</b> → <b>Recording Files</b> .
<b>RemoteConnect Mode</b>	If enabled, the RemoteConnect related parameters will be set synchronously. Please make the trunk host is allocated by GDMS or it supports TLS.
<b>Direct Callback</b>	Allows external numbers the option to get directed to the extension that last called them. For Example: User 2002 has dialed external number 061234575 but they were not reachable, once they have received missed call notification on their phone, they would mostly call back the number, if the option "Direct Callback" is enabled then they will be directly bridged to user 2002 regardless of the configured inbound destination.



Advanced Settings	
<b>Codec Preference</b>	Select 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.265, H.263, H.263p and VP8.
<b>Send PPI Header</b>	If enabled, the SIP INVITE message sent to the trunk will contain PPI (P-Preferred-Identity) header. The default setting is “No”. <b>Note:</b> “Send PPI Header” and “Send PAI Header” cannot be enabled at the same time. Only one of the two headers can be contained in SIP INVITE message.
<b>PPI Mode</b>	<b>Default</b> – Include the trunk’s preferred CID (configured in <i>Basic Settings</i> ) in the PPI Header. <b>Original CID</b> – Include the original CID in the PPI Header. <b>DOD Number</b> – Include the trunk’s DOD number in the PPI Header. If no DOD number has been set, the trunk’s preferred CID will be used.
<b>Send PAI Header</b>	If enabled, the SIP INVITE message sent to the trunk will contain PAI (P-Asserted-Identity) header including configured PAI Header. The default setting is “No”. <b>Note:</b> “Send PPI Header” and “Send PAI Header” cannot be enabled at the same time. Only one of the two headers can be contained in the SIP INVITE message.
<b>PAI Header</b>	If “Send PAI Header” is enabled and “PAI Header” is configured as “123456” for instance, the PAI header in the SIP message sent from the UCM will contain “123456”. If “Send PAI Header” is enabled and “PAI Header” is configured as “empty”, the PAI header in the SIP message sent from the UCM will contain the original CID. <b>Note:</b> “Send PAI Header” needs to be enabled to use this feature
<b>Send Anonymous</b>	If checked, the "From" header in outgoing INVITE message will be set to anonymous.
<b>DOD As From Name</b>	If enabled and "From User" is configured, the INVITE's From header will contain the DOD number.
<b>Passthrough PAI Header</b>	If checked and option "Send PAI Header" not checked, the PAI header will be passthrough from one side to the other side.
<b>Send PANI Header</b>	If checked, the INVITE and REGISTER sent to the trunk will contain P-Access-Network-Info header.
<b>Access Network Info</b>	The access network information in the P-Access-Network-Info header.



<b>Send Anonymous</b>	If checked, the "From" header in outgoing INVITE message will be set to anonymous.
<b>Outbound Proxy Support</b>	Select to enable outbound proxy in this trunk. The default setting is "No".
<b>Outbound Proxy</b>	When outbound proxy support is enabled, enter the IP address or URL of the outbound proxy.
<b>Remove OBP from Route</b>	It is used to set if the phone system will remove outbound proxy URI from the route header. If is set to "Yes", it will remove the route header from SIP requests. The default setting is "No".
<b>DID Mode</b>	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".
<b>GIN Registration</b>	If enabled, the UCM will send a GIN REGISTER (generate implicit numbers).
<b>DTMF Mode</b>	Configure the default DTMF mode when sending DTMF on this trunk. <ul style="list-style-type: none"> <li>• <b>Default:</b> The global setting of DTMF mode will be used. The global setting for DTMF Mode setting is under Web GUI→<b>PBX Settings</b>→<b>SIP Settings</b>→<b>ToS</b>.</li> <li>• <b>RFC4733:</b> Send DTMF using RFC4733.</li> <li>• <b>Info:</b> Send DTMF using SIP INFO message.</li> <li>• <b>Inband:</b> Send DTMF using inband audio. This requires 64-bit codec, i.e., PCMU and PCMA.</li> <li>• <b>Auto:</b> Send DTMF using RFC4733 if offered. Otherwise, inband will be used.</li> </ul>
<b>Enable Heartbeat Detection</b>	If enabled, the UCM630xA will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
<b>Heartbeat Frequency</b>	When "Enable Heartbeat Detection" 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.
<b>The Maximum Number of Call Lines</b>	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limit.
<b>Packet Loss Retransmission</b>	Configure to enable Packet Loss Retransmission.
<b>Audio FEC</b>	Configure to enable Forward Error Correction (FEC) for audio.
<b>Video FEC</b>	Configure to enable Forward Error Correction (FEC) for video.
<b>ICE support</b>	Toggles ICE support. For peer trunks, ICE support will need to be enabled on the other end.



<b>FECC</b>	Configure to enable Far-end Camera Control
<b>SRTP</b>	Enable SRTP for the VoIP trunk. The default setting is "No".
<b>Enable T.38 UDPTL</b>	Enable or disable T.38 UDPTL support.
<b>STIR/SHAKEN</b>	Block disturbance calls, this function needs to be supported by the opposite end.
<b>CC Settings</b>	
<b>Enable CC</b>	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.
<b>CC Max Agents</b>	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.
<b>CC Max Monitors</b>	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 59: SIP Peer Trunk Configuration Parameters**

<b>Basic Settings</b>	
<b>Provider Name</b>	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider's server of the trunk.
<b>Auto Record</b>	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→ <b>CDR</b> → <b>Recording Files</b> .
<b>Keep Original CID</b>	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.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
<b>NAT</b>	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 is related to NAT configuration or SIP/RTP port configuration on the firewall.
<b>Disable This Trunk</b>	If selected, the trunk will be disabled. <b>Note:</b> If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
<b>TEL URI</b>	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.
<b>Caller ID Number</b>	<p>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.</p> <p><b>Important Note:</b> When making outgoing calls, the following priority order rule will be used to determine which CallerID will be set before sending out the call :</p> <ul style="list-style-type: none"> <li>• CID from inbound call (<b>Keep Original CID</b> Enabled) → Trunk Username/CallerID (<b>Keep Trunk CID</b> Enabled) → DOD → Extension CallerID Number → Trunk Username/CallerID (<b>Keep Trunk CID</b> Disabled) → Global Outbound CID.</li> </ul>
<b>CallerID Name</b>	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
<b>Transport</b>	<p>Configure the SIP transport protocol to be used in this trunk. The default setting is "UDP".</p> <ul style="list-style-type: none"> <li>• UDP</li> <li>• TCP</li> <li>• TLS</li> </ul>
<b>RemoteConnect Mode</b>	If enabled, the RemoteConnect related parameters will be set synchronously. Please make the trunk host is allocated by GDMS or it supports TLS.
<b>Direct Callback</b>	<p>Allows external numbers the option to get directed to the extension that last called them.</p> <p>For Example: User 2002 has dialed external number 061234575 but they were not reachable, once they have received missed call notification on their phone, they would mostly call back the number, if the option "Direct Callback" is enabled then they will be directly bridged to user 2002 regardless of the configured inbound destination.</p>
<b>Advanced Settings</b>	
<b>Codec Preference</b>	Select 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.265, H.263, H.263p and VP8.
<b>Send PPI Header</b>	If checked, the invite message sent to trunks will contain PPI (P-Preferred-Identity) Header.
<b>Send PAI Header</b>	If checked, the INVITE, 18x and 200 SIP messages sent to trunks will contain P-Asserted-Identity (PAI) header. It is not possible to send both PPI and PAI headers.
<b>Passthrough PAI Header</b>	If enabled and "Send PAI Header" is disabled, PAI headers will be preserved as calls pass through the UCM.



<b>Send PANI Header</b>	If checked, the INVITE sent to the trunk will contain P-Access-Network-Info header.
<b>Send Anonymous</b>	If checked, the "From" header in outgoing INVITE message will be set to anonymous.
<b>DID Mode</b>	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".
<b>DTMF Mode</b>	<p>Configure the default DTMF mode when sending DTMF on this trunk.</p> <ul style="list-style-type: none"> <li>• <b>Default:</b> The global setting of DTMF mode will be used. The global setting for DTMF Mode setting is under Web GUI→<b>PBX Settings</b>→<b>SIP Settings</b>→<b>ToS</b>.</li> <li>• <b>RFC4733:</b> Send DTMF using RFC4733.</li> <li>• <b>Info:</b> Send DTMF using SIP INFO message.</li> <li>• <b>Inband:</b> Send DTMF using inband audio. This requires 64-bit codec, i.e., PCMU and PCMA.</li> <li>• <b>Auto:</b> Send DTMF using RFC4733 if offered. Otherwise, inband is used.</li> </ul>
<b>Enable Heartbeat Detection</b>	If enabled, the UCM630xA will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
<b>Heartbeat Frequency</b>	When "Enable Heartbeat Detection" 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.
<b>Maximum Number of Call Lines</b>	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limit.
<b>Packet Loss Retransmission</b>	Configure to enable Packet Loss Retransmission.
<b>Audio FEC</b>	Configure to enable Forward Error Correction (FEC) for audio.
<b>Video FEC</b>	Configure to enable Forward Error Correction (FEC) for video.
<b>ICE support</b>	Toggles ICE support. For peer trunks, ICE support will need to be enabled on the other end.
<b>FECC</b>	Configure to enable Far-end Camera Control
<b>SRTP</b>	Enable SRTP for the VoIP trunk. The default setting is "No".
<b>IPVT Mode</b>	Configures the UCM to be used exclusively for IPVT. <b>Warning:</b> This will lock out certain UCM features.
<b>Sync LDAP Enable</b>	Automatically sync local LDAP phonebooks to a remote peer (SIP peer trunk only). To ensure successful syncing, the remote peer must also enable this service and set the same password as the local UCM. Port 873 is used by default.



<b>Sync LDAP Password</b>	Password used for LDAP phonebook encryption and decryption. The password must be the same for both peers to ensure successful syncing.
<b>LDAP Outbound Rule</b>	Specify an outbound rule for LDAP sync feature. The UCM630xA will automatically modify the remote contacts by adding prefix parsed from this rule.
<b>LDAP Dialed Prefix</b>	Specify the prefix for LDAP sync feature. The UCM630xA will automatically modify the remote contacts by adding this prefix.
<b>LDAP Last Sync Date</b>	The last successful sync date.
<b>Enable T.38 UDPTL</b>	Enable or disable T.38 UDPTL support.
<b>STIR/SHAKEN</b>	Block disturbance calls, this function needs to be supported by the opposite end.
<b>CC Settings</b>	
<b>Enable CC</b>	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.
<b>CC Max Agents</b>	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 can make. The minimum value is 1.
<b>CC Max Monitors</b>	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 60: Create New IAX Trunk**

<b>Type</b>	Select the VoIP trunk type. <ul style="list-style-type: none"> <li>Peer IAX Trunk</li> <li>Register IAX Trunk</li> </ul>
<b>Provider Name</b>	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider's server of the trunk.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
<b>Username</b>	Enter the username to register to the trunk from the provider when "Register IAX Trunk" type is selected.
<b>Password</b>	Enter the password to register to the trunk from the provider when "Register IAX Trunk" type is selected.
<b>Disable This Trunk</b>	If selected, the trunk will be disabled.
<b>Caller ID Number</b>	Number that the trunk will try to use when making outbound calls. CID priority from highest to lowest is as follows: <b>From User</b> (register trunk only) >>> <b>Inbound Call CID</b> (if Keep Original CID is





	<p>enabled and the call is originally from another trunk) &gt;&gt;&gt; <b>Trunk CID</b> (Keep Trunk CID enabled) &gt;&gt;&gt; <b>DOD CID</b> &gt;&gt;&gt; <b>Extension CID</b> &gt;&gt;&gt; <b>Register Trunk Username</b> (Keep Trunk CID disabled) &gt;&gt;&gt; <b>Global Outbound CID</b>.</p> <p>Note 1: Certain providers may ignore this CID.</p> <p>Note 2: If this CID contains asterisk (*), call recordings from this trunk might be lost when saving them to NAS storage.</p>
<b>CallerID Name</b>	Configure the new name of the caller when the extension has no CallerID Name configured.

**Table 61: IAX Register Trunk Configuration Parameters**

Basic Settings	
<b>Provider Name</b>	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider's server of the trunk.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
<b>Disable This Trunk</b>	If selected, the trunk will be disabled.
<b>Caller ID</b>	<p>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.</p> <p><b>Important Note:</b> When making outgoing calls, the following priority order rule will be used to determine which CallerID will be set before sending out the call :</p> <p>From user (Register Trunk Only) → CID from inbound call (<b>Keep Original CID</b> Enabled) → Trunk Username/CallerID (<b>Keep Trunk CID</b> Enabled) → DOD → Extension CallerID Number → Trunk Username/CallerID (<b>Keep Trunk CID</b> Disabled) → Global Outbound CID.</p>
<b>CallerID Name</b>	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
<b>Username</b>	Enter the username to register to the trunk from the provider.
<b>Password</b>	Enter the password to register to the trunk from the provider.
Advanced Settings	
<b>Codec Preference</b>	Select 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.265, H.263, H.263p and VP8.



<b>Enable Heartbeat Detection</b>	If enabled, the UCM630xA will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
<b>Heartbeat Frequency</b>	When "Enable Heartbeat Detection" 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.
<b>Maximum Number of Call Lines</b>	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limited.

**Table 62: IAX Peer Trunk Configuration Parameters**

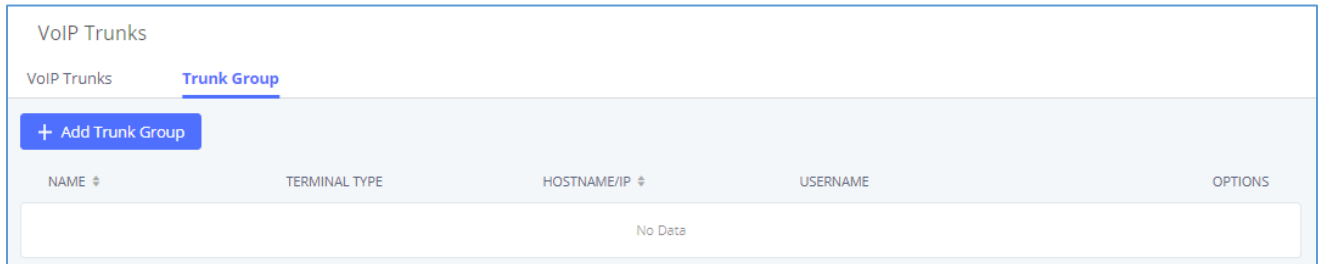
Basic Settings	
<b>Provider Name</b>	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider's server of the trunk.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
<b>Disable This Trunk</b>	If selected, the trunk will be disabled.
<b>Caller ID</b>	<p>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.</p> <p><b>Important Note:</b> When making outgoing calls, the following priority order rule will be used to determine which CallerID will be set before sending out the call :</p> <p>CID from inbound call (<b>Keep Original CID</b> Enabled) → Trunk Username/CallerID (<b>Keep Trunk CID</b> Enabled) → DOD → Extension CallerID Number → Trunk Username/CallerID (<b>Keep Trunk CID</b> Disabled) → Global Outbound CID..</p>
<b>CallerID Name</b>	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
Advanced Settings	
<b>Codec Preference</b>	Select 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.265, H.263, H.263p and VP8.
<b>Enable Heartbeat Detection</b>	If enabled, the UCM630xA will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
<b>Heartbeat Frequency</b>	When "Enable Heartbeat Detection" 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.




<b>Maximum Number of Call Lines</b>	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limited.
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## Trunk Groups

Users can create VoIP Trunk Groups to register and easily apply the same settings on multiple accounts within the same SIP server. This can drastically reduce the amount of time needed to manage accounts for the same server and improve the overall cleanliness of the web UI.



**Figure 97: Trunk Group**

Once creating the new trunk group and configuring the SIP settings, users can add multiple accounts within the configured SIP server by pressing  button and configuring the username, password, and authentication ID fields.



**Create New Trunk Group**

Type:

\* Provider Name:

\* Host Name:

Transport:

Keep Original CID:

Keep Trunk CID:

NAT:

Disable This Trunk:

TEL URI:

Need Registration:

Allow outgoing calls if registration fails:

CallerID Name:

\* Username:  /  /  -

[Add Username](#) +

AuthTrunk:

Auto Record:

Direct Callback:

RemoteConnect Mode:

**Figure 98: Trunk Group Configuration**

## Direct Outward Dialing (DOD)

The UCM630xA 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:

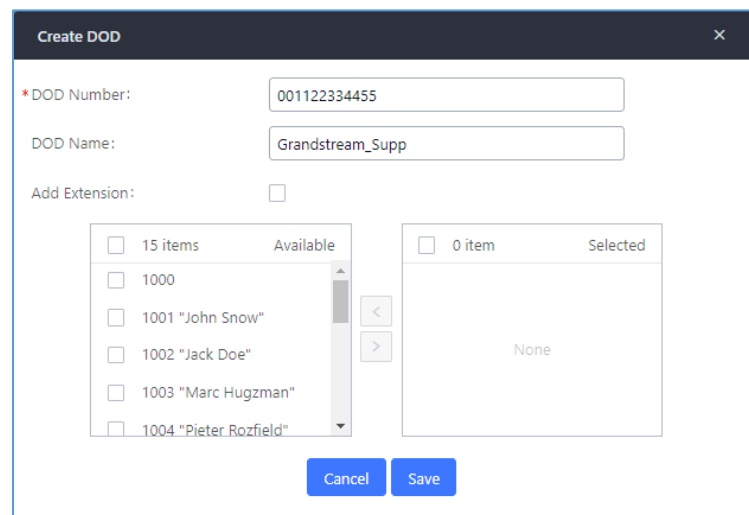
Company ABC has a SIP trunk. This SIP trunk has 4 DIDs associated to it. The main number of the office is routed to an auto attendant. The other three numbers are direct lines to specific users of the company. Now when a user makes an outbound call their caller ID shows up as the main office number. 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 to configure DOD on the UCM630xA:

1. To setup DOD go to UCM630xA Web GUI → **Extension/Trunk** → **VoIP Trunks** page.



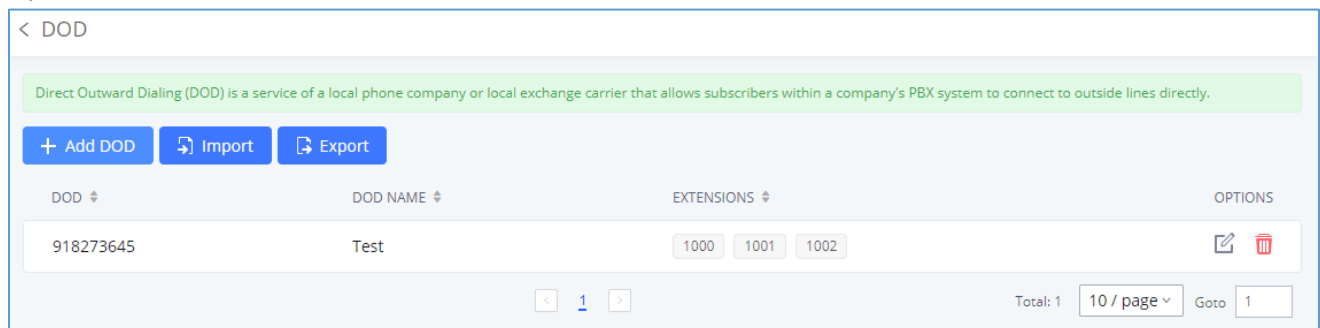
2. Click  to access the DOD options for the selected SIP Trunk.
3. Click "Add DOD" to begin your DOD setup
4. For "DOD Number" enter one of the numbers (DIDs) from your SIP trunk provider. In the example above Company ABC received 4 DIDs from their provider. ABC will enter in the number for the CEO's direct line.
5. Set the DOD name and If extension number need to be appended to the DID number click on "Add Extension".
6. 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 99: DOD extension selection**

7. 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.



DOD	DOD NAME	EXTENSIONS	OPTIONS
918273645	Test	1000 1001 1002	 

**Figure 100: Edit DOD**

**Note:** Users can import and export DOD files.



## SLA STATION

The UCM630xA 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 UCM630xA is like BLF but SLA is used to monitor external line i.e., analog trunk on the UCM630xA. 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 UCM630xA.

### Create/Edit SLA Station

SLA Station can be configured on Web GUI → **Extension/Trunk** → **SLA Station**.

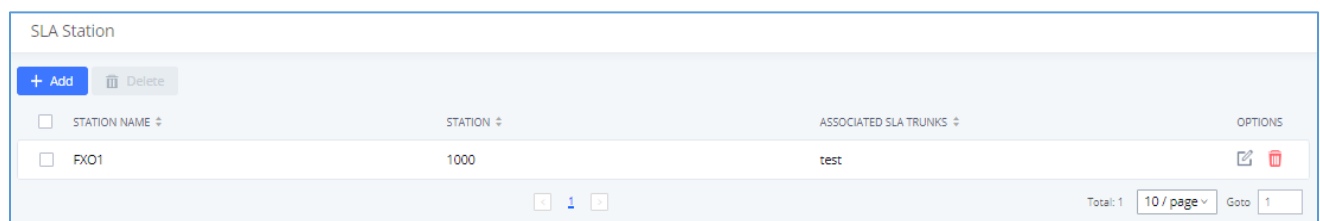


Figure 101: SLA Station

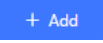




- Click on  to add an SLA Station.
- Click on  to edit the SLA Station. The following table shows the SLA Station configuration parameters.
- Click on  to delete the SLA Station.

Table 63: SLA Station Configuration Parameters

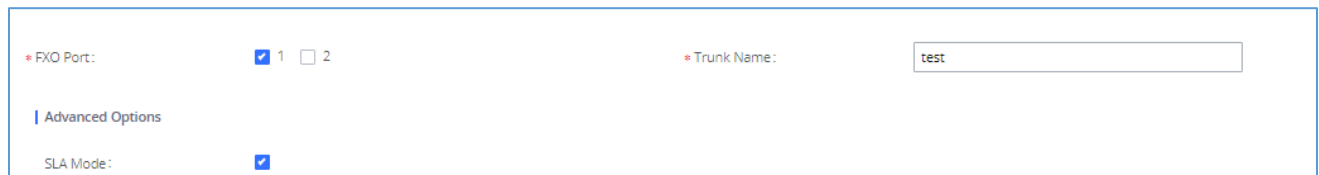
<b>Station Name</b>	Configure a name to identify the SLA Station.
<b>Station</b>	Specify a SIP extension as a station that will be using SLA.
<b>Available SLA Trunks</b>	Existing Analog Trunks with SLA Mode enabled will be listed here.
<b>Associated SLA Trunks</b>	Select a trunk for this SLA from the 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.
<b>SLA Station Options</b>	
<b>Ring Timeout</b>	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.



<b>Ring Delay</b>	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.
<b>Hold Access</b>	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 UCM630xA, go to Web GUI→**Extension/Trunk**→**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 GUI→**Extension/Trunk**→**SLA Station** page.



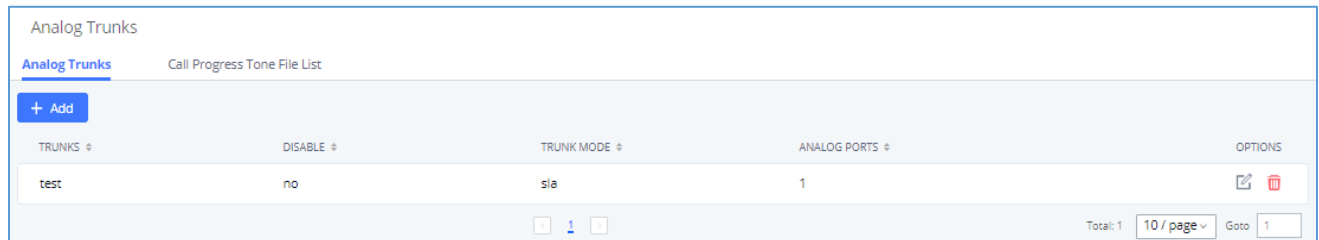
\* FXO Port:  1  2      \* Trunk Name:

| Advanced Options

SLA Mode:

**Figure 102: Enable SLA Mode for Analog Trunk**

2. Click on “Save”. The analog trunk will be listed with trunk mode “SLA”.

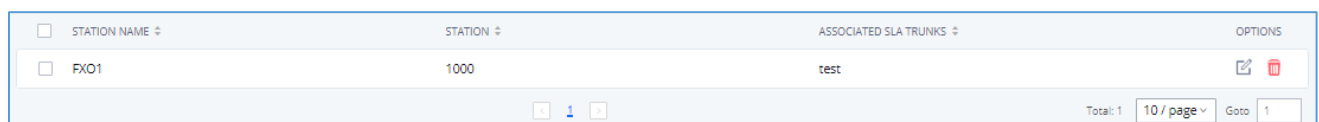


TRUNKS	DISABLE	TRUNK MODE	ANALOG PORTS	OPTIONS
test	no	sla	1	

Total: 1    10 / page    Goto 1

**Figure 103: Analog Trunk with SLA Mode Enabled**

3. On the UCM630xA, go to Web GUI→**Extension/Trunk**→**SLA Station** page, click on “Add”. 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
<input type="checkbox"/> FXO1	1000	test	

Total: 1    10 / page    Goto 1

**Figure 104: SLA Example - SLA Station**



4. On the SIP phone 1, configure to register UCM630xA extension 1002. Configure the MPK as BLF mode and the value must be set to “extension\_trunkname”, which is 1002\_fxo1 in this case.
5. On the SIP phone 2, configure to register UCM630xA extension 1005. Configure the MPK as BLF mode and value must be set to “extension\_trunkname”, which is 1005\_fxo1 in this case.

Mode	Account	Description	Value
MPK 1	Busy Lamp Field (BLF)	Account 2	1005_fxo1

Figure 105: 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.





# CALL ROUTES

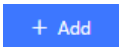


## Outbound Routes

In the following sections, we will discuss the steps and parameters used to configure and manage outbound rules in UCM630xA, these rules are the regulating points for all external outgoing calls initiated by the UCM through all types of trunks: SIP, Analog and Digital.

### Configuring Outbound Routes

In the UCM630xA, an outgoing calling rule pairs an extension pattern with a trunk used to dial the pattern. This allows different patterns to be dialed through different trunks (e.g., "Local" 7-digit dials through an FXO while "Long distance" 10-digit dials through a low-cost SIP trunk). Users can also set up a failover trunk to be used when the primary trunk fails.

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

- Click on  to add a new outbound route.
- Click on  to edit the outbound route.
- Click on  to delete the outbound route.

On the UCM630xA, the outbound route priority is based on "Best matching pattern". For example, the UCM630xA has outbound route A with pattern 1xxx and outbound route B with pattern 10xx configured. When dialing 1000 for outbound call, outbound route B will always be used first. This is because pattern 10xx is a better match than pattern 1xxx. Only when there are multiple outbound routes with the same pattern configured.

**Table 64: Outbound Route Configuration Parameters**

<b>Outbound Rule Name</b>	Configure the name of the calling rule (e.g., local, long_distance, and etc.). Letters, digits, _ and - are allowed.
<b>Pattern</b>	<p>All patterns are prefixed by "_" character, but please do not enter more than one "_" at the beginning. All patterns can add comments, such as "_pattern /* comment */". In patterns, some characters have special meanings:</p> <ul style="list-style-type: none"> <li>• [12345-9] ... Any digit in the brackets. In this example, 1,2,3,4,5,6,7,8,9 is allowed.</li> <li>• N ... Any digit from 2-9.</li> <li>• . ... Wildcard, matching one or more characters.</li> <li>• ! ... Wildcard, matching zero or more characters immediately.</li> <li>• X ... Any digit from 0-9.</li> <li>• Z ... Any digit from 1-9.</li> <li>• - ... Hyphen is to connect characters and it will be ignored.</li> </ul>



	<ul style="list-style-type: none"> <li>• [] Contain special characters ([x], [n], [z]) represent letters x, n, z.</li> </ul>
<b>Disable This Route</b>	After creating the outbound route, users can choose to enable and disable it. If the route is disabled, it will not take effect anymore. However, the route settings will remain in UCM. Users can enable it again when it is needed.
<b>Password</b>	Configure the password for users to use this rule when making outbound calls.
<b>Local Country Code</b>	If your local country code is affected by the outbound blacklist, please enter it here to bypass the blacklist.
<b>Call Duration Limit</b>	Enable to configure the maximum duration for the call using this outbound route.
<b>Maximum Call Duration</b>	Configure the maximum duration of the call (in seconds). The default setting is 0, which means no limit.
<b>Warning Time</b>	Configure the warning time for the call using this outbound route. If set to x seconds, the warning tone will be played to the caller when x seconds are left to end the call.
<b>Auto Record</b>	If enabled, calls using this route will automatically be recorded.
<b>Warning Repeat Interval</b>	Configure the warning repeat interval for the call using this outbound route. If set to X seconds, the warning tone will be played every x seconds after the first warning.
<b>PIN Groups</b>	Select a PIN Group
<b>PIN Groups with Privilege Level</b>	If enabled and PIN Groups are used, Privilege Levels and Filter on Source Caller ID will also be applied.
<b>Privilege Level</b>	<p>Select privilege level for the outbound rule.</p> <ul style="list-style-type: none"> <li>• <b>Internal:</b> The lowest level required. All users can use this rule.</li> <li>• <b>Local:</b> Users with Local, National, or International level can use this rule.</li> <li>• <b>National:</b> Users with National or International level can use this rule.</li> <li>• <b>International:</b> The highest level required. Only users with international level can use this rule.</li> <li>• <b>Disable:</b> The default setting is "Disable". If selected, only the matched source caller ID will be allowed to use this outbound route.</li> </ul> <p>Please be aware of the potential security risks when using "Internal" level, which means all users can use this outbound rule to dial out from the trunk.</p>
<b>Enable Filter on Source Caller ID</b>	<p>When enabled, users could specify extensions allowed to use this outbound route. "Privilege Level" is automatically disabled if using "Enable Filter on Source Caller ID".</p> <p>The following two methods can be used at the same time to define the extensions as the source caller ID.</p>



	<ol style="list-style-type: none"> <li>1. Select available extensions/extension groups from the list. This allows users to specify arbitrary single extensions available in the PBX.</li> <li>2. Custom Dynamic Route: define the pattern for the source caller ID. This allows users to define extension range instead of selecting them one by one.           <ul style="list-style-type: none"> <li>• All patterns are prefixed with the "_".</li> <li>• Special characters:               <ul style="list-style-type: none"> <li><b>X</b>: Any Digit from 0-9.</li> <li><b>Z</b>: Any Digit from 1-9.</li> <li><b>N</b>: Any Digit from 2-9.</li> <li><b>."</b>: Wildcard. Match one or more characters.</li> <li><b>"!"</b>: Wildcard. Match zero or more characters immediately.</li> <li>Example: <b>[12345-9]</b> - Any digit from 1 to 9.</li> </ul> </li> </ul> <p><u>Note</u>: Multiple patterns can be used. Patterns should be separated by comma ",", Example: <b>_X. , _NNXXNXXXXX , _818X.</b></p> </li> </ol>
<b>Outbound Route CID</b>	Attempt to use the configured outbound route CID. This CID will not be used if DOD is configured.
<b>Send This Call Through Trunk</b>	
<b>Trunk</b>	Select the trunk for this outbound rule.
<b>Strip</b>	<p>Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk.</p> <p><u>Example:</u> The users will dial 9 as the first digit of a long-distance calls. However, 9 should not be sent out via analog lines and the PSTN line. In this case, 1 digit should be stripped before the call is placed.</p>
<b>Prepend</b>	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
<b>Use Failover Trunk</b>	
<b>Failover Trunk</b>	<p>Failover trunks can be used to make sure that a call goes through an alternate route when the primary trunk is busy or down. If "Use Failover Trunk" is enabled and "Failover trunk" is defined, the calls that cannot be placed via the regular trunk may have a secondary trunk to go through.</p> <p>UCM630xA support up to 10 failover trunks.</p> <p><u>Example:</u> The user's primary trunk is a VoIP trunk and the user would like to use the PSTN when the VoIP trunk is not available. The PSTN trunk can be configured as the failover trunk of the VoIP trunk.</p>



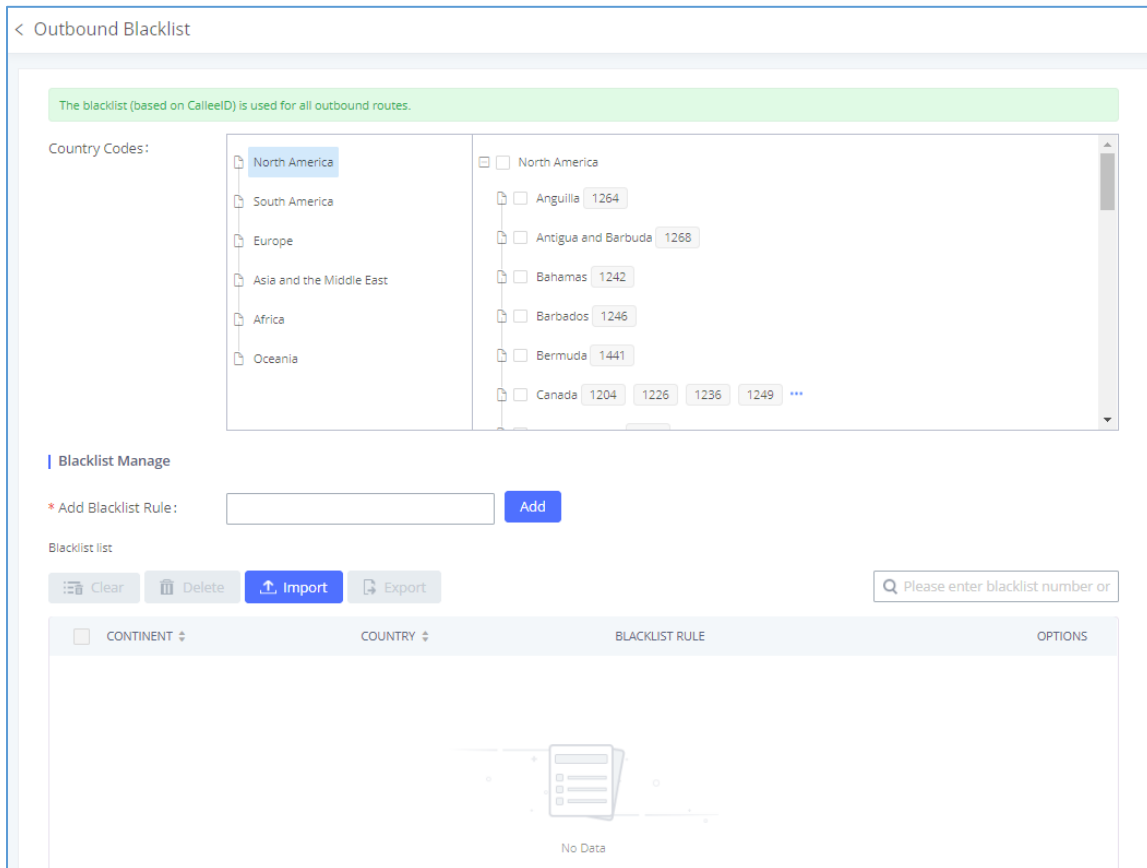
<b>Strip</b>	<p>Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk.</p> <p><u>Example:</u> The users will dial 9 as the first digit of a long-distance calls. However, 9 should not be sent out via analog lines and the PSTN line. In this case, 1 digit should be stripped before the call is placed.</p>
<b>Prepend</b>	<p>Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.</p>
<b>Time Condition</b>	
<b>Time Condition Mode</b>	<p>Use Main Trunk or Failover Trunk: Use the Main Trunk and its settings during the configured time conditions. If the main trunk is unavailable, the Failover Trunk and its settings will be used instead.</p> <p>Use Specific Trunks: Use specific trunks during the configured time conditions. The Strip and Prepend settings of the Main Trunk will be used. If a trunk is unavailable during its time condition, no failover trunks will be used.</p>

## Outbound Blacklist

The UCM630xA allows users to configure blacklist for outbound routes. If the dialing number matches the blacklist numbers or patterns, the outbound call will not be allowed. The outbound blacklist can be configured under UCM Web GUI → **Extension/Trunk** → **Outbound Routes**: Outbound Blacklist.

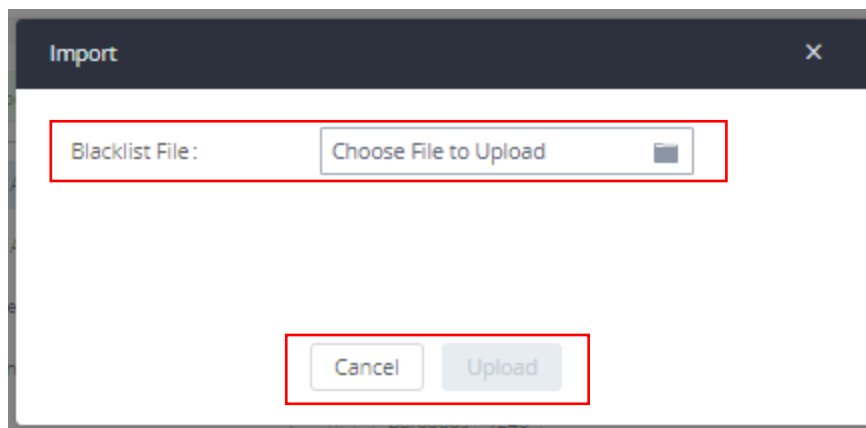
Users can configure numbers, patterns or select country code to add in the blacklist. Please note that the blacklist settings apply to all outbound routes.





**Figure 106: Country Codes**

**Note:** Users can export outbound route blacklists and delete all blacklist entries. Additionally, users can also import blacklists for outbound routes.



**Figure 107: Blacklist Import/Export**



## Scheduled Sync

The UCM630xA allows users to synchronize the outbound routes, this feature can be found on the WebGUI→**Extension/Trunk**→**Outbound Routes**→ **Scheduled Sync**.

**Table 65: Outbound Routes/Scheduled Sync**

<b>Scheduled Sync</b>	Enable the Scheduled Sync feature
<b>Server Address</b>	Enter the TFTP server address. For example, "192.168.1.2:69".
<b>File Name</b>	Specify the file name
<b>Sync Time</b>	Enter the sync time (24hr format). Valid range is 0-23.
<b>Sync Frequency</b>	Create new sync every x day(s). The valid range is 1 to 30.

## PIN Groups

The UCM630xA supports pin group. Once this feature is configured, users can apply pin group to specific outbound routes. When placing a call on pin protected outbound routes, caller will be asked to input the group pin number, this feature can be found on the WebGUI→**Extension/Trunk**→**Outbound Routes**→**PIN Groups**.

**Table 66: Outbound Routes/PIN Group**

<b>Name</b>	Specify the name of the group
<b>Record In CDR</b>	Specify whether to enable/disable record in CDR
<b>PIN Number</b>	Specify the code that will asked once dialing via a trunk
<b>PIN Name</b>	Specify the name of the PIN

Once user click on [PIN Groups](#) the following figure shows to configure the new PIN.



### Create New PIN Group

\* Name:

Record in CDR:

**Members**

\* PIN Number:

\* PIN Name:

**Figure 108: Create New PIN Group**

The following screenshot shows an example of created PIN Groups and members:

< PIN Groups

NAME	RECORD IN CDR	OPTIONS						
▼ GSEMEA	yes	<input type="button" value="✎"/> <input type="button" value="↓"/> <input type="button" value="✖"/>						
<table border="1" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left;">PIN NUMBER</th> <th style="text-align: left;">PIN NAME</th> </tr> </thead> <tbody> <tr> <td>125478963</td> <td>Dao</td> </tr> <tr> <td>1596324</td> <td>Emily</td> </tr> </tbody> </table>			PIN NUMBER	PIN NAME	125478963	Dao	1596324	Emily
PIN NUMBER	PIN NAME							
125478963	Dao							
1596324	Emily							

Total: 1 | 10 / page | Goto 1

**Figure 109: PIN Members**

**Note:**

If PIN group is enabled on outbound route level, password, privilege level and enable filter on source caller ID will be disabled, unless if you check the option “PIN Groups with Privilege Level” where you can use the PIN Groups and Privilege Level or PIN Groups and Enable Filter on Source Caller ID.

**General**

\* Calling Rule Name:  Disable This Route:

\* Pattern:

PIN Groups:  PIN Groups with Privilege Level:

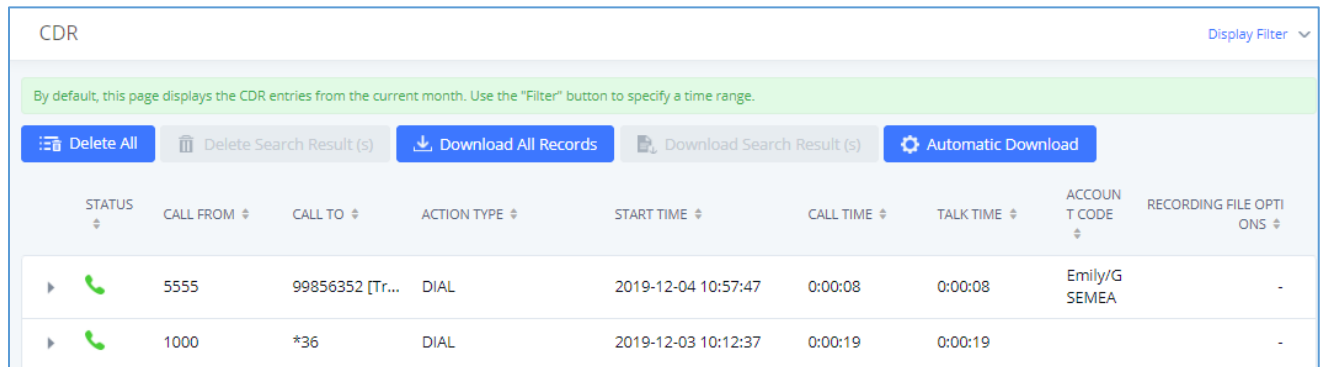
Password:

Privilege Level:

**Figure 110: Outbound PIN**



If PIN group CDR is enabled, the call with PIN group information will be displayed as part of CDR under Account Code field.



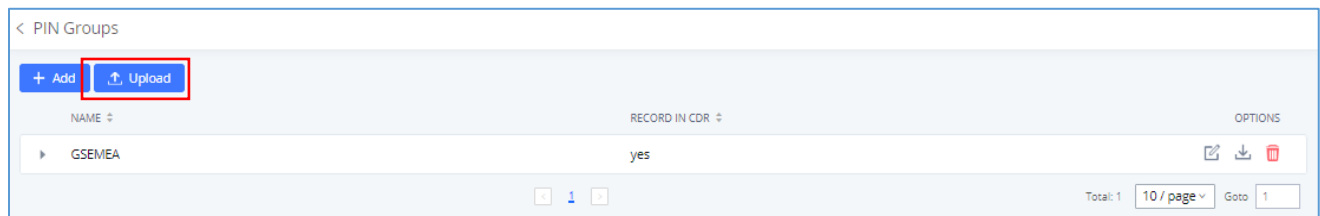
STATUS	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	TALK TIME	ACCOUNT CODE	RECORDING FILE OPTIONS
	5555	99856352 [Tr...]	DIAL	2019-12-04 10:57:47	0:00:08	0:00:08	Emily/G SEMEA	-
	1000	*36	DIAL	2019-12-03 10:12:37	0:00:19	0:00:19		-

Figure 111: CDR Record

### - Importing PIN Groups from CSV files:

User can also import PIN Groups by uploading CSV files for each group. To do this:

1. Navigate to **Extension/Trunk** → **Outbound Routes** → **PIN Groups** and click on the “Upload” button.



NAME	RECORD IN CDR	OPTIONS
GSEMEA	yes	

Figure 112: Importing PIN Groups from CSV files

2. Select the CSV file to upload. Incorrect file formats and improperly formatted CSV files will result in error messages such as the one below:

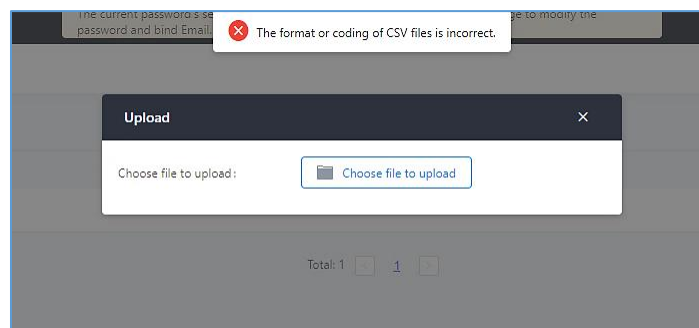


Figure 113: Incorrect CSV File

3. To ensure a successful import, please follow the format in the sample image below

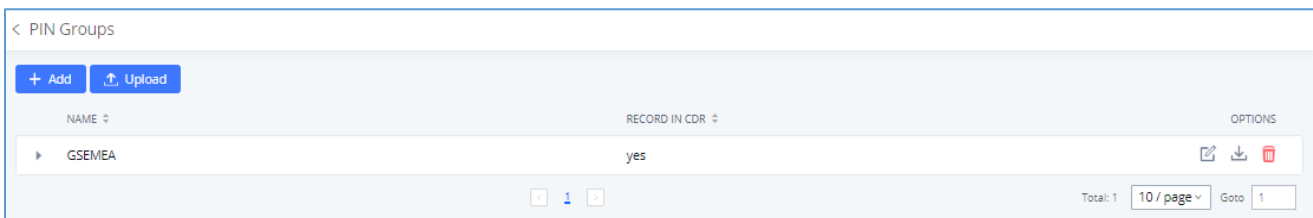




	A	B	C	D
1	ALPHA			
2	pin	pin_name		
3	1625	test1		
4	9497	test2		
5	5872	test3		
6				
7				

**Figure 114: CSV File Format**

- The top-left value (A1) is the PIN Group name. In this case, it is “ALPHA”.
- Row 2 contains the labels for the modifiable fields: pin and pin\_name. These values should not be changed and will cause an upload error otherwise.
- Rows 3+ contain the user-defined values with Column A holding the PINs and Column B holding the PIN names. PIN values must consist of at least four digits.
- Once the file is successfully uploaded, the entry will be added to the list of PIN Groups.

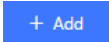




NAME	RECORD IN CDR	OPTIONS
GSEMEA	yes	[Edit] [Download] [Delete]

**Figure 115: CSV File Successful Upload**

## Inbound Routes

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

- Click on  to add a new inbound route.
- Click on "Blacklist" to configure blacklist for all inbound routes.
- Click on  to edit the inbound route.
- Click on  to delete the inbound route.



## Inbound Rule Configurations

**Table 67: Inbound Rule Configuration Parameters**

<b>Trunks</b>	Select the trunk to configure the inbound rule.						
<b>Inbound Route Name</b>	Configure the name of the Inbound Route. For example, "Local", "LongDistance" and etc.						
<b>Pattern</b>	<ul style="list-style-type: none"> <li>All patterns are prefixed with the "_".</li> <li>Special characters:           <ul style="list-style-type: none"> <li><b>X</b>: Any Digit from 0-9.</li> <li><b>Z</b>: Any Digit from 1-9.</li> <li><b>N</b>: Any Digit from 2-9.</li> <li><b>"."</b>: Wildcard. Match one or more characters.</li> <li><b>"!"</b>: Wildcard. Match zero or more characters immediately.</li> </ul> </li> <li>Example: [12345-9] - Any digit from 1 to 9.</li> <li>The pattern can be composed of two parts, <i>Pattern</i> and <i>CallerID Pattern</i>. The first part is used to specify the dialed number while the second part is used to specify the caller ID and it is optional, if set it means only the extension with the specific caller ID can call in or call out. For example, pattern '_2XXX/1234' means the only extension with the caller ID '1234' can use this rule.</li> </ul> <p><b>Note:</b></p> <ul style="list-style-type: none"> <li>Multiple patterns can be used. Each pattern should be entered in new line.</li> <li>Users can add comments to the end of patterns to better organize and keep track of complex rules by typing "/" and "/" before and after each comment respectively</li> </ul> <p>Example:</p> <table border="1"> <thead> <tr> <th>Pattern</th> <th>CallerID Pattern</th> </tr> </thead> <tbody> <tr> <td>_X.</td> <td>1000</td> </tr> <tr> <td>_ NNXXNXXXXX /* 10-digit long distance */</td> <td>1001</td> </tr> </tbody> </table>	Pattern	CallerID Pattern	_X.	1000	_ NNXXNXXXXX /* 10-digit long distance */	1001
Pattern	CallerID Pattern						
_X.	1000						
_ NNXXNXXXXX /* 10-digit long distance */	1001						
<b>Disable This Route</b>	After creating the inbound route, users can choose to enable and disable it. If the route is disabled, it will not take effect anymore. However, the route settings will remain in UCM. Users can enable it again when it is needed.						
<b>Seamless Transfer Whitelist</b>	Allows the selected extension to use this function. If an extension is busy, and a mobile phone is bound to that extension, the mobile phone can pick up calls to that extension.						
<b>Ringback tone</b>	Choose the custom ring back tone to play when caller reaches the route.						



<b>Auto Record</b>	If enabled, calls using this route will automatically be recorded.
<b>Block Collect Call</b>	If enabled, collect calls will be blocked. Note: Collect calls are indicated by the header "P-Asserted-Service-Info: service-code=Backward Collect Call, P-Asserted-Service-Info: service-code=Collect Call".
<b>Alert-Info</b>	Configure the Alert-Info, when UCM receives an INVITE request, the Alert-Info header field specifies an alternative ring tone to the UAS.
<b>Fax Detection</b>	If enabled, fax signals from the trunk during a call will be detected.
<b>Fax Destination</b>	Configures the destination of faxes. <ul style="list-style-type: none"> <li>• <b>Extension:</b> send the fax to the designated FXS/SIP extension (fax machine) or a FAX extension.</li> <li>• <b>Fax to Email:</b> send the fax as an email attachment to the designated extension's email address. If the selected extension does not have an associated email address, it will be sent to the default email address configured in the <i>Call Features-&gt;Fax/T.38-&gt;Fax Settings</i> page.</li> </ul> <p>Note: please make sure the sending email address is correctly configured in <b>System Settings-&gt;Email Settings</b>.</p>
<b>Prepend Trunk Name</b>	If enabled, the trunk name will be added to the caller id name as the displayed caller id name.
<b>Set Caller ID Info</b>	Manipulates Caller ID (CID) name and/or number within the call flow to help identify who is calling. When enabled two field will show allowing to manipulate the CalleID Number and the Caller ID Name.
<b>CalleID Number</b>	Configure the pattern-matching format to manipulate the numbers of incoming callers or to set a fixed CallerID number for calls that go through this inbound route. <ul style="list-style-type: none"> <li>• <b>`\${CALLERID(num)}`:</b> Default value which indicates the number of an incoming caller (CID). The CID will not be modified.</li> <li>• <b>`\${CALLERID(num):n}`:</b> Skips the first n characters of a CID number, where n is a number.</li> <li>• <b>`\${CALLERID(num):-n}`:</b> Takes the last n characters of a CID number, where n is a number.</li> <li>• <b>`\${CALLERID(num):s:n}`:</b> Takes n characters of a CID number starting from s+1, where n is a number and s is a character position (e.g. `\${CALLERID(num):2:7}` takes 7 characters after the second character of a CID number).</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>n\${CALLERID(num)}</b>: Prepends n to a CID number, where n is a number.</li> </ul>
<b>CallerID Name</b>	<p>Default string is <b>\${CALLERID(name)}</b>, which means the name of an incoming caller, it is a pattern-matching syntax format.</p> <p><b>A\${CALLERID(name)}B</b> means Prepend a character 'A' and suffix a character 'B' to <b>\${CALLERID(name)}</b>.</p> <p>Not using pattern-matching syntax means setting fix name to incoming caller.</p>
<b>Enable Route-Level Inbound Mode</b>	<p>Gives uses the ability to configure inbound mode per individual route. When enabled two field will show allowing to set the Inbound mode and the Inbound mode Suffix.</p> <p><b>Note:</b> Global inbound mode must be enabled before users can configure route-level inbound mode</p>
<b>Inbound Mode</b>	<p>Choose the inbound mode for this route.</p> <p><b>Note:</b> Toggling the global inbound mode will not affect routes that have Route-level Inbound Mode enabled. If all routes have the option enabled, toggling the global inbound mode via BLF will trigger a voice prompt indicating that none of the routes will be affected by the global inbound mode change.</p>
<b>Inbound Mode Suffix</b>	<p>Dial "Global Inbound Mode feature code + Inbound Mode Suffix" or a route's assigned suffix to toggle the route's inbound mode.</p> <p>The BLF subscribed to the inbound mode suffix can monitor the current inbound mode.</p>
<b>Inbound Multiple Mode</b>	<p>Multiple mode allows user to switch between destinations of the inbound rule by feature codes. Configure related feature codes as described in <b>[Inbound Route: Multiple Mode]</b>. If this option is enabled, user can use feature code to switch between different modes/destinations.</p>
<b>Dial Trunk</b>	<p>This option shows up only when "By DID" is selected. If enabled, the external users dialing in to the trunk via this inbound route can dial outbound call using the UCM's trunk.</p>
<b>Privilege Level</b>	<p>This option shows up only when "By DID" is selected.</p> <ul style="list-style-type: none"> <li>• <b>Disable:</b> Only the selected Extensions or Extension Groups are allowed to use this rule, when enabled Filter on Source Caller ID.</li> <li>• <b>Internal:</b> The lowest level required. All users are allowed to use this rule, check this level might be risky for security purpose.</li> <li>• <b>Local:</b> User with Local level, National or International level are allowed to use this rule.</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>National:</b> Users with National or International Level are allowed to use this rule.</li> <li>• <b>International:</b> The highest level required. Only users with international level are allowed to use this rule.</li> </ul>
<b>Allowed DID Destination</b>	<p>This option shows up only when "By DID" is selected. This controls the destination that can be reached by the external caller via the inbound route. The DID destination are:</p> <ul style="list-style-type: none"> <li>• Extension</li> <li>• Meeting</li> <li>• Call Queue</li> <li>• Ring Group</li> <li>• Paging/Intercom Groups</li> <li>• IVR</li> <li>• Voicemail Groups</li> <li>• Dial By Name</li> <li>• All</li> </ul>
<b>Default Destination</b>	<p>Select the default destination for the inbound call.</p> <ul style="list-style-type: none"> <li>• Extension</li> <li>• Voicemail</li> <li>• Meeting Room</li> <li>• Call Queue</li> <li>• Ring Group</li> <li>• Paging/Intercom</li> <li>• Voicemail Group</li> <li>• DISA</li> <li>• IVR</li> <li>• External Number</li> <li>• By DID</li> </ul> <p>When "By DID" is used, the UCM will look for the destination based on the number dialed, which could be local extensions, meeting, call queue, ring group, paging/intercom group, IVR and voicemail groups as configured in "DID destination". If the dialed number matches the DID pattern, the call will be allowed to go through.</p> <ul style="list-style-type: none"> <li>• Dial By Name</li> <li>• Callback</li> </ul>
<b>Strip</b>	<p>Specify the number of digits to strip from the beginning of the DID. This is used when "By DID" is selected in "Default Destination".</p>



<b>Prepend</b>	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
<b>Time Condition</b>	
<b>Start Time</b>	Select the start time "hour:minute" for the trunk to use the inbound rule.
<b>End Time</b>	Select the end time "hour:minute" for the trunk to use the inbound rule.
<b>Date</b>	Select "By Week" or "By Day" and specify the date for the trunk to use the inbound rule.
<b>Week</b>	Select the day in the week to use the inbound rule.
<b>Destination</b>	<p>Select the destination for the inbound call under the defined time condition.</p> <ul style="list-style-type: none"> <li>• Extension</li> <li>• Voicemail</li> <li>• Meeting Room</li> <li>• Call Queue</li> <li>• Ring Group</li> <li>• Paging/Intercom</li> <li>• Voicemail Group</li> <li>• DISA</li> <li>• IVR</li> <li>• By DID</li> </ul> <p>When "By DID" is used, the UCM will look for the destination based on the number dialed, which could be local extensions, meeting, call queue, ring group, paging/intercom group, IVR and voicemail groups as configured in "DID destination". If the dialed number matches the DID pattern, the call will be allowed to go through.</p> <p>Configure the number of digits to be stripped in "Strip" option.</p> <ul style="list-style-type: none"> <li>• Dial By Name</li> <li>• External Number</li> <li>• Callback</li> </ul>

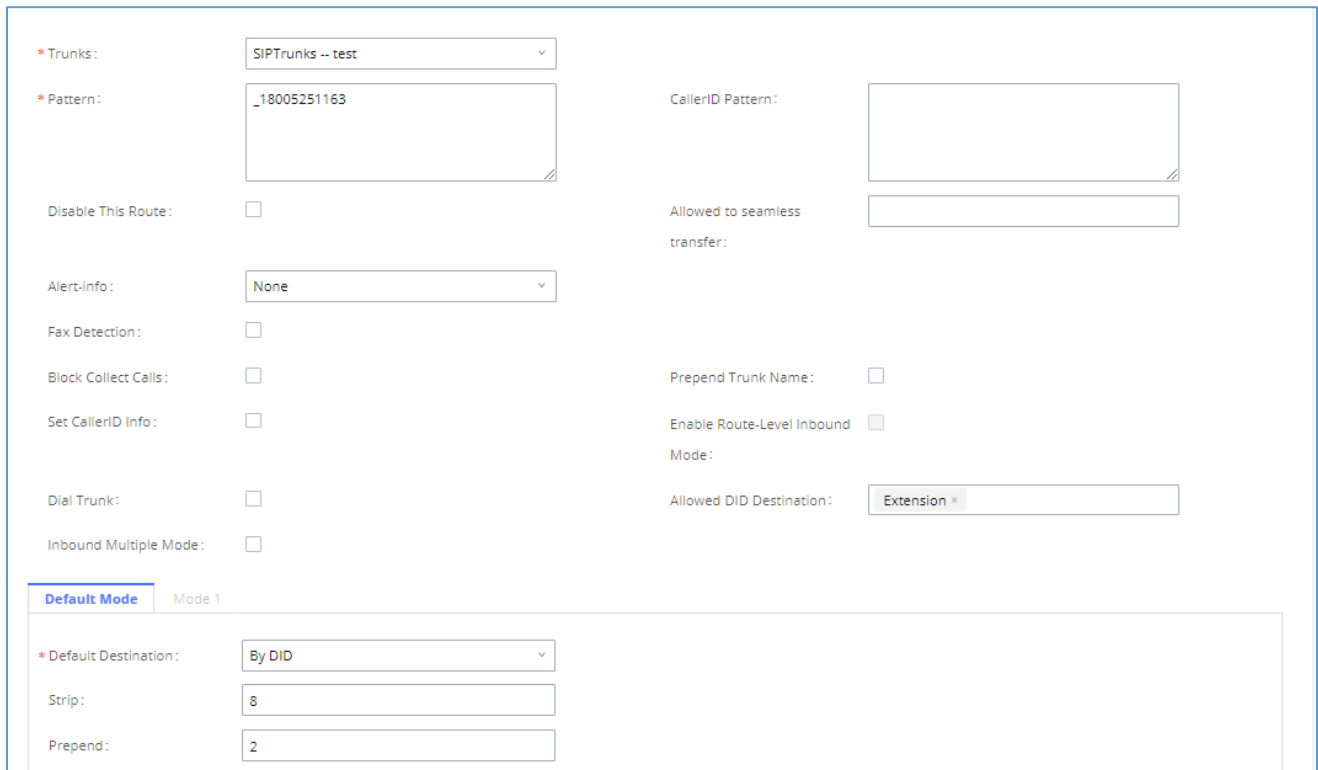
### Inbound Route: Prepend Example

UCM630xA now allows user to prepend digits to an inbound DID pattern, with strip taking precedence over prepend. With the ability to prepend digits in inbound route DID pattern, user no longer needs to create multiple routes for the same trunk to route calls to different extensions. The following example demonstrates the process:

1. If Trunk provides a DID pattern of 18005251163.
2. If **Strip** is set to 8, UCM630xA will strip the first 8 digits.



3. If **Prepend** is set to 2, UCM630xA will then prepend a 2 to the stripped number, now the number become 2163.
4. UCM630xA will now forward the incoming call to extension 2163.



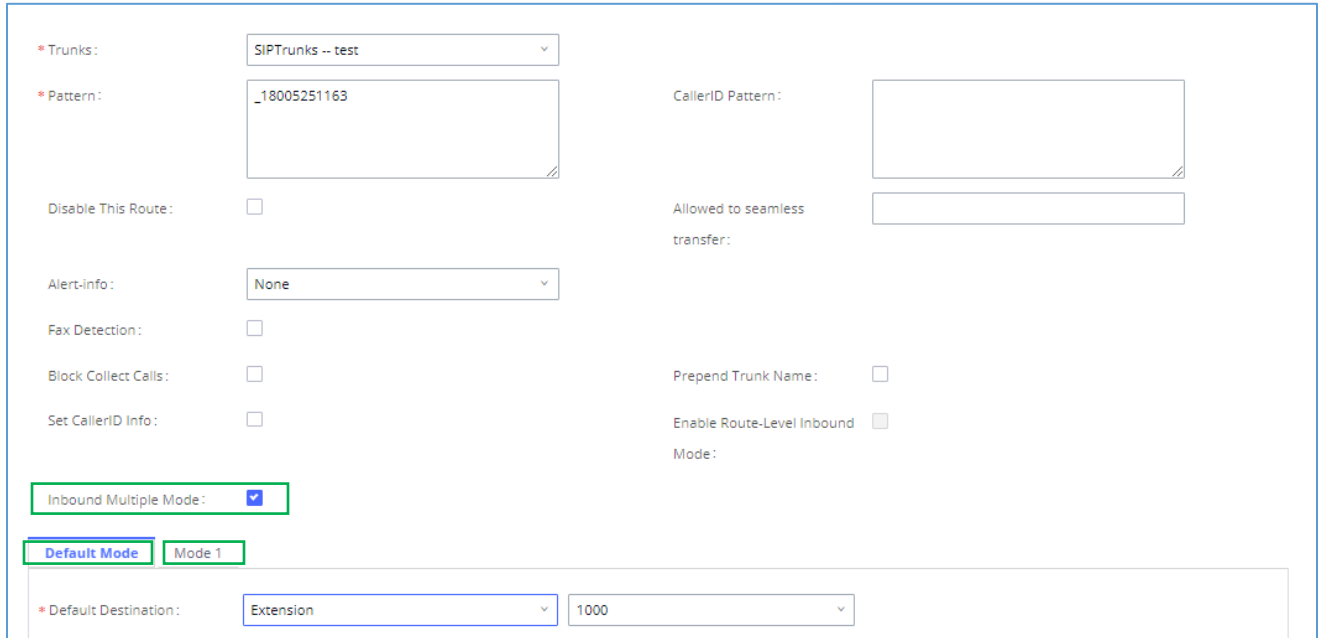
The screenshot displays the configuration interface for an Inbound Route. The 'Trunks' dropdown is set to 'SIPTrunks -- test'. The 'Pattern' field contains '\_18005251163'. The 'Prepend' field is set to '2'. Other settings include 'Alert-info' set to 'None', 'Fax Detection', 'Block Collect Calls', 'Set CallerID Info', 'Dial Trunk', and 'Inbound Multiple Mode' all unchecked. The 'Default Destination' is set to 'By DID', 'Strip' is '8', and 'Prepend' is '2'. The 'Allowed DID Destination' field shows 'Extension x'.

**Figure 116: Inbound Route feature: Prepend**

### Inbound Route: Multiple Mode

In the UCM630xA, the user can configure inbound route to enable multiple mode to switch between different destinations. The inbound multiple mode can be enabled under Inbound Route settings.





\* Trunks: SIPTrunks -- test

\* Pattern: \_18005251163

CallerID Pattern:

Disable This Route:

Allowed to seamless transfer:

Alert-info: None

Fax Detection:

Block Collect Calls:

Prepend Trunk Name:

Set CallerID Info:

Enable Route-Level Inbound Mode:

Inbound Multiple Mode:

Default Mode Mode 1


\* Default Destination: Extension 1000

**Figure 117: Inbound Route - Multiple Mode**

When Multiple Mode is enabled for the inbound route, the user can configure a “Default Destination” and a “Mode 1” destination for all routes. By default, the call coming into the inbound routes will be routed to the default destination.

SIP end devices that have registered on the UCM630xA can dial feature code \*62 to switch to inbound route “Mode 1” and dial feature code \*61 to switch back to “Default Destination”. Switching between different mode can be easily done without Web GUI login.

For example, the customer service hotline destination has to be set to a different IVR after 7PM. The user can dial \*62 to switch to “Mode 1” with that IVR set as the destination before off work.

 [Set Global Inbound Mode](#)

To customize feature codes for “Default Mode” and “Mode 1”, click on [Set Global Inbound Mode](#) under “Inbound Routes” page, check “Enable Inbound Multiple Mode” option and change “Inbound Default Mode” and “Inbound Mode 1” values (By default, \*61 and \*62 respectively).





### Set Global Inbound Mode

Caution: Disabling Inbound Multiple Mode will switch the inbound mode to default mode.

Enable Inbound Multiple

Mode:

Inbound Mode:

\* Inbound Default Mode:

\* Inbound Mode 1:

**Figure 118: Inbound Route - Multiple Mode Feature Codes**

## Inbound Route: Route-Level Mode

In the UCM630xA, users can enable Route-Level Inbound Mode to switch between different destinations for each individual inbound route. The inbound Route-Level mode can be enabled under Inbound Route settings.

\* Trunks:

\* Pattern:

Disable This Route:

Alert-info:

Fax Detection:

Block Collect Calls:

Set CallerID Info:

Inbound Mode:

Inbound Multiple Mode:

CallerID Pattern:

Allowed to seamless transfer:

Prepend Trunk Name:

Enable Route-Level Inbound Mode:

\* Inbound Mode Suffix:

**Figure 119: Inbound Route - Route-Level Mode**

Global inbound mode must be enabled before configuring Route-Level Inbound Mode. Additionally, the Mode 1 must be configured as well.

When Route-Level Inbound Mode is enabled, the user can configure a “Default Destination” and a “Mode 1” destination for each specific route. By default, the call coming into this specific inbound route will be routed to the default destination.



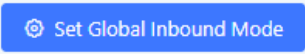
Users can toggle the route's inbound mode by dialing "Global Inbound Mode feature code + Inbound Mode Suffix" and the current inbound route can be monitored by subscribing a BLF to the Inbound Mode Suffix.

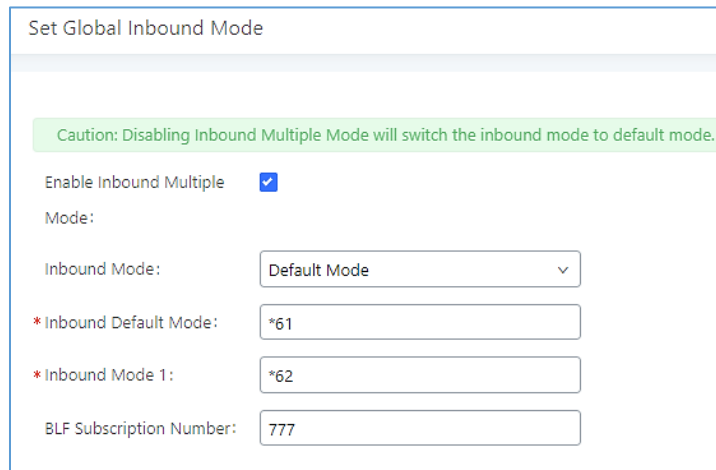
For example, Inbound Default Mode feature code is set to \*61 and the Inbound Mode suffix for route 1 is set to 1010. To switch the mode of route 1 to Default Mode, users can dial \*611010.

**Note:** Toggling the global inbound mode will not affect routes that have *Route-level Inbound Mode* enabled. If all routes have the option enabled, toggling the global inbound mode via BLF will trigger a voice prompt indicating that none of the routes will be affected by the global inbound mode change.

### Inbound Route: Inbound Mode BLF Monitoring

Users can assign MPKs and VPKs to monitor and toggle the current global inbound mode of the UCM. To do this, please refer to the following steps:

1. Access the UCM web GUI and navigate to Extension/Trunk→Inbound Routes.
2. Click on the  button and enable Inbound Multiple Mode.
3. Edit the subscribe number field to the desired BLF value.



Set Global Inbound Mode

Caution: Disabling Inbound Multiple Mode will switch the inbound mode to default mode.

Enable Inbound Multiple

Mode:

Inbound Mode:

\*Inbound Default Mode:

\*Inbound Mode 1:

BLF Subscription Number:

**Figure 120: Global Inbound Mode**

4. Configure the BLF value on a phone's MPK/VPK. As an example, a GXP2140 with the BLF configured will show the Inbound Mode status on its screen once configured. The 777 BLF is lit green, indicating that the current inbound mode is "Default Mode".





**Figure 121: Inbound Mode - Default Mode**

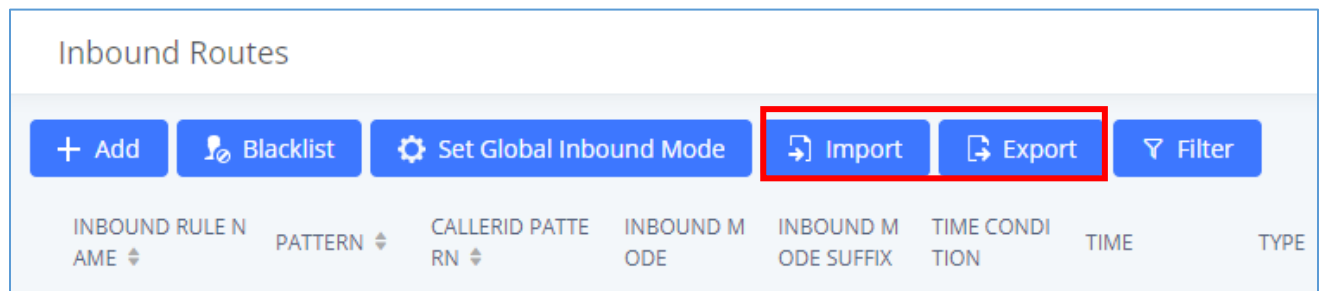
- Pressing the key will toggle the inbound mode to “Mode 1”, and the button’s color will change to red.



**Figure 122: Inbound Mode - Mode 1**

## Inbound Route: Import/Export Inbound Route

Users can now import and export inbound routes to quickly set up inbound routing on a UCM or to back up an existing configuration. An exported inbound route configuration can be directly imported without needing any manual modifications.



**Figure 123: Import/Export Inbound Route**

The imported file should be on CSV format and using UTF-8 encoding, the imported file should contain below columns, and each column should be separated by a comma (It is recommended to use Notepad++ for the imported file creation):

- Disable This Route: Yes/No.
- Pattern: Always prefixed with \_
- CallerID Pattern: Always prefixed with \_
- Prepend Trunk Name: Yes/No.





- Prepend User Defined Name Enable: Yes/No.
- Prepend User Defined Name: A string.
- Alert-info: None, Ring 1, Ring 2... User should enter an Alert-info string following the values we have in the Inbound route Alert-Info list.
- Allowed to seamless transfer: [Extension\_number]
- Inbound Multiple Mode: Yes/No.
- Default Destination: By DID, Extension, Voicemail... User should enter a Default Destination string following the values we have in the Inbound route Default Destination list.
- Destination: An Extension number, Ring Group Extension...
- Default Time Condition.
- Mode 1: By DID, Extension, Voicemail... User should enter a Default Destination string following the values we have in the mode 1 Default Destination list.
- Mode 1 Destination: An Extension number, Ring Group Extension...
- Mode 1 Time Condition.

## FAX with Two Media

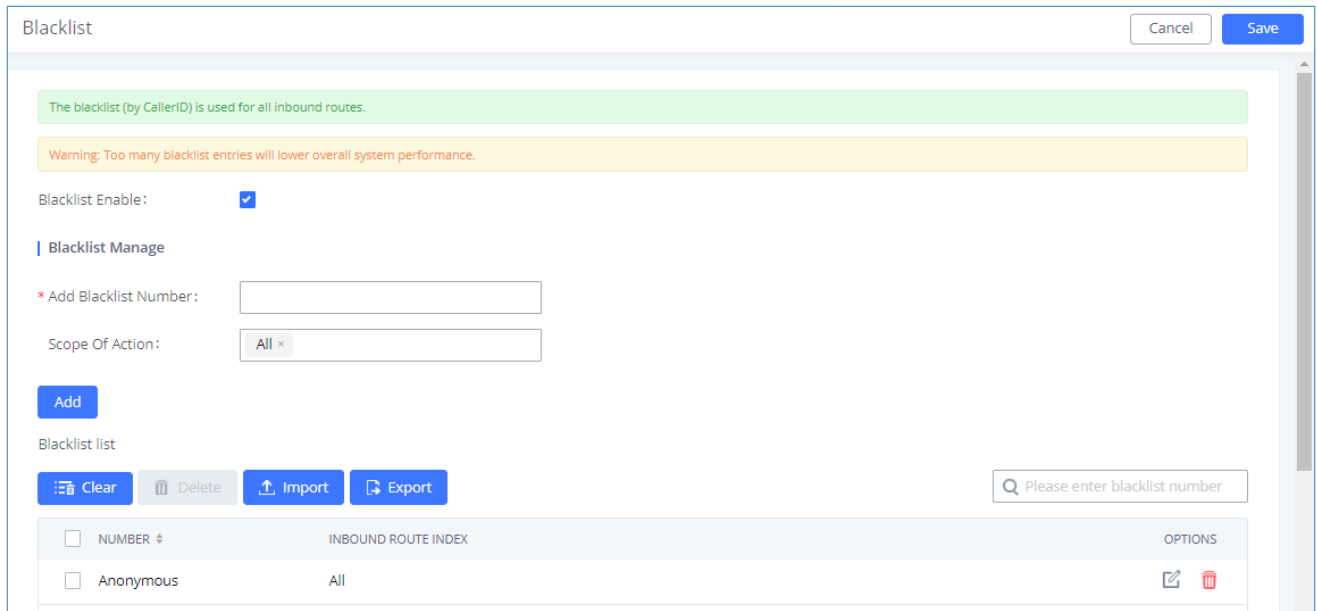
The UCM630xA supports Fax re-INVITE with multiple codec negotiation. If a Fax re-INVITE contains both T.38 and PCMA/PCMU codec, UCM630xA will choose T.38 codec over PCMA/PCMU.

## Blacklist Configurations

In the UCM630xA, Blacklist is supported for all inbound routes. Users could enable the Blacklist feature and manage the Blacklist by clicking on "Blacklist".

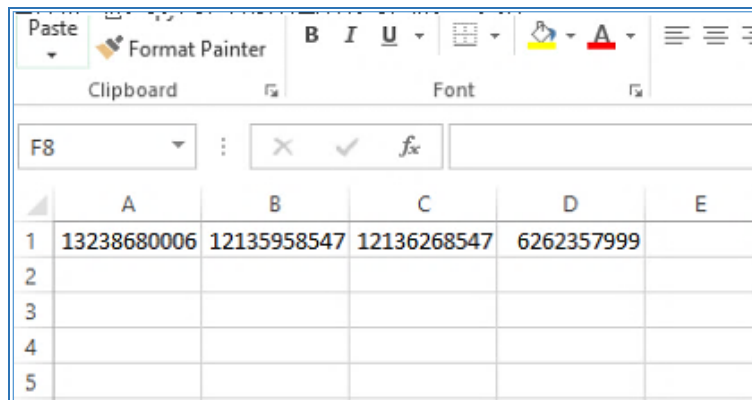
- Select the checkbox for "Blacklist Enable" to turn on Blacklist feature for all inbound routes. Blacklist is disabled by default.
- Enter a number in "Add Blacklist Number" field and then click "Add" to add to the list. Anonymous can also be added as a Blacklist Number by typing "Anonymous" in Add Blacklist Number field.
- To remove a number from the Blacklist, select the number in "Blacklist list" and click on  or click on "Clear" button to remove all the numbers on the blacklist.
- User can also export the inbound route blacklist by pressing on  button.





**Figure 124: Blacklist Configuration Parameters**

- To add blacklist number in batch, click on “Import” to upload blacklist file in csv format. The supported csv format is as below.



**Figure 125: Blacklist csv File**

**Note:**



Users could also add a number to the Blacklist or remove a number from the Blacklist by dialing the feature code for "Blacklist Add" (default: \*40) and "Blacklist Remove" (default: \*41) from an extension. The feature code can be configured under Web GUI → **Call Features** → **Feature Codes**.



## FAX SERVER

The UCM6300A series supports T.30/T.38 Fax and Fax Pass-through. It can convert the received Fax to PDF format and send it to the configured Email address. Fax/T.38 settings can be accessed via Web GUI→**Call Features**→**FAX/T.38**. The list of received Fax files will be displayed in the same web page for users to view, retrieve and delete.

### Configure Fax/T.38

- Click on "Create New Fax Extension". In the popped-up window, fill the extension, name and Email address to send the received Fax to.
- Click on "**Fax Settings**" to configure the Fax parameters.
- Click on  to edit the Fax extension.
- Click on  to delete the Fax extension.

**Fax Settings**

---

- Enable Error Correction Mode:
- Maximum Transfer Rate:
- Minimum Transfer Rate:
- Max Concurrent Sending Fax:
- Fax Queue Length:
- User Information in Fax Header:
- Fax Header Information:
- Default Email Address:  [Email Template](#)
- Send PDF Files Only:
- Enable Fax Resend:
- Max Resend Attempts:
- Fax Resend Frequency:

**Figure 126: Fax Settings**



**Table 68: FAX/T.38 Settings**

<b>Enable Error Correction Mode</b>	Configure to enable Error Correction Mode (ECM) for the Fax. The default setting is "Yes".
<b>Maximum Transfer Rate</b>	Configure the maximum transfer rate during the Fax rate negotiation. The possible values are 2400, 4800, 7200, 9600, 12000 and 14400. The default setting is 14400.
<b>Minimum Transfer Rate</b>	Configure the minimum transfer rate during the Fax rate negotiation. The possible values are 2400, 4800, 7200, 9600, 12000 and 14000. The default setting is 2400.
<b>Max Concurrent Sending Fax</b>	Configure the concurrent fax that can be sent by UCM6300A. Two modes "Only" and "More" are supported. <ul style="list-style-type: none"> <li>• <b>Only</b> Under this mode, the UCM6300A allows only single user to send fax at a time.</li> <li>• <b>More</b> Under this mode, the UCM6300A supports multiple concurrent fax sending by the users.</li> </ul> By default, this option is set to "only".
<b>Fax Queue Length</b>	Configure the maximum length of Fax Queue from 6 to 10. The default setting is 6.
<b>User Information in Fax Header</b>	If enabled this this will give users the option to send a special header in SIP fax messages.
<b>Fax Header Information</b>	Adds fax header into the fax file.
<b>Default Email Address</b>	Configure the Email address to send the received Fax to if user's Email address cannot be found.  <b>Note:</b> The extension's Email address or the Fax's default Email address needs to be configured in order to receive Fax from Email. If neither of them is configured, Fax will not be received from Email.
<b>Template Variables</b>	Fill in the "Subject:" and "Message:" content, to be used in the Email when sending the Fax to the users.



	<p>The template variables are:</p> <ul style="list-style-type: none"> <li>• <code>{CALLERIDNUM}</code> : Caller ID Number</li> <li>• <code>{CALLERIDNAME}</code> : Caller ID Name</li> <li>• <code>{RECEIVEEXTEN}</code> : The extension to receive the Fax</li> <li>• <code>{FAXPAGES}</code> : Number of pages in the Fax</li> <li>• <code>{VM_DATE}</code> : The date and time when the Fax is received</li> </ul>
<b>Send PDF Files Only</b>	If enabled, fax emails will no longer attach TIFF files. Only PDF files will be attached.
<b>Enable Fax Resend</b>	Enables the fax resend option which allow the UCM to keep attempting to send faxes up to a specified amount of times. Additionally, if a fax still fails to send, a <i>Resend</i> button will appear in the File Send Progress list in <i>Value-Added Features</i> → <i>Fax Sending</i> to allow manual resending.
<b>Max Resend Attempts</b>	Configures the maximum attempts number to resend the fax. Default value is set to 5.
<b>Fax Resend Frequency</b>	Configures the Fax Resend Frequency. Default value is set to 50.

## Receiving Fax

### Example Configuration to Receive Fax from PSTN Line

The following instructions describe how to use the UCM6300A to receive fax from PSTN line on the Fax machine connected to the UCM6300A FXS port.

1. Connect Fax machine to the UCM6300A FXS port.
2. Connect PSTN line to the UCM6300A FXO port.
3. Go to Web GUI → **Extension/Trunk** page.
4. Create or edit the analog trunk for Fax as below.
  - Fax Detection:** Make sure “Fax Detection” option is set to “NO”.





Edit Analog Trunk: FAX Cancel Save

\* FXO Port:  1  2      \* Trunk Name: FAX

**Advanced Options**

SLA Mode:

Enable Polarity Reversal:

Current Disconnect Threshold (ms):  200      \* Ring Timeout: 8000

\* RX Gain: 0dB      \* TX Gain: 0dB

Use CallerID:       **Fax Mode: None**

Caller ID Scheme: Bellcore/Telcordia      \* FXO Dial Delay (ms): 0

Auto Record:       Disable This Trunk:

DAHDI Out Line Selection: Ascend      \* The Maximum Number of Call Lines: 0

Echo Cancellation Mode: Default      Direct Callback:

**Figure 127: Configure Analog Trunk**

5. Go to UCM6300A Web GUI → **Extension/Trunk** → **Extensions** page.
6. Create or edit the extension for FXS port.
  - **Analog Station:** Select FXS port to be assigned to the extension. By default, it is set to "None".
  - Once selected, analog related settings for this extension will show up in "**Analog Settings**" section.

Create New Extension Cancel Save

**Basic Settings**    Media    Features    Specific Time    Follow Me

\* Select Extension Type: FXS Extension

Select Add Method: Single

---

**General**

\* Extension: 1005      Analog Station: FXS 1

CallerID Number:      \* Privilege: Internal

Voicemail: Local Voicemail      \* Voicemail Password: 96902350

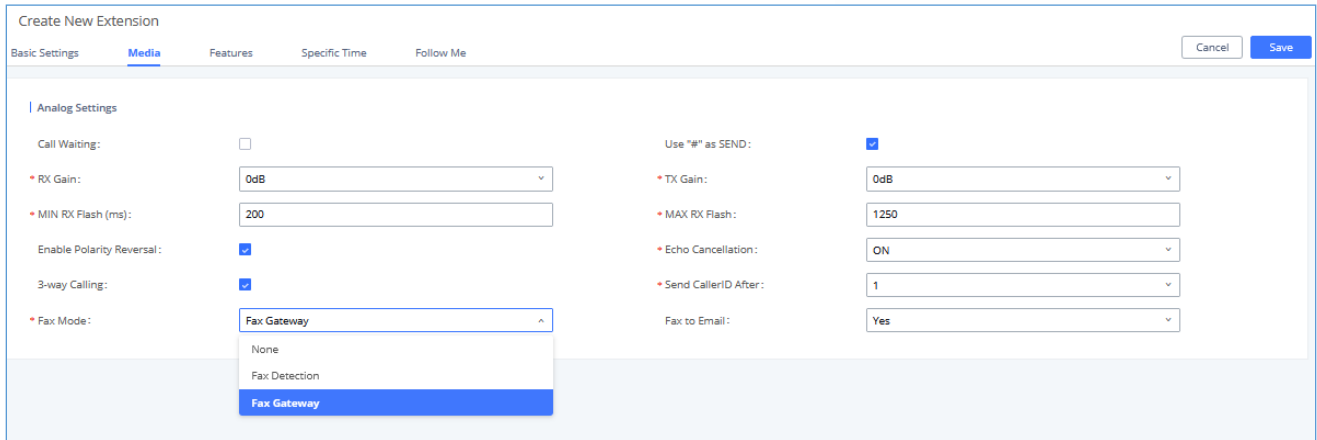
Skip Voicemail Password Verification:       Send Voicemail Email Notification: Default

Attach Voicemail to Email: Default      Keep Voicemail after Emailing: Default

Disable This Extension:       Emergency Calls CID:

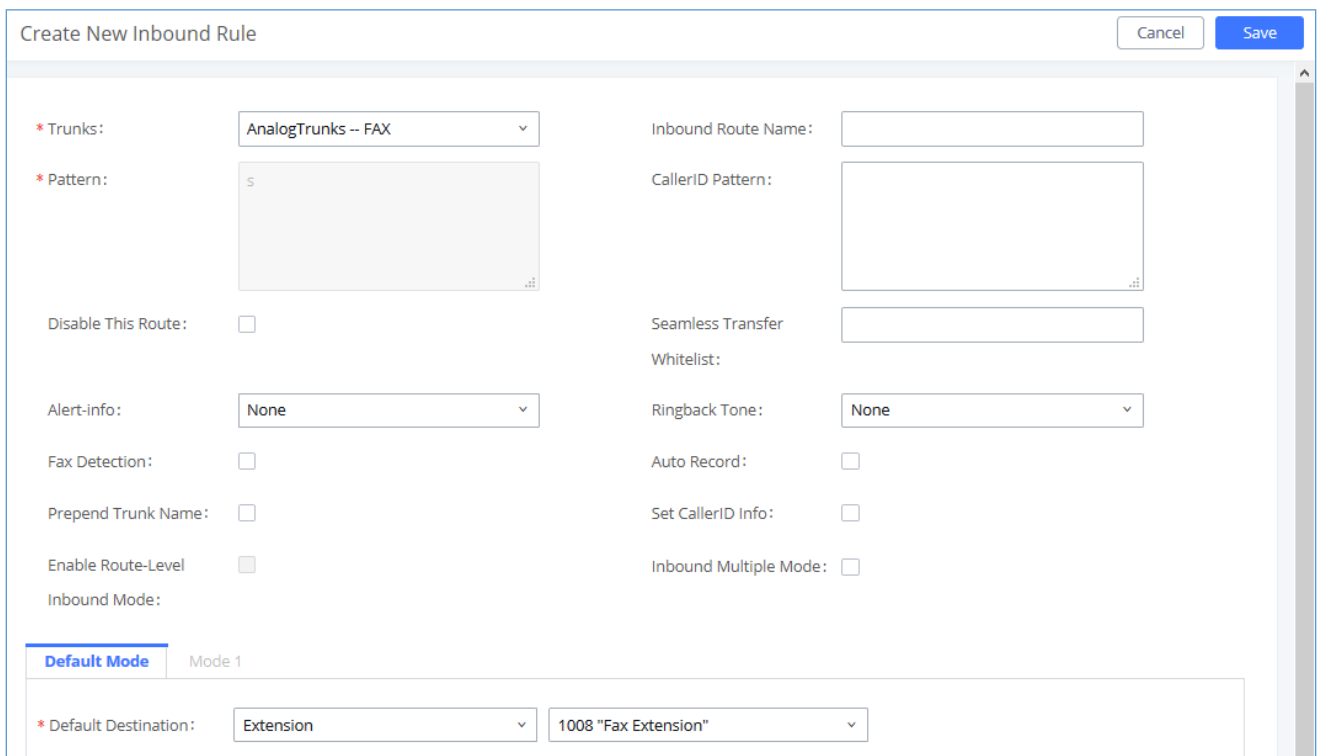
**Figure 128: Configure Extension for Fax Machine: FXS Extension**





**Figure 129: Configure Extension for Fax Machine: Analog Settings**

7. Go to Web GUI→**Extension/Trunk**→**Inbound Routes** page.
8. Create an inbound route to use the Fax analog trunk. Select the created extension for Fax machine in step 4 as the default destination.



**Figure 130: Configure Inbound Rule for Fax**

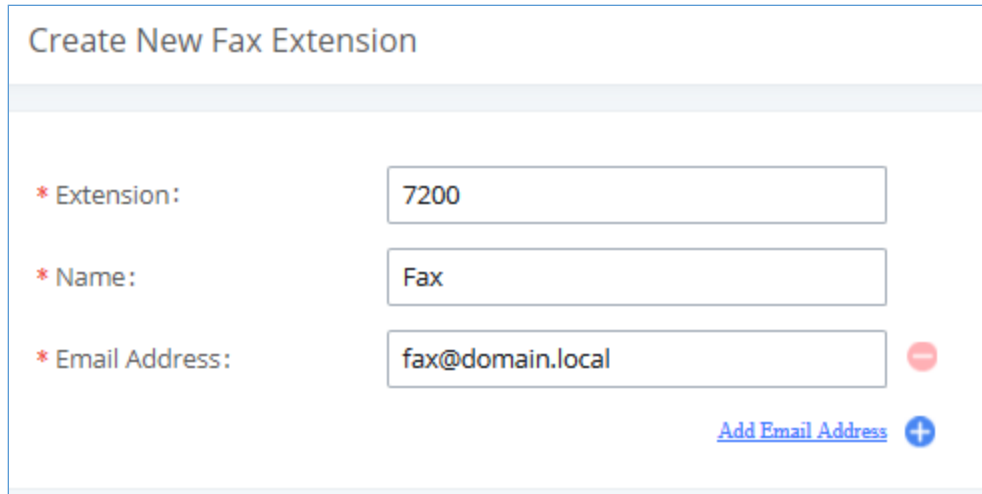
Now the Fax configuration is done. When there is an incoming Fax calling to the PSTN number for the FXO port, it will send the Fax to the Fax machine.



## Example Configuration for Fax-To-Email

The following instructions describe a sample configuration on how to use Fax-to-Email feature on the UCM6300A.

1. Connect PSTN line to the UCM6300A FXO port.
2. Go to UCM6300A Web GUI→**Call Features**→**Fax/T.38** page. Create a new Fax extension.



Create New Fax Extension

\* Extension:

\* Name:

\* Email Address:  ⊖

[Add Email Address](#) ⊕

Figure 131: Create Fax Extension

3. Go to UCM6300A Web GUI→**Extension/Trunk**→**Analog Trunks** page. Create a new analog trunk. Please make sure "Fax Detection" is set to "No".
4. Go to UCM6300A Web GUI→**Extension/Trunk**→**Inbound Routes** page. Create a new inbound route and set the default destination to the Fax extension.



Create New Inbound Rule
Cancel Save

\* Trunks: AnalogTrunks -- FAX

\* Pattern: 

S

Disable This Route:

Alert-info: None

Fax Detection:

Prepend Trunk Name:

Enable Route-Level:

Inbound Mode:

Inbound Route Name:

CallerID Pattern:

Seamless Transfer:

Whitelist:

Ringback Tone: None

Auto Record:

Set CallerID Info:

Inbound Multiple Mode:

Default Mode Mode 1

\* Default Destination: Fax Fax

**Figure 132: Inbound Route to Fax Extension**

- Once successfully configured, the incoming Fax from external Fax machine to the PSTN line number will be converted to PDF+Tiff file and sent to the extension 7200 and email address **fax@domain.local** as attachment.

**Note:** In order for the file to be sent to the email address configured on the external extension, please make sure that the email settings are well configured. Please refer to **[Email Settings]** section.

List of Fax Files

Delete
Clear

NAME	DATE	SIZE	OPTIONS
<input type="checkbox"/> VFAX-7200-20210125-112246-1611570166.49.pdf	2021-01-25 11:22:46 UTC+01:00	1.49 KB	<a href="#">Download</a> <a href="#">Delete</a>
<input type="checkbox"/> VFAX-7200-20210125-112246-1611570166.49.tiff	2021-01-25 11:22:46 UTC+01:00	5.69 KB	<a href="#">Download</a> <a href="#">Delete</a>

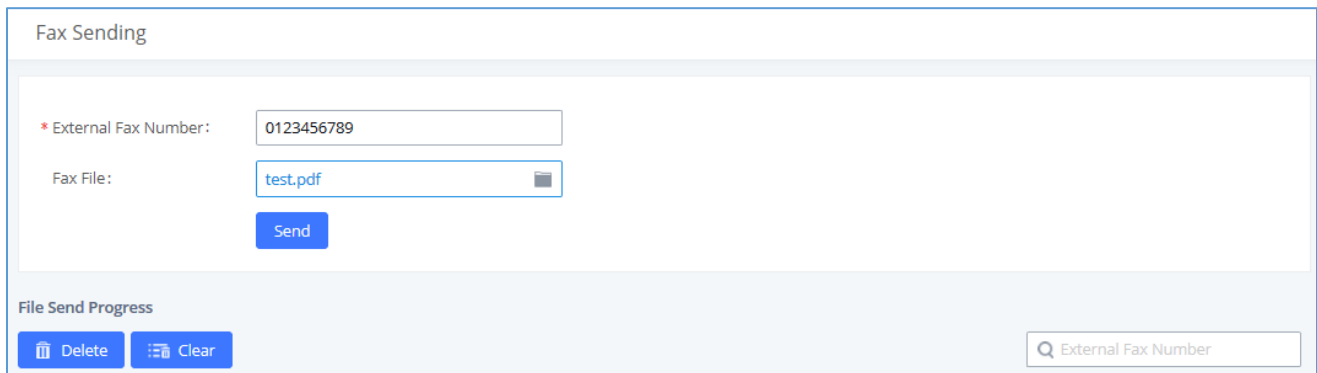
**Figure 133: List of Fax Files**



## FAX Sending UCM6300A

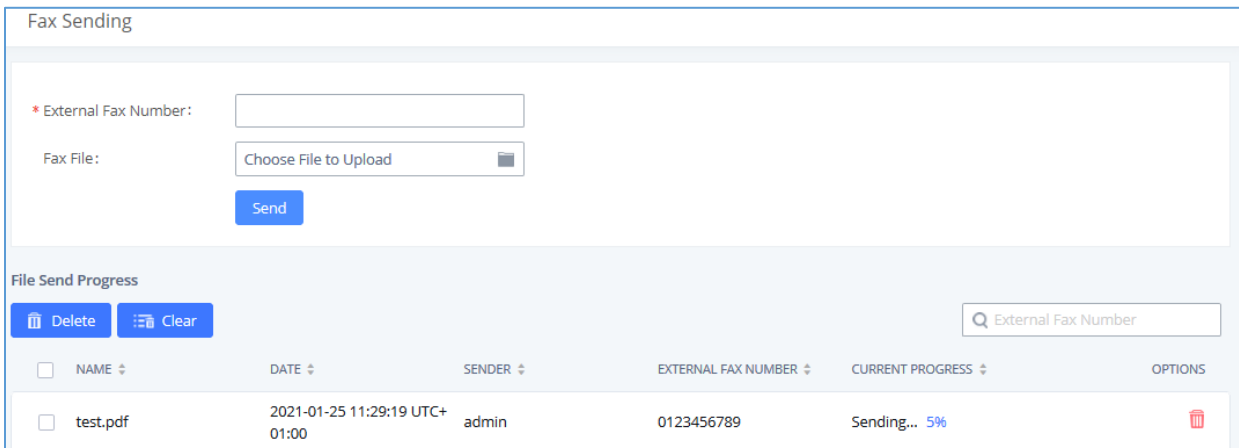
Besides the support of Fax machines, The supports also sending Fax via Web GUI access. This feature can be found on Web GUI→**Value-added Features**→**Fax Sending** page. To send fax, pre-setup for analog trunk and outbound route is required. Please refer to **[ANALOG TRUNKS]**, **[VOIP TRUNKS]** and **[Outbound Routes]** sections for configuring analog trunk and outbound route.


After making sure analog trunk or VoIP Trunk is setup properly and UCM6300A can reach out to PSTN numbers via the trunk, on Fax Sending page, enter the fax number and upload the file to be faxed. Then click on “Send” to start. The progress of sending fax will be displayed in Web GUI. Users can also view the sending history is in the same web page.



**Figure 134: Fax Sending in Web GUI**

After that you can see the ongoing sending operation on the progress bar.



NAME	DATE	SENDER	EXTERNAL FAX NUMBER	CURRENT PROGRESS	OPTIONS
<input type="checkbox"/> test.pdf	2021-01-25 11:29:19 UTC+01:00	admin	0123456789	Sending... 5%	

**Figure 135: Fax Send Progress**

**Note:** Only A3, A4, and B4 paper sizes are supported for the Fax Sending.





## MEETING

With the UCM you can easily create, schedule, manage, and join meeting calls, from your desktop or laptop computer. UCM conferencing uses WebRTC technology, so all the participants don't have to download and install any additional software or plugins. UCM conferencing must be enabled by the administrator for the concerned extensions. The meeting configurations can be accessed under Web GUI→**Call Features**→**Meeting**. In this page, users could enable, set the Basic setting, create, edit, view, manage, delete meeting rooms, and edit the Meeting Schedule.

Below are the UCM meeting specifications supported for each model:

UCM630x Aseries	Number of meeting room	Participant limit
UCM6300A	3	50
UCM6302A	5	75
UCM6304A	7	120
UCM6308A	9	150

### Room

- Click on "Add" to add a new meeting room.
- Click on  to edit the meeting room.
- Click on  to delete the meeting room.

**Table 69: Meeting room Configuration Parameters**

<b>Extension</b>	Configure the meeting number for the users to dial into the meeting. <b>Note:</b> Up to 64 characters.
<b>Password</b>	When configured, the users who would like to join the meeting call must enter this password before accessing the meeting room. <b>Note:</b> <ul style="list-style-type: none"> <li>• Only digits are allowed.</li> <li>• The password has to be at least 4 characters. All repetitive and</li> </ul>



	sequential digits (e.g., 0000, 1111, 1234 and 2345) or common digits (e.g., 111222 and 321321) are not allowed.
<b>Host Password</b>	Configure the Host password.
<b>Privilege</b>	Please select the permission level for outgoing calls.
<b>Allow User Invite</b>	If enabled, participants can invite other users to the meeting.

Meeting Settings contains the following options:

**Table 70: Meeting Settings**

<b>Enable Talk detection</b>	If enabled, the AMI will send the corresponding event when a user starts or ends talking.
<b>DSP Talking Threshold</b>	The time in milliseconds of sound above what the dsp has established as base line silence for a user before a user is considered to be talking. This value affects several operations and should not be changed unless the impact on call quality is fully understood, the default value is 200.
<b>DSP Silence Threshold</b>	The time in milliseconds of sound falling within the dsp has established as base line silence before a user is considered to be silent. This value affects several operations and should not be changed unless the impact on call quality is fully understood, the default value is 2500.
<b>Audio Codec Preference</b>	Configures the preferred codecs for temporary accounts such as meeting participants who joined via link.
<b>Packet Loss Retransmission</b>	Packet Loss Retransmission configuration for temporary accounts (meeting participants without registered extensions who entered the meeting via link)
<b>Jitter Buffer</b>	Select jitter buffer method for temporary accounts such as meeting participants who joined via link. <b>Disable:</b> Jitter buffer will not be used. <b>Fixed:</b> Jitter buffer with a fixed size (equal to the value of "Jitter Buffer Size") <b>Adaptive:</b> Jitter buffer with an adaptive size that will not exceed the value of "Max Jitter Buffer"). <b>NetEQ:</b> Dynamic jitter buffer via NetEQ.



## Meeting Schedule

Meeting Schedule can be found under UCM **Web GUI** → **Call Features** → **Meeting** → **Meeting Schedule**. Users can create, edit, view, and delete a Meeting Schedule.

- Click on “Add” to add a new Meeting Schedule.
- Click on the scheduled meeting to edit or delete the event.

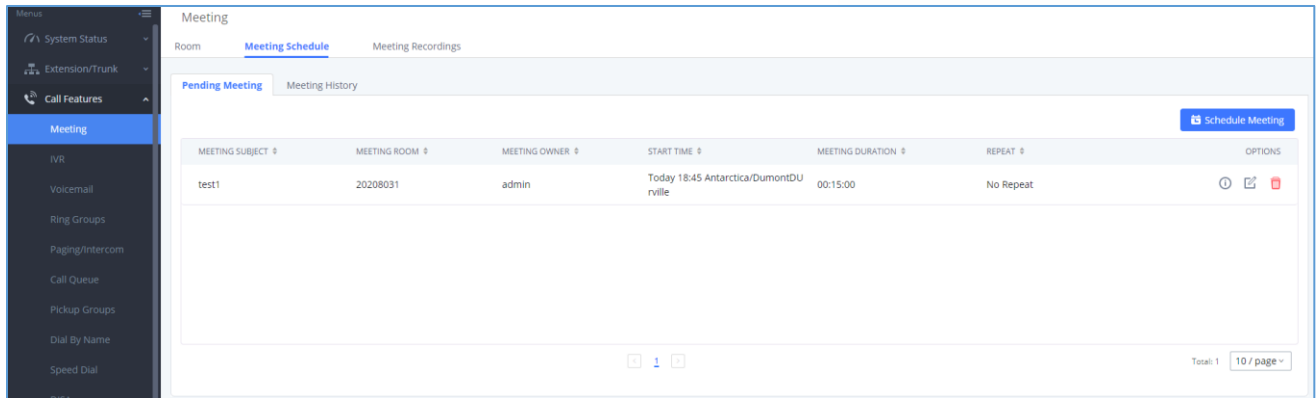
Table 71: Meeting Schedule Parameters

Schedule Options	
<b>Meeting Subject</b>	Configure the name of the scheduled meeting. Letters, digits, _ and - are allowed.
<b>Meeting Room</b>	Select a meeting room for this scheduled meeting.
<b>Password</b>	Configure Meeting login password.
<b>Time</b>	Set the beginning date and duration of this scheduled meeting. Please be aware to avoid time conflicts in the same meeting room.
<b>Time Zone</b>	Configure the time zone
<b>Host</b>	Configure the meeting's host
<b>Host password</b>	Configure the meeting's host password
<b>Repeat</b>	Choose when to repeat a scheduled meeting.
<b>Email Reminder (m)</b>	Email reminders will be sent out x minutes prior to the start of the meeting. Valid range is 5-1440. 60 is the default value. 0 indicates not to send out email reminders for the meeting.
<b>Allow User Invite</b>	If enabled, participants can invite other users to the meeting.
<b>Call Participants</b>	If enabled, invited participants will be called when the meeting starts.
<b>Invitees</b>	Select the participants to invite to the meeting. Enter either extension numbers or email addresses.
<b>Description</b>	Set a description of scheduled meeting.

Once created, at the scheduled meeting time, UCM630xA will send INVITE to the extensions that have been selected for meeting.





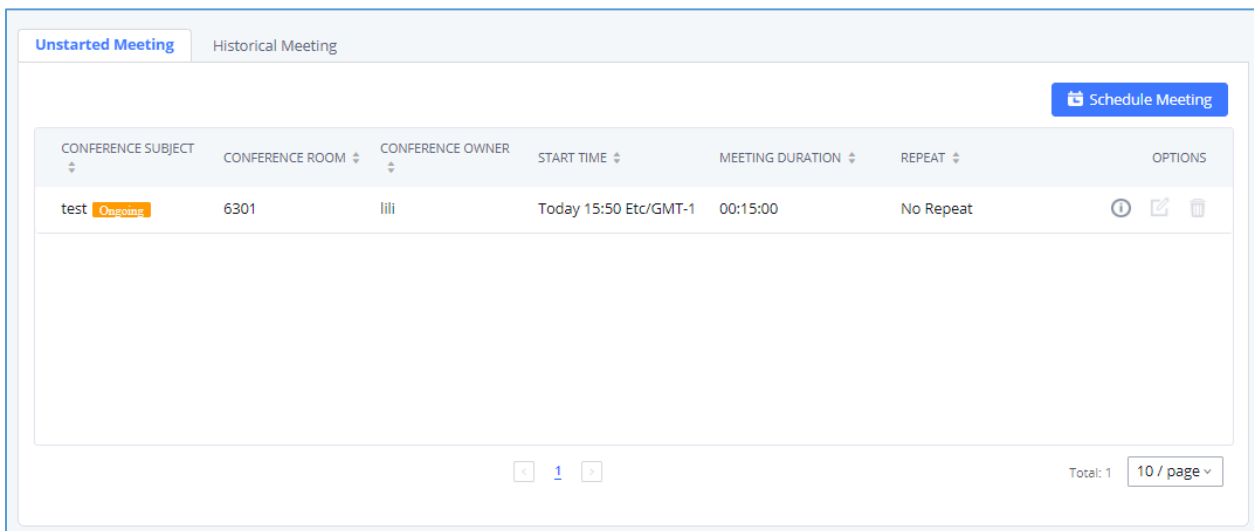


Meeting Schedule interface showing a pending meeting. The table contains one entry:

MEETING SUBJECT	MEETING ROOM	MEETING OWNER	START TIME	MEETING DURATION	REPEAT	OPTIONS
test1	20208031	admin	Today 18:45 Antarctica/DumontDU rville	00:15:00	No Repeat	[Info] [Edit] [Delete]

**Figure 136: Meeting Schedule**

Once the meeting starts, it will be displayed under **Unstarted Meeting** with an “Ongoing” status, as displayed vidbelow.






Unstarted Meeting interface showing an ongoing meeting. The table contains one entry:

CONFERENCE SUBJECT	CONFERENCE ROOM	CONFERENCE OWNER	START TIME	MEETING DURATION	REPEAT	OPTIONS
test <span style="background-color: orange; color: white; padding: 2px;">Ongoing</span>	6301	lili	Today 15:50 Etc/GMT-1	00:15:00	No Repeat	[Info] [Edit] [Delete]

**Figure 137: Meeting Scheduled-Ongoing**

Once the meeting is finished, the meeting will be displayed under Historical meeting as below:



Unstarted Meeting		Historical Meeting					
CONFERENCE SUBJECT	CONFERENCE ROOM	CONFERENCE OWNER	START TIME	MEETING DURATION	REPEAT	OPTIONS	
test	6301	lili	2021-01-20 15:50	00:15:00	No Repeat	  	

Total: 1    10 / page

**Figure 138: Meeting Scheduled-Completed**

In addition, once the meeting ends, the system will send a meeting report email to the host including PDF file where he/she can view the meeting, participant information, device type and trend graph of participant levels



**Notes:**

- Conferencing can be resource-intensive and may cause performance issues with the UCM when used.
- To ensure the best experience, please use Google Chrome (v67 or higher) or Mozilla Firefox (v60).

**Meeting Recordings**

The UCM630xA allows users to record the meeting call and retrieve the recording from Web GUI → **Call Features** → **Meeting** → **Meeting Recordings**.

To record the meeting call, when the meeting room is in idle, enable "Record Meeting" from the meeting room configuration dialog. Save the setting and apply the change. When the meeting call starts, the call will be automatically recorded in .wav format.

The recording files will be listed as below once available. Users could click on  to download the recording or click on  to delete the recording. Users could also delete all recording files by clicking on "Delete All Recording Files" or delete multiple recording files at once by clicking on "Delete" after selecting the recording files.



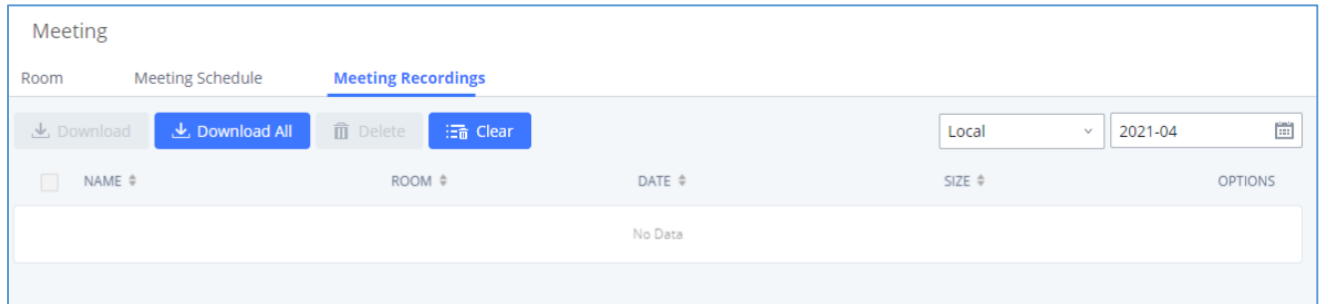


Figure 139: Meeting Recordings

## Wave WebRTC Video Calling

Web calls and conferencing can now be achieved through the UCM's new WebRTC page. To get started with this new feature, please make sure to:

1. Navigate to **Value-Added Features** → **WebRTC** and enable WebRTC support.

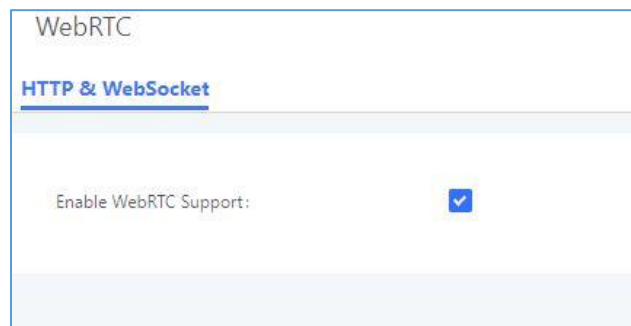


Figure 140: Enabling WebRTC Feature

2. Enable the WebRTC on the extensions that would use this feature under **Extension / Trunk** → **Extensions** by editing the concerned extensions.



**Other Settings**

Ring Timeout:

\* Skip Trunk Auth:

Support Hot-desking Mode:

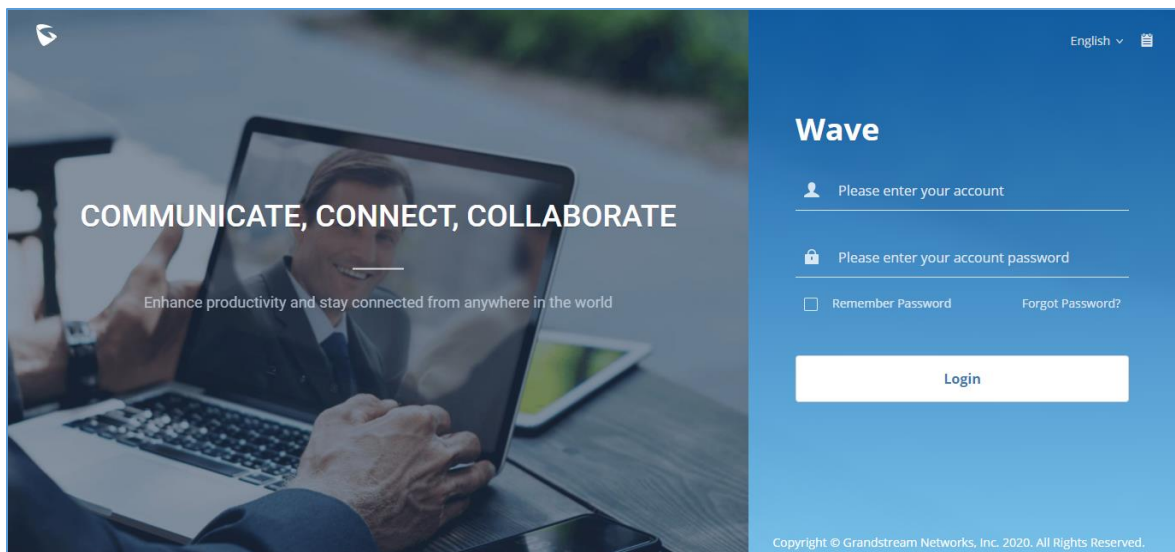
**Enable WebRTC Support:**

Call Duration Limit:

Call Waiting:

**Figure 141: Enabling WebRTC on Extensions**

The UCM offers the possibility to login to an extension via Grandstream Wave Portal using user portal password in addition to SIP registration password, where it offers a sleek interface to host meetings, receive email reminders for scheduled meetings, manage contacts, initiate calls, call transfer, chat functionality and more. Access the page by entering the UCM's server address and port which is 8090. (e.g. <https://my.ucm.com:8090>).



**Figure 142: Grandstream Wave Interface**

For more details about the Wave Web, please refer to the following guides:

[https://www.grandstream.com/hubfs/Product\\_Documentation/UCM630xA\\_Wave\\_Web\\_administration\\_guide.pdf?hsLang=en](https://www.grandstream.com/hubfs/Product_Documentation/UCM630xA_Wave_Web_administration_guide.pdf?hsLang=en)

[https://www.grandstream.com/hubfs/Product\\_Documentation/GS\\_Wave\\_Web\\_user\\_guide.pdf](https://www.grandstream.com/hubfs/Product_Documentation/GS_Wave_Web_user_guide.pdf)

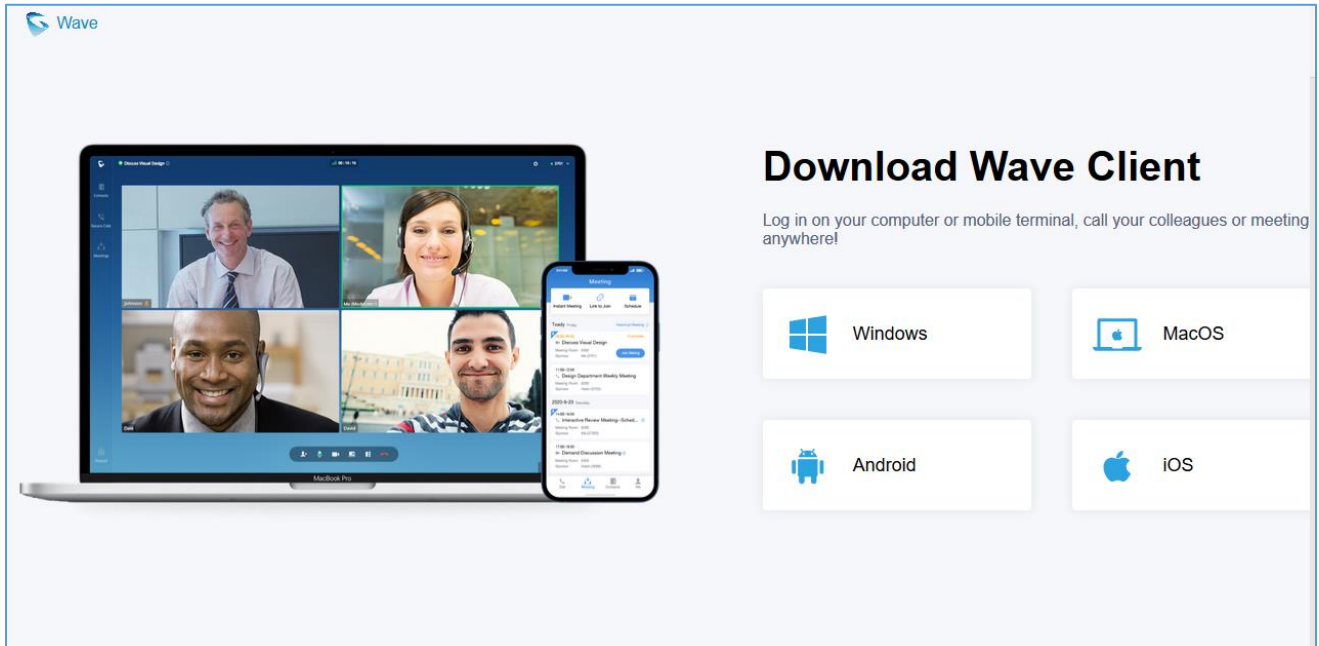


Grandstream Wave is also available in Android/iOS version where it can be downloaded from Google Play or AppStore. For more details, please refer to the following guide:

[https://www.grandstream.com/hubfs/Product\\_Documentation/GSWave\\_mobile\\_app\\_User\\_Guide.pdf](https://www.grandstream.com/hubfs/Product_Documentation/GSWave_mobile_app_User_Guide.pdf)

Desktop version is also supported in Windows and MacOS and can be downloaded from:

<https://fw.gdms.cloud/wave/download/>



**Download Wave Client**

Log in on your computer or mobile terminal, call your colleagues or meeting anywhere!

Windows MacOS

Android iOS



**Figure 143: Download Wave Client**



# IVR

## Configure IVR

IVR configurations can be accessed under the UCM630xA Web GUI → **Call Features** → **IVR**. Users could create, edit, view, and delete an IVR.

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

**Create New IVR**

[Basic Settings](#)    [Key Pressing Events](#)

---

• **Name:**

• **Extension:**

Dial Trunk:

Auto Record:

Dial Other Extensions:  All  Extension  Audio Conference  Video Conference  Call Queue  
 Ring Group  Paging/Intercom Groups  Voicemail Groups  Fax Extension  
 Dial By Name


• **IVR Black/Whitelist:**

Replace Display Name:

Return to IVR Menu:

Alert-info:

• **Prompt:**  [Upload Audio File](#)

[Add Prompt](#) 

• **Digit Timeout (s):**

• **Response Timeout:**

• **Response Timeout Prompt:**  [Upload Audio File](#)

• **Invalid Input Prompt:**  [Upload Audio File](#)

• **Response Timeout Prompt Repeats:**

• **Invalid Input Prompt Repeats:**

Language:

**Figure 144: Create New IVR**



**Table 72: IVR Configuration Parameters**

Basic Settings	
<b>Name</b>	Configure the name of the IVR. Letters, digits, _ and - are allowed.
<b>Extension</b>	Enter the extension number for users to access the IVR.
<b>Dial Trunk</b>	If enabled, all callers to the IVR can use trunk. The permission must be configured for the users to use the trunk first. The default setting is "No".
<b>Auto Record</b>	If enabled, calls to this IVR will automatically be recorded.
<b>Permission</b>	<p>Assign permission level for outbound calls if "Dial Trunk" is enabled. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level.</p> <p>The default setting is "Internal". If the user tries to dial outbound calls after dialing into the IVR, the UCM630xA will compare the IVR's permission level with the outbound route's privilege level.</p> <p>If the IVR's permission level is higher than (or equal to) the outbound route's privilege level, the call will be allowed to go through.</p>
<b>Dial Other Extensions</b>	<p>This controls the destination that can be reached by the external caller via the inbound route. The available destinations are:</p> <ul style="list-style-type: none"> <li>• Extension</li> <li>• Meeting</li> <li>• Call Queue</li> <li>• Ring Group</li> <li>• Paging/Intercom Groups</li> <li>• Voicemail Groups</li> <li>• Dial by Name</li> <li>• All</li> </ul>
<b>IVR Black/Whitelist</b>	If enabled only numbers inside of the Whitelist or outside of the Blacklist can be called from IVR.
<b>Internal Black/Whitelist</b>	Contain numbers, either of Blacklist or Whitelist.
<b>External Black/Whitelist</b>	This feature can be used only when Dial Trunk is enabled, it contains external numbers allowed or denied calling from the IVR, the allowed format is the following: Number1, number2, number3...
<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with IVR name.
<b>Return to IVR Menu</b>	If enabled and if a call to an extension fails, the caller will be redirected to the IVR menu.
<b>Alert Info</b>	When present in an INVITE request, the alert-Info header field specifies and alternative ring tone to the UAS.



<b>Prompt</b>	Select an audio file to play as the welcome prompt for the IVR. Click on "Prompt" to add additional audio file under Web GUI→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Custom Prompt</b> .
<b>Digit Timeout</b>	Configure the timeout between digit entries. After the user enters a digit, the user needs to enter the next digit within the timeout. If no digit is detected within the timeout, the UCM630xA will consider the entries complete. The default timeout is 3s.
<b>Response Timeout</b>	After playing the prompts in the IVR, the UCM630xA will wait for the DTMF entry within the timeout (in seconds). If no DTMF entry is detected within the timeout, a timeout prompt will be played. The default setting is 10 seconds.
<b>Response Timeout Prompt</b>	Select the prompt message to be played when timeout occurs.
<b>Invalid Input Prompt</b>	Select the prompt message to be played when an invalid extension is pressed.
<b>Response Timeout Prompt Repeats</b>	Configure the number of times to repeat the prompt if no DTMF input is detected. When the loop ends, it will go to the timeout destination if configured, or hang up. The default setting is 3.
<b>Invalid Input Prompt Repeats</b>	Configure the number of times to repeat the prompt if the DTMF input is invalid. When the loop ends, it will go to the invalid destination if configured, or hang up. The default setting is 3.
<b>Language</b>	Select the voice prompt language to be used for this IVR. The default setting is "Default" which is the selected voice prompt language under Web GUI→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> . The dropdown list shows all the current available voice prompt languages on the UCM630xA. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under Web GUI→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> .

### Key Pressing Events

<b>Key Press Event:</b>	Select the event for each key pressing for 0-9, *, Timeout and Invalid. The event options are:
<b>Press 0</b>	
<b>Press 1</b>	<ul style="list-style-type: none"> <li>• Extension</li> </ul>
<b>Press 2</b>	<ul style="list-style-type: none"> <li>• Voicemail</li> </ul>
<b>Press 3</b>	<ul style="list-style-type: none"> <li>• Meeting Rooms</li> </ul>
<b>Press 4</b>	<ul style="list-style-type: none"> <li>• Voicemail Group</li> </ul>
<b>Press 5</b>	<ul style="list-style-type: none"> <li>• IVR</li> </ul>
<b>Press 6</b>	<ul style="list-style-type: none"> <li>• Ring Group</li> </ul>
<b>Press 7</b>	<ul style="list-style-type: none"> <li>• Queues</li> </ul>
<b>Press 8</b>	<ul style="list-style-type: none"> <li>• Page Group</li> </ul>





<b>Press 9</b> <b>Press *</b>	<ul style="list-style-type: none"> <li>• Custom Prompt</li> <li>• Hangup</li> <li>• DISA</li> <li>• Dial by Name</li> <li>• External Number</li> <li>• Callback</li> </ul>
<b>Timeout</b>	When exceeding the number of defined answer timeout, IVR will enter the configured event when timeout. If not configured, then it will Hangup.
<b>Invalid</b>	Configure the destination when the Invalid Repeat Loop is done.

Edit IVR: OfficeOpen

Basic Settings      Key Pressing Events

Press 0:	Extension ▾	2000 ▾
Press 1:	IVR ▾	Sales ▾
Press 2:	IVR ▾	Support ▾
Press 3:	Select an Op... ▾	
Press 4:	Select an Op... ▾	
Press 5:	Select an Op... ▾	
Press 6:	Select an Op... ▾	
Press 7:	Select an Op... ▾	
Press 8:	Select an Op... ▾	
Press 9:	Select an Op... ▾	
Press *:	Select an Op... ▾	
Timeout:	Custom Pro... ▾	goodbye ▾
Invalid:	Custom Pro... ▾	goodbye ▾

**Figure 145: Key Pressing Events**

## Black/White List in IVR

In some scenarios, the IPPBX administrator needs to restrict the extensions that can be reached from IVR. For example, the company CEO and directors prefer only receiving calls transferred by the secretary, some special extensions are used on IP surveillance end points which should not be reached from external calls via

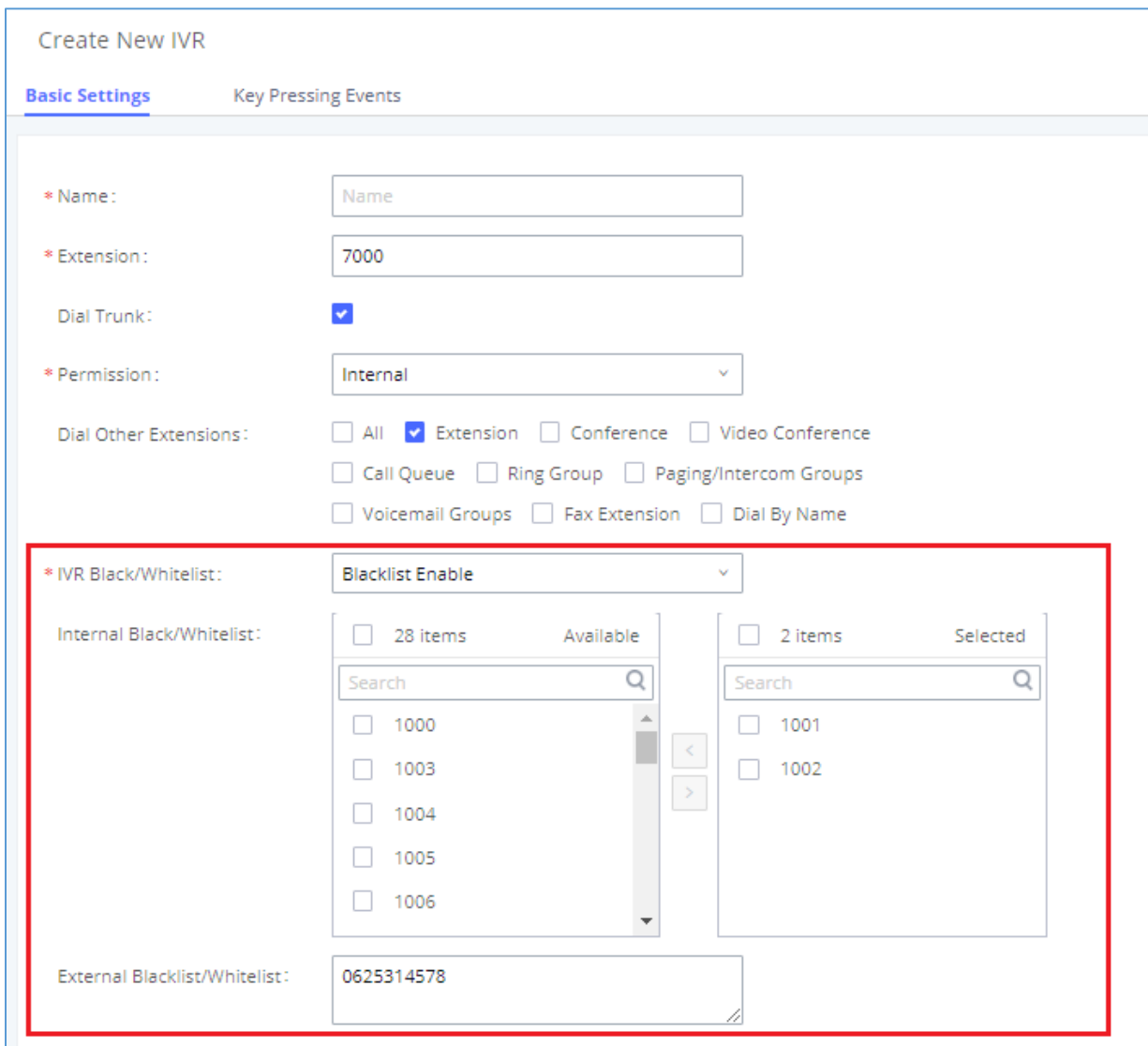


IVR for privacy reason. UCM has now added blacklist and whitelist in IVR settings for users to manage this.

**Note:** up to 500 extensions are allowed on the back/whitelist.

To use this feature, log in UCM Web GUI and navigate to **Call Features**→**IVR**→**Create/Edit IVR: IVR Black/Whitelist**.

- If the user selects “Blacklist Enable” and adds extension in the list, the extensions in the list will not be allowed to be reached via IVR.
- If the user selects “Whitelist Enable” and adds extension in the list, only the extensions in the list can be allowed to be reached via IVR.



**Create New IVR**

**Basic Settings**      Key Pressing Events

\* Name:

\* Extension:

Dial Trunk:

\* Permission:

Dial Other Extensions:  All  Extension  Conference  Video Conference  
 Call Queue  Ring Group  Paging/Intercom Groups  
 Voicemail Groups  Fax Extension  Dial By Name

\* IVR Black/Whitelist:

Internal Black/Whitelist:  28 items Available       2 items Selected

Search

1000     1001  
 1003     1002  
 1004  
 1005  
 1006

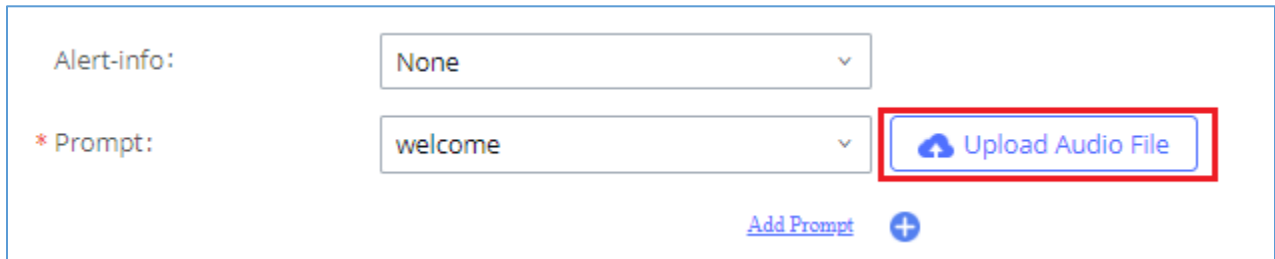
External Blacklist/Whitelist:

**Figure 146: Black/Whitelist**



## Create Custom Prompt

To record new IVR prompt or upload IVR prompt to be used in IVR, click on “Upload Audio File” next to the “Welcome Prompt” option and the users will be redirected to Custom Prompt page. Or users could go to Web GUI→**PBX Settings**→**Voice Prompt**→**Custom Prompt** page directly.



Alert-info:  ▼

\* Prompt:  ▼ [Upload Audio File](#)

[Add Prompt](#) +

Figure 147: Click on Prompt to Create IVR Prompt

Once the IVR prompt file is successfully added to the UCM630xA, it will be added into the prompt list options for users to select in different IVR scenarios.



## LANGUAGE SETTINGS FOR VOICE PROMPT

The UCM630xA supports multiple languages in Web GUI as well as system voice prompt. Currently, there are 16 languages supported in system voice prompt: **English (United States), Arabic, Chinese, Dutch, English (United Kingdom), French, German, Greek, Hebrew, Italian, Polish, Portuguese, Russian, Spanish, Catalan, Swedish and Turkish.**

English (United States) and Chinese voice prompts are built in with the UCM630xA already. The other languages provided by Grandstream can be downloaded and installed from the UCM630xA Web GUI directly. Additionally, users could customize their own voice prompts, package them and upload to the UCM630xA.

Language settings for voice prompt can be accessed under Web GUI→**PBX Settings**→**Voice Prompt**→**Language Settings**.

### Download and Install Voice Prompt Package

To download and install voice prompt package in different languages from UCM630xA Web GUI, click on "Add Voice Prompt Package" button.

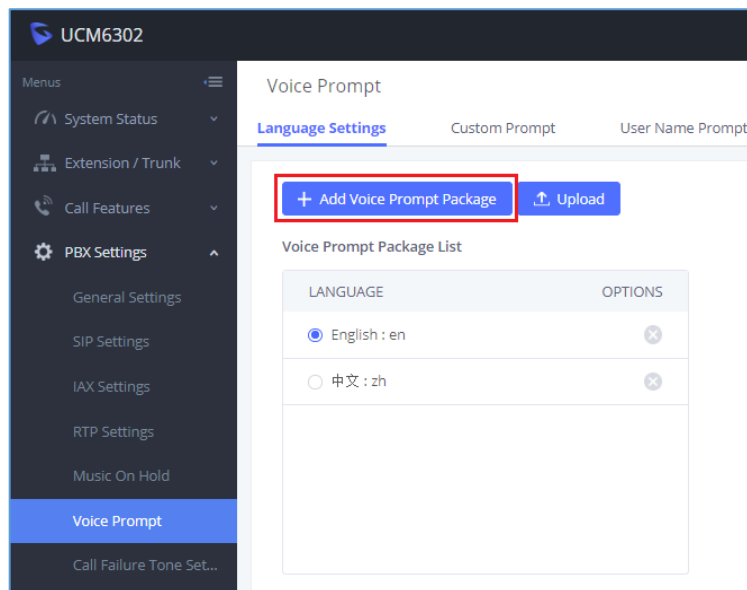



Figure 148: Language Settings for Voice Prompt




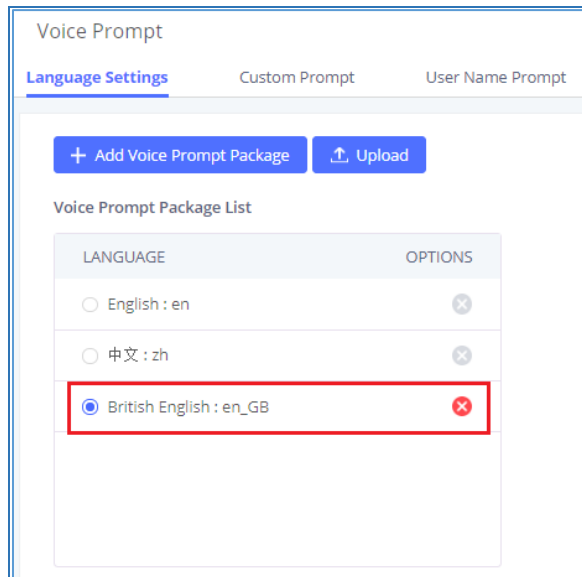
A new dialog window of voice prompt package list will be displayed. Users can see the version number (latest version available V.S. current installed version), package size and options to upgrade or download the language.



VOICE PROMPT PACKAGE LIST	VERSION (REMOTE / LOCAL)	SIZE	OPTIONS
British English	1.9/-	4.2M	↓
Deutsch	1.8/-	4.2M	↓
English	1.11/1.8	6.0M	↑
Español	1.10/-	4.4M	↓
Español(Català)	1.8/-	3.1M	↓
Español(Español)	1.8/-	4.2M	↓
Ελληνικά	1.8/-	4.4M	↓
Français	1.8/-	4.1M	↓
Italiano	1.8/-	4.0M	↓

**Figure 149: Voice Prompt Package List**

Click on  to download the language to the UCM630xA. The installation will be automatically started once the downloading is finished.



Voice Prompt

Language Settings    Custom Prompt    User Name Prompt

+ Add Voice Prompt Package    Upload

Voice Prompt Package List

LANGUAGE	OPTIONS
<input type="radio"/> English : en	×
<input type="radio"/> 中文 : zh	×
<input checked="" type="radio"/> British English : en_GB	×

**Figure 150: New Voice Prompt Language Added**



A new language option will be displayed after successfully installed. Users then could select it to apply in the UCM630xA system voice prompt or delete it from the UCM630xA.

## Customize Specific Prompt

On the UCM630xA, if the user needs to replace some specific customized prompt, the user can upload a single specific customized prompt from Web GUI→**PBX Settings**→**Voice Prompt**→**Language Settings** and click on “**Upload**” instead of the entire language pack.

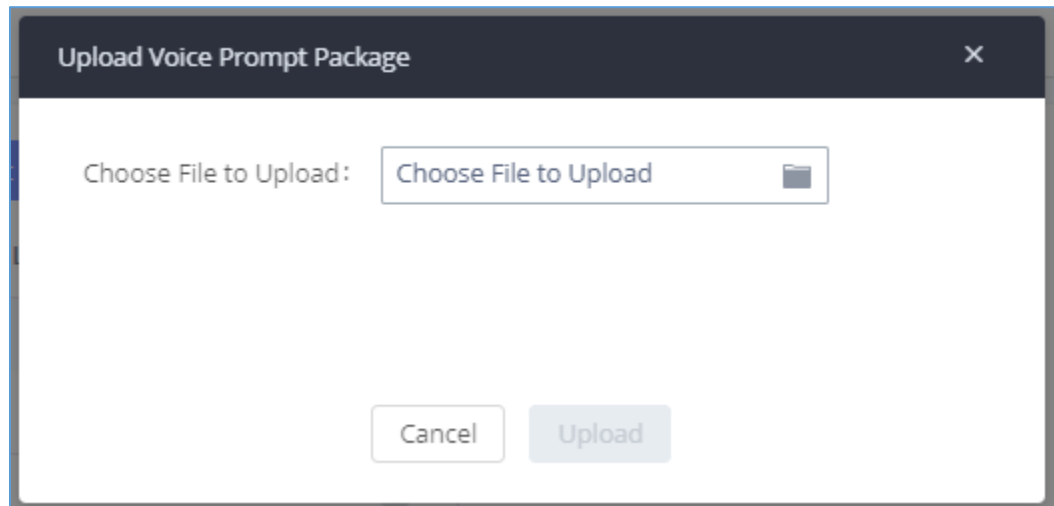


Figure 151: Upload Single Voice Prompt for Entire Language Pack



## Username Prompt Customization

There are two ways to customize/set new username prompt:

### Upload Username Prompt File from Web GUI

1. First, Users should have a pre-recorded file respecting the following format:
  - PCM encoded / 16 bits / 8000Hz mono.
  - In .tar/.tar.gz/.tgz format
  - File size under 30M.
  - Filename must be set as the extension number with 18 characters max. For example, the recorded file name 1000.wav will be used for extension 1000.
2. Go under web GUI **PBX Settings** → **Voice Prompt** → **Username Prompt** and click on “**Upload**” button.



3. Select the recorded file to upload it and press Save and Apply Settings.
- Click on  to record again the username prompt.
  - Click on to play recorded username prompt.
  - Select username prompts and press  to delete specific file or select multiple files for deletion using the button "**Delete**".

### **Record Username via Voicemail Menu**

The second option to record username is using voicemail menu, please follow below steps:

- Dial \*98 to access the voicemail
- After entering the desired extension and voicemail password, dial "0" to enter the recordings menu and then "3" to record a name.

Another option is that each user can record their own name by following below steps:

- The user dials \*97 to access his/her voicemail
- After entering the voicemail password, the user can press "0" to enter the recordings menu and then "3" to record his name.



# VOICEMAIL

## Configure Voicemail

If the voicemail is enabled for UCM630xA extensions, the configurations of the voicemail can be globally set up and managed under Web GUI→**Call Features**→**Voicemail**.

* Max Greeting Time (s):	<input type="text" value="60"/>
Dial "0" for Operator:	<input type="checkbox"/>
Operator Type:	<input type="text" value="Extension"/>
Operator Extension:	<input type="text" value="None"/>
* Max Messages Per Folder:	<input type="text" value="50"/>
Max Message Time:	<input type="text" value="15 minutes"/>
Min Effective Message Time:	<input type="text" value="3 seconds"/>
Announce Message Caller-ID:	<input type="checkbox"/>
Announce Message Duration:	<input type="checkbox"/>
Play Envelope:	<input checked="" type="checkbox"/>
Play Most Recent First:	<input type="checkbox"/>
Allow User Review:	<input type="checkbox"/>
Voicemail Remote Access:	<input type="checkbox"/>
Forward Voicemail to Peered UCMs:	<input type="checkbox"/>
Voicemail Password:	<input type="text"/>
Format:	<input type="text" value="GSM"/>

Figure 152: Voicemail Settings





**Table 73: Voicemail Settings**

<b>Max Greeting Time (s)</b>	Configure the maximum number of seconds for the voicemail greeting. The default setting is 60 seconds.
<b>Dial '0' For Operator</b>	If enabled, the caller can press 0 to exit the voicemail application and connect to the configured operator's extension.
<b>Operator Type</b>	Configure the operator type; either an extension or a ring group.
<b>Operator Extension</b>	Select the operator extension, which will be dialed when users press 0 to exit voicemail application. The operator extension can also be used in IVR.
<b>Max Messages Per Folder</b>	Configure the maximum number of messages per folder in users' voicemail. The valid range 10 to 1000. The default setting is 50.
<b>Max Message Time</b>	<p>Select the maximum duration of the voicemail message. The message will not be recorded if the duration exceeds the max message time. The default setting is 15 minutes. The available options are:</p> <ul style="list-style-type: none"> <li>• 1 minute</li> <li>• 2 minutes</li> <li>• 5 minutes</li> <li>• 15 minutes</li> <li>• 30 minutes</li> <li>• Unlimited</li> </ul>
<b>Min Effective Message Time</b>	<p>Configure the minimum duration (in seconds) of a voicemail message. Messages will be automatically deleted if the duration is shorter than the Min Message Time. The default setting is 3 seconds. The available options are:</p> <ul style="list-style-type: none"> <li>• No minimum</li> <li>• 1 second</li> <li>• 2 seconds</li> <li>• 3 seconds</li> <li>• 4 seconds</li> <li>• 5 seconds</li> </ul> <p><b>Note:</b> Silence and noise duration are not counted in message time.</p>
<b>Announce Message Caller-ID</b>	If enabled, the caller ID of the user who has left the message will be announced at the beginning of the voicemail message. The default setting is "No".
<b>Announce Message Duration</b>	If enabled, the message duration will be announced at the beginning of the voicemail message. The default setting is "No".
<b>Play Envelope</b>	If enabled, a brief introduction (received time, received from, and etc.) of each message will be played when accessed from the voicemail application. The default setting is "Yes".
<b>Play Most Recent First</b>	If enabled, it will play the most recent message first.



<b>Allow User Review</b>	If enabled, users can review the message following the IVR before sending.
<b>Voicemail Remote Access</b>	<p>If enabled, external callers routed by DID and reaching VM will be prompted by the UCM with 2 options:</p> <ul style="list-style-type: none"> <li>• <b>Press 1 to leave a message.</b> To leave a message for the extension reached by DID.</li> <li>• <b>Press 2 to access voicemail management system.</b> This will allow caller to access any extension VM after entering extension number and its VM password.</li> </ul> <p><b>Note:</b> This option applies to inbound call routed by DID only. The default setting is “Disabled”.</p>
<b>Forward Voicemail to Peered UCMs</b>	Enables the forwarding of voicemail to remote extensions on peered SIP trunks. The default setting is “Disabled”.
<b>Voicemail Password</b>	Configures the default voicemail password that will be used when an extension is reset.
<b>Format</b>	Warning: WAV files take up significantly more storage space than GSM files.

**Note:** Resetting an extension will reset Voicemail Password, Send Voicemail to Email, and Keep Voicemail after Emailing values to default. Previous custom voicemail prompts and messages will be deleted.

## Access Voicemail

If the voicemail is enabled for UCM630xA extensions, the users can dial the voicemail access number (by default \*97) to access their extension’s voicemail. The users will be prompted to enter the voicemail password and then can enter digits from the phone keypad to navigate in the IVR menu for different options.

Otherwise the user can dial the voicemail access code (by default \*98) followed by the extension number and password in order to access to that specific extension’s voicemail.

Table 74: Voicemail IVR Menu

Main Menu	Sub Menu 1	Sub Menu 2
<b>1 – New messages</b>	3 - Advanced options	1 - Send a reply 2 - Call the person who sent this message 3 - Hear the message envelop 4 - Leave a message * - Return to the main menu
	5 - Repeat the current message	



	7 - Delete this message	
	8 - Forward the message to another user	
	9 - Save	
	* - Help	
	# - Exit	
<b>2 - Change folders</b>	0 - New messages	
	1 - Old messages	
	2 - Work messages	
	3 - Family messages	
	4 - Friend messages	
	# - Cancel	
<b>3 - Advanced options</b>	1 - Send a reply	
	2 - Call the person who sent this message	
	3 - Hear the message envelop	
	4 - Leave a message	
	* - Return to the main menu	
<b>0 - Mailbox options</b>	1 - Record your unavailable message	1 - Accept this recording
		2 - Listen to it
		3 - Re-record your message
	2 - Record your busy message	1 - Accept this recording
		2 - Listen to it
		3 - Re-record your message
	3 - Record your name	1 - Accept this recording
		2 - Listen to it
		3 - Re-record your message
	4 - Record temporary greeting	1 - Accept this recording
		2 - Listen to it
		3 - Re-record your message
	5 - Change your password	
* - Return to the main menu		

## Leaving Voicemail

If an extension has voicemail enabled under basic settings “**Extension/Trunk → Extensions → Basic Settings**”



and after a ring timeout or user not available, the caller will be automatically redirected to the voicemail in order to leave a message on which case they can press # in order to submit the message.

In case if the caller is calling from an internal extension, they will be directly forwarded to the extension's voicemail box. But if the caller is calling from outside the system and the incoming call is routed by DID to the destination extension, then the caller will be prompted with the choice to either press 1 to access voicemail management or press 2 to leave a message for the called extension. This feature could be useful for remote voicemail administration.

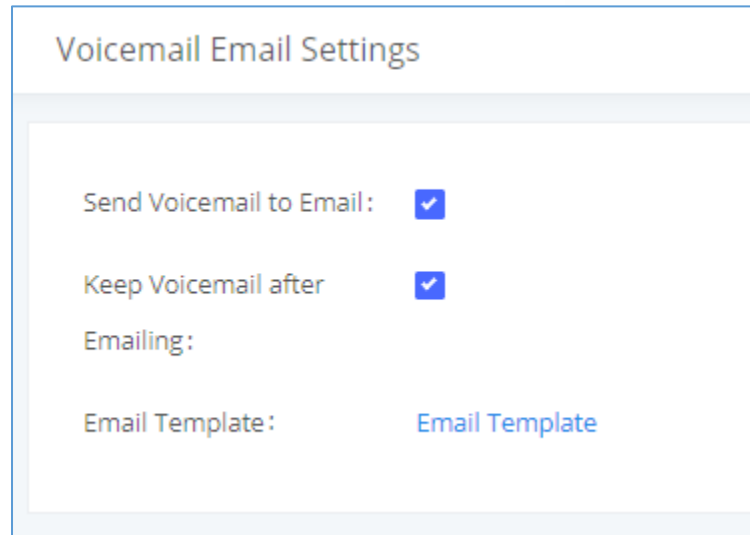
## Voicemail Email Settings

The UCM630xA can be configured to send the voicemail as attachment to Email. Click on "Voicemail Email Settings" button to configure the Email attributes and content.

**Table 75: Voicemail Email Settings**

<b>Send Voicemail to Email</b>	If enabled, voicemail will be sent to the user's email address. Note: SMTP server must be configured to use this option.
<b>Keep Voicemail after Emailing</b>	Enable this option if you want to keep recording files after the Email is sent. The default setting is Enable.
<b>Email Template</b>	<p>Fill in the "Subject:" and "Message:" content, to be used in the Email when sending to the user.</p> <p>The template variables are:</p> <ul style="list-style-type: none"> <li>• \t: TAB</li> <li>• \${VM_NAME}: Recipient's first name and last name</li> <li>• \${VM_DUR}: The duration of the voicemail message</li> <li>• \${VM_MAILBOX}: The recipient's extension</li> <li>• \${VM_CALLERID}: The caller ID of the person who has left the message</li> <li>• \${VM_MSGNUM}: The number of messages in the mailbox</li> <li>• \${VM_DATE}: The date and time when the message is left</li> </ul>



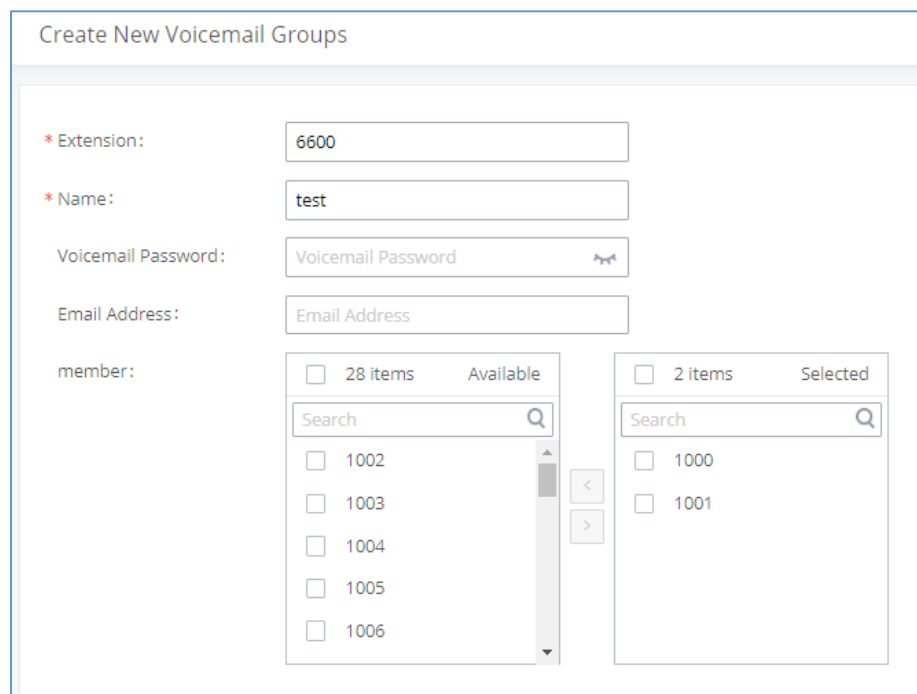


**Figure 153: Voicemail Email Settings**

Click on "Email Template" button to view the default template as an example.

## Configure Voicemail Group

The UCM630xA supports voicemail group and all the extensions added in the group will receive the voicemail to the group extension. The voicemail group can be configured under Web GUI → **Call Features** → **Voicemail** → **Voicemail Group**. Click on "Add" to configure the group.



**Figure 154: Voicemail Group**



**Table 76: Voicemail Group Settings**

<b>Extension</b>	Enter the Voicemail Group Extension. The voicemail messages left to this extension will be forwarded to all the voicemail group members.
<b>Name</b>	Configure the Name to identify the voicemail group. Letters, digits, _ and - are allowed.
<b>Voicemail Password</b>	Configure the voicemail password for the users to check voicemail messages.
<b>Email Address</b>	Configure the Email address for the voicemail group extension.
<b>Member</b>	Select available mailboxes from the left list and add them to the right list. The extensions need to have voicemail enabled to be listed in available mailboxes list.

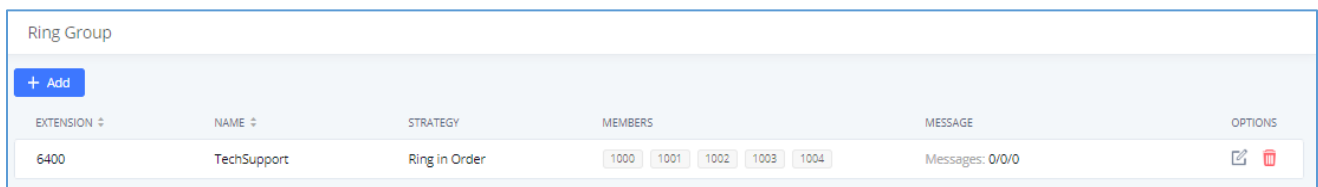


# RING GROUP

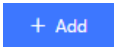


The UCM630xA supports ring group feature with different ring strategies applied to the ring group members. This section describes the ring group configuration on the UCM630xA.

## Configure Ring Group





Ring group settings can be accessed via Web GUI → Call Features → Ring Group.



**Figure 155: Ring Group**

- Click on  to add ring group.
- Click on  to edit the ring group. The following table shows the ring group configuration parameters.
- Click on  to delete the ring group.

**Table 77: Ring Group Parameters**

<b>Ring Group Name</b>	Configure ring group name to identify the ring group. Letters, digits, _ and – are allowed.
<b>Extension</b>	Configure the ring group extension.
<b>Members</b>	Select available users from the left side to the ring group member list on the right side. Click on   to arrange the order.
<b>LDAP Phonebook</b>	Select available remote users from the left side to the ring group member list on the right side. Click on   to arrange the order. Note: LDAP Sync must be enabled first.
<b>Ring Strategy</b>	Select the ring strategy. The default setting is “Ring in order”. <ul style="list-style-type: none"> <li>• <b>Ring simultaneously.</b> Ring all the members at the same time when there is incoming call to the ring group extension. If any of the member answers the call, it will stop</li> </ul>



	<p>ringing.</p> <ul style="list-style-type: none"> <li>• <b>Ring in order.</b> Ring the members with the order configured in ring group list. If the first member does not answer the call, it will stop ringing the first member and start ringing the second member.</li> </ul>
<b>Music On Hold</b>	Select the “Music On Hold” Class of this Ring Group, “Music On Hold” can be managed from the “Music On Hold” panel on the left.
<b>Custom Prompt</b>	<p>This option is to set a custom prompt for a ring group to announce to caller. Click on ‘Prompt’, it will direct the users to upload the customized voice prompts.</p> <p><b>Note:</b> Users can also refer to the page <b>PBX Settings→Voice Prompt→Custom Prompt</b>, where they could record new prompt or upload prompt files.</p>
<b>Ring Timeout on Each Member</b>	<p>Configure the number of seconds to ring each member. If set to 0, it will keep ringing. The default setting is 60 seconds.</p> <p><b>Note:</b> The actual ring timeout might be overridden by users if the phone has ring timeout settings as well.</p>
<b>Auto Record</b>	If enabled, calls on this ring group will be automatically recorded. The default setting is No. The recording files can be accessed from Web GUI→CDR→Recording Files.
<b>Endpoint Call Forwarding Support</b>	<p>This allows the UCM to work with endpoint-configured call forwarding settings to redirect calls to ring group. For example, if a member wants to receive calls to the ring group on his mobile phone, he will have to set his endpoint’s call forwarding settings to his mobile number. By default, it is disabled.</p> <p>However, this feature has the following limitations:</p> <ul style="list-style-type: none"> <li>• This feature will work only when call forwarding is configured on endpoints, not on the UCM.</li> <li>• If the forwarded call goes through an analog trunk, and polarity reversal is disabled, the other ring group members will no longer receive the call after it is forwarded.</li> <li>• If the forwarded call goes through a VoIP trunk, and the outbound route for it is PIN-protected and requires authentication, the other ring group members will no longer receive the call after it is forwarded.</li> </ul>





	<ul style="list-style-type: none"> <li>If the forwarded call hits voicemail, the other ring group members will no longer receive the call.</li> </ul>
<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with the Ring Group name the caller know whether the call is incoming from a direct extension or a Ring Group.
<b>Skip Busy Agent</b>	If enabled, skip busy agents regardless of call waiting settings.
<b>Enable Destination</b>	If enabled, users could select extension, voicemail, ring group, IVR, call queue, voicemail group as the destination if the call to the ring group has no answer. Secret and Email address are required if voicemail is selected as the destination.
<b>Default Destination</b>	<p>The call would be routed to this destination if no one in this ring group answers the call.</p> <p><b>Note:</b> Users can now set the voicemail of ring groups as routing destinations and IVR key press event destinations and to do so ring group must have their Default Destination set to Voicemail with Ring Group Extensions.</p>



### Create New Ring Groups

\* Ring Group Name:

\* Extension:

Members:

<input type="checkbox"/> 27 items Available <input type="text" value="Search"/> <input type="button" value="Q"/> <input type="checkbox"/> 1000 <input type="checkbox"/> 1001 <input type="checkbox"/> 1005 <input type="checkbox"/> 1006 <input type="checkbox"/> 1007	<input type="button" value="←"/> <input type="button" value="→"/> <input type="button" value="↑"/> <input type="button" value="↓"/>	<input type="checkbox"/> 3 items Selected <input type="text" value="Search"/> <input type="button" value="Q"/> <input type="checkbox"/> 1002 <input type="checkbox"/> 1003 <input type="checkbox"/> 1004
--	--	--

LDAP Phonebook:

<input type="checkbox"/> 0 item Available <input type="text" value="Search"/> <input type="button" value="Q"/> <div style="text-align: center; padding: 20px;">None</div>	<input type="button" value="←"/> <input type="button" value="→"/> <input type="button" value="↑"/> <input type="button" value="↓"/>	<input type="checkbox"/> 0 item Selected <input type="text" value="Search"/> <input type="button" value="Q"/> <div style="text-align: center; padding: 20px;">None</div>
---	--	--

**Ring Group Options**

Ring Strategy:


Music On Hold:

Custom Prompt:

**Figure 156: Ring Group Configuration**

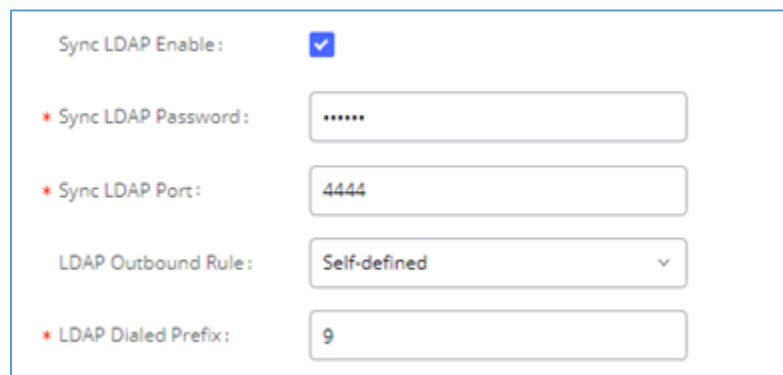
## Remote Extension in Ring Group

Remote extensions from the peer trunk of a remote UCM630xA can be included in the ring group with local extension. An example of Ring Group with peer extensions is presented in the following:

1. Creating SIP Peer Trunk between both UCM630xA\_A and UCM630xA\_B. **SIP Trunk** can be found under Web GUI→**Extension/Trunk**→**VoIP Trunks**. Also, please configure their Inbound/Outbound routes accordingly.
2. Click edit button in the menu , and check if **Sync LDAP Enable** is selected, this option will allow UCM630xA\_A update remote LDAP server automatically from peer UCM630xA\_B. In addition, **Sync LDAP Password** must match for UCM630xA\_A and UCM630xA\_B to sync LDAP contact automatically. Port



number can be anything between 0~65535, and use the outbound rule created in step 1 for the **LDAP Outbound Rule** option.



Sync LDAP Enable:

\* Sync LDAP Password:

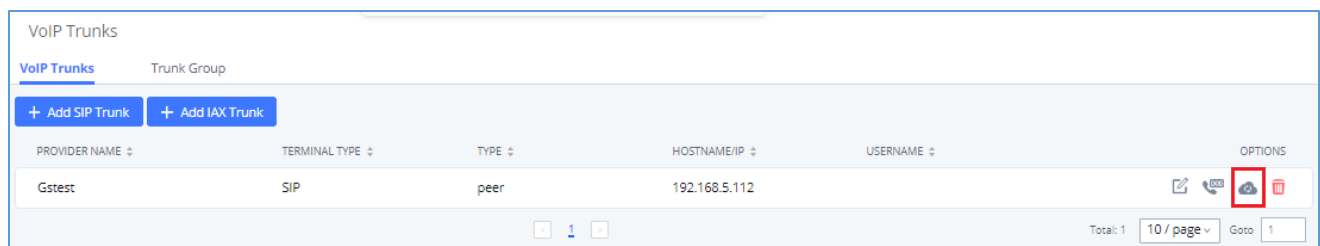
\* Sync LDAP Port:

LDAP Outbound Rule:

\* LDAP Dialed Prefix:

**Figure 157: Sync LDAP Server option**


- In case if LDAP server does not sync automatically, user can manually sync LDAP server. Under **VoIP Trunks** page, click sync button shown in the following figure to manually sync LDAP contacts from peer UCM630xA.



VoIP Trunks

VoIP Trunks Trunk Group

+ Add SIP Trunk + Add IAX Trunk

PROVIDER NAME	TERMINAL TYPE	TYPE	HOSTNAME/IP	USERNAME	OPTIONS
Gstest	SIP	peer	192.168.5.112		

Total: 1 10 / page Goto 1

**Figure 158: Manually Sync LDAP Server**

- Under **Ring Groups** setting page, click “Add”. **Ring Groups** can be found under Web GUI→**Call Features**→**Ring Groups**.
- If LDAP server is synced correctly, **Available LDAP Numbers** box will display available remote extensions that can be included in the current ring group. Please also make sure the extensions in the peer UCM630xA can be included into that UCM630xA's LDAP contact.



### Create New Ring Group

• Ring Group Name:

• Extension:

Members:

Available Extensions		Selected Extensions	
<input type="checkbox"/> 106		<input type="checkbox"/> 0	
<input type="text" value="Search"/>		<input type="text" value="Search"/>	
<input type="checkbox"/> 1000		None	
<input type="checkbox"/> 1001			
<input type="checkbox"/> 1002			
<input type="checkbox"/> 1003			

Available LDAP		Selected LDAP	
<input type="checkbox"/> 15		<input type="checkbox"/> 0	
<input type="text" value="Search"/>		<input type="text" value="Search"/>	
<input type="checkbox"/> 5000(ou=ucm6510,dc=pbx,dc=com)		None	
<input type="checkbox"/> 5001(ou=ucm6510,dc=pbx,dc=com)			
<input type="checkbox"/> 5002(ou=ucm6510,dc=pbx,dc=com)			
<input type="checkbox"/> 5003(ou=ucm6510,dc=pbx,dc=com)			

LDAP Phonebook:

Ring Group Options



**Figure 159: Ring Group Remote Extension**



## PAGING AND INTERCOM GROUP


Paging and Intercom Group can be used to make an announcement over the speaker on a group of phones. Targeted phones will answer immediately using speaker. The UCM630xA paging and intercom can be used via feature code to a single extension or a paging/intercom group. This section describes the configuration of paging/intercom group under Web GUI→**Call Features**→**Paging/Intercom**.

### Configure Paging/Intercom Group

- Click on "Add" to add paging/intercom group.
- Click on  to edit the paging/intercom group.
- Click on  to delete the paging/intercom group.
- Click on "Paging/Intercom Group Settings" to edit Alert-Info Header. This header will be included in the SIP INVITE message sent to the callee in paging/intercom call.

### Configure Multicast Paging

Create New Paging/Intercom Groups

<b>* Name:</b>	<input type="text" value="Name"/>
<b>* Type:</b>	<input style="border-bottom: 1px solid #ccc;" type="text" value="Multicast Paging"/>
<b>* Extension:</b>	<input type="text" value="Extension"/>
<b>* Maximum Call Duration.:</b>	<input type="text" value="0"/>
<b>Custom Prompt:</b>	<input style="border-bottom: 1px solid #ccc;" type="text" value="None"/> <span style="float: right; border: 1px solid #ccc; padding: 2px 5px; color: #007bff; text-decoration: none;">  Upload Audio File         </span>
<b>* Multicast IP Address:</b>	<input type="text" value="Configure multicast IP address"/>
<b>* Port:</b>	<input type="text" value="Configure the port number"/>

**Figure 160: Multicast Paging**



**Table 78: Multicast Paging Configuration Parameters**

<b>Name</b>	Configure paging/intercom group name.
<b>Type</b>	Select “Multicast Paging”.
<b>Extension</b>	Configure the paging/intercom group extension.
<b>Multicast IP Address</b>	The allowed multicast IP address range is 224.0.1.0 - 238.255.255.255. <b>Note:</b> This field appears only when “Type” is set to “Multicast Paging”.
<b>Maximum Call Duration</b>	Specify the maximum call duration in seconds. The default value 0 means no limit.
<b>Custom Prompt</b>	This option is to set a custom prompt for a paging/intercom group to announce to caller. Click on ‘Prompt’, it will direct the users to upload the customized voice prompts. <b>Note:</b> Users can also refer to the page PBX Settings→Voice Prompt→Custom Prompt, where they could record new prompt or upload prompt files.
<b>Multicast IP Address</b>	Select which extensions are allowed to use the paging/intercom feature for this paging group.
<b>Port</b>	Specify port for multicast paging. <b>Note:</b> This field appears only when “Type” is set to “Multicast Paging”.



## Configure 2-way Intercom

### Create New Paging/Intercom Groups

---

\* Name:

\* Type:

\* Extension:

Auto Record:

Replace Display Name:

\* Maximum Call Duration   
(s):

Custom Prompt:  [Upload Audio File](#)

Members:

22 items Available  
   
 1000  
 1001

0 item Selected

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**Figure 161: 2-way Intercom**

**Table 79: 2-way Intercom Configuration Parameters**

<b>Name</b>	Configure paging/intercom group name.
<b>Type</b>	Select "2-way Intercom".
<b>Extension</b>	Configure the paging/intercom group extension.
<b>Auto Record</b>	Enable this option to record in WAV format.
<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with Paging/Intercom name.
<b>Maximum Call Duration</b>	Specify the maximum call duration in seconds. The default value 0 means no limit.



<b>Custom Prompt</b>	<p>This option is to set a custom prompt for a paging/intercom group to announce to caller. Click on 'Prompt', it will direct the users to upload the customized voice prompts.</p> <p><b>Note:</b> Users can also refer to the page <b>PBX Settings→Voice Prompt→Custom Prompt</b>, where they could record new prompt or upload prompt files.</p>
<b>Members</b>	<p>Select available users from the left side to the paging/intercom group member list on the right.</p>
<b>Paging/Intercom Whitelist</b>	<p>Select which extensions are allowed to use the paging/intercom feature for this paging group.</p>

### Configure 1-way Paging

#### Create New Paging/Intercom Groups

---

\* Name:

\* Type:

\* Extension:

Auto Record:

Replace Display Name:

\* Maximum Call Duration   
(s):

Custom Prompt:  [Upload Audio File](#)

Members:

22 items Available

Search

1000

1001

0 item Selected

Search

**Figure 162: 1-way Paging**





**Table 80: 1-way Paging Configuration Parameters**

<b>Name</b>	Configure paging/intercom group name.
<b>Type</b>	Select "1-way Paging".
<b>Extension</b>	Configure the paging/intercom group extension.
<b>Auto Record</b>	Enable this option to record in WAV format.
<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with Paging/Intercom name.
<b>Maximum Call Duration</b>	Specify the maximum call duration in seconds. The default value 0 means no limit.
<b>Custom Prompt</b>	<p>This option is to set a custom prompt for a paging/intercom group to announce to caller. Click on 'Prompt', it will direct the users to upload the customized voice prompts.</p> <p><b>Note:</b> Users can also refer to the page <b>PBX Settings→Voice Prompt→Custom Prompt</b>, where they could record new prompt or upload prompt files.</p>
<b>Members</b>	Select available users from the left side to the paging/intercom group member list on the right.
<b>Paging/Intercom Whitelist</b>	Select which extensions are allowed to use the paging/intercom feature for this paging group.



## Configure Announcement Paging

Create New Paging/Intercom Groups

Enable:

\* Name:

\* Type:

Extension:

Custom Prompt:  [Upload Audio File](#)

Repeat:

\* Date:

\* Time:

Transmission Method:

Members:

30 items Available

Search

- 1000
- 1001
- 1002
- 1003
- 1004

0 item Selected

Search

None

**Figure 163: Announcement Paging**

**Table 81: Announcement Paging Configuration Parameters**

<b>Enable</b>	Enable/Disable Announcement Paging.
<b>Name</b>	Configure paging/intercom group name.
<b>Type</b>	Select "Announcement Paging"
<b>Custom Prompt</b>	<p>This option is to set a custom prompt for a paging/intercom group to announce to caller. Click on 'Prompt', it will direct the users to upload the customized voice prompts.</p> <p><b>Note:</b> Users can also refer to the page <b>PBX Settings→Voice Prompt→Custom Prompt</b>, where they could record new prompt or upload prompt files.</p>
<b>Repeat</b>	If enabled, the announcement page will be repeated for the selected weekdays.
<b>Date</b>	Configure Announcement Paging Date.



<b>Time</b>	Configure Announcement Paging Time.
<b>Transmission Method</b>	Configure Announcement Paging transmission method. <b>Unicast:</b> Depending on members selection <b>Multicast:</b> Depending on Multicast IP address and Port
<b>Members</b>	Select available users from the left side to the paging/intercom group member list on the right.

## Paging/Intercom Group Settings

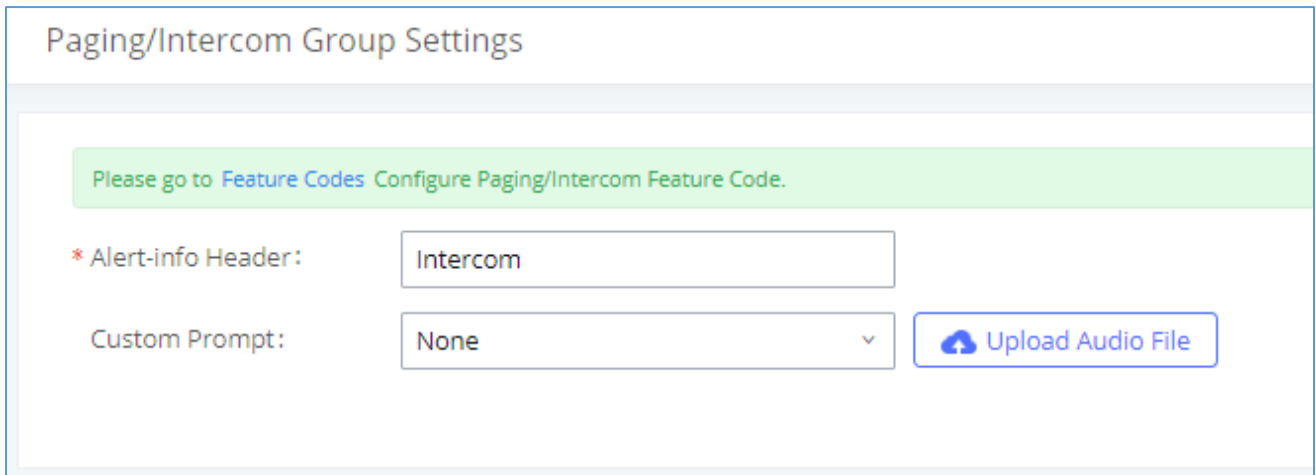


Figure 164: Page/Intercom Group Settings

The UCM630xA has pre-configured paging/intercom feature code. By default, the Paging Prefix is \*81 and the Intercom Prefix is \*80. To edit page/intercom feature code, click on "Feature Codes" in the "Paging/Intercom Group Settings" dialog. Or users could go to Web GUI→**Call Features**→**Feature Codes** directly.

## Configure a Scheduled Paging/Intercom

Users can schedule paging/intercom calls by using the Schedule Paging/Intercom page. To schedule, click the Add button on the new page and configure the caller, the group to use, and the time to call out.

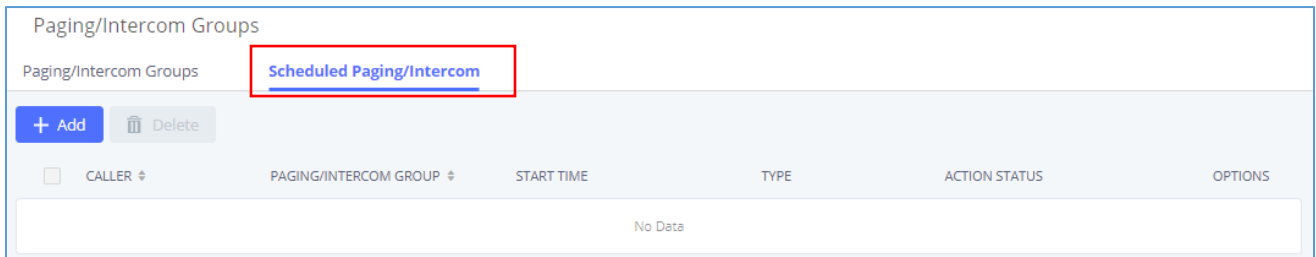


Figure 165: Schedule Paging/Intercom page



**Table 82: Schedule Paging / Intercom Settings**

<b>Caller</b>	Configure the caller ID for the paging / intercom group.
<b>Paging/Intercom Group</b>	Select the paging / intercom group from the list of the available groups.
<b>Start Time</b>	Configure the start time of the scheduled paging / intercom call.
<b>Type</b>	Select the type for the scheduled paging / intercom call. The available types are: <b>Single</b> time or <b>Daily</b> basis. Default is “Single”.
<b>Action Status</b>	Display the action status of the scheduled paging / intercom call.

Edit Scheduled Paging/Intercom: 9999

---

\* Caller:

1001 ▼

\* Paging/Intercom Group:

9999 -- test ▼

Type:

Single ▼

\* Start Time:

2020-07-04 📅

12:00 🕒

**Figure 166: Creating a scheduled paging/intercom call**

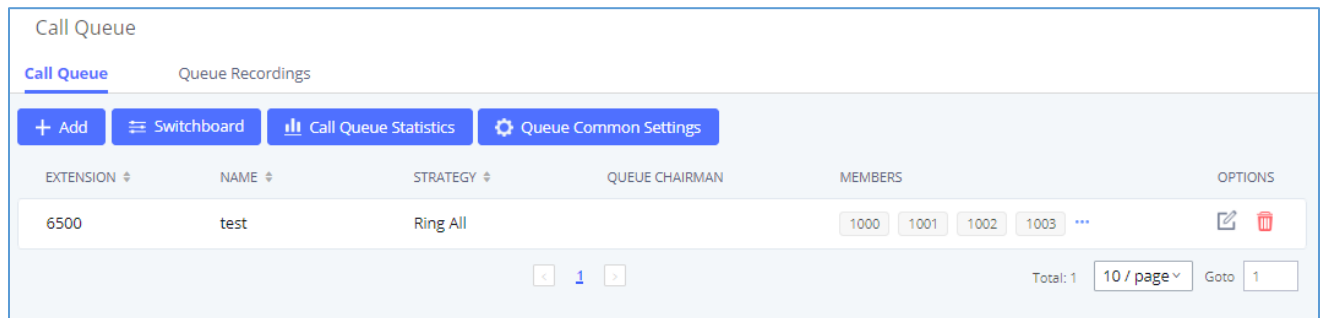


## CALL QUEUE

The UCM630xA supports call queue by using static agents or dynamic agents. Call Queue system can accept more calls than the available agents. Incoming calls will be held until next representative is available in the system. This section describes the configuration of call queue under Web GUI→**Call Features**→**Call Queue**.

### Configure Call Queue



Call queue settings can be accessed via Web GUI→**Call Features**→**Call Queue**.



**Figure 167: Call Queue**

UCM630xA supports custom prompt feature in call queue. This custom prompt will active after the caller waits for a period of time in the Queue. Then caller could choose to leave a message/ transfer to default extension or keep waiting in the queue.

To configure this feature, please go to UCM Web GUI→**Call Features**→**Call Queue**→Create New Queue/Edit Queue→Queue Options→set Enable Destination to Enter Destination with Voice Prompt. Users could configure the wait time with Voice Prompt Cycle.

- Click on "Add" to add call queue.
- Click on  to edit the call queue. The call queue configuration parameters are listed in the table below.
- Click on  to delete the call queue.

**Table 83: Call Queue Configuration Parameters**

Basic Settings	
<b>Extension</b>	Configure the call queue extension number.
<b>Name</b>	Configure the call queue name to identify the call queue.
<b>Strategy</b>	Select the strategy for the call queue. <ul style="list-style-type: none"> <li>• <b>Ring All</b></li> </ul>




	<p>Ring all available Agents simultaneously until one answers.</p> <ul style="list-style-type: none"> <li>• <b>Linear</b> Ring agents in the specified order.</li> <li>• <b>Least Recent</b> Ring the agent who has been called the least recently.</li> <li>• <b>Fewest Calls</b> Ring the agent with the fewest completed calls.</li> <li>• <b>Random</b> Ring a random agent.</li> <li>• <b>Round Robin</b> Ring the agents in Round Robin scheduling with memory.</li> </ul> <p>The default setting is "Ring All".</p>
<b>Music On Hold</b>	<p>Select the Music On Hold class for the call queue.</p> <p><b>Note:</b> Music On Hold classes can be managed from Web GUI→<b>PBX Settings</b>→<b>Music On Hold</b>.</p>
<b>Max Queue Length</b>	<p>Configure the maximum number of calls to be queued at once. This number does not include calls that have been connected with agents. It only includes calls not connected yet. The default setting is 0, which means unlimited. When the maximum value is reached, the caller will be treated with busy tone followed by the next calling rule after attempting to enter the queue.</p>
<b>Wrapup Time</b>	<p>Configure the number of seconds before a new call can ring the queue after the last call on the agent is completed. If set to 0, there will be no delay between calls to the queue. The default setting is 10 seconds.</p>
<b>Retry Time</b>	<p>Configure the number of seconds to wait before ringing the next agent.</p>
<b>Ring Time</b>	<p>Configure the number of seconds an agent will ring before the call goes to the next agent. The default setting is 30 seconds.</p>
<b>Auto Record</b>	<p>If enabled, the calls on the call queue will be automatically recorded. The recording files can be accessed in Queue Recordings under Web GUI→<b>Call Features</b>→<b>Call Queue</b>.</p>
<b>Max Wait Time</b>	<p>Configure the timeout after which users will be disconnected from the call queue. The default setting is "60". 0 means unlimited.</p> <p><b>Note:</b> It is recommended to configure "Wait Time" longer than the "Wrapup Time".</p>
<b>Welcome Prompt</b>	<p>If enabled, users can upload an audio file that will be played as an Initial tone when dialing the queue number.</p>
<b>Destination</b>	<p>Once Max Wait Time has been configured, select to which destination send the calls that have timed out. The default is to "Hang up" the call.</p>



<b>Destination Prompt Cycle</b>	Configure the voice prompt cycle (in seconds) of the call queue. Once all agents are busy and the voice prompt will be played, and you can press the appropriate key to transfer to failover destination.
<b>Custom Prompt</b>	When playing a custom prompt, press 1 to transfer to failover destination.
<b>Destination</b>	Select failover destination to send callers after pressing 1 upon hearing the custom prompt.
<b>Advanced Settings</b>	
<ul style="list-style-type: none"> <li>- <b>Virtual Queue</b></li> <li>- <b>Caller Announcement</b></li> <li>- <b>Queue Chairman</b></li> </ul>	Refer to <b>Call Center Settings and Enhancements</b> section for detailed information about these features.
<b>Enable Position Announcement</b>	If enabled, the system will inform callers waiting in the queue of their positions in line.
<b>Enable Wait Time Announcement</b>	If enabled, the estimated wait time for the call to get answered will periodically be announced to the caller. Note: Wait time will not be announced if less than one minute.
<b>Announcement Interval</b>	The interval at which caller positions and estimated wait times will be announced.
<b>Enable Agent Login</b>	Enables agent login/logout feature for static agents (supported only on GXP21XX phones with firmware higher than 1.0.9.18).
<b>Leave When Empty</b>	<p>Configure whether the callers will be disconnected from the queue or not if the queue has no agent anymore. The default setting is "Strict".</p> <ul style="list-style-type: none"> <li>• <b>Yes</b> Callers will be disconnected from the queue if all agents are paused or invalid.</li> <li>• <b>No</b> Never disconnect the callers from the queue when the queue is empty.</li> <li>• <b>Strict</b> Callers will be disconnected from the queue if all agents are paused, invalid or unavailable.</li> </ul>
<b>Dial in Empty Queue</b>	<p>Configure whether the callers can dial into a call queue if the queue has no agent. The default setting is "No".</p> <ul style="list-style-type: none"> <li>• <b>Yes</b> Callers can always dial into a call queue.</li> <li>• <b>No</b> Callers cannot dial into a queue if all agents are paused or invalid.</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>Strict</b> Callers cannot dial into a queue if the agents are paused, invalid or unavailable.</li> </ul>
<b>Failover Destination</b>	<p>Choose the destination where the call will be directed when the queue is empty or when all the agents are not logged in, here are the destinations that can be configured:</p> <ul style="list-style-type: none"> <li>• Play Sound.</li> <li>• Extension.</li> <li>• Voicemail.</li> <li>• Queues.</li> <li>• Ring Group.</li> <li>• Voicemail Group.</li> <li>• IVR.</li> <li>• External Number.</li> </ul>
<b>Enable Agent Login</b>	Enabling agent login will cause the dynamic agents to be unavailable.
<b>Queue Chairman</b>	The queue chairman can log into his web portal to operate the queue.
<b>Report Hold Time</b>	If enabled, the UCM630xA will report (to the agent) the duration of time of the call before the caller is connected to the agent. The default setting is "No".
<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with the Call Queue name so that the caller knows the call is incoming from a Call Queue.
<b>Enable Feature Codes</b>	Enable feature codes option for call queue. For example, *83 is used for "Agent Pause"
<b>Autofill</b>	Configure to enable autofill.
<b>Dynamic Login Password</b>	If enabled, the configured PIN number is required for dynamic agent to log in. The default setting is disabled.
<b>Alert-Info</b>	When present in an INVITE request, the Alert-info header field specifies an alternative ring tone to the UAS.
<b>Agents</b>	
<b>Static Agents</b>	Go to "Agents" Tab and Select the available users to be the static agents in the call queue. Choose from the available users on the left to the static agents list on the right. Click on   to choose. And use UP and Down arrow to select the order of the agent within the call queue.

### Static Agents limitation:

To guarantee a high level of audio quality with the call queue feature, UCMs will limit the number of static agents allowed to be assigned depending on the UCM model used. If the user attempts to configure the number of static agents to be more than the maximum allowed number, a warning message will appear.





The following table lists the maximum number of static agents for each UCM model:

**Table 84: Static Agent Limitation**

UCM Model	Max Static Agents in Call Queue
UCM6300A	10
UCM6302A	60
UCM6304A	100
UCM6308A	120

Click on "Global Queue Settings" to configure Agent Login Extension Postfix and Agent Logout Extension Postfix. Once configured, users could log in the call queue as dynamic agent.

**Dynamic Agent Login Settings**

Agent Login Code Suffix:

Agent Logout Code Suffix:



Example: If 6500 is the queue extension,  
 Agent Login Extension Suffix is \*,  
 Agent Logout Extension Suffix is \*\*,  
 dial 6500\* to log in and 6500\*\* to log out.

**Figure 168: Agent Login Settings**

For example, if the call queue extension is 6500, Agent Login Extension Postfix is \* and Agent Logout Extension Postfix is \*\*, users could dial 6500\* to login to the call queue as dynamic agent and dial 6500\*\* to logout from the call queue. Dynamic agent does not need to be listed as static agent and can log in/log out at any time.

- Call queue feature code "Agent Pause" and "Agent Unpause" can be configured under Web GUI → **Call Features** → **Feature Codes**. The default feature code is \*83 for "Agent Pause" and \*84 for "Agent Unpause".

**Note:** When dialing the "Agent Pause" feature code, users can specify the reason for it. The following reasons are available: (1) Lunch, (2) Hourly Break, (3) Backoffice, (4) Email, and (5) Wrap.

- Queue recordings are shown on the Call Queue page under "Queue Recordings" Tab. Click on  to download the recording file in .wav format; click on  to delete the recording file. To delete multiple



recording files by one click, select several recording files to be deleted and click on “Delete Selected Recording Files” or click on “Delete All Recording Files” to delete all recording files.

## Call Center Settings and Enhancements

UCM supports light weight call center features including virtual queue and position announcement, allowing the callers to know their position on the call queue and giving them the option to either stay on the line waiting for their turn or activate a callback which will be initiated by the UCM once an agent is free.

To configure call center features, press  on an existing call queue and go under the advanced settings tab.

Following parameters are available:

**Table 85: Call Center Parameters**

<b>Enable Virtual Queue</b>	Enable virtual queue to activate call center features.
<b>Virtual Queue Period</b>	Configure the time in (s) after which the virtual queue will take effect and the menu will be presented to the caller to choose an option. Default is 20s.
<b>Virtual Queue Mode</b>	<p><b>Offered to caller after timeout:</b> After the virtual queue period passes, the caller will enter the virtual call queue and be presented with a menu to choose an option, the choices are summarized below:</p> <ul style="list-style-type: none"> <li>• Press * to set current number as callback number.</li> <li>• Press 0 to set a callback number different than current caller number.</li> <li>• Press # to keep waiting on the call queue.</li> </ul> <p><b>Triggered on user request:</b> In this mode, the callers can activate the virtual queue by pressing 2, then they will be presented with the menu to choose an option as below:</p> <ul style="list-style-type: none"> <li>• Press * to set current number as callback number.</li> <li>• Press 0 to set a callback number different than current caller number.</li> <li>• Press # to keep waiting on the call queue.</li> </ul>
<b>Virtual Queue Outbound Prefix</b>	System will add this prefix to dialed numbers when calling back users.
<b>Enable Virtual Queue Timeout</b>	When this option is enabled and after a caller registers a call back request on the virtual queue. While all the agents are busy, the UCM will call an agent once he/she is idle again, this timeout is used for how long the UCM continues calling the agent and if the agent doesn't answer the call then the callback request will timeout and expire.
<b>Write Timeout</b>	Configure the virtual queue callback timeout period in seconds.
<b>Enable Virtual Queue Position Announcement</b>	<p>Enable the announcement of the caller's position periodically.</p> <p><b>Note:</b> Queue position will now be announced to the caller upon entering the queue.</p>



<b>Position Announcement Interval</b>	Configure the period of time in (s) during which the UCM will announce the caller's position in the call queue.
<b>Enable Virtual Queue Wait Time Announcement</b>	When enabled the UCM will announce the estimated queue wait time to callers if the estimated wait time is longer than 1 minute.
<b>Queue Chairman</b>	Select the extension to act as chairman of the queue (monitoring).
<b>Virtual Queue Welcome Prompt</b>	Click on "Upload Audio File" to upload the VQ welcome prompt.
<b>Enable Agent Login</b>	<p>When enabled, statics agents can conveniently log in and out of a queue by configuring a programmable key on their phones as a shortcut.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>✓ This feature is currently available only for GXP21xx phones on firmware 1.0.9.18 or greater.</li> <li>✓ After enabling the feature, users need to set the option on GXP21XX phone under "<b>Account→SIP Settings→Advanced Features→Special Feature</b>" to "<b>UCM Call Center</b>". A softkey labeled "UCM-CC" will appear on the bottom of the phone's screen.</li> <li>✓ When this option is enabled, dynamic agent login will be no longer supported.</li> <li>✓ In case of concurrent registrations, changing agent status on one phone (login/logout) will be reflected on all phones.</li> </ul>

### Queue Auto fill enhancement:

The waiting callers are connecting with available members in a parallel fashion until there are no more available members or no more waiting callers.

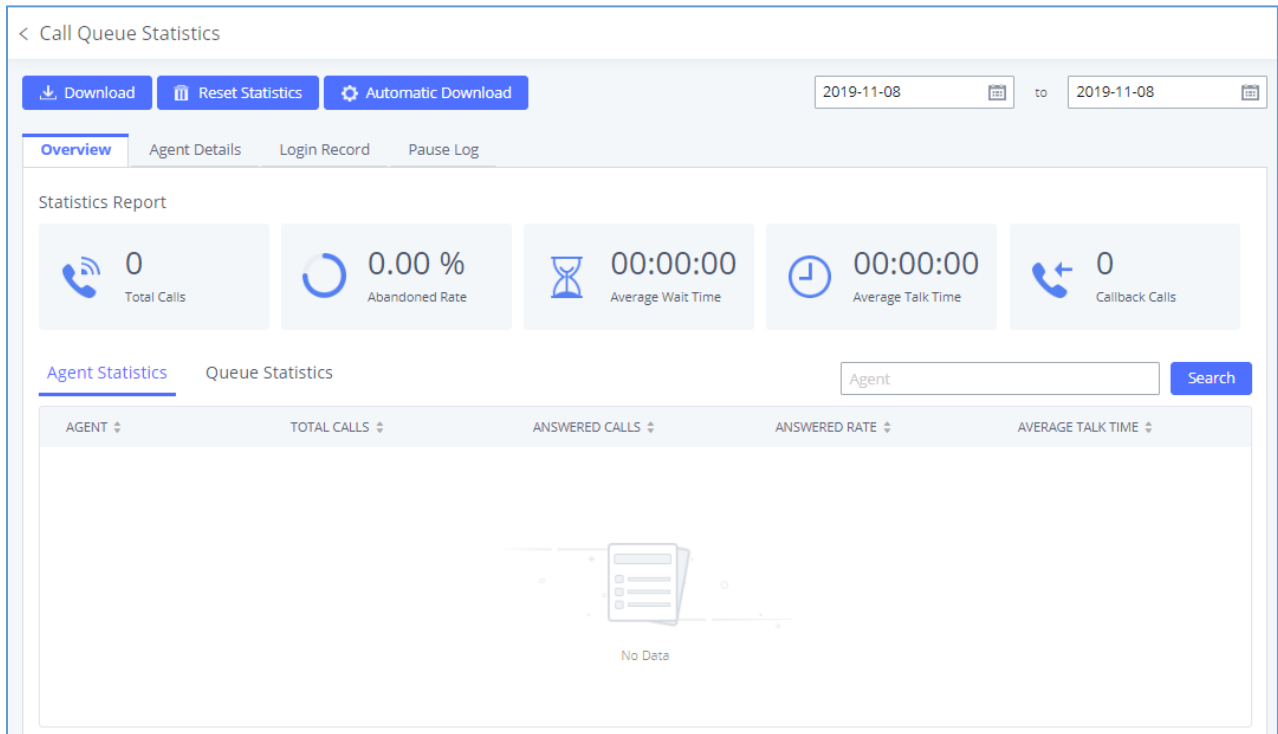
For example, in a call queue with linear method, if there are two available agents, when two callers call in the queue at the same time, UCM will assign the two callers to each of the two available agents at the same time, rather than assigning the second caller to second available agent after the first agent answers the call from the first caller.

### Queue Statistics

Along with call center features, users can also gather detailed call queue statistics allowing them to make better changes/decision to manage better the call distribution and handling based on time, agent, and queue.

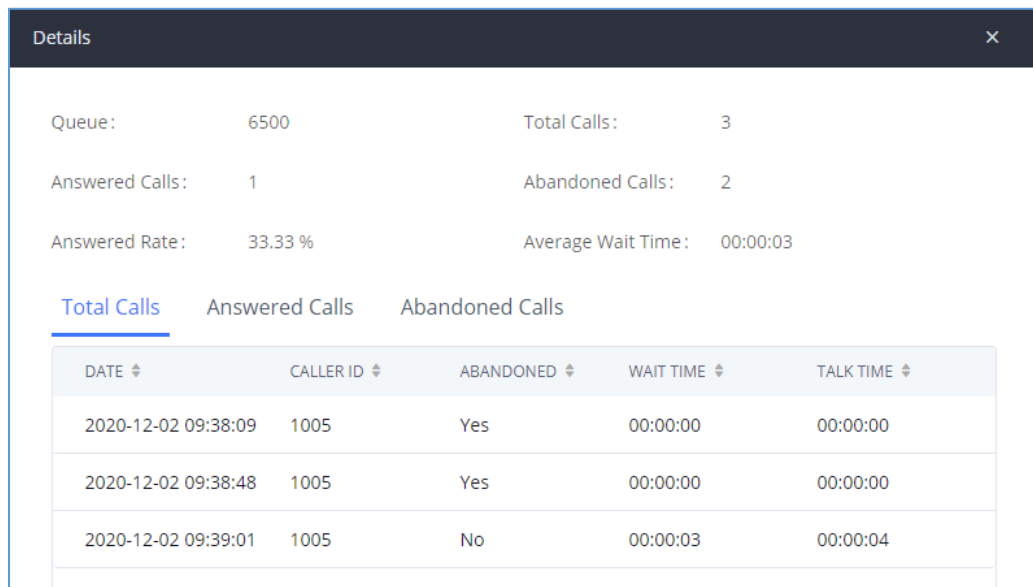
To access call queue statistics, go to Web GUI→**Call Features→Call Queue** and click on "Call Queue Statistics", the following page will be displayed:





**Figure 169: Call Queue Statistics**

By selecting a time interval, administrators can get detailed statistics for agent(s) such as total calls, answered calls etc, as well as for the queue(s) such as ABANDONED CALLS also a detailed information for the queue's call log by clicking on **Options**→**Information** button and the below window will pop up:



**Figure 170: Queue's call log details**



User can download statistics on CSV format by clicking on the “**Download**”, also the statistics can be cleared using “**Reset Statistics**” button.

The statistics can be automatically sent to a specific email address on a preconfigured Period, this can be done by clicking on “**Automatic Download**”, and user will be directed to below page where he can configure the download period (Day/Week/Month) and the Email where the statistics will be sent (Email settings should be configured correctly):

### Automatic Download

Automatically send call queue statistics to the configured email address at the specified frequency and time.

Automatic Download:

Report Type:  All  Overview  Agent Details  Login Record  Pause Log

Automatic Download:

Period:

Email:  [Email Template](#)

**Figure 171: Automatic Download Settings - Queue Statistics**

Significantly more information is now available UCM’s queue statistics page. In addition to the information presented in previous firmware, users can now view a call log that displays calls to all agents and queues, a dynamic agent login/logout record, and a pause log. Statistics reports for these new pages can be obtained by pressing the Download button in the top left corner of the Call Queue Statistics page. The reports are in .CSV format and will be packaged into a single tar.gz file upon download.

**Agent Details** is a call log that shows every call to each individual agent from all queues. The following information is available:

- Time – the date and time the call was received.
- Agent – the agent that was rung for the call.
- Queue – the queue that the call went to.
- Caller ID Number – the CID of the caller
- Abandoned – indicates whether the call was picked up or not by that specific agent. If the call rang several agents simultaneously, and this specific agent did not pick up the call, the call will be considered abandoned even if a different agent in the same queue picked it up.
- Wait Time – the amount of time that the call was waiting in queue after dialing in.
- Talk Time – the duration of the call after it was picked up by agent.



TIME	AGENT	QUEUE	CALLER ID NUMBER	ABANDONED	WAIT TIME	TALK TIME
2019-11-08 10:56:36	2000	6500	1000	No	00:00:05	00:00:29
2019-11-08 11:09:07	2000	6500	1000	No	00:00:07	00:01:51
2019-11-08 11:18:17	2000	6500	1000	Yes	00:00:04	00:00:00

**Figure 172: Agent details**

**Login Record** is a report that shows the timestamps of dynamic agent logins and logouts and calculates the amount of time the dynamic agents were logged in. Dynamic agents are extensions that log in and out either via agent login/logout codes (configured in Global Queue Settings page) or by using the GXP21xx call queue softkey. A new record will be created only when an agent logs out. The following information is available:

- Agent – the extension that logged in and out.
- Queue – the queue that the extension logged in and out of.
- Login Time – the time that the extension logged into the queue.
- Logout Time – the time that the extension logged out of the queue.
- Login Duration – the total length of time that the extension was logged in.

AGENT	QUEUE	LOGIN TIME	LOGOUT TIME	LOGIN DURATION
2000	6500	2019-11-08 09:48:53	2019-11-08 09:53:00	00:04:07
2000	6500	2019-11-08 09:53:10	2019-11-08 09:55:22	00:02:12

**Figure 173: Login Record**

**Pause Log** is a report that shows the times of agent pauses and unpauses and calculates the amount of time that agents are paused. If an agent is part of several queues, an entry will be created for each queue. An entry will only be created after an agent unpauses. The following information is available:

- Agent – the extension that paused and unpaused.
- Queue – the queue that the agent is in.
- Pause Time – the time that the agent paused.
- Resume Time – the time that the agent unpaused.
- Pause Duration – the total length of time the agent was paused for.



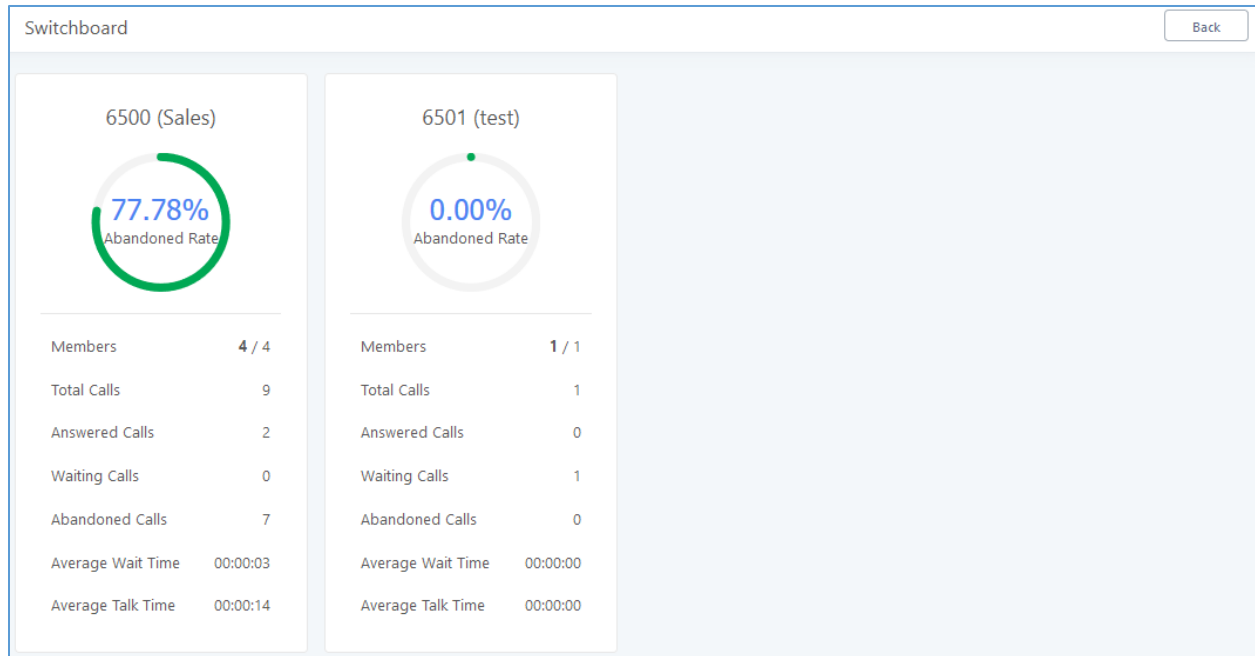
Overview Agent Details Login Record <b>Pause Log</b>				
Statistics Report				
AGENT	QUEUE	PAUSE TIME	RESUME TIME	PAUSE DURATION
2000	6500	2019-11-08 11:32:00	2019-11-08 11:33:33	00:01:33
2000	6500	2019-11-08 11:32:00	2019-11-08 11:33:33	00:01:33

**Figure 174: Pause Log**

## Switchboard

Switchboard is a Web GUI tool for call queue monitoring and management, admin can access to it from the menu **Call Features**→**Call Queue** then press “Switchboard”.

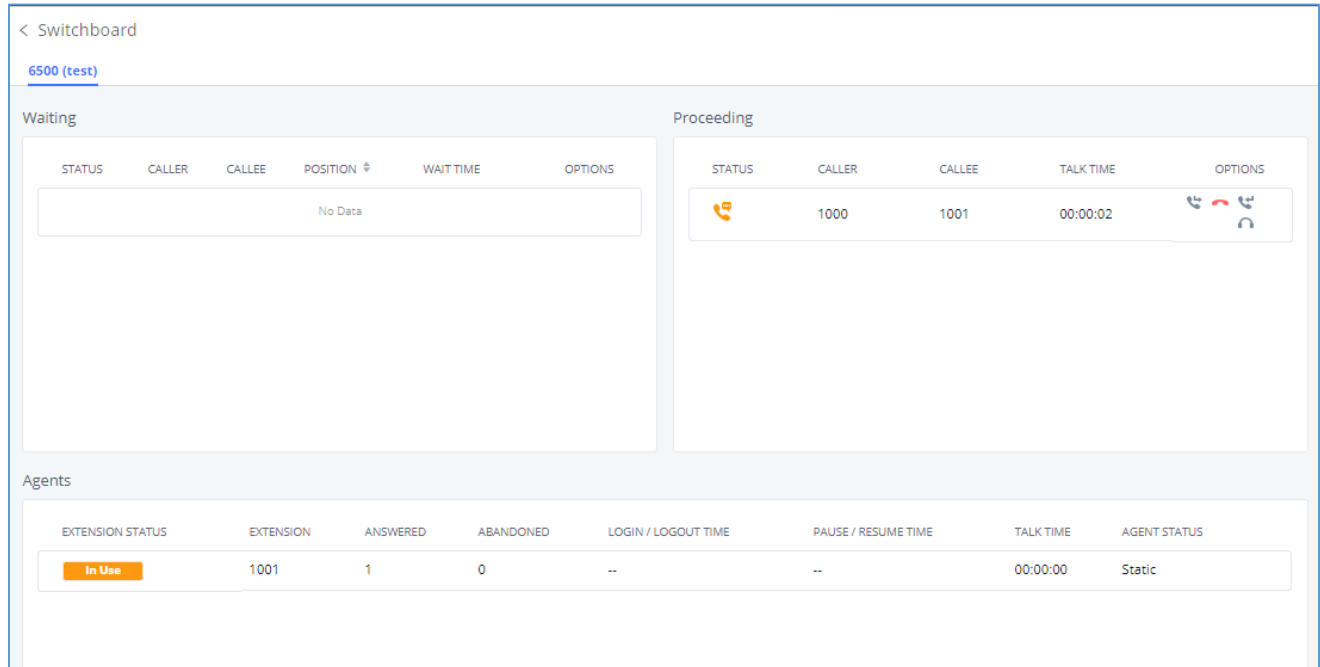
Following page will be displayed:



**Figure 175: Switchboard Summary**

Page above summarizes the available queues statistics and if one of the queues is clicked the user will be directed to page below:






**Figure 176: Call Queue Switchboard**

The table below gives a brief description for the main menus:

**Table 86: Switchboard Parameters**

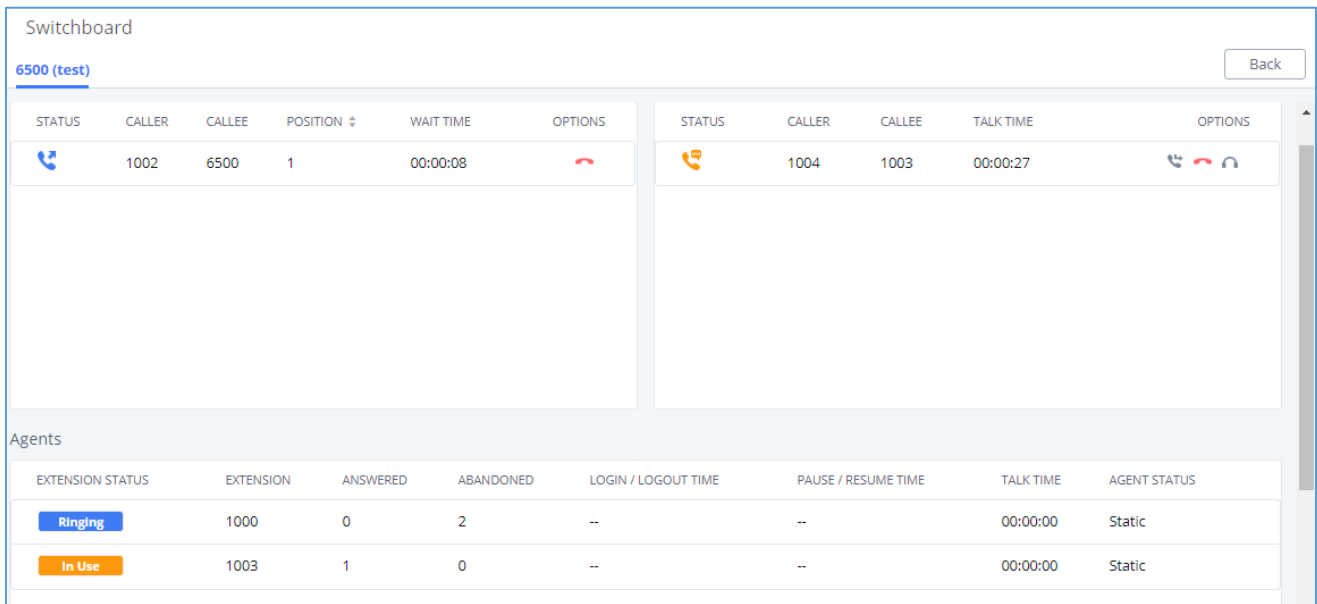
<b>Waiting</b>	This menu shows the current waiting calls along with the caller id and the option to hang-up call by pressing on the  button.
<b>Proceeding</b>	Shows the current established calls along with the caller id and the callee (agent) as well as the option to hang-up, transfer, add meeting or barge-in the call.
<b>Agents</b>	Displays the list of agents in the queue and the extension status (idle, ringing, in use or unavailable) along with some basic call statistics and agent's mode (static or dynamic).  <b>Note:</b> the dashboard will show the number of calls (answered and abandoned) of each agent. For dynamic agents, it will count the number of calls starting from the last login time.

There are three different privilege levels for Call Queue management from the switchboard: Super Admin, Queue Chairman, and Queue Agent.





- **Super Admin** - Default admin of the UCM. Call queue privileges include being able to view and edit all queue agents, monitor, and execute actions for incoming and ongoing calls for each extension in Switchboard, and generate Call Queue reports to track performance.
- **Queue Chairman** - User appointed by Super Admin to monitor and manage an assigned queue extension via Switchboard. The Queue Chairman can log into the UCM user portal with his extension number and assigned user password. To access the Switchboard, click on “*Value-added Features*” in the side menu and click on “*Call Queue*”. In the image below, User 1001 is the Queue Chairman appointed to manage Queue Extension 6500 and can see all the agents of the queue in the Switchboard.



The screenshot shows the Switchboard interface for Queue Extension 6500 (test). It features two call detail tables and an Agents table.

STATUS	CALLER	CALLEE	POSITION	WAIT TIME	OPTIONS
	1002	6500	1	00:00:08	

STATUS	CALLER	CALLEE	TALK TIME	OPTIONS
	1004	1003	00:00:27	

EXTENSION STATUS	EXTENSION	ANSWERED	ABANDONED	LOGIN / LOGOUT TIME	PAUSE / RESUME TIME	TALK TIME	AGENT STATUS
	1000	0	2	--	--	00:00:00	Static
	1003	1	0	--	--	00:00:00	Static

**Figure 177: Queue Chairman**

- **Queue Agent** - User appointed by Super Admin to be a member of a queue extension. A queue agent can log into the UCM user portal with his extension number and assigned user password. To access the Switchboard, click on “*Value-added Features*” in the side menu and click on “*Call Queue*”. However, a queue agent can view and manage only his own calls and statistics, but not other agents’ in the queue extension. In the image below, User 1000 is a queue agent and can see only his own information in the Switchboard.



< Switchboard

[6500 \(test\)](#)

Waiting						Proceeding				
STATUS	CALLER	CALLEE	POSITION	WAIT TIME	OPTIONS	STATUS	CALLER	CALLEE	TALK TIME	OPTIONS
	1000	6500	1	00:00:01		No Data				

EXTENSION STATUS	EXTENSION	ANSWERED	ABANDONED	LOGIN / LOGOUT TIME	PAUSE / RESUME TIME	TALK TIME	AGENT STATUS
<b>Ringling</b>	1001	1	1	--	--	00:00:04	Static

**Figure 178: Queue Agent**

## Global Queue Settings

As explained before, under this section users can configure the feature codes for Dynamic agent login and logout, and also can now customize the keys for virtual queue options like shown below.

### Global Queue Settings

---

#### Dynamic Agent Login Settings

Agent Login Code Suffix:

Agent Logout Code Suffix:

Example: If 6500 is the queue extension,  
 Agent Login Extension Suffix is \*,  
 Agent Logout Extension Suffix is \*\*,  
 dial **6500\*** to log in and **6500\*\*** to log out.

---

#### Virtual Queue Callback Key Settings

\* Call Back Current Number:

\* Custom Callback Number:

\* Continue Waiting:

**Figure 179: Global Queue Settings**



**Table 87: Global Queue Settings**

Dynamic Agent Login Settings	
<b>Agent Login Code Suffix</b>	Configure the code to dial after the queue extension to log into the queue (i.e. queue extension + suffix). If no suffix is configured, dynamic agents will not be able to log in
<b>Agent Logout Code Suffix</b>	Configure the code to dial after the queue extension to log out of the queue (i.e. queue extension + suffix). If no suffix is configured, dynamic agents will not be able to log out.
Virtual Queue Callback Key Settings	
<b>Call Back Current Number</b>	Press the feature key configured to set your current number as callback number.
<b>Custom Callback Number</b>	Press these feature key configured to set a custom callback number.
<b>Continue Waiting</b>	Press the feature key configured to continue waiting.

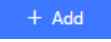




## PICKUP GROUPS

The UCM630xA supports pickup group feature which allows users to pick up incoming calls for other extensions if they are in the same pickup group, by dialing "Pickup Extension" feature code (by default \*8).

### Configure Pickup Groups

Pickup groups can be configured via Web GUI → **Call Features** → **Pickup Groups**.

- Click on  to create a new pickup group.
- Click on  to edit the pickup group.
- Click on  to delete the pickup group.

Select extensions from the list on the left side to the right side.

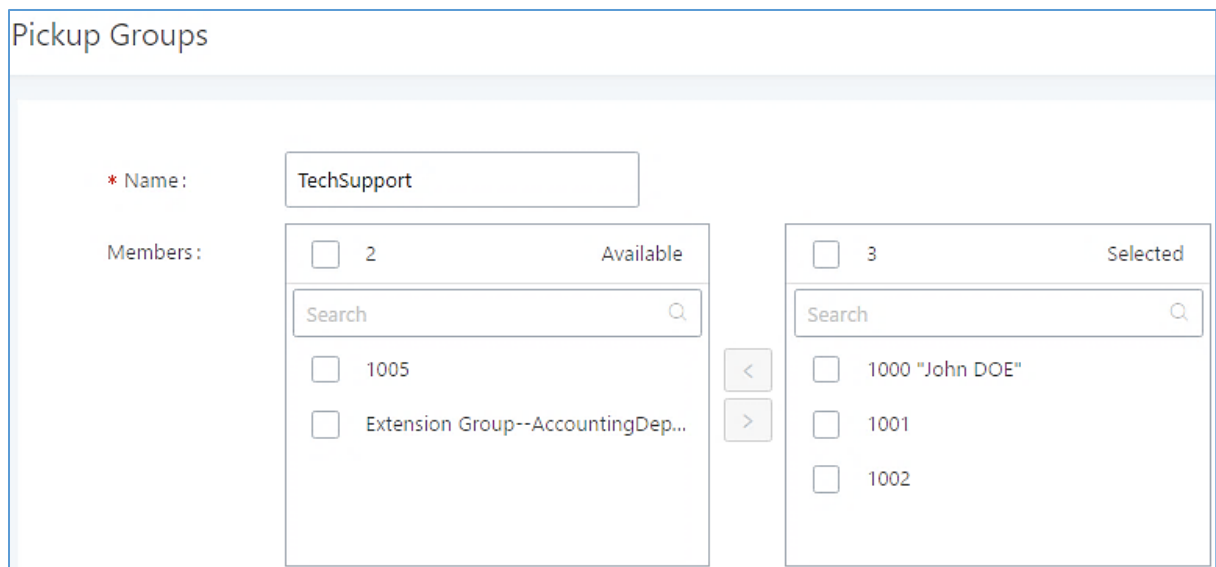


Figure 180: Edit Pickup Group

### Configure Pickup Feature Code

When picking up the call for the pickup group member, the user only needs to dial the pickup feature code. It is not necessary to add the extension number after the pickup feature code. The pickup feature code is configurable under Web GUI → **Call Features** → **Feature Codes**.

The default feature code for call pickup extension is \*8, otherwise if the person intending to pick up the call knows the ringing extension they can use \*\* followed by the extension number in order to perform the call pickup operation. The following figure shows where you can customize these features codes



Feature Codes

Feature Maps    DND/Call Forward    Feature Codes

[Reset All](#)    [Default All](#)

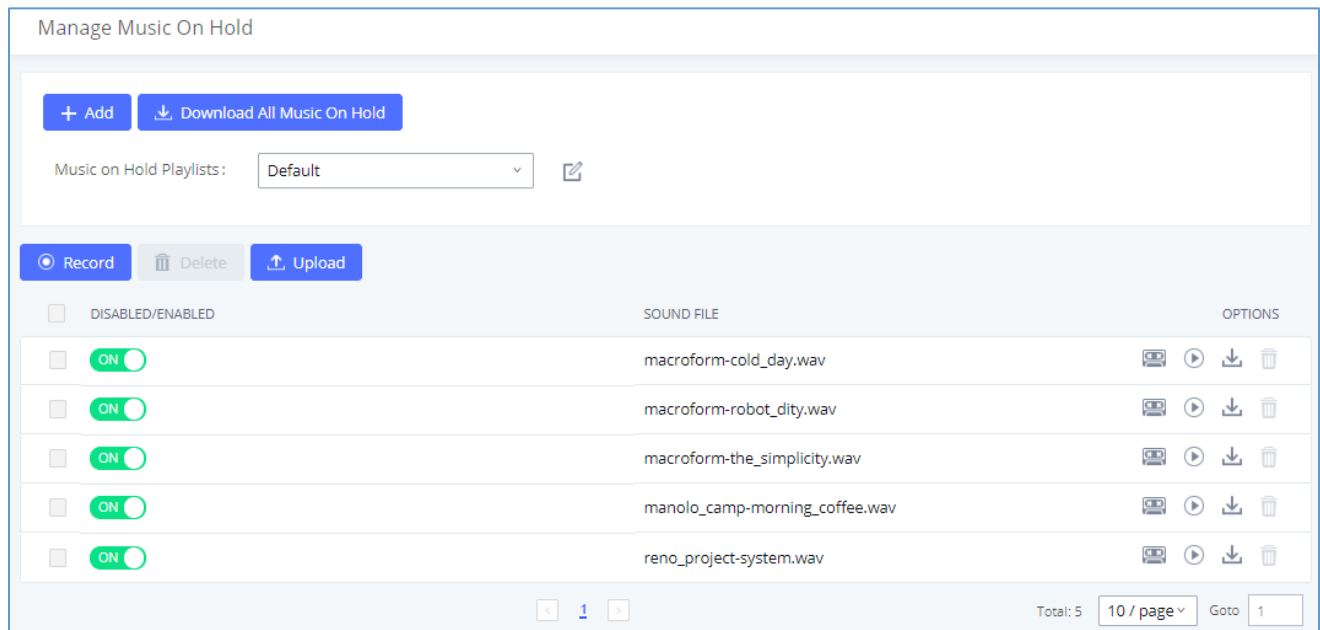
* Voicemail Access Code:	<input type="text" value="*98"/>	<input checked="" type="checkbox"/>	* My Voicemail:	<input type="text" value="*97"/>	<input checked="" type="checkbox"/>
* Agent Pause:	<input type="text" value="*83"/>	<input checked="" type="checkbox"/>	* Agent Unpause:	<input type="text" value="*84"/>	<input checked="" type="checkbox"/>
* Paging Prefix:	<input type="text" value="*81"/>	<input checked="" type="checkbox"/>	* Intercom Prefix:	<input type="text" value="*80"/>	<input checked="" type="checkbox"/>
* Blacklist Add:	<input type="text" value="*40"/>	<input checked="" type="checkbox"/>	* Blacklist Remove:	<input type="text" value="*41"/>	<input checked="" type="checkbox"/>
* Pickup on Ringing Prefix:	<input type="text" value="**"/>	<input checked="" type="checkbox"/>	* Pickup In-call Prefix:	<input type="text" value="*45"/>	<input type="checkbox"/>
* Pickup Extension:	<input type="text" value="*8"/>	<input checked="" type="checkbox"/>	* Direct Dial Voicemail Prefix:	<input type="text" value="*"/>	<input checked="" type="checkbox"/>
* Direct Dial Mobile Phone Prefix:	<input type="text" value="*88"/>	<input checked="" type="checkbox"/>	* Call Completion Request:	<input type="text" value="*11"/>	<input checked="" type="checkbox"/>
* Call Completion Cancel:	<input type="text" value="*12"/>	<input checked="" type="checkbox"/>	Enable Spy:	<input type="checkbox"/>	
* Listen Spy:	<input type="text" value="*54"/>		* Whisper Spy:	<input type="text" value="*55"/>	
* Barge Spy:	<input type="text" value="*56"/>		* Wakeup Service:	<input type="text" value="*36"/>	<input checked="" type="checkbox"/>
* PMS Wakeup Service:	<input type="text" value="*35"/>	<input checked="" type="checkbox"/>	* Update PMS Room Status:	<input type="text" value="*23"/>	<input checked="" type="checkbox"/>
* Presence Status:	<input type="text" value="*48"/>	<input checked="" type="checkbox"/>	* Dynamic Agent Logout:	<input type="text" value="*85"/>	<input checked="" type="checkbox"/>
* Voicemail Group Access Code:	<input type="text" value="*99"/>	<input checked="" type="checkbox"/>			

**Figure 181: Edit Pickup Feature Code**



## MUSIC ON HOLD




Music On Hold settings can be accessed via Web GUI → **PBX Settings** → **Music On Hold**. In this page, users could configure music on hold class and upload music files. The "default" Music On Hold class already has 5 audio files defined for users to use.



**Figure 182: Music On Hold Default Class**



- Click on "Create New MOH Class" to add a new Music On Hold class.
- Click on to configure the MOH class sort method to be "Alpha" or "Random" for the sound files.
- Click on next to the selected Music On Hold class to delete this Music On Hold class.
- Click on Upload to start uploading. Users can upload:
  - Single files with 8KHz Mono Music file, or
  - Music on hold files in a compressed package with .tar, .tar.gz and .tgz as the suffix. The file name can only be letters, digits, or special characters -\_
  - the size for the uploaded file should be less than 30M, the compressed file will be applied to the entire MoH.
- Users could also download all the music on hold files from UCM. In the Music On Hold page, click on Download All Music On Hold and the file will be downloaded to your local PC.

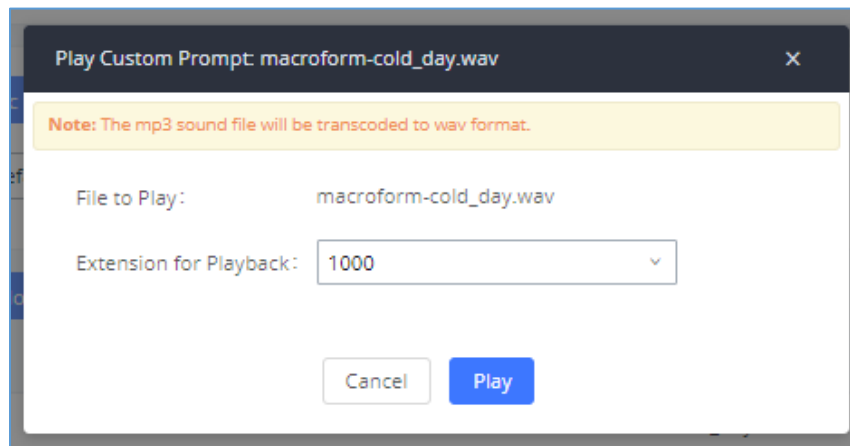


- Click on  to disable it from the selected Music On Hold Class.
- Click on  to enable it from the selected Music On Hold Class.
- Select the sound files and click on  to delete all selected Music On Hold files.

The UCM630xA allows Users to select the Music On Hold file from WebGUI to play it. The UCM630xA will initiate a call to the selected extension and play this Music On Hold file once the call is answered.

Steps to play the Music On Hold file:


1. Click on the  button for the Music On Hold file.
2. In the prompted window, select the extension to playback and click .



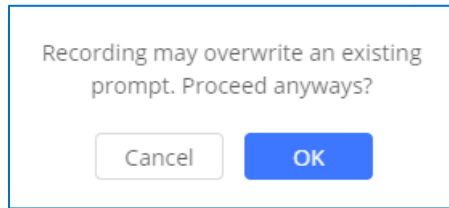
**Figure 183: Play Custom Prompt**

3. The selected extension will ring.
4. Answer the call to listen to the music playback.



Users could also record their own Music On Hold to override an existing custom prompt, this can be done by following those steps:

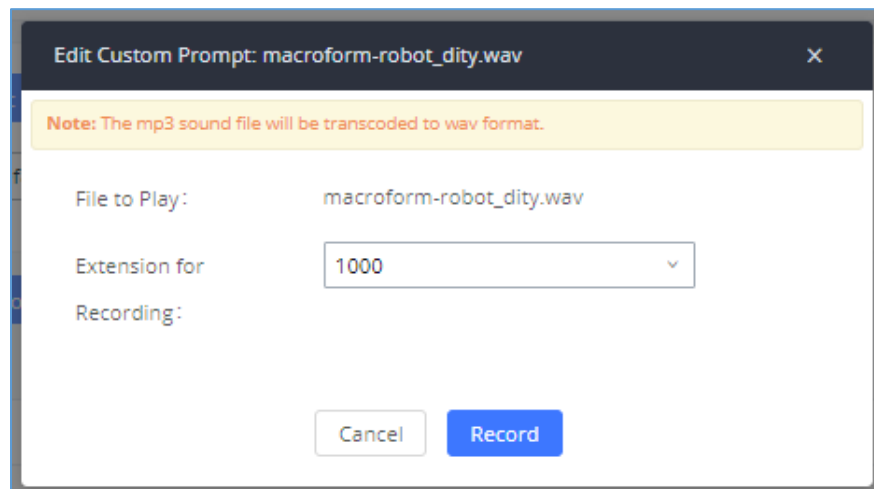
1. Click on .
2. A message of confirmation will pop up, as shown below.





**Figure 184: Information Prompt**

3. Click .
4. In the prompted window, select the extension to playback and click .



**Figure 185: Record Custom Prompt**

5. Answer the call and start to record your new music on hold.
6. Hangup the call and refresh Music On Hold page then you can listen to the new recorded file.

---

 **Notes:**

Once the MOH file is deleted, there are two ways to recover the music files.

- Users could download the MOH file from this link:  
<http://downloads.asterisk.org/pub/telephony/sounds/releases/asterisk-moh-opsound-wav-2.03.tar.gz>  
 After downloading and unzip the pack, users could then upload the music files to UCM.
  - Factory reset could also recover the MOH file on the UCM.
- 





## BUSY CAMP-ON

The UCM630xA supports busy camp-on/call completion feature that allows the PBX to camp on a called party and inform the caller as soon as the called party becomes available given the previous attempted call has failed.

The configuration and instructions on how to use busy camp-on/call completion feature can be found in the following guide:

[https://www.grandstream.com/hubfs/Product\\_Documentation/ucm6xxx\\_busy\\_camp\\_on\\_guide.pdf](https://www.grandstream.com/hubfs/Product_Documentation/ucm6xxx_busy_camp_on_guide.pdf)



## PRESENCE

UCM does support SIP presence feature which allows users to advertise their current availability status and willingness to receive calls, this way other users can use their phones in order to monitor the presence status of each user and decide whether to call them or not based on their advertised availability.

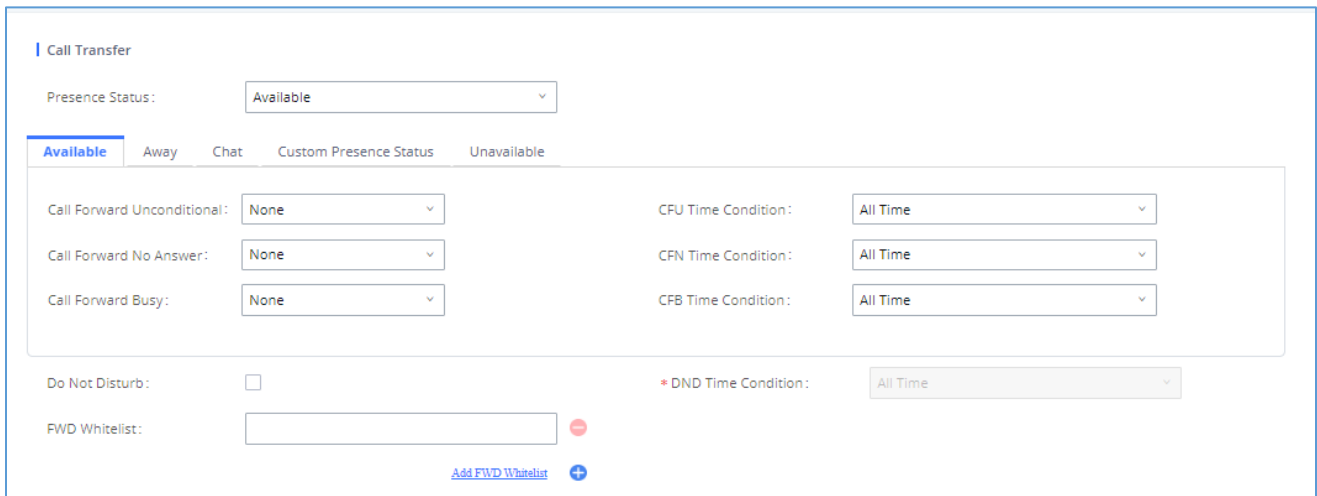
This feature is different than BLF which is used to monitor the dialog status for each extension (Ringing, Idle or Busy). Instead the SIP presence module gives more options for users to choose which state they want to put themselves in.

In order to configure the presence status of an extension from the web GUI, users can access the menu of configuration using one of the two following methods:

- From admin account, go under the menu **Extension/Trunk**→**Extensions** and choose the desired extension to edit then navigate to the “Features” tab.

OR

- From the User Portal, go under the menu Basic **Information**→**Extensions** and navigate to the Features tab to have the following options.



**Figure 186: SIP Presence Configuration**

Select which status to set from the presence status selection drop list, six options are available and below is a brief description of these states:



**Table 88: SIP Presence Status**

<b>Available</b>	The contact is online and can participate in conversations/phone calls.
<b>Away</b>	The contact is currently away (ex: for lunch break).
<b>Chat</b>	The contact has limited conversation flexibility and can only be reached via chat.
<b>Do Not Disturb</b>	The Contact is on DND (Do Not Disturb) mode.
<b>Custom Presence Status</b>	Please enter the presence status for this mode on the Web GUI. Up to 64 characters.
<b>Unavailable</b>	The contact is unreachable for the moment, please try to contact later.

Another option to set the presence status and which is more practical is using the feature code from the user's phone, one the user dials the feature code (default is \*48), a prompt will be played to select which status they want to put themselves in, by pressing the corresponding key.

The feature code can be enabled and customized from the Web GUI → **Call Features** → **Feature Codes**.

* Voicemail Access Code:	<input type="text" value="*98"/>	<input checked="" type="checkbox"/>	* My Voicemail:	<input type="text" value="*97"/>	<input checked="" type="checkbox"/>
* Agent Pause:	<input type="text" value="*83"/>	<input checked="" type="checkbox"/>	* Agent Unpause:	<input type="text" value="*84"/>	<input checked="" type="checkbox"/>
* Paging Prefix:	<input type="text" value="*81"/>	<input checked="" type="checkbox"/>	* Intercom Prefix:	<input type="text" value="*80"/>	<input checked="" type="checkbox"/>
* Blacklist Add:	<input type="text" value="*40"/>	<input checked="" type="checkbox"/>	* Blacklist Remove:	<input type="text" value="*41"/>	<input checked="" type="checkbox"/>
* Call Pickup on Ringing:	<input type="text" value="**"/>	<input checked="" type="checkbox"/>	* Pickup In-call:	<input type="text" value="*45"/>	<input type="checkbox"/>
* Pickup Extension:	<input type="text" value="*8"/>	<input checked="" type="checkbox"/>	* Direct Dial Voicemail Prefix:	<input type="text" value="*"/>	<input checked="" type="checkbox"/>
* Direct Dial Mobile Phone Prefix:	<input type="text" value="*88"/>	<input checked="" type="checkbox"/>	* Call Completion Request:	<input type="text" value="*11"/>	<input checked="" type="checkbox"/>
* Call Completion Cancel:	<input type="text" value="*12"/>	<input checked="" type="checkbox"/>	Enable Spy:	<input type="checkbox"/>	
* Listen Spy:	<input type="text" value="*54"/>		* Whisper Spy:	<input type="text" value="*55"/>	
* Barge Spy:	<input type="text" value="*56"/>		* Wakeup Service:	<input type="text" value="*36"/>	<input checked="" type="checkbox"/>
* PMS Wakeup Service:	<input type="text" value="*35"/>	<input checked="" type="checkbox"/>	* Update PMS Room Status:	<input type="text" value="*23"/>	<input checked="" type="checkbox"/>
* Presence Status:	<input type="text" value="*48"/>	<input checked="" type="checkbox"/>	* Dynamic Agent Logout:	<input type="text" value="*85"/>	<input checked="" type="checkbox"/>

**Figure 187: SIP Presence Feature Code**

When a user does change his/her SIP presence status by making a call using presence feature code, the UCM will create a corresponding CDR entry showing the call as **Action type = PRSENCE\_STATUS**.






CDR Display Filter

By default, this page displays the CDR entries from the current month. Use the "Filter" button to specify a time range.

Recording files are currently stored in USB Disk . Do you want to change the location? This will change the file storage location for basic call, conference, queue, SCA, and emergency call recordings.

Delete All
Delete Search Result (s)
Download All Records
Download Search Result (s)
Automatic Download

STATUS	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	TALK TIME	ACCOUNT C ODE	RECORDING FILE OPTIONS
	"1000" 1000	*48	PRESENCE_STATUS	2019-12-11 17:55:33	0:00:13	0:00:13		-
	"admin" VFAX [T...	998653221	DIAL	2019-12-11 17:51:09	0:00:22	0:00:22		-
	"1000" 1000	6500	QUEUE[6500]	2019-12-11 17:32:45	0:00:04	0:00:00		-


**Figure 188: Presence Status CDR**

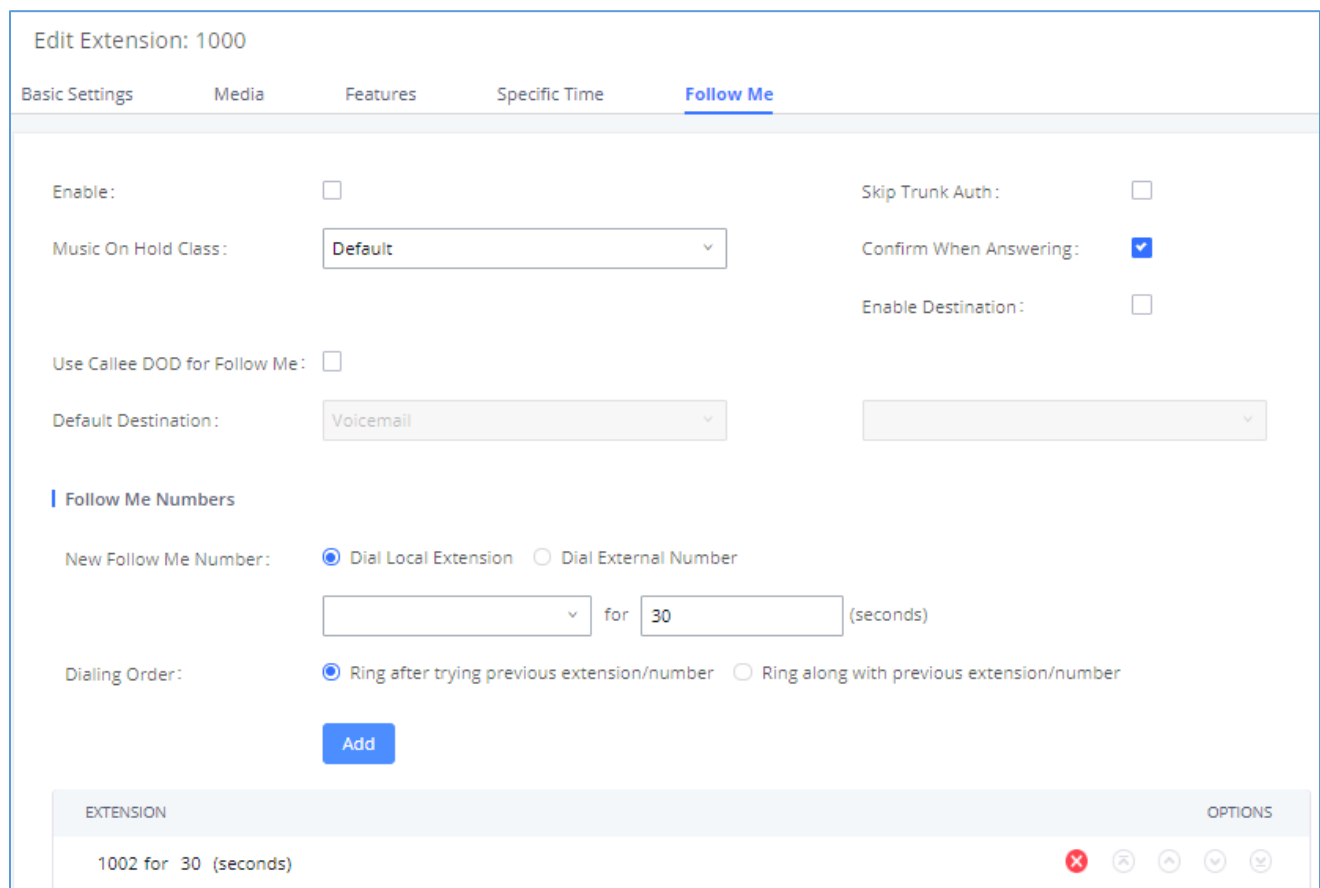


## FOLLOW ME

Follow Me is a feature on the UCM630xA that allows users to direct calls to other phone numbers and have them ring all at once or one after the other. Calls can be directed to users' home phone, office phone, mobile and etc. The calls will get to the user no matter where they are. Follow Me option can be found under extension settings page Web GUI → **Extension/Trunk** → **Extensions**.

To configure follow me:

1. Choose the extension and click on .
2. Go to the Follow me tab to add destination numbers and enable the feature.



**Edit Extension: 1000**

Basic Settings   Media   Features   Specific Time   **Follow Me**

Enable:

Music On Hold Class:

Use Callee DOD for Follow Me:

Default Destination:

Skip Trunk Auth:

Confirm When Answering:

Enable Destination:

**Follow Me Numbers**

New Follow Me Number:  Dial Local Extension    Dial External Number

for  (seconds)

Dialing Order:  Ring after trying previous extension/number    Ring along with previous extension/number

EXTENSION	OPTIONS
1002 for 30 (seconds)	<input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>





**Figure 189: Edit Follow Me**

3. Click on  to add local extensions or external numbers to be called after ringing the extension selected in the first step.
4. Once created, it will be displayed on the follow me list. And you can click on  to delete the Follow Me.



The following table shows the Follow Me configuration parameters:

**Table 89: Follow Me Settings**

<b>Enable</b>	Configure to enable or disable Follow Me for this user.
<b>Skip Trunk Auth</b>	If external number is added in the Follow Me, please make sure this option is enabled or the “Skip Trunk Auth” option of the extension is enabled, otherwise the external Follow Me number cannot be reached.
<b>Music On Hold Class</b>	Configure the Music On Hold class that the caller would hear while tracking users
<b>Confirm When Answering</b>	By default, it is enabled, and user will be asked to press 1 to accept the call or to press 2 to reject the call after answering a Follow Me call. If it is disabled, the Follow Me call will be established once after the user answers.
<b>Enable Destination</b>	When enabled, the call will be routed to the default destination if no one in the Follow Me extensions answers the call.
<b>Default Destination</b>	Configure the destination if no one in the Follow Me extensions answers the call. The available options are: <ul style="list-style-type: none"> <li>• Extension</li> <li>• Voicemail</li> <li>• Queues</li> <li>• Ring Group</li> <li>• Voicemail Group</li> <li>• IVR</li> <li>• External Number</li> </ul>
<b>Follow Me Numbers</b>	The added numbers are listed here. Click on   to arrange the order. Click on  to delete the number. Click on  to add new numbers.
<b>New Follow Me Number</b>	Add a new Follow Me number which could be a ‘Local Extension’ or ‘External Number’. The selected dial plan should have permissions to dial the defined external number.
<b>Dialing Order</b>	Select the order in which the Follow Me destinations will be dialed to reach the user: ring all at once or ring one after the other.

Click on “Follow Me Options” under Web GUI→**Extension/Trunk**→**Extension** page to enable or disable the options listed in the following table.



**Table 90: Follow Me Options**

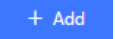
<b>Playback Incoming Status Message</b>	If enabled, the PBX will playback the incoming status message before starting the Follow Me steps.
<b>Record the Caller's Name</b>	If enabled, the PBX will record the caller's name from the phone so it can be announced to the callee in each step.
<b>Playback Unreachable Status Message</b>	If enabled, the PBX will playback the unreachable status message to the caller if the callee cannot be reached.

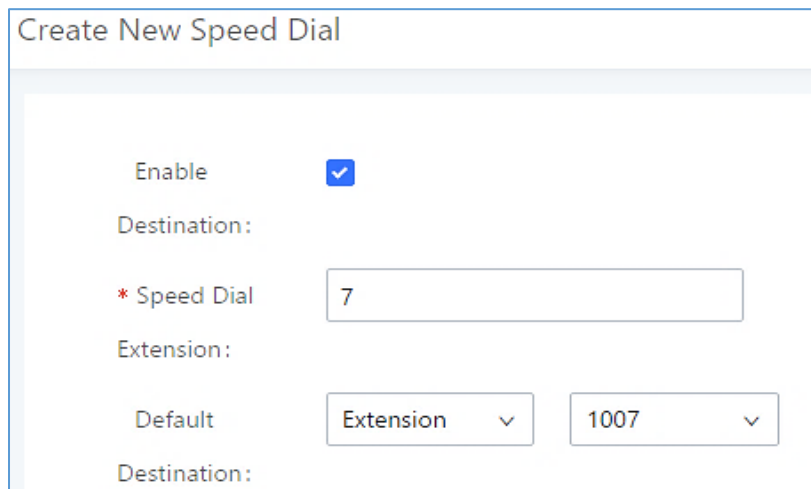


## SPEED DIAL

The UCM630xA supports Speed Dial feature that allows users to call a certain destination by pressing one or four digits on the keypad. This creates a system-wide speed dial access for all the extensions on the UCM630xA.

To enable Speed Dial, on the UCM630xA Web GUI, go to page Web GUI→**Call Features**→**Speed Dial**.

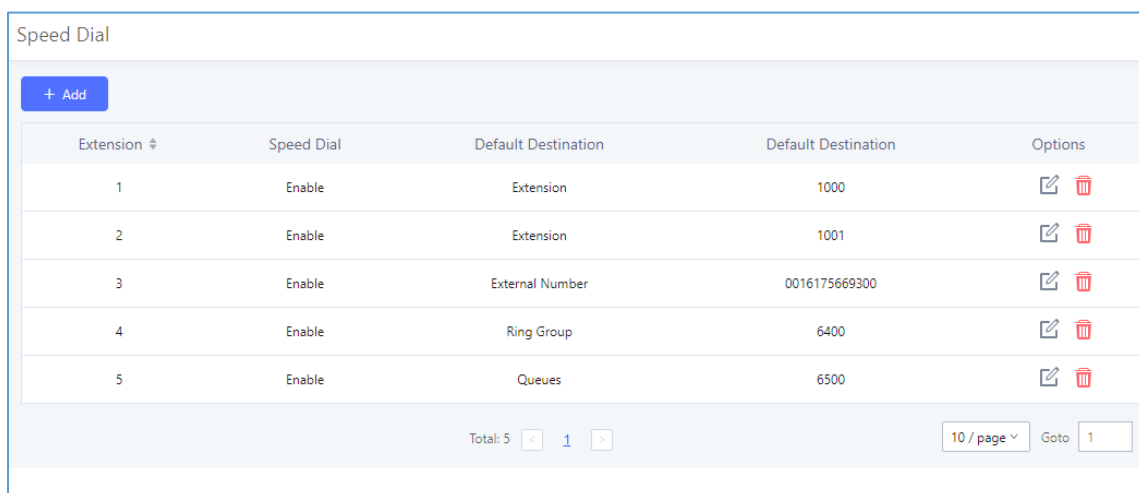
User should first click on . Then decide from one digit up to four digits combination used for Speed Dial and select a dial destination from “Default Destination”. The supported destinations include extension, voicemail, meeting room, voicemail group, IVR, ring group, call queue, page group, DISA, Dial by Name and external number.



The screenshot shows a 'Create New Speed Dial' form with the following fields:

- Enable:** A checkbox that is checked.
- Destination:** A label above the input fields.
- \* Speed Dial:** A text input field containing the digit '7'.
- Extension:** A label above the dropdown menus.
- Default:** Two dropdown menus. The first is set to 'Extension' and the second is set to '1007'.
- Destination:** A label below the dropdown menus.

**Figure 190: Speed Dial Destinations**



The screenshot shows a table titled 'Speed Dial' with a '+ Add' button and a table with 5 columns: Extension, Speed Dial, Default Destination, Default Destination, and Options. The table contains 5 rows of data.

Extension	Speed Dial	Default Destination	Default Destination	Options
1	Enable	Extension	1000	
2	Enable	Extension	1001	
3	Enable	External Number	0016175669300	
4	Enable	Ring Group	6400	
5	Enable	Queues	6500	

At the bottom of the table, there is a pagination control showing 'Total: 5', a page number '1', and a '10 / page' dropdown menu.

**Figure 191: List of Speed Dial**

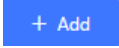




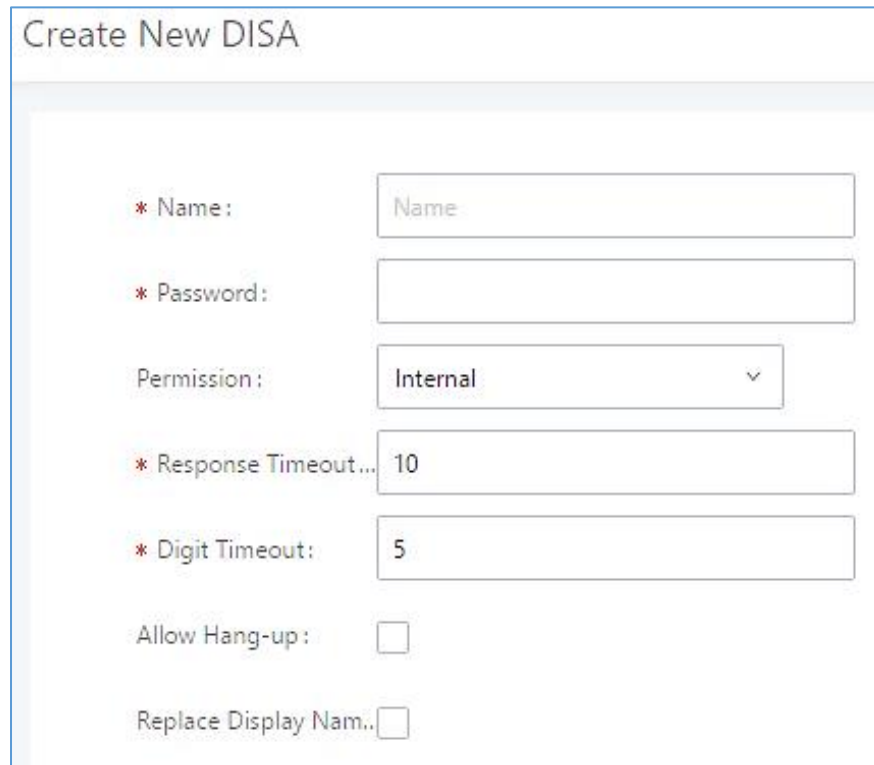


## DISA

In many situations, the user will find the need to access his own IP PBX resources, but he is not physically near one of his extensions. However, he does have access to his own cell phone. In this case, we can use what is commonly known as DISA (Direct Inward System Access). Under this scenario, the user will be able to call from the outside, whether it is using his cell phone, pay phone, regular PSTN, etc. After calling into UCM630xA, the user can then dial out via the SIP trunk or PSTN trunk connected to UCM630xA as it is an internal extension.

The UCM630xA supports DISA to be used in IVR or inbound route. Before using it, create new DISA under Web GUI→**Call Features**→**DISA**.

- Click on  to add a new DISA.
- Click on  to edit the DISA configuration.
- Click on  to delete the DISA.



**Create New DISA**

\* Name:

\* Password:

Permission:

\* Response Timeout:

\* Digit Timeout:

Allow Hang-up:

Replace Display Nam.:

**Figure 192: Create New DISA**

The following table details the parameters to set and configure DISA feature on UCM630xA PBX.



**Table 91: DISA Settings**

<b>Name</b>	Configure DISA name to identify the DISA.
<b>Password</b>	Configure the password (digit only) required for the user to enter before using DISA to dial out.  <b>Note:</b> The password must be at least 4 digits.
<b>Permission</b>	Configure the permission level for DISA. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". If the user tries to dial outbound calls after dialing into the DISA, the UCM630xA will compare the DISA's permission level with the outbound route's privilege level. If the DISA's permission level is higher than (or equal to) the outbound route's privilege level, the call will be allowed to go through.
<b>Response Timeout</b>	Configure the maximum amount of time the UCM630xA will wait before hanging up if the user dials an incomplete or invalid number. The default setting is 10 seconds.
<b>Digit Timeout</b>	Configure the maximum amount of time permitted between digits when the user is typing the extension. The default setting is 5 seconds.
<b>Allow Hangup</b>	If enabled, during an active call, users can enter the UCM630xA Hangup feature code (by default it is *0) to disconnect the call or hang up directly. A new dial tone will be heard shortly for the user to make a new call. The default setting is "No".
<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with the DISA name.

Once successfully created, users can configure the inbound route destination as "DISA" or IVR key event as "DISA". When dialing into DISA, users will be prompted with password first. After entering the correct password, a second dial tone will be heard for the users to dial out.



## EMERGENCY

UCM supports configuration and management of numbers to be called in emergency situation, thus bypassing the regular outbound call routing process and allowing users in critical situation to dial out for emergency help with the possibility to have redundant trunks as point of exit in case one of the lines is down.

UCM6xxx series are also now in full compliance with Kari's Law and Ray Baum's Act, for more information, please refer to the following links:

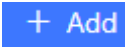
<https://www.fcc.gov/mlts-911-requirements>

[https://www.grandstream.com/hubfs/Product\\_Documentation/UCM\\_Emergency\\_Calls\\_Guide.pdf](https://www.grandstream.com/hubfs/Product_Documentation/UCM_Emergency_Calls_Guide.pdf)

In addition, Emergency calls can be automatically recorded by toggling on the new Auto Record and recordings can be viewed in the new Emergency Recordings tab on the same page. Additionally, users can have these recordings be sent to the configured email address(es).

Email alerts are also supported after enabling the notification for the event under “**Maintenance → System Events**”

To configure emergency numbers, users need to follow below steps:

1. Navigate on the web GUI under “**Call Features → Emergency Calls**”
2. Click on  to add a new emergency number.
3. Configure the required fields “Name, Emergency Number and Trunk(s) to be used to reach the number”.
4. Save and apply the configuration.



### Create New Emergency Call

\* Name:

\* Emergency Number:

Emergency Level:  ▾

Disable Hunt on Busy:

Custom Prompt:  ▾ [Prompt](#)

\* Use Trunks:

\* Members Notified:

Available	Selected
<input type="checkbox"/> 11 items <input type="text" value="Search"/> <ul style="list-style-type: none"> <li><input type="checkbox"/> 1001 "John Doe"</li> <li><input type="checkbox"/> 1002</li> <li><input type="checkbox"/> 1003</li> <li><input type="checkbox"/> 1004</li> </ul>	<input type="checkbox"/> 1 item <input type="text" value="Search"/> <ul style="list-style-type: none"> <li><input type="checkbox"/> 1000 "James tuan"</li> </ul>

Strip:

Prepend:

Auto Record:

Send Recording File:

Email Address:  [+](#)

**Figure 193: Emergency Number Configuration**



The table below gives more description of the configuration Parameters when creating emergency numbers.

**Table 92: Emergency Numbers Parameters**



<b>Name</b>	Configure the name of the emergency call. For example, "emergency911", "emergency211" and etc.
<b>Emergency Number</b>	Config the emergency service number. For example, "911", "211" and etc.
<b>Emergency Level</b>	Select the emergency level of the number. Level "3" means the most urgent.
<b>Disable Hunt on Busy</b>	If this option is not enabled, when the lines of trunks which the coming emergency call routes by are completely occupied, the line-grabbing function will automatically cut off a line from all busy lines so that the coming emergency call can seize it for dialing out. This option is not enabled by default.
<b>Custom Prompt</b>	This option sets a custom prompt to be used as an announcement to the person receiving an emergency call. The file can be uploaded from the page "Custom Prompt". Click "Prompt" to add additional record.
<b>Use Trunks</b>	Select the trunks for the emergency call. Select one trunk at least and select five trunks at most.
<b>Members Notified</b>	Select the members who will be notified when an emergency call occurs.
<b>Strip</b>	Specify the number of digits that will be Stripped from the beginning of the dialed number before the call is placed via the selected trunk.
<b>Prepend</b>	Specify the digits to be Prepend before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
<b>Auto Record</b>	When enabled, emergency call will be automatically recorded.
<b>Send Recording File</b>	When enabled recording files will be sent to the configured email address.
<b>Email Address</b>	The email address to where the recording files will be sent.



Emergency Calls

[Emergency Calls](#)   [Emergency Recordings](#)   [Emergency Location Mapping](#)

[+ Add](#)

NAME ↓	EMERGENCY NUMBER ↓	EMERGENCY LEVEL ↓	DISABLE HUNT ON BUSY ↓	OPTIONS
911	911	1	No	 

< 1 >

Total: 1   10 / page ▼   Goto: 1

**Figure 194: 911 Emergency Sample**



## CALLBACK

Callback is designed for users who often use their mobile phones to make long distance or international calls which may have high service charges. The callback feature provides an economic solution for reduce the cost from this.

The callback feature works as follows:

1. Configure a new callback on the UCM630xA.
2. On the UCM630xA, configure destination of the inbound route for analog trunk to callback.
3. Save and apply the settings.
4. The user calls the PSTN number of the UCM630xA using the mobile phone, which goes to callback destination as specified in the inbound route.
5. Once the user hears the ringback tone from the mobile phone, hang up the call on the mobile phone.
6. The UCM630xA will call back the user.
7. The user answers the call.
8. The call will be sent to DISA or IVR which directs the user to dial the destination number.
9. The user will be connected to the destination number.

In this way, the calls are placed and connected through trunks on the UCM630xA instead of to the mobile phone directly. Therefore, the user will not be charged on mobile phone services for long distance or international calls.

To configure callback on the UCM630xA, go to Web GUI→**Call Features**→**Callback** page and click on

[+ Create New Callback](#)

. Configuration parameters are listed in the following table.

**Table 93: Callback Configuration Parameters**

<b>Name</b>	Configure a name to identify the Callback. (Enter at least two characters)
<b>CallerID Pattern</b>	Configure the pattern of the callers allowed to use this callback. The caller who places the inbound call needs to have the CallerID match this pattern so that the caller can get callback after hanging up the call. <b>Note:</b> If leaving as blank, all numbers are allowed to use this callback.
<b>Outbound Prepend</b>	Configure the prepend digits to be added at before dialing the outside number. The number with prepended digits will be used to match the outbound route. '-' is the connection character which will be ignored.
<b>Delay Before Callback</b>	Configure the number of seconds to be delayed before calling back the user.
<b>Destination</b>	Configure the destination which the callback will direct the caller to. Two destinations are available: <ul style="list-style-type: none"> <li>• IVR</li> <li>• DISA</li> </ul> The caller can then enter the desired number to dial out via UCM630xA trunk.



## BLF AND EVENT LIST

### BLF

The UCM630xA supports BLF monitoring for extensions, ring group, call queue, meeting room and parking lot. For example, on the user's phone, configure the parking lot number 701 as the BLF monitored number. When there is a parked call on 701, the LED for this BLF key will light up in red, meaning a call is parked against this parking lot. Pressing this BLF key can pick up the call from this parking lot.





#### Note:

On the Grandstream GXP series phones, the MPK supports "Call Park" mode, which can be used to park the call by configuring the MPK number as call park feature code (e.g., 700). MPK "Call Park" mode can also be used to monitor and pickup parked call if the MPK number is configured as parking lot (e.g., 701).

### Event List

Besides BLF, users can also configure the phones to monitor event list. In this way, both local extensions on the same UCM630xA and remote extensions on the VOIP trunk can be monitored. The event list setting is under Web GUI→**Call Features**→**Event List**.

- Click on "Add" to add a new event list.
- Sort selected extensions manually in the Eventlist
- Click on  to edit the event list configuration.
- Click on  to delete the event list.

**Table 94: Event List Settings**

<b>URI</b>	Configure the name of this event list (for example, office_event_list). Please note the URI name cannot be the same as the extension name on the UCM630xA. The valid characters are letters, digits, _ and -.
<b>Local Extensions</b>	Select the available extensions/Extension Groups listed on the local UCM630xA to be monitored in the event list.
<b>Remote Extensions</b>	If LDAP sync is enabled between the UCM630xA and the peer UCM630xA, the remote extensions will be listed under "Available Extensions". If not, manually enter the remote extensions under "Special Extensions" field.
<b>Special Extensions</b>	Manually enter the remote extensions in the peer/register trunk to be monitored in the event list. Valid format: 5000,5001,9000





### Create New Event List

\* URI:

Event Type: Dialog

Local Extensions:

9 items Available

Search here

- 1003 "Betty"
- 1004
- 1005 "Will"
- 1006 "Iala"
- 1007 "Kiki"

<

>

↑

↓

↑

↓

3 items Selected

Search here

- 1000 "Mia"
- 1001 "John"
- 1002 "Chris"

Remote Extensions:

0 item Available

Search here

None

<

>

↑

↓

↑

↓

0 item Selected

Search here

None

Special Extensions:

**Figure 195: Create New Event List**

Remote extension monitoring works on the UCM630xA via event list BLF, among Peer SIP trunks or Register SIP trunks (register to each other). Therefore, please properly configure SIP trunks on the UCM630xA first before using remote BLF feature. Please note the SIP end points need support event list BLF in order to monitor remote extensions.

When an event list is created on the UCM630xA and remote extensions are added to the list, the UCM630xA will send out SIP SUBSCRIBE to the remote UCM630xA to obtain the remote extension status. When the SIP end points register and subscribe to the local UCM630xA event list, it can obtain the remote extension status from this event list. Once successfully configured, the event list page will show the status of total extension and



subscribers for each event list. Users can also select the event URI to check the monitored extension's status and the subscribers' details.

---

 **Notes:**

- To configure LDAP sync, please go to UCM630xA Web GUI → **Extension/Trunk** → **VoIP Trunk**. You will see "Sync LDAP Enable" option. Once enabled, please configure password information for the remote peer UCM630xA to connect to the local UCM630xA. Additional information such as port number, LDAP outbound rule, LDAP Dialed Prefix will also be required. Both the local UCM630xA and remote UCM630xA need enable LDAP sync option with the same password for successful connection and synchronization.
  - Currently LDAP sync feature only works between two UCM630xAs.
  - (Theoretically) Remote BLF monitoring will work when the remote PBX being monitored is non-UCM630xA PBX. However, it might not work the other way around depending on whether the non-UCM630xA PBX supports event list BLF or remote monitoring feature.
- 

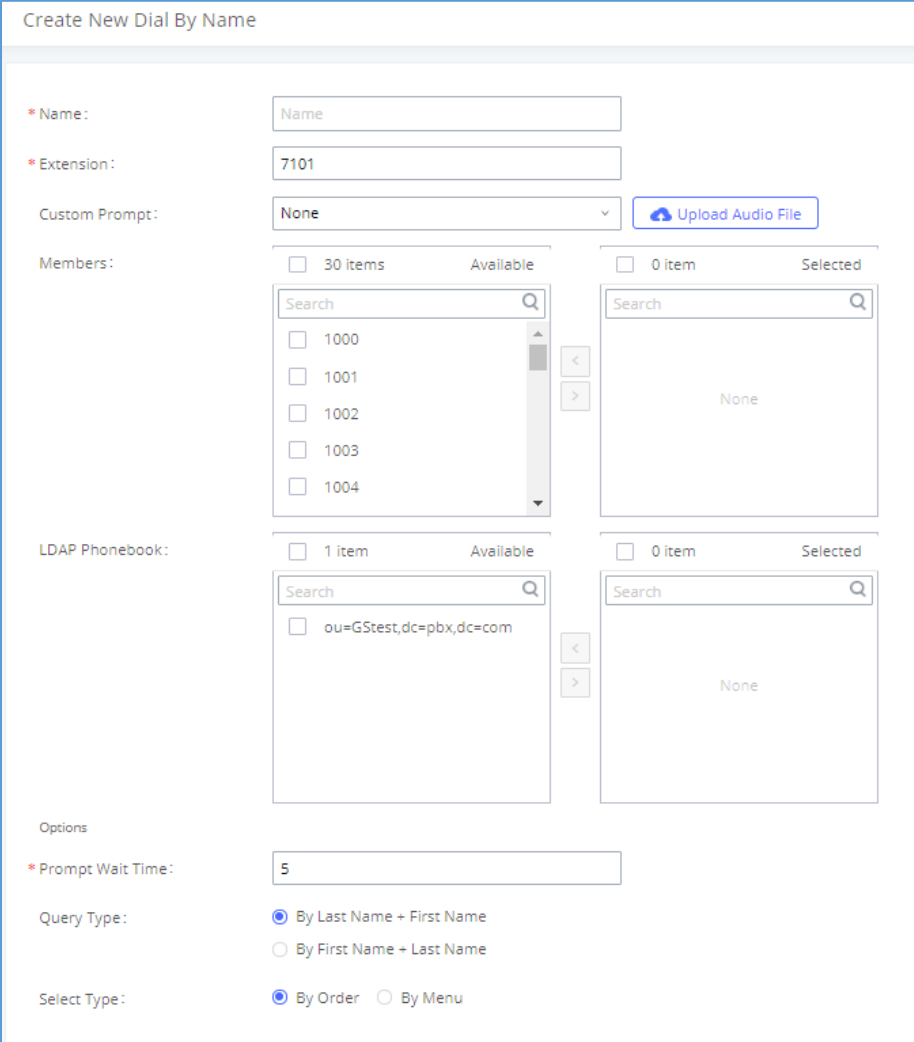


## DIAL BY NAME

Dial by Name is a feature on the PBX that allows caller to search a person by first or last name via his/her phone's keypad. The administrator can define the Dial by Name directory including the desired extensions in the directory and the searching type by "first name" or "last name". After dialing in, the PBX IVR/Auto Attendant will guide the caller to spell the digits to find the person in the Dial by Name directory. This feature allows customers/clients to use the guided automatic system to contact the enterprise employees without having to know the extension number, which brings convenience and improves business image for the enterprise.

### Dial by Name Configuration

The administrators can create the dial by name group under Web GUI→**Call Features**→**Dial By Name**.

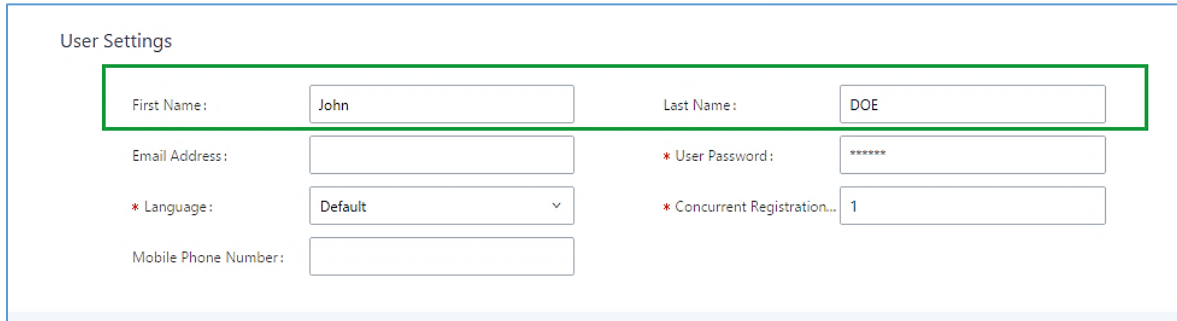


The screenshot shows the 'Create New Dial By Name' configuration interface. It includes the following fields and sections:

- Name:** A text input field containing the word 'Name'.
- Extension:** A text input field containing the number '7101'.
- Custom Prompt:** A dropdown menu set to 'None' and an 'Upload Audio File' button.
- Members:** Two columns of selection boxes. The 'Available' column shows 30 items with a search bar and a list of extensions: 1000, 1001, 1002, 1003, and 1004. The 'Selected' column shows 0 items and is currently empty.
- LDAP Phonebook:** Two columns of selection boxes. The 'Available' column shows 1 item: 'ou=GStest,dc=pbx,dc=com'. The 'Selected' column shows 0 items and is currently empty.
- Options:**
  - Prompt Wait Time:** A text input field containing the number '5'.
  - Query Type:** Radio buttons for 'By Last Name + First Name' (selected), 'By First Name + Last Name', and 'By Order'.
  - Select Type:** Radio buttons for 'By Order' (selected) and 'By Menu'.

**Figure 196: Create Dial by Name Group**





The screenshot shows a 'User Settings' form with the following fields:

- First Name: John
- Last Name: DOE
- Email Address: (empty)
- \* User Password: \*\*\*\*\*
- \* Language: Default
- \* Concurrent Registration: 1
- Mobile Phone Number: (empty)

**Figure 197: Configure Extension First Name and Last Name**

### 1. Name

Enter a Name to identify the Dial by Name group.

### 2. Extension

Configure the direct dial extension for the Dial By Name group.

### 3. Custom Prompt

This option sets a custom prompt for directory to announce to a caller. The file can be uploaded from the page "Custom Prompt". Click "Upload Audio File" to add additional record.

### 4. Available Extensions/Selected Extensions

Select available extensions from the left side to the right side as the directory for the Dial By Name group. Only the selected extensions here can be reached by the Dial By Name IVR when dialing into this group. The extensions here must have a valid first name and last name configured under Web GUI → **Extension/Trunk** → **Extensions** in order to be searchable in Dial By Name directory through IVR. By specifying the extensions here, the administrators can make sure unscreened calls will not reach the company employee if he/she does not want to receive them directly.

### 5. Prompt Wait Time

Configure "Prompt Wait Time" for Dial By Name feature. During Dial By Name call, the caller will need to input the first letters of First/Last name before this wait time is reached. Otherwise, timeout will occur, and the call might hang up. The timeout range is between 3 and 60 seconds.

### 6. Query Type

Specify the query type. This defines how the caller will need to enter to search the directory.

By First Name: enter the first 3 digits of the first name to search the directory.

By Last Name: enter the first 3 digits of the last name to search the directory.



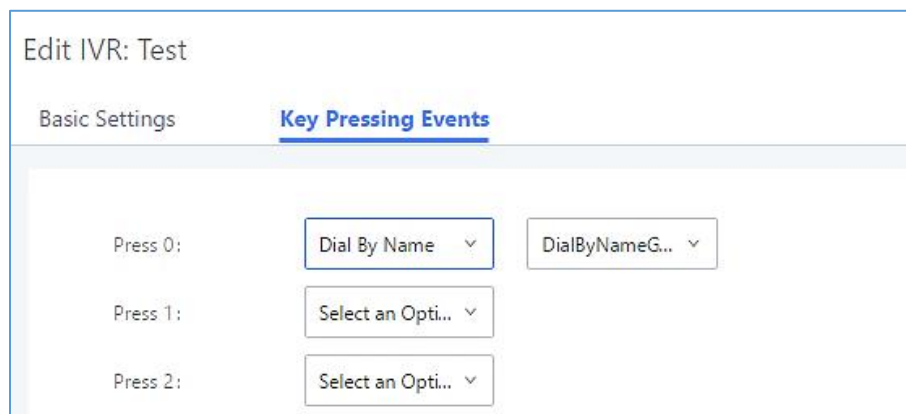
## 7. Select Type

Specify the select type on the searching result. The IVR will confirm the name/number for the party the caller would like to reach before dialing out.

**By Order:** After the caller enters the digits, the IVR will announce the first matching party's name and number. The caller can confirm and dial out if it is the destination party, or press \* to listen to the next matching result if it is not the desired party to call.

**By Menu:** After the caller enters the digits, the IVR will announce 8 matching results. The caller can press number 1 to 8 to select and call or press 9 for results in next page.

The Dial by Name group can be used as the destination for inbound route and key pressing event for IVR. The group name defined here will show up in the destination list when configuring IVR and inbound route. If Dial by Name is set as a key pressing event for IVR, user could use '\*' to exit from Dial by Name, then re-enter IVR and start a new event. The following example shows how to use this option.



Edit IVR: Test  
 Basic Settings      **Key Pressing Events**

Press 0:	Dial By Name ▾	DialByNameG... ▾
Press 1:	Select an Opti... ▾	
Press 2:	Select an Opti... ▾	

**Figure 198: Dial By Name Group In IVR Key Pressing Events**



### Edit Inbound Rule Save

* Pattern:	<input type="text" value="."/>	CallerID Pattern:	<input type="text" value="Separate patterns by commas, such as *_*"/>
Disable This Route:	<input type="checkbox"/>	Prepend Trunk Name:	<input type="checkbox"/>
Prepend User Defined Nam...:	<input type="checkbox"/> <input type="text"/>	Inbound Multiple Mode:	<input type="checkbox"/>
Alert-info:	<input type="text" value="None"/>	Dial Trunk:	<input type="checkbox"/>
Privilege Level:	<input type="text" value="Internal"/>	DID Destination:	<input type="text"/>
Allowed to seamless transfe...:	<input type="text"/>		
Default Mode			
* Default Destination:	<input type="text" value="Dial By Name"/>	<input type="text" value="DialByNameGP1"/>	

**Figure 199: Dial By Name Group In Inbound Rule**

Please refer to [Username Prompt Customization] for User Name Prompt Customization.



## ACTIVE CALLS AND MONITOR

The active calls on the UCM630xA are displayed in Web GUI→**System Status**→**Active Calls** page. Users can monitor the status, hang up the call as well as barge in the active calls in real time manner.

### Active Calls Status

To view the status of active calls, navigate to Web GUI→**System Status**→**Active Calls**. The following figure shows extension 1004 is calling 1000. 1000 is ringing.

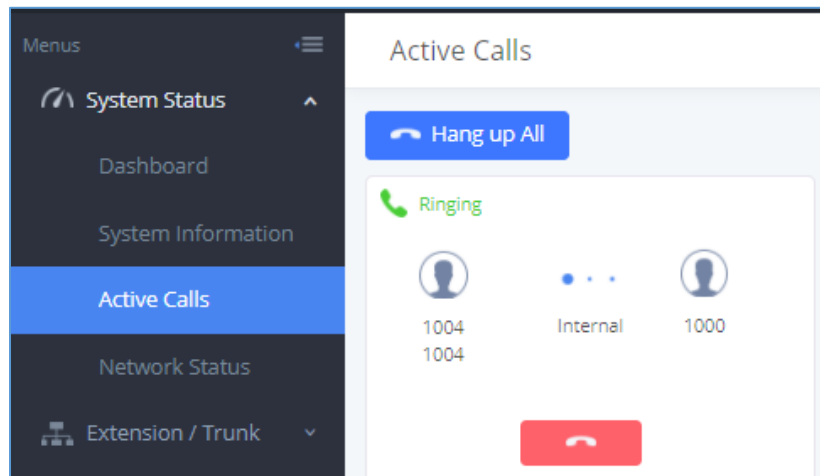


Figure 200: Status→PBX Status→Active Calls - Ringing

The following figure shows the call between 1000 and 5555 is established.

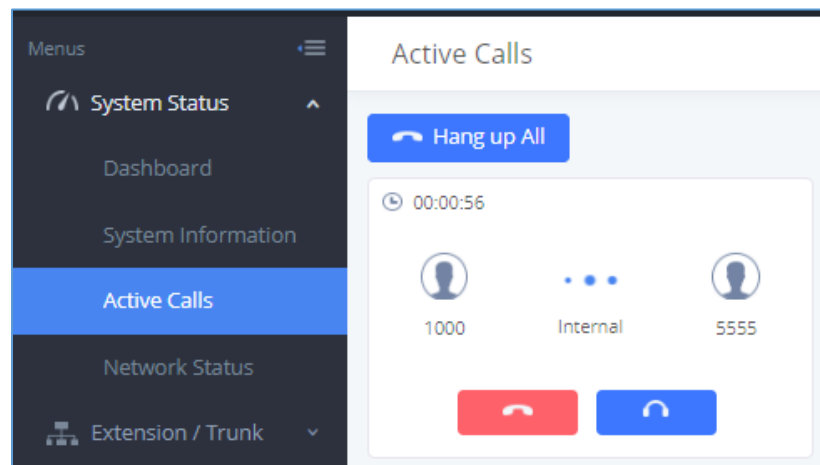
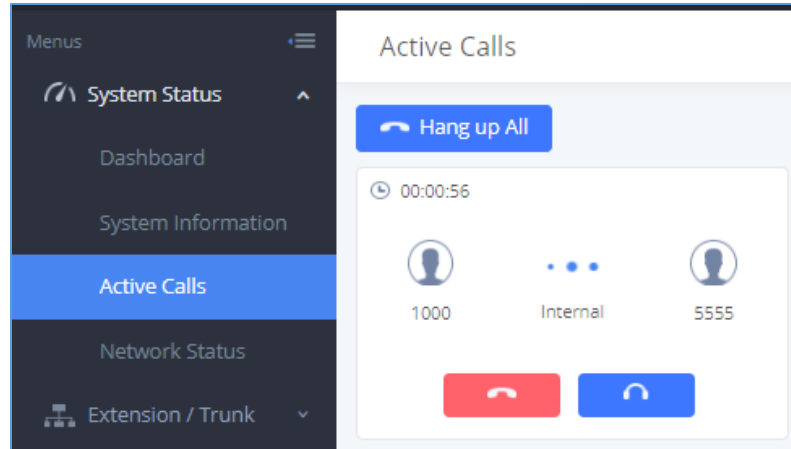


Figure 201: Status→PBX Status→Active Calls – Call Established

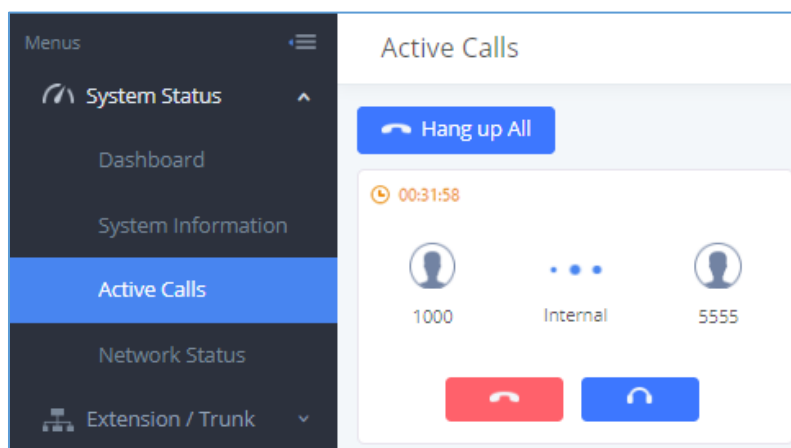


The gray color of the active call means the connection of call time is less than half an hour. It means this call is normal.



**Figure 202: Call Connection less than half hour**

The orange color of the active call means the connection of call time is greater than half an hour but less than one hour. It means this call is a bit long.

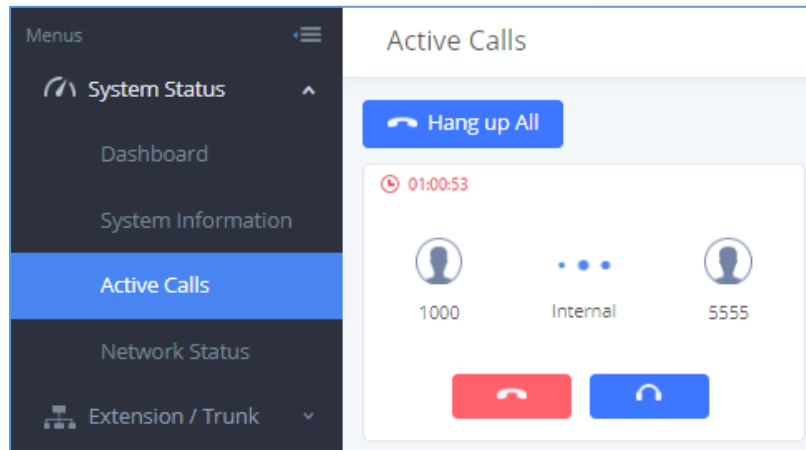


**Figure 203: Call Connection between half an hour and one hour**

The red color of the active call means the connection of call time is more than one hour. It means this call could be abnormal.


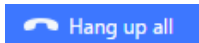






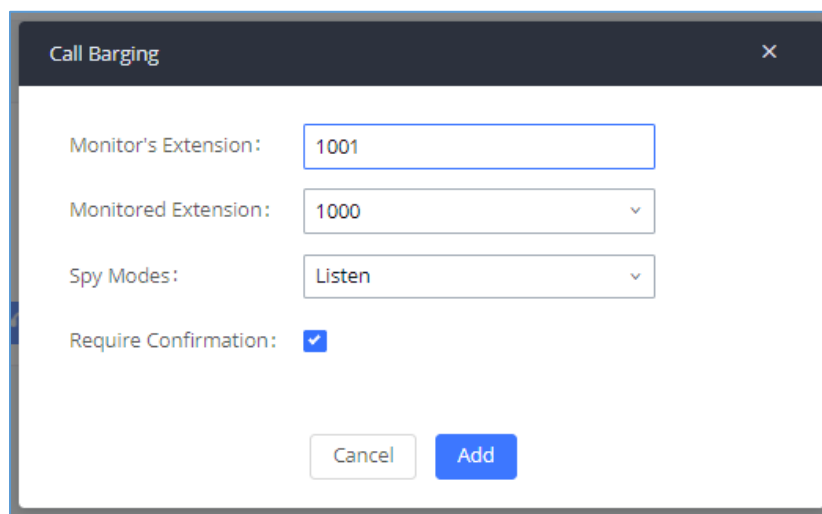
**Figure 204: Call Connection more than one hour**

## Hang Up Active Calls

To hang up an active call, click on  icon in the active call dialog. Users can also click on  to hang up all active calls.

## Call Monitor

During an active call, click on icon  and the monitor dialog will pop up.



**Figure 205: Configure to Monitor an Active Call**

In the “Monitor” dialog, configure the following to monitor an active call:

1. Enter an available extension for “Monitor’s Extension” which will be used to monitor the active call.



2. "Monitored Extension" must be one of the parties in the active call to be monitored.
3. Select spy mode. There are three options in "Spy Mode".
  - **Listen**  
In "Listen" mode, the extension monitoring the call can hear both parties in the active call but the audio of the user on this extension will not be heard by either party in the monitored active call.
  - **Whisper**  
In "Whisper" mode, the extension monitoring the call can hear both parties in the active call. The user on this extension can only talk to the selected monitored extension and he/she will not be heard by the other party in the active call. This can be usually used to supervise calls.
  - **Barge**  
In "Barge" mode, the extension monitoring the call can talk to both parties in the active call. The call will be established similar to three-way meeting.
4. Enable or disable "Require Confirmation" option. If enabled, the confirmation of the invited monitor's extension is required before the active call can be monitored. This option can be used to avoid adding participant who has auto-answer configured, or call forwarded to voicemail.
5. Click on "Add". An INVITE will be sent to the monitor's extension. The monitor can answer the call and start monitoring. If "Require Confirmation" is enabled, the user will be asked to confirm to monitor the call.

Another way to monitor active calls is to dial the corresponding feature codes from an extension. Please refer to **[Table 95: UCM630xA Feature Codes]** and **[Call Recording]** section for instructions.



## CALL FEATURES

The UCM630xA supports call recording, transfer, call forward, call park and other call features via feature code. This section lists all the feature codes in the UCM630xA and describes how to use the call features.

### Feature Codes

Table 95: UCM630xA Feature Codes

Feature Maps	
<b>Blind Transfer</b>	<ul style="list-style-type: none"> <li>• Default code: <b>#1</b></li> <li>• Enter the code during active call. After hearing "Transfer", you will hear dial tone. Enter the number to transfer to. Then the user will be disconnected, and transfer is completed.</li> <li>• Options:               <ul style="list-style-type: none"> <li><b>Disable</b></li> <li><b>Allow Caller:</b> Enable the feature code on caller side only.</li> <li><b>Allow Callee:</b> Enable the feature code on callee side only.</li> <li><b>Allow Both:</b> Enable the feature code on both caller and callee.</li> </ul> </li> </ul>
<b>Attended Transfer</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*2</b></li> <li>• Enter the code during active call. After hearing "Transfer", you will hear the dial tone. Enter the number to transfer to and the user will be connected to this number. Hang up the call to complete the attended transfer. In case of the called party does not answer, users could press *0 to cancel the call and retrieve the first call leg.</li> <li>• Options:               <ul style="list-style-type: none"> <li><b>Disable</b></li> <li><b>Allow Caller:</b> Enable the feature code on caller side only.</li> <li><b>Allow Callee:</b> Enable the feature code on callee side only.</li> <li><b>Allow Both:</b> Enable the feature code on both caller and callee.</li> </ul> </li> </ul>
<b>Seamless Transfer</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*44</b> (Disabled by default).</li> <li>• Seamless Transfer allows user to perform blind transfer using UCM feature code without having music on hold presented during the transfer process, it minimizes the interruption during transfer, making the process smooth and simple.</li> <li>• During an active call use the feature code (*44 by default) followed by the number you want to transfer to in order to perform the seamless transfer.</li> </ul>



<b>Disconnect</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*0</b></li> <li>• Enter the code during active call. It will disconnect the call.</li> <li>• Options:  <b>Disable</b>  <b>Allow Caller:</b> Enable the feature code on caller side only.  <b>Allow Callee:</b> Enable the feature code on callee side only.  <b>Allow Both:</b> Enable the feature code on both caller and callee.</li> </ul>
<b>Call Park</b>	<ul style="list-style-type: none"> <li>• Default code: <b>#72</b></li> <li>• Enter the code during active call to park the call.</li> <li>• Options:  <b>Disable</b>  <b>Allow Caller:</b> Enable the feature code on caller side only.  <b>Allow Callee:</b> Enable the feature code on callee side only.  <b>Allow Both:</b> Enable the feature code on both caller and callee.</li> </ul>
<b>Start/Stop Call Recording</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*3</b></li> <li>• Enter the code followed by # or SEND to start recording the audio call and the UCM630xA will mix the streams natively on the fly as the call is in progress.</li> <li>• Options:  <b>Disable</b>  <b>Allow Caller:</b> Enable the feature code on caller side only.  <b>Allow Callee:</b> Enable the feature code on callee side only.  <b>Allow Both:</b> Enable the feature code on both caller and callee.</li> </ul>
<b>Enable Recording Whitelist</b>	Enable the Recording Whitelist feature
<b>Recording Operation Whitelist</b>	Select extension in the whitelist that can use the *3 recording function.
<b>Feature Code Digits Timeout</b>	Set the maximum interval (ms) between digits for feature code activation
<b>DND/Call Forward</b>	
<b>Do Not Disturb (DND) Activate</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*77</b></li> </ul>
<b>Do Not Disturb (DND) Deactivate</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*78</b></li> </ul>
<b>Call Forward Busy Activate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*90</b></li> <li>• Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.</li> </ul>



<b>Call Forward Busy Deactivate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*91</b></li> </ul>
<b>Call Forward No Answer Activate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*92</b></li> <li>• Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.</li> </ul>
<b>Call Forward No Answer Deactivate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*93</b></li> </ul>
<b>Call Forward Unconditional Activate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*72</b></li> <li>• Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.</li> </ul>
<b>Call Forward Unconditional Deactivate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*73</b></li> </ul>
<b>Remote Call Forward Enable</b>	<p>Enable this option and configure the Remote Call Forward Whitelist below to allow specific extensions to dial the remote call forwarding feature codes to set call forwarding for any extension.</p>
<b>Feature Codes</b>	
<b>Voicemail Access Code</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*98</b></li> <li>• Enter *98 and follow the voice prompt. Or dial *98 followed by the extension and # to access the entered extension's voicemail box.</li> </ul>
<b>My Voicemail</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*97</b></li> <li>• Press *97 to access the voicemail box.</li> </ul>
<b>Agent Pause</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*83</b></li> <li>• Pause the agent in all call queues.</li> </ul>
<b>Agent Unpause</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*84</b></li> <li>• Unpause the agent in all call queues.</li> </ul>
<b>Paging Prefix</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*81</b></li> <li>• To page an extension, enter the code followed by the extension number.</li> </ul>
<b>Intercom Prefix</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*80</b></li> <li>• To intercom an extension, enter the code followed by the extension number.</li> </ul>
<b>Blacklist Add</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*40</b></li> <li>• To add a number to blacklist for inbound route, dial *40 and follow the voice prompt to enter the number.</li> </ul>



<b>Blacklist Remove</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*41</b></li> <li>• To remove a number from current blacklist for inbound route, dial *41 and follow the voice prompt to remove the number.</li> </ul>
<b>Call Pickup on Ringing</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>**</b></li> <li>• To pick up a call for any extension xxxx, enter the code followed by the extension number xxxx.</li> </ul>
<b>Pickup In-call</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*45</b> (Disabled by default).</li> <li>• If “Pickup In-call” feature is enabled, only the extensions added in “Allowed to seamless transfer” in the extension’s Seamless Transfer Privilege Control List” can pick up the call.</li> </ul>
<b>Pickup Extension</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*8</b></li> <li>• This code is for the pickup group, which can be assigned for each extension on the extension configuration page.</li> <li>• If there is an incoming call to an extension, the other extensions within the same pickup group can dial *8 directly to pick up the call.</li> </ul>
<b>Direct Dial Voicemail Prefix</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*</b></li> <li>• This code is for the user to directly dial or transfer to an extension's voicemail.</li> <li>• For example, directly dial *5000 will have to call go into the extension 5000's voicemail. If the user would like to transfer the call to the extension 5000's voicemail, enter *5000 as the transfer target number.</li> </ul>
<b>Direct Dial Mobile Phone Prefix</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*88</b></li> <li>• If you have the permission to call mobile phone number, use this prefix plus the extension number can dial the mobile phone number of this extension directly.</li> </ul>
<b>Call Completion Request</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*11</b></li> <li>• This code is for the user who wants to use Call Completion to complete a call.</li> </ul>
<b>Call Completion Cancel</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*12</b></li> <li>• This code is for the user who wants to cancel Call Completion request.</li> </ul>
<b>Enable Spy</b>	Check this box to enable spy feature codes. Disabled by default.



<b>Listen Spy</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*54</b> ("Enable Spy" needs to be checked)</li> <li>• This is the feature code to listen in on a call to monitor performance. Monitor's line will be muted, and neither party will hear from the monitor's extension.</li> </ul>
<b>Whisper Spy</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*55</b> ("Enable Spy" needs to be checked)</li> <li>• This is the feature code to speak to one side of the call (for example, whisper to employees to help them handle a call). Only one side will be able to hear from the monitor's extension.</li> </ul>
<b>Barge Spy</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*56</b> ("Enable Spy" needs to be checked)</li> <li>• This is the feature code to join in on the call to assist both parties.</li> </ul>
<b>Wakeup Service</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*36</b></li> <li>• Dial this code to access UCM wakeup service, you can add, update, activate or deactivate wakeup service.</li> </ul>
<b>PMS Wakeup Service</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*35</b></li> <li>• Dial this code to access UCM PMS wakeup service, you can add, update, activate or deactivate PMS wakeup service.</li> </ul>
<b>PMS Remote Wakeup Service</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*37</b></li> <li>• Allows the user to add, update, activate, and deactivate PMS wakeup service for other extensions.</li> </ul>
<b>Update PMS Room Status</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*23</b></li> <li>• Use this code with maid code to update PMS room status. Choose the status to set after hearing the prompt, for example: for maid 001 dial *23001 and then 1 after hearing the prompt.</li> </ul>
<b>Presence Status</b>	<ul style="list-style-type: none"> <li>• Dial this code to set the presence status of the extension.</li> <li>• Possible options are:           <ul style="list-style-type: none"> <li>1:"unavailable"</li> <li>2:"available"</li> <li>3:"away"</li> <li>4:"chat"</li> <li>5:"dnd"</li> <li>6:"userdef"</li> </ul> </li> </ul>
<b>Dynamic Agent Logout</b>	<ul style="list-style-type: none"> <li>• Default Code: *85</li> <li>• Use this code to logout the dynamic agent from all queues.</li> </ul>

The UCM630xA also allows user to one click enable / disable specific feature code as shown below:



Feature Codes
Save Cancel

Feature Maps
DND/Call Forward
Feature Codes

Reset All
Default All

<ul style="list-style-type: none"> <li>* Voicemail Access Code: <input type="text" value="*98"/> <input checked="" type="checkbox"/></li> <li>* Agent Pause: <input type="text" value="*83"/> <input checked="" type="checkbox"/></li> <li>* Paging Prefix: <input type="text" value="*81"/> <input checked="" type="checkbox"/></li> <li>* Blacklist Add: <input type="text" value="*40"/> <input checked="" type="checkbox"/></li> <li>* Call Pickup on Ringing: <input type="text" value="**"/> <input checked="" type="checkbox"/></li> <li>* Pickup Extension: <input type="text" value="*8"/> <input checked="" type="checkbox"/></li> <li>* Direct Dial Mobile Phone Prefix: <input type="text" value="*88"/> <input checked="" type="checkbox"/></li> <li>* Call Completion Cancel: <input type="text" value="*12"/> <input checked="" type="checkbox"/></li> <li>* Listen Spy: <input type="text" value="*54"/> <input type="checkbox"/></li> <li>* Barge Spy: <input type="text" value="*56"/> <input type="checkbox"/></li> <li>* PMS Wakeup Service: <input type="text" value="*35"/> <input checked="" type="checkbox"/></li> <li>* Presence Status: <input type="text" value="*48"/> <input checked="" type="checkbox"/></li> </ul>	<ul style="list-style-type: none"> <li>* My Voicemail: <input type="text" value="*97"/> <input checked="" type="checkbox"/></li> <li>* Agent Unpause: <input type="text" value="*84"/> <input checked="" type="checkbox"/></li> <li>* Intercom Prefix: <input type="text" value="*80"/> <input checked="" type="checkbox"/></li> <li>* Blacklist Remove: <input type="text" value="*41"/> <input checked="" type="checkbox"/></li> <li>* Pickup In-call: <input type="text" value="*45"/> <input type="checkbox"/></li> <li>* Direct Dial Voicemail Prefix: <input type="text" value="*"/> <input checked="" type="checkbox"/></li> <li>* Call Completion Request: <input type="text" value="*11"/> <input checked="" type="checkbox"/></li> <li>Enable Spy: <input type="checkbox"/></li> <li>* Whisper Spy: <input type="text" value="*55"/> <input type="checkbox"/></li> <li>* Wakeup Service: <input type="text" value="*36"/> <input checked="" type="checkbox"/></li> <li>* Update PMS Room Status: <input type="text" value="*23"/> <input checked="" type="checkbox"/></li> <li>* Dynamic Agent Logout: <input type="text" value="*85"/> <input checked="" type="checkbox"/></li> </ul>
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**Figure 206: Enable/Disable Feature codes**

## Parking Lot

User can create parking lots and their related slots under Web GUI → **Call Features** → **Parking Lot**. In the Parking Lot page, users can create lots of their own. This allows different groups within an organization to have their own parking lots instead of sharing one large parking lot with others. While creating a new parking lot, users can assign it a range that they think is appropriate for the group that will use the parking lot.

Parking Lot
Parking Lot Status

Parking Lot Settings

+ Add

EXTENSION	NAME	SLOTS	OPTIONS
700	DefaultLot	701-720	✎ ✖
7000	Dales	7001-7022	✎ ✖

Total: 2
10 / page
Goto 1

**Figure 207: Parking Lot**

User can create a new Parking lot by clicking on button “Add” :





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**Create New Parking Lot**

<p>* Parking Lot Extension: <input type="text"/></p> <p>* Parking Slots: <input type="text"/></p> <p>* Parking Timeout (s): <input type="text" value="300"/></p> <p>Fallover Destination: <input type="text"/></p> <p>Forward to Destination on Timeout: <input type="checkbox"/></p>	<p>* Parking Lot Name: <input type="text"/></p> <p>Use parklot as extension: <input type="checkbox"/></p> <p>Music on Hold Playlists: <input type="text" value="Default"/></p> <p>Ring-All Callback on Timeout: <input type="checkbox"/></p>
---	--

**Figure 208: New Parking Lot**

**Table 96 : Parking Lot**

<b>Parking Lot Extension</b>	<ul style="list-style-type: none"> <li>• Default Extension: <b>700</b></li> <li>• During an active call, initiate blind transfer and then enter this code to park the call.</li> </ul>
<b>Parking Lot Name</b>	<ul style="list-style-type: none"> <li>• Set a name to the parking lot</li> </ul>
<b>Parked Slots</b>	<ul style="list-style-type: none"> <li>• Default Extension: <b>701-720</b></li> <li>• These are the extensions where the calls will be parked, i.e., parking lots that the parked calls can be retrieved.</li> </ul>
<b>Use Parklot as Extension</b>	<ul style="list-style-type: none"> <li>• If checked, the parking lot number can be used as extension. The user can transfer the call to the parking lot number to park the call. Please note this parking lot number range might conflict with extension range.</li> </ul>
<b>Parking Timeout (s)</b>	<ul style="list-style-type: none"> <li>• Default setting is <b>300</b> seconds, and the maximum limit is <b>99.999</b> seconds.</li> <li>• This is the timeout allowed for a call to be parked. After the timeout, if the call is not picked up, the extension who parks the call will be called back.</li> </ul>
<b>Music On Hold Classes</b>	Select the Music on Hold Class.
<b>Fallover Destination</b>	Configures a callback failover destination when the extension that is called back is busy. The call will be routed to the destination number and this reduces the chance of dropping parked calls.
<b>Ring All Callback on Timeout</b>	If enabled, all registered endpoints of the extension will ring when callback occurs. Otherwise, only the original endpoint will be called back.
<b>Forward to destination on timeout</b>	If enabled, the call will be routed to the configured destination upon timeout. Otherwise, the call will be routed back to the original caller.



<b>Timeout Destination</b>	This option appears once Forward to Destination on Timeout is enabled. Upon park timeout, the call will be routed to the configured destination.
<b>Parking Lot Timeout Alert-Info</b>	Adds an Alert-Info header to parking lot callbacks after the Parking Timeout has been reached.

## Call Park

The UCM630xA provides call park and call pickup features via feature code.

### Park a Call

There are two feature codes that can be used to park the call.

- **Feature Maps→Call Park (Default code #72)**  
 During an active call, press #72 and the call will be parked. Parking lot number (default range 701 to 720) will be announced after parking the call.
- **Feature Misc→Call Park (Default code 700)**  
 During an active call, initiate blind transfer (default code #1) and then dial 700 to park the call. Parking lot number (default range 701 to 720) will be announced after parking the call.

### Retrieve Parked Call

To retrieve the parked call, simply dial the parking lot number and the call will be established. If a parked call is not retrieved after the timeout, the original extension who parks the call will be called back.

### Monitor Call Park CID Name Information (GXP21xx, GRP261x Phones Only)

Users can see the CID name information of parked calls. VPK/MPKs must be configured as "Monitored Call Park" with the desired parking lot extension. The display will alternate between displaying the parking lot extension and the call's CID name. There is no need to configure anything on the UCM.





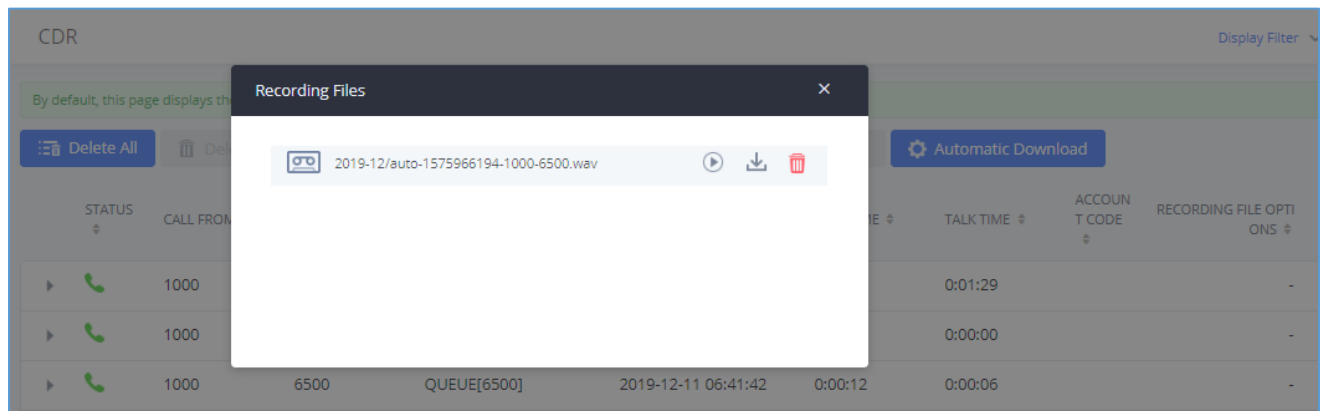
Figure 209: Monitored Call Park CID name

## Call Recording

The UCM630xA allows users to record audio during the call. If "Auto Record" is turned on for an extension, ring group, call queue or trunk, the call will be automatically recorded when there is established call with it. Otherwise, please follow the instructions below to manually record the call.

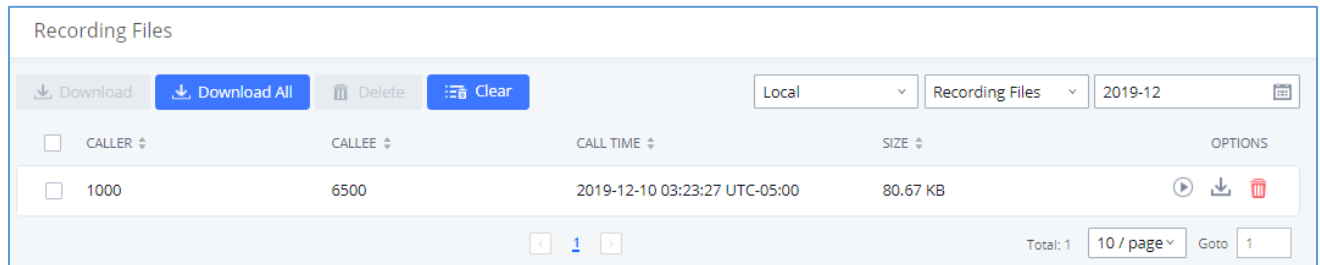


1. Make sure the feature code for "Start/Stop Call Recording" is configured and enabled.
2. After establishing the call, enter the "Start/Stop Call Recording" feature code (by default it is \*3) followed by # or SEND to start recording.
3. To stop the recording, enter the "Start/Stop Call Recording" feature code (by default it is \*3) followed by # or SEND again. Or the recording will be stopped once the call hangs up.
4. The recording file can be retrieved under Web GUI→**CDR**. Click on  to show and play the recording or click on  to download the recording file.



**Figure 210: Download Recording File from CDR Page**

The above recorded call's recording files are also listed under the UCM630xA Web GUI→**CDR**→**Recording Files**.



**Figure 211: Download Recording File from Recording Files Page**

## Enable Spy

If “Enable Spy” option is enabled, feature codes for Listen Spy, Whisper Spy and Barge Spy are available for users to dial from any extension to perform the corresponding actions.

Assume a call is on-going between extension A and extension B, user could dial the feature code from extension C to listen on their call (\*54 by default), whisper to one side (\*55 by default), or barge into the call (\*56 by default). Then the user will be asked to enter the number to call, which should be either side of the active call, extension A or B in this example.



---

 **Caution:**

“Enable Spy” allows any user to listen to any call by feature codes. This may result in the leakage of user privacy.

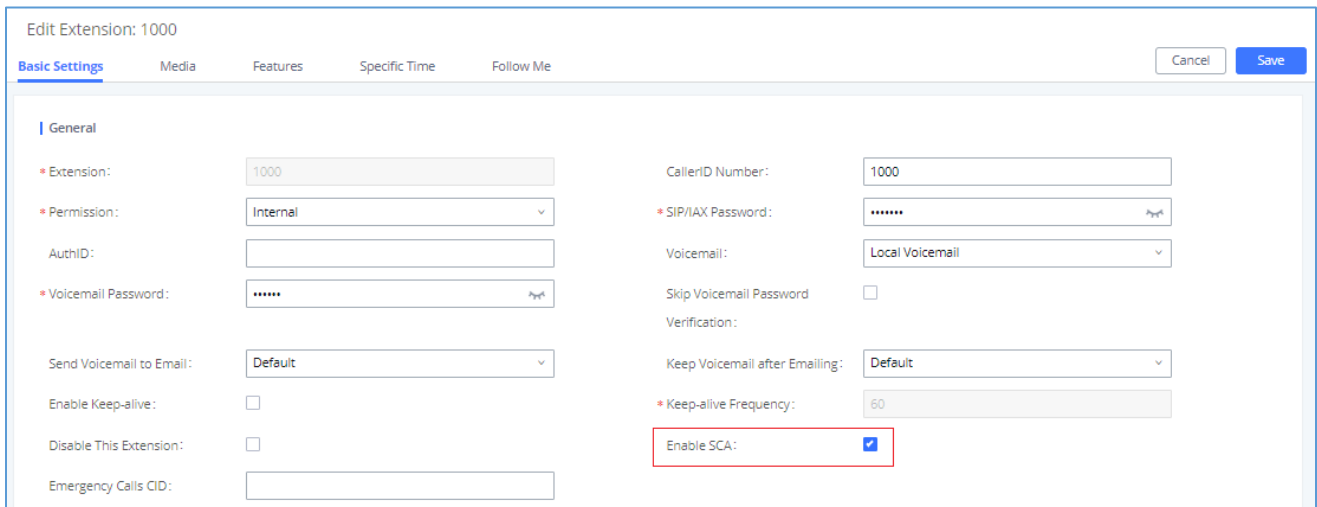
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## Shared Call Appearance (SCA)

Shared Call Appearance (SCA) functionality has been added to the UCM. With SCA, users can assign multiple devices to one extension, configure endpoints to monitor that extension, make actions on behalf of that extension such as viewing call status and placing and receiving calls, and even barging into existing calls. To configure the SCA functionality, please follow the steps below:

1. Users can enable SCA by navigating to the Extensions page, editing the desired extension, and enabling the option SCA.

**Note:** With SCA enabled, the Concurrent Registrations field can only have a value of 1.





The screenshot shows the 'Edit Extension: 1000' configuration page. The 'Basic Settings' tab is selected. The 'General' section contains the following fields:

- Extension: 1000
- Permission: Internal
- AuthID: (empty)
- Voicemail Password: (masked)
- Send Voicemail to Email: Default
- Enable Keep-alive:
- Disable This Extension:
- Emergency Calls CID: (empty)
- CallerID Number: 1000
- SIP/IAX Password: (masked)
- Voicemail: Local Voicemail
- Skip Voicemail Password Verification:
- Keep Voicemail after Emailing: Default
- Keep-alive Frequency: 60
- Enable SCA:  (highlighted with a red box)

**Figure 212: Enabling SCA option under Extension’s Settings**

2. After enabling the option, navigate to *Call Features* → *SCA*. The newly enabled SCA extension will be listed. Click the “+” button under the Options column to add a number that will share the main extension’s call appearance, which will be called private numbers.



SCA					
SCA Number Group		SCA Line Status			
STATUS	SHARED LINE	ROLE	IP AND PORT	SUBSCRIBED	OPTIONS
Unavailable	1000	shared	--	no	 

Total: 1 | 10 / page | Goto: 1

**Figure 213: SCA Number Configuration**

- Configure the private number as desired.

**Add Private Number** ✕

\* Private Number:

Related Shared Line:

Enable This Number:

Allow Origination from This Number:

Allow Termination to This Number:

**Figure 214: SCA Private Number Configuration**

- Once the private number has been created, users must now register a device to it. To properly register a device to the private number, use the configured private number as the SIP User ID. Auth ID and Password will be the same as the main extensions. Once registration is complete, SCA is now configured.



**Edit SCA Number Group:**

---

\* Shared Line Number:

Allow Call Retrieve from

Another Location:

Alert All Appearances for

Group Paging Calls:

Multiple Call Arrangement:

Allow Bridging between

Locations:

Bridge Warning Tone:

**Figure 215: SCA Options**

- Next, configure the VPK or MPK to Shared for both the main extension and the private number. SCA is now configured for both endpoint devices.

The following table describe the SCA Number configuration setting:

**Table 97: Add SCA Private Number**

<b>Private Number</b>	Configures the private number for the SCA.
<b>Related Shared Line</b>	Display the related shared line.
<b>Enable This Number</b>	Whether enable this private number. If not enabled, this private number is only record in DB, it will not affect other system feature.
<b>Allow Origination from This Number</b>	Enable this option will allow calling from this private number. By default, it is enabled.
<b>Allow Termination to This Number</b>	Enable this option will allows calls to this private number. By default, it is enabled.

The following table describes the options available when editing the SCA number:



**Table 98: Editing the SCA Number**

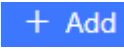
<b>Shared Line Number</b>	While SCA is enabled, this number will be the same as the extension number.
<b>Allow Call Retrieve from Another Location</b>	Allows remote call retrieval. Must be enabled in public hold. By default, it is enabled.
<b>Alert All Appearances for Group Paging Calls</b>	Allows all SCA group members to ring when the SCA shared number is paged. If disabled, only the SCA shared number will ring when paged. By default, it is disabled.
<b>Multiple Call Arrangement</b>	Allows simultaneous calls in an SCA group. By default, it is disabled.
<b>Allow Bridging between Locations</b>	Allows location bridging for SCA group. Must be enabled when using the Barge-In feature. By default, it is disabled.
<b>Bridge Warning Tone</b>	<p>Configures the notification in the bridge when another party join.</p> <ul style="list-style-type: none"> <li>• <b>None:</b> No notification sound.</li> <li>• <b>Barge-In only:</b> Notification sound will play when another party join.</li> <li>• <b>Barge-In and Repeat:</b> Notification sound will play when another party joins and repeat every 30 seconds.</li> </ul> <p>By default, it is set to “Barge-In Only”.</p>



## ANNOUNCEMENT

The Announcement feature (not to be confused with Announcement Paging and Announcement Center) is a feature that allows users to set an unskippable audio file to play to callers before routing them to a configured destination. Announcements can be configured as a destination in the Inbound Routes page.


To configure Announcement, users need to follow below steps:

1. Navigate on the web GUI under “Call Features → Announcement”
2. Click on  to add a new Announcement.
3. Configure the required fields Name, Prompt, Default Destination to be used for the announcement.

Save and apply the configuration.

**Create New Announcement**

\* Name:

Prompt:  

Default Destination:

**Figure 216: Announcement settings**

The table below gives more description of the configuration parameters when creating Announcement.

**Table 99: Announcement Parameters**

<b>Name</b>	Configure the name of the Announcement.
<b>Prompt</b>	Audio file that needs to be uploaded in order to be played for a specific destination.
<b>Default Destination</b>	Select the destination where to play the audio file.





## PBX SETTINGS

This section describes internal options that have not been mentioned in previous sections yet. The settings in this section can be applied globally to the UCM630xA, including general configurations, jitter buffer, RTP settings, ports config and STUN monitor. The options can be accessed via Web GUI→**PBX Settings**→**General Settings**.

### PBX Settings/General Settings

Table 100: Internal Options/General

General Preferences	
<b>Global Outbound CID</b>	Configure the global CallerID used for all outbound calls when no other CallerID is defined with higher priority. If no CallerID is defined for extension or trunk, the global outbound CID will be used as CallerID.
<b>Global Outbound CID Name</b>	Configure the global CallerID Name used for all outbound calls. If configured, all outbound calls will have the CallerID Name set to this name. If not, the extension's CallerID Name will be used.
<b>Ring Timeout</b>	Configure the number of seconds to ring an extension before the call goes to the user's voicemail box. The default setting is 60. <b>Note:</b> This is the global value used for each extension if "Ring Timeout" field is left empty on the extension configuration page.
<b>Call Duration Limit</b>	Configure the maximum duration of call-blocking.
<b>Maximum Call Duration (s)</b>	The maximum call duration (in seconds). The default value 0 means no limit.
<b>Warning Time (s)</b>	The amount of seconds before the maximum call duration is reached to play the warning tone to the caller.
<b>Warning Repeat Interval (s)</b>	The amount of seconds that must pass after the first warning tone before another warning tone is played.
<b>Enable 486 to Failover Trunk</b>	Reroutes failed outbound calls that receive a 486 response through the failover trunk to retry the call. If disabled, calls that receive a 486 response will be terminated.
<b>Record Prompt</b>	If enabled, users will hear voice prompt before recording is started or stopped. For example, before recording, the UCM630xA will play voice prompt "The call will be recorded". The default setting is "No".
<b>Device Name</b>	The name of the UCM you are using.



<b>International Call Prefix</b>	When this configuration is empty, International Call Prefix can be empty or +.
<b>Meeting Max Concurrent Audio</b>	Maximum number of participants that can be heard simultaneously in meetings. If the number of participants talking at any given point exceeds this value, the audio of the excess participants will not be heard.
<b>Meeting Voice Indicator Sensitivity</b>	Configures the sensitivity of the talking indicator in meetings. Setting this higher will make the talking indicator appear more easily for lower volumes of audio. Note: This does not adjust audio input sensitivity itself. Lower volumes of sounds may still be heard even if the talking indicator does not show the source.
<b>Meeting Voice Quality</b>	Voice quality of meetings
<b>Extension Preferences</b>	
<b>Enforce Strong Passwords</b>	<p>If enabled, strong password will be enforced for the password created on the UCM630xA. The default setting is "No".</p> <p>Strong Password Rules:</p> <ol style="list-style-type: none"> <li>1. Password for voicemail, voicemail group, outbound route, DISA, call queue and meeting require non-repetitive and non-sequential digits, with a minimum length of 4 digits. Repetitive digits pattern (such as 0000, 1111, 1234, 2345, and etc.), or common digits pattern (such as 111222, 321321 and etc.) are not allowed to be configured as password.</li> <li>2. Password for extension registration, Web GUI admin login, LDAP and LDAP sync requires alphanumeric characters containing at least two categories of the following, with a minimum length of 4 characters. <ul style="list-style-type: none"> <li>• Numeric digits</li> <li>• Lowercase alphabet characters</li> <li>• Uppercase alphabet characters</li> <li>• Special characters</li> </ul> </li> </ol>
<b>Enable Random Password</b>	If enabled, random password will be generated when the extension is created. The default setting is "Yes". It is recommended to enable it for security purpose.
<b>Enable Auto E-mail To User</b>	If enabled, UCM630xA will send Email notification to user automatically after editing extension settings or adding a new extension.
<b>Disable Extension Range</b>	If set to "Yes", users could disable the extension range pre-configured/configured on the UCM630xA. The default setting is "No".



	<p><b>Note:</b> It is recommended to keep the system assignment to avoid inappropriate usage and unnecessary issues.</p>
<p><b>Extension Ranges</b></p>	<p>The default extension range assignment is:</p> <ul style="list-style-type: none"> <li>• <u>User Extensions</u>: 1000-6299 User Extensions is referring to the extensions created under Web GUI→<b>Extension/Trunk</b>→<b>Extensions</b> page.</li> <li>• <u>Pick Extensions</u>: 4000-4999 This refers to the extensions that can be manually picked from end device when being provisioned by the UCM630xA. There are two related options in zero config page→Zero Config Settings, "Pick Extension Segment" and "Enable Pick Extension". If "Enable Pick Extension" under zero config settings is selected, the extension list defined in "Pick Extension Segment" will be sent out to the device after receiving the device's request. This "Pick Extension Segment" should be a subset of the "Pick Extensions" range here. This feature is for the GXP series phones that support selecting extension to be provisioned via phone's LCD.</li> <li>• <u>Auto Provision Extensions</u>: 5000-6299 This sets the range for "Zero Config Extension Segment" which is the extensions can be assigned on the UCM630xA to provision the end device.</li> <li>• <u>Meeting Extensions</u>: 6300-6399</li> <li>• <u>Ring Group Extensions</u>: 6400-6499</li> <li>• <u>Queue Extensions</u>: 6500-6599</li> <li>• <u>Voicemail Group Extensions</u>: 6600-6699</li> <li>• <u>IVR Extensions</u>: 7000-7100</li> <li>• <u>Dial By Name Extensions</u>: 7101-7199</li> </ul>

## PBX Settings/RTP Settings

### RTP Settings

Table 101: Internal Options/RTP Settings



<b>RTP Start</b>	Configure the RTP port starting number. The default setting is 10000.
<b>RTP End</b>	Configure the RTP port ending address. The default setting is 20000.
<b>Strict RTP</b>	Configure to enable or disable strict RTP protection. If enabled, RTP packets that do not come from the source of the RTP stream will be dropped. The default setting is "Disable".
<b>RTP Checksums</b>	Configure to enable or disable RTP Checksums on RTP traffic. The default setting is "Disable".
<b>ICE Support</b>	Configure whether to support ICE. The default setting is enabled. ICE is the integrated use of STUN and TURN structure to provide reliable VoIP or video calls and media transmission, via a SIP request/ response model or multiple candidate endpoints exchanging IP addresses and ports, such as private addresses and TURN server address.
<b>STUN Server</b>	Configure STUN server address. STUN protocol is a Client/Server and also a Request/Response protocol. It is used to check the connectivity between the two terminals, such as maintaining a NAT binding entries keep-alive agreement. The default STUN Server is stun.ipvideotalk.com.  Valid format: [(hostname   IP-address) [:' port] The default port number is 3478 if not specified.
<b>BFCP UDP Start</b>	Configure BFCP UDP port starting number. The default setting is 50000.
<b>BFCP UDP End</b>	Configure BFCP UDP port ending number. The default setting is 52999.
<b>BFCP TCP Start</b>	Configure BFCP TCP port starting number. The default setting is 53000.
<b>BFCP TCP End</b>	Configure BFCP TCP port ending number. The default setting is 55999.
<b>TURN Server</b>	Configure TURN server address. TURN is an enhanced version of the STUN protocol and is dedicated to the processing of symmetric NAT problems.
<b>TURN Server Name</b>	Configure turn server account name
<b>TURN Server Password</b>	Configure turn server account password.
<b>Connection Protocol</b>	Protocol used to connect to the TURN server.

## Payload

The UCM630xA payload type for audio codecs and video codes can be configured here.

**Table 102: Internal Options/Payload**

<b>AAL2-G.726</b>	Configure payload type for ADPCM (G.726, 32kbps, AAL2 codeword packing).
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	The default setting is 112.
<b>DTMF</b>	Configured payload type for DTMF. The default setting is 101.
<b>G.721 Compatible</b>	Configure to enable/disable G.721 compatible. The default setting is Yes.
<b>G.726</b>	Configure the payload type for G.726 if "G.721 Compatible" is disabled. The default setting is 111.
<b>iLBC</b>	Configure the payload type for iLBC. The default setting is 97.
<b>OPUS</b>	Configure the payload type for OPUS. The default setting is 123.
<b>Audio FEC Payload Type</b>	Configure the Audio FEC Payload Type. The default setting is 127
<b>Audio RED Payload Type</b>	Configure the Audio RED Payload Type. Default setting is 122
<b>H.264</b>	Configure the payload type for H.264. The default setting is 99.
<b>H.265</b>	Configure the payload type for H.264. The default setting is 114.
<b>H.263P</b>	Configure the payload type for H.263+. The default setting is 100 103.
<b>VP8</b>	Configure the payload type for VP8. The default setting is 108.
<b>Main Video FEC</b>	Configure the Main Video FEC
<b>RTP FECC</b>	Configure the RTP FECC
<b>RTX</b>	Configure the RTX
<b>G.722.1</b>	G.722.1: Low-complexity coder, 24kbps.
<b>G.722.1C</b>	G.722.1C: Low-complexity coder, 48kbps.

## PBX Settings/Voice Prompt Customization

### Record New Custom Prompt

In the UCM630xA Web GUI→**PBX Settings**→**Voice Prompt**→**Custom Prompt** page, click on “Record” and follow the steps below to record new IVR prompt.



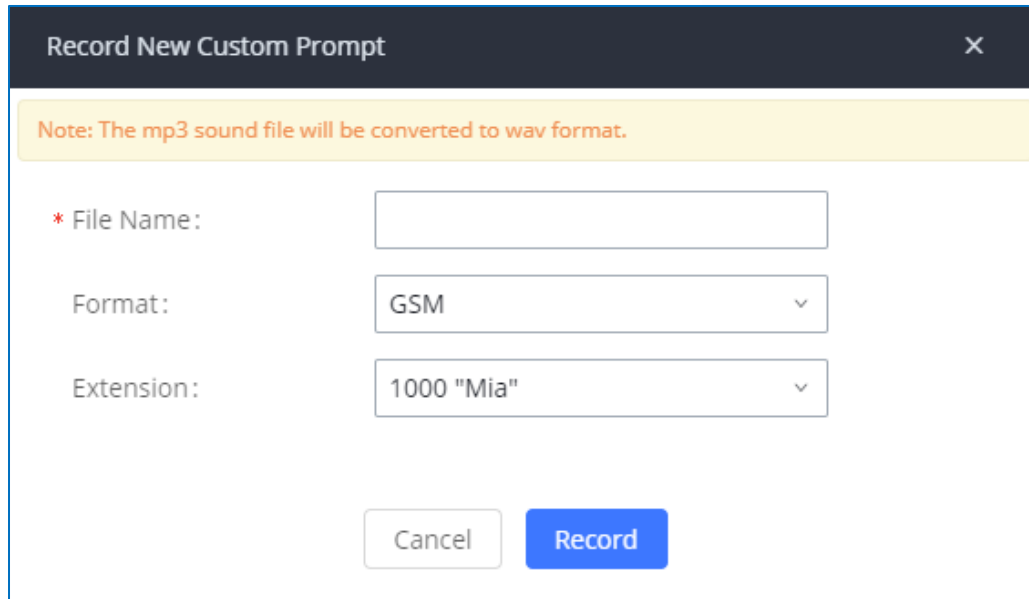


Figure 217: Record New Custom Prompt

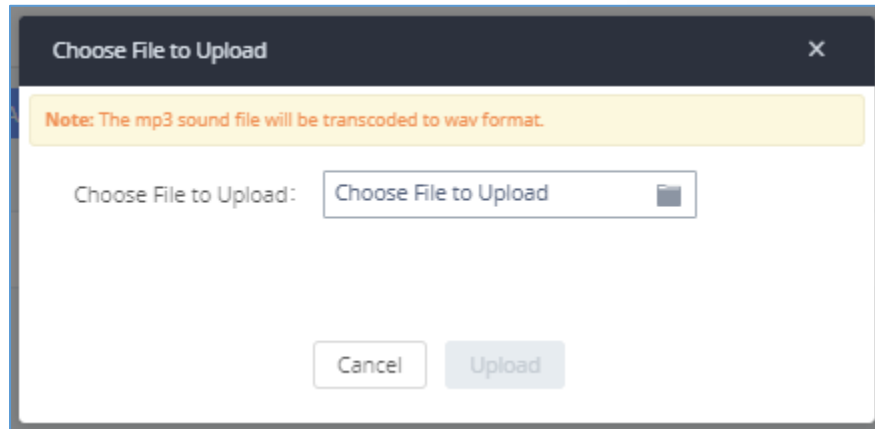
1. Specify the IVR file name.
2. Select the format (GSM or WAV) for the IVR prompt file to be recorded.
3. Select the extension to receive the call from the UCM630xA to record the IVR prompt.
4. Click the “Record” button. A request will be sent to the UCM630xA. The UCM630xA will then call the extension for recording the IVR prompt from the phone.
5. Pick up the call from the extension and start the recording following the voice prompt.
6. The recorded file will be listed in the IVR Prompt web page. Users could select to re-record, play, or delete the recording.

### Upload Custom Prompt

If the user has a pre-recorded IVR prompt file, click on “Upload” in Web GUI→PBX Settings→**Voice Prompt**→**Custom Prompt** page to upload the file to the UCM630xA. The following are required for the IVR prompt file to be successfully uploaded and used by the UCM630xA:

- PCM encoded.
- 16 bits.
- 8000Hz mono.
- In .mp3 or .wav format; or raw/ulaw/alaw/gsm file with .ulaw or .alaw suffix.
- File size under 5M.



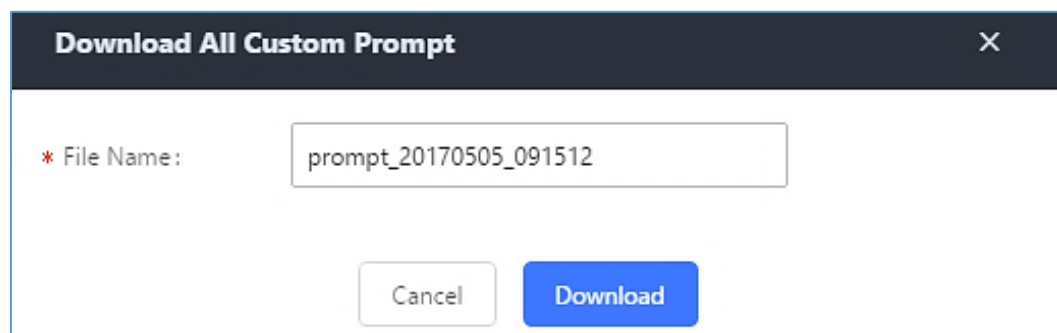


**Figure 218: Upload Custom Prompt**

Click on “choose file to upload” to start uploading. Once uploaded, the file will appear in the Custom Prompt web page.

### Download All Custom Prompt

On the UCM630xA, the users can download all custom prompts from UCM Web GUI to local PC. To download all custom prompt, log in UCM Web GUI and navigate to **PBX Settings**→**Voice Prompt**→**Custom Prompt** and click on “Download All”. The following window will pop up in order to set a name for the downloaded file.



**Figure 219: Download All Custom Prompt**

**Note:** The downloaded file will have a .tar extension.

## PBX Settings/ Call Failure Tone Settings

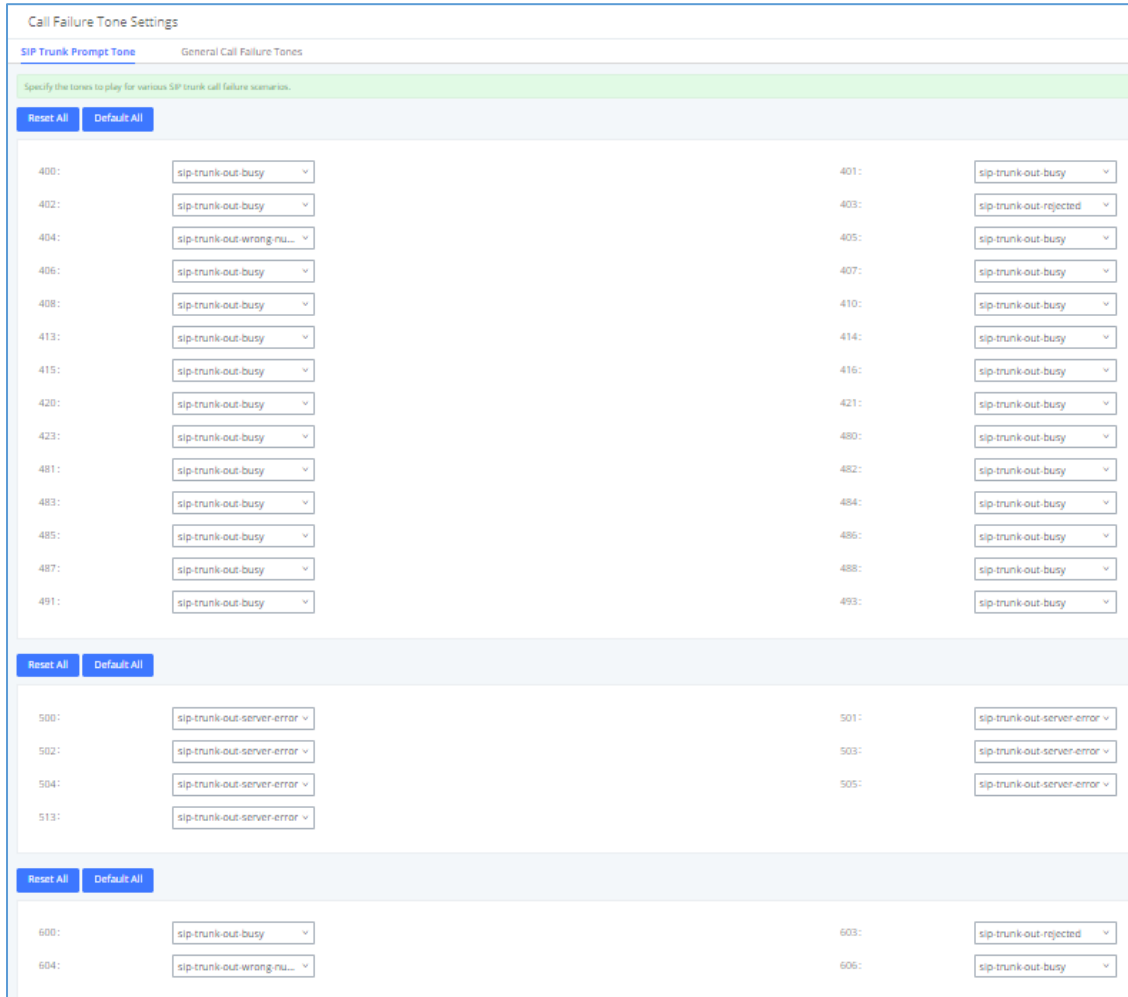
### SIP Trunk Prompt Tone

**Prompt Tone Settings** tab has been added to the UCM to help users choose which prompt will be played by the UCM during call failure, the following voice message responses have been added and can be set to be played for 4XX, 5XX, and 6XX call failures:



- Default for 404 and 604 status codes: *“Your call can’t be completed as dialed. Please check the number and dial again.”*
- Default for 5xx status codes: *“Server error. Please check your device.”*
- Default for 403 and 603 status codes: *“The call was rejected by the server. Please try again later.”*
- Default for all other status codes: *“All circuits are busy now. Please try again later.”*

Additionally, custom voice messages recorded and uploaded in **PBX Settings**→**Voice Prompt**→**Custom Prompt** can be used for these failure responses instead of the default messages.



The screenshot shows the 'Call Failure Tone Settings' interface, specifically the 'SIP Trunk Prompt Tone' section. It features a table of status codes with corresponding dropdown menus for selecting tones. The interface includes 'Reset All' and 'Default All' buttons at the top of each section.

Status Code	Selected Tone	Status Code	Selected Tone
400	sip-trunk-out-busy	401	sip-trunk-out-busy
402	sip-trunk-out-busy	403	sip-trunk-out-rejected
404	sip-trunk-out-wrang.nu...	405	sip-trunk-out-busy
406	sip-trunk-out-busy	407	sip-trunk-out-busy
408	sip-trunk-out-busy	410	sip-trunk-out-busy
413	sip-trunk-out-busy	414	sip-trunk-out-busy
415	sip-trunk-out-busy	416	sip-trunk-out-busy
420	sip-trunk-out-busy	421	sip-trunk-out-busy
423	sip-trunk-out-busy	480	sip-trunk-out-busy
481	sip-trunk-out-busy	482	sip-trunk-out-busy
483	sip-trunk-out-busy	484	sip-trunk-out-busy
485	sip-trunk-out-busy	486	sip-trunk-out-busy
487	sip-trunk-out-busy	488	sip-trunk-out-busy
491	sip-trunk-out-busy	493	sip-trunk-out-busy
500	sip-trunk-out-server-error	501	sip-trunk-out-server-error
502	sip-trunk-out-server-error	503	sip-trunk-out-server-error
504	sip-trunk-out-server-error	505	sip-trunk-out-server-error
513	sip-trunk-out-server-error		
600	sip-trunk-out-busy	603	sip-trunk-out-rejected
604	sip-trunk-out-wrang.nu...	606	sip-trunk-out-busy

**Figure 220: SIP Trunk Prompt Tone**

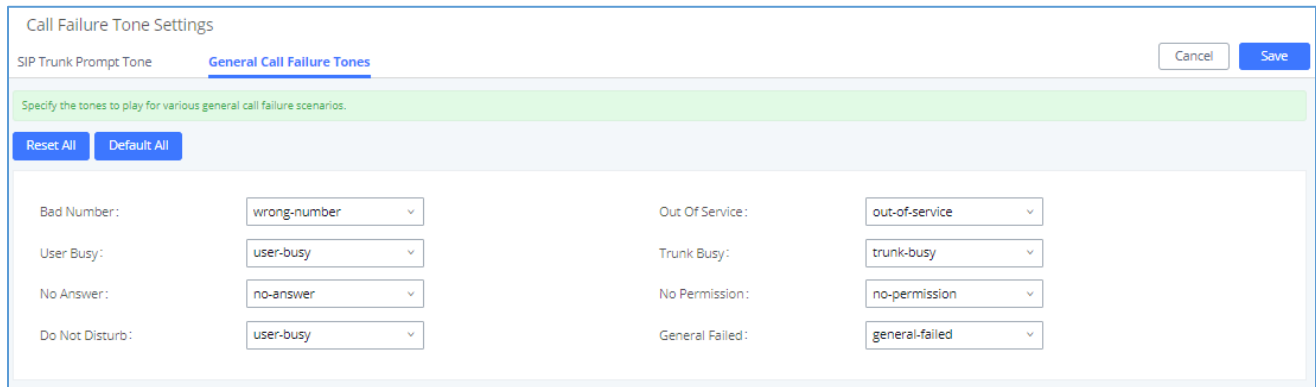
## General Call Prompt Tone

Moreover, users also have the possibility to customize the prompt for typical call failure reasons like (no permission to allow outbound calls, busy lines, incorrect number dialed ...Etc.).





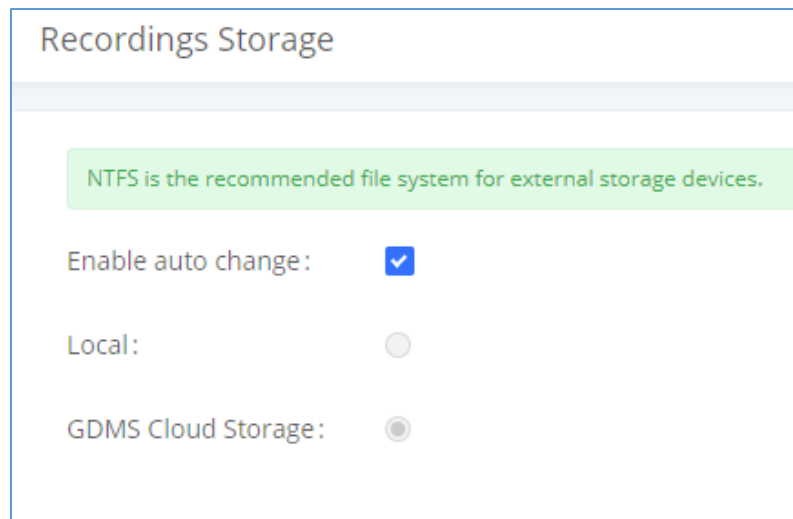
To customize these prompts user could record and upload their own files under “**PBX Settings → Voice Prompt → Custom Prompts**” then select each one for specific call failure case under “**PBX Settings -> Call Failure Tone Settings → General Call Prompt Tone**” page as shown on the following figure:



**Figure 221: General call Failure Prompts**

## PBX Settings/Recordings Storage

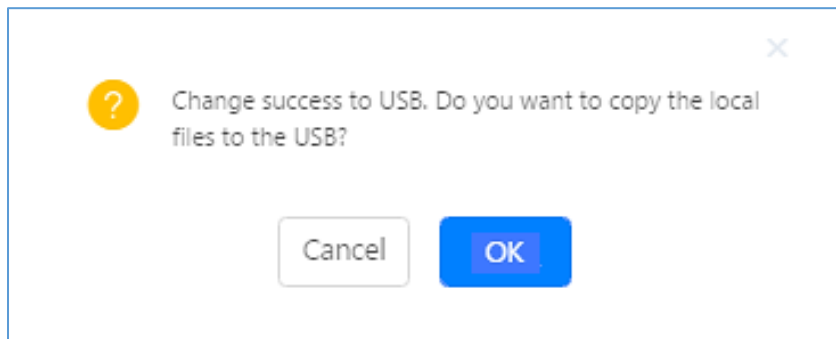
The UCM630xA supports call recordings automatically or manually and the recording files can be saved in external storage plugged in the UCM630xA or on the UCM630xA locally. To manage the recording storage, users can go to UCM630xA Web GUI→**PBX Settings→Recordings Storage** page and select whether to store the recording files in USB Disk, SD card, GDMS or locally on the UCM630xA.



**Figure 222: Settings→Recordings Storage**



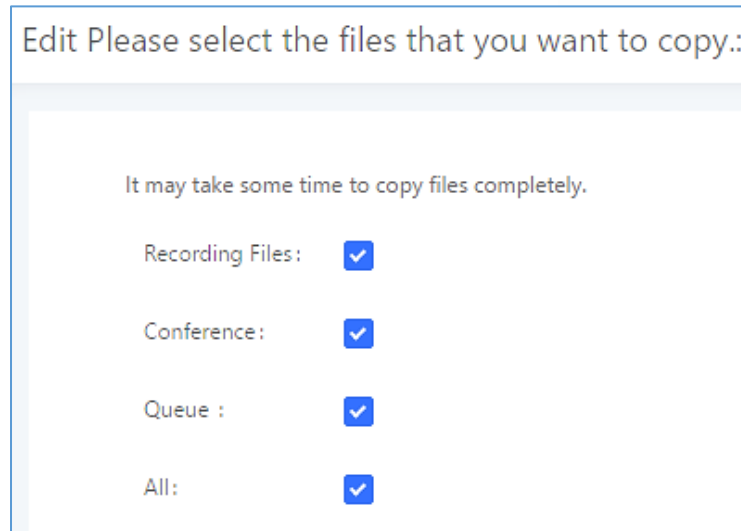
- If **“Enable Auto Change”** is selected, the recording files will be automatically saved in the available USB Disk or SD card plugged into the UCM630xA. If both USB Disk and SD card are plugged in, the recording files will be always saved in the USB Disk.
- If **“Local”** is selected, the recordings will be stored in UCM630xA internal storage.
- If **“GDMS Cloud Storage”** is selected, recording data will no longer be stored locally and if you need to listen to the recording, download the recording file to the computer side and play it offline.
- 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 UCM630xA.  
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 223: 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 224: Recording Storage Category**

On the UCM630xA, recording files are generated and exist in 3 categories: normal call recording files, meeting 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.
- **Meeting:** Copy the meeting 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.

## PBX Settings/NAS

The UCM supports adding and backing up recordings to a network-attached storage (NAS) server. Following table describes NAS settings:

**Table 103: NAS Settings**

<b>Enable</b>	Enabled / Disable the NAS recording functionality.
<b>Host</b>	Configure the Domain or IP address of the NAS server. <b>Note:</b> Currently, only IP addresses are supported in the Host/IP field.
<b>Share Name</b>	Specify the name of the shared folder.
<b>Username</b>	Specify the account username to access the NAS server.
<b>Password</b>	Configure the account password to access the NAS server.
<b>Status</b>	If configured correctly, the Status field will show “Mounted”, and the newly



added NAS server will be shown on the Mounted Netdisk List. Additionally, the NAS will appear as a selectable storage option in the PBX Settings→Recording Storage page and CDR→Recording Files page.



## SIP SETTINGS

The UCM630xA SIP global settings can be accessed via Web GUI→**PBX Settings**→**SIP Settings**.

### SIP Settings/General

**Table 104: SIP Settings/General**

<b>Realm For Digest Authentication</b>	Configure the host name or domain name for the UCM630xA. Realms MUST be globally unique according to RFC3261. The default setting is Grandstream.
<b>Bind UDP Port</b>	Configure the UDP port used for SIP. The default setting is 5060.
<b>Bind IPv4 Address</b>	Configure the IPv4 address to bind to. The default setting is 0.0.0.0, which means binding to all addresses.
<b>Bind IPv6 Address</b>	Configure the IPv6 address to bind to. The default is : "[::]" and it means to bind to all IP addresses.
<b>Allow Guest Calls</b>	If enabled, the UCM630xA allows unauthorized INVITE coming into the PBX and the call can be made. The default setting is "No". <b>Warning:</b> Please be aware of the potential security risk when enabling "Allow Guest Calls" as this will allow any user with the UCM630xA address to dial into the UCM630xA.
<b>Allow Transfer</b>	If set to "No", all transfers initiated by the endpoint in the UCM630xA will be disabled (unless enabled in peers or users). The default setting is "Yes".
<b>MWI From</b>	When sending MWI NOTIFY requests, this value will be used in the "From:" header as the "name" field. If no "From User" is configured, the "user" field of the URI in the "From:" header will be filled with this value.
<b>Enable Diversion Header</b>	If disabled, the UCM will not forward the diversion header.
<b>Block Collect Calls</b>	If enabled, collect calls will be blocked. <b>Note:</b> Collect calls are indicated by the header "P-Asserted-Service-Info: service-code=Backward Collect Call, P-Asserted-Service-Info: service-code=Collect Call".

### SIP Settings/MISC

**Table 105: SIP Settings/Misc**

Outbound SIP Registrations	
<b>Register Timeout</b>	Configure the register retry timeout (in seconds). The default setting is 20.
<b>Register Attempts</b>	Configure the number of registration attempts before the UCM630xA gives up.



	The default setting is 0, which means the UCM630xA will keep trying until the server side accepts the registration request.
<b>Video</b>	
<b>Max Bit Rate (kb/s)</b>	Configure the maximum bit rate (in kb/s) for video calls. The default setting is 384.
<b>Support SIP Video</b>	Select to enable video support in SIP calls. The default setting is "Yes".
<b>Reject Non-Matching INVITE</b>	If enabled, when rejecting an incoming INVITE or REGISTER request, the UCM630xA will always reject with "401 Unauthorized" instead of notifying the requester whether there is a matching user or peer for the request. This reduces the ability of an attacker to scan for valid SIP usernames. The default setting is "No".
<b>SDP Attribute Passthrough</b>	
<b>Enable Attribute Passthrough</b>	If enable, and if the service does not know the attribute of FEC/FECC/BFCP, then the attribute will be passthrough.
<b>Early Media</b>	
<b>Enable Use Final SDP</b>	If enabled, call negotiation will use final response SDP.
<b>Blind Transfer</b>	
<b>Allow callback when blind transfer fails</b>	If enabled, the UCM will call back to the transferrer when blind transfer fails (reason of failure includes: busy and no answer). <b>Note:</b> This feature takes effect only on internal calls.
<b>Blind transfer timeout</b>	Configure the timeout in (s) for the transferrer waiting for the destination to answer. Default is 60s.
<b>Hold</b>	
<b>Forward HOLD Requests</b>	Configure the UCM to forward HOLD requests instead of processing holds internally. This serves to meet the standards set by some providers that require HOLD requests to be passed along from endpoint to endpoint. This option is disabled by default. <b>Note:</b> Enabling this option may cause hold retrieval issues and MOH to not be heard.

## SIP Settings/Session Timer

**Table 106: SIP Settings/Session Timer**

<b>Force Timer</b>	If checked, always request, and run session timer.
<b>Timer</b>	If checked, run session timer only when requested by other UA.
<b>Session Expire</b>	Configure the maximum session refresh interval (in seconds). Default is 1800.
<b>Min SE</b>	Configure the minimum session refresh interval (in seconds). The default setting is 90.



## SIP Settings/TCP and TLS

**Table 107: SIP Settings/TCP and TLS**

<b>TCP Enable</b>	Configure to allow incoming TCP connections with the UCM630xA. The default setting is "No".
<b>TCP Bind IPv4 Address</b>	Configure the IP address for TCP server to bind to. "0.0.0.0" means binding to all interfaces. The port number is optional, and the default port number is 5060. For example, 192.168.1.1:5062.
<b>TCP Bind IPv6 Address</b>	Configure the IPv6 address for TCP server to bind to. ":::" means bind to all interfaces. The port number is optional with the default being 5060. For example, [2001:0DB8:0000:0000:0000:0000:1428:0000]:5060.
<b>TLS Enable</b>	Configure to allow incoming TLS connections with the UCM630xA. The default setting is "Yes".
<b>TLS Bind IPv4 Address</b>	Configure the IPv4 address for TLS server to bind to. "0.0.0.0" means binding to all interfaces. The port number is optional, and the default port number is 5061. For example, 192.168.1.1:5063. Note: The IP address must match the common name (host name) in the certificate so that the TLS socket will not bind to multiple IP addresses.
<b>TLS Bind IPv6 Address</b>	Configure the IPv6 address for TLS server to bind to. ":::" means bind to all interfaces. The port number is optional with default being 5061. For example, [2001:0DB8:0000:0000:0000:0000:1428:0000]:5061. Note: The IP address must match the common name (host name) in the certificate so that the TLS socket will not bind to multiple IP addresses.
<b>TLS Do Not Verify</b>	If enabled, the TLS server's certificate will not be verified when acting as a client. The default setting is "Yes".
<b>TLS Self-Signed CA</b>	This is the CA certificate if the TLS server being connected to requires self-signed certificate, including server's public key. This file will be renamed as "TLS.ca" automatically. <b>Note:</b> The size of the uploaded ca file must be under 2MB.
<b>TLS Cert</b>	This is the Certificate file (*.pem format only) used for TLS connections. It contains private key for client and signed certificate for the server. This file will be renamed as "TLS.pem" automatically. <b>Note:</b> The size of the uploaded certificate file must be under 2MB.
<b>TLS CA Cert</b>	This file must be named with the CA subject name hash value. It contains CA's (Certificate Authority) public key, which is used to verify the accessed servers. <b>Note:</b> The size of the uploaded CA certificate file must be under 2MB.
<b>TLS CA List</b>	Display a list of files under the CA Cert directory.



## SIP Settings/NAT

**Table 108: SIP Settings/NAT**

<b>External Host</b>	Configure a static IP address and port (optional) used in outbound SIP messages if the UCM630xA is behind NAT. If it is a host name, it will only be looked up once.
<b>Use IP address in SDP</b>	If enabled, the SDP connection will use the IP address resolved from the external host.
<b>External UDP Port</b>	Configure externally mapped UDP port when the PBX is behind a static NAT or PAT.
<b>External TCP Port</b>	Configure the externally mapped TCP port when the UCM630xA is behind a static NAT or PAT.
<b>External TLS Port</b>	Configures the externally mapped TLS port when UCM630xA is behind a static NAT or PAT.
<b>Local Network Address</b>	Specify a list of network addresses that are considered inside of the NAT network. Multiple entries are allowed. If not configured, the external IP address will not be set correctly. A sample configuration could be as follows: 192.168.0.0/16

## SIP Settings/ToS

**Table 109: SIP Settings/ToS**

<b>ToS for SIP</b>	Configure the Type of Service for SIP packets. The default setting is None.
<b>ToS for RTP Audio</b>	Configure the Type of Service for RTP audio packets. The default setting is None.
<b>ToS for RTP Video</b>	Configure the Type of Service for RTP video packets. The default setting is None.
<b>Default Incoming/Outgoing Registration Time</b>	Configure the default duration (in seconds) of incoming/outgoing registration. The default setting is 120.
<b>Max Registration/Subscription Time</b>	Configure the maximum duration (in seconds) of incoming registration and subscription allowed by the UCM630xA. The default setting is 3600.
<b>Min Registration/Subscription Time</b>	Configure the minimum duration (in seconds) of incoming registration and subscription allowed by the UCM630xA. The default setting is 60.



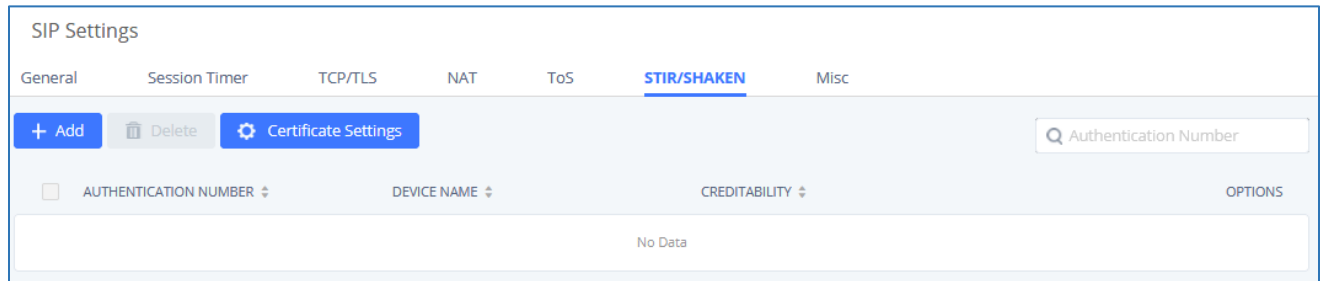


<b>Enable Relaxed DTMF</b>	Select to enable relaxed DTMF handling. The default setting is "No".
<b>DTMF Mode</b>	Select DTMF mode to send DTMF. The default setting is RFC4733. If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, a-law or u-law are required. When "Auto" is selected, "RFC4733" will be used if offered, otherwise "Inband" will be used. The default setting is "RFC4733".
<b>RTP Timeout</b>	During an active call, if there is no RTP activity within the timeout (in seconds), the call will be terminated. The default setting is no timeout. <b>Note:</b> This setting does not apply to calls on hold.
<b>RTP Hold Timeout</b>	When the call is on hold, if there is no RTP activity within the timeout (in seconds), the call will be terminated. This value of RTP Hold Timeout should be larger than RTP Timeout. The default setting is no timeout.
<b>RTP Keep-alive</b>	This feature can be used to avoid abnormal call drop when the remote provider requires RTP traffic during proceeding. For example, when the call goes into voicemail and there is no RTP traffic sent out from UCM, configuring this option can avoid voicemail drop. When configured, RTP keep-alive packet will be sent to remote party at the configured interval. If set to 0, RTP keep-alive is disabled.
<b>100rel</b>	Configure the 100rel setting on UCM630xA. The default setting is "Yes".
<b>Trust Remote Party ID</b>	Configure whether the Remote-Party-ID should be trusted. The default setting is "No".
<b>Send Remote Party ID</b>	Configure whether the Remote-Party-ID should be sent or not. The default setting is "No".
<b>Generate In-Band Ringing</b>	Configure whether the UCM630xA should generate Inband ringing or not. The default setting is "Never". <ul style="list-style-type: none"> <li>• <b>Yes:</b> The UCM630xA will send 180 Ringing followed by 183 Session Progress and in-band audio.</li> <li>• <b>No:</b> The UCM630xA will send 180 Ringing if 183 Session Progress has not been sent yet. If audio path is established already with 183 then send in-band ringing.</li> <li>• <b>Never:</b> Whenever ringing occurs, the UCM630xA will send 180 Ringing as long as 200OK has not been set yet. Inband ringing will not be generated even the end point device is not working properly.</li> </ul>
<b>Server User Agent</b>	Configure the user agent string for the UCM630xA.
<b>Send Compact SIP Headers</b>	If enabled, compact SIP headers will be sent. The default setting is "No".
<b>Passthrough PAI Header</b>	Passthrough PAI Header



## SIP Settings/STIR/SHAKEN

To prevent robocalls, UCM now supports STIR/SHAKE protocols. Related options have been added as a new tab in the **SIP Settings** page.



Clicking on the **Add** button will show the following window:

Add Authentication Number
×

\* Authentication Number:

\* Device Name:

\* Creditability:

Figure 225: SIP Settings/STIR/SHAKEN - Add Authentication Number

Table 110: SIP Settings/STIR/SHAKEN - Add Authentication Number Settings

<b>Authentication Number</b>	Configure the Authentication Number.
<b>Device Name</b>	Configure the device name.
<b>Creditability</b>	Configure the attestation level, which is the level of confidence of the carrier that the CID has not been spoofed. The following options are available:



- **A (Full attestation)** - The carrier is associated with the caller and the number. There is high confidence that the CID has not been spoofed.
- **B (Partial attestation)** - The carrier is associated with the caller but not the number. There is uncertainty about whether the CID has been spoofed or not.
- **C (Gateway attestation)** - The carrier is not associated with the caller and has no confidence at all about the number. Generally used for traceback.

Clicking on the **Certificate Settings** button will bring up the following window:

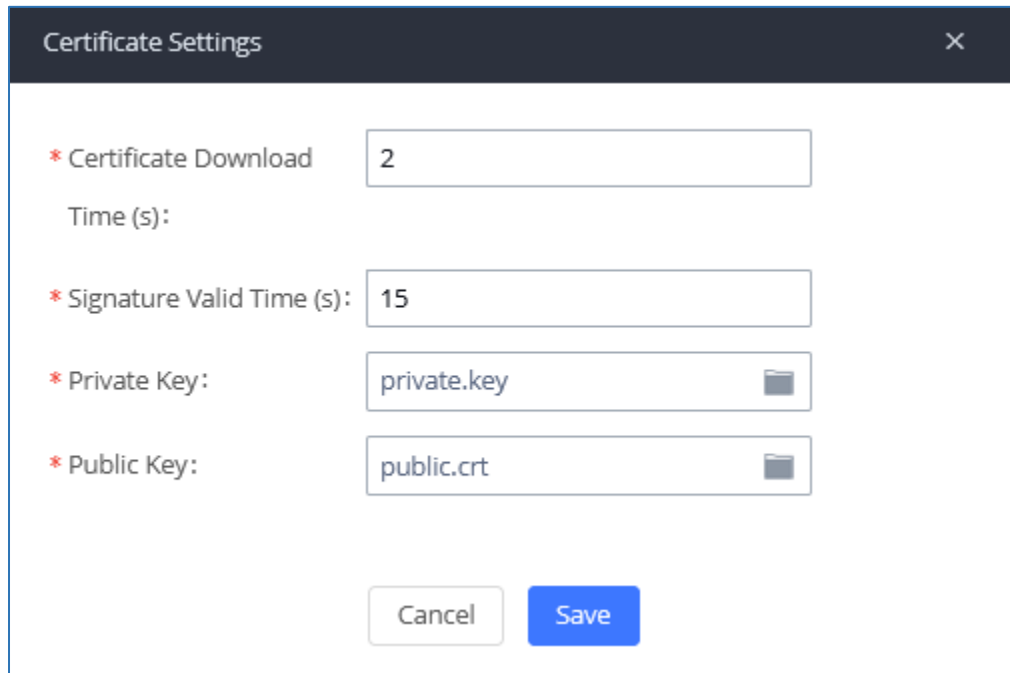


Figure 226: SIP Settings/STIR/SHAKEN – Certificate Settings

Table 111: SIP Settings/STIR/SHAKEN – Certificate Settings

<b>Certificate Download Time (s)</b>	Configure the public key download timeout period, the default value is 2 seconds.
<b>Signature Valid Time (s)</b>	Configure the validity period of the digital signature, the default value is 15 seconds.



<b>Private Key</b>	Configure the Private key. <b>Note:</b> The uploaded file must be less than 2MB in file size, only supports the .key format and must be ECC type. This file will automatically be renamed to "private.key".
<b>Public Key</b>	Configure the Public Key. <b>Note:</b> The uploaded file must be less than 2MB in file size, only supports the .crt format and must be ECC type. This file will automatically be renamed to "public.crt".

## Transparent Call-Info header

UCM supports transparent call info header in order to integrate GDS door system with GXP21XX/GRP261X phones, the UCM will forward the call-info header to the phone in order to request the live view from GDS door system and give the option to open the door via softkey.

```

Session Initiation Protocol (INVITE)
  Request-Line: INVITE sip:3001@192.168.6.36:5064 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP 192.168.6.187:5060;rport;branch=z9hG4bKPj3217e67b-e74f-4f9f-b06f-afd3dcbbe29b
    From: "3002" <sip:3002@192.168.6.187>;tag=202dca4f-2b9d-4880-924c-d48cea7d0596
    To: <sip:3001@192.168.6.36>
    Contact: <sip:68aae6ea-f1d4-4e62-9987-446e718a2448@192.168.6.187:5060>
    Call-ID: 7f66bb20-0b9f-4828-a355-698853b8d9fb
    CSeq: 17559 INVITE
    Allow: OPTIONS, INFO, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, MESSAGE, REGISTER, REFER
    Supported: 100rel, timer, replaces, norefersub
    Session-Expires: 1800
    Min-SE: 90
    Call-Info: <https://192.168.6.186:443/capture/8001> ;purpose=GDS-view
    Max-Forwards: 70
    User-Agent: Grandstream UCM6202V1.5A 1.0.13.15
    Content-Type: application/sdp
    Content-Length: 547
  Message Body
  
```

Figure 227: Transparent Call-Info



## IAX SETTINGS

The UCM630xA IAX global settings can be accessed via Web GUI→**PBX Settings**→**IAX Settings**.

### IAX Settings/General

Table 112: IAX Settings/General

<b>Bind Port</b>	Configure the port number that the IAX2 will be allowed to listen to. The default setting is 4569.
<b>Bind IPv4 Address</b>	Force IAX2 to bind to a specific address instead of all addresses.
<b>Bind IPv6 address</b>	Configure the IPv6 address to bind to. "[::]" means to bind to all IP addresses.
<b>IAX1 Compatibility</b>	Select to configure IAX1 compatibility. The default setting is "No".
<b>No Checksums</b>	If selected, UDP checksums will be disabled and no checksums will be calculated/checked on systems supporting this feature. The default setting is "No".
<b>Delay Reject</b>	If enabled, the IAX2 will delay the rejection of calls to avoid DOS. The default setting is "No".
<b>ADSI</b>	Select to enable ADSI phone compatibility. The default setting is "No".
<b>Music On Hold Interpret</b>	Specify which Music On Hold class this channel would like to listen to when being put on hold. This music class is only effective if this channel has no music class configured and the bridged channel putting the call on hold has no "Music On Hold Suggest" setting.
<b>Music On Hold Suggest</b>	Specify which Music On Hold class to suggest to the bridged channel when putting the call on hold.
<b>Bandwidth</b>	Configure the bandwidth for IAX settings. The default setting is "Low".

### IAX Settings/Registration

Table 113: IAX Settings/Registration

IAX Registration Options	
<b>Min Reg Expire</b>	Configure the minimum period (in seconds) of registration. The default setting is 60.
<b>Max Reg Expire</b>	Configure the maximum period (in seconds) of registration. The default setting is 3600.
<b>IAX Thread Count</b>	Configure the number of IAX helper threads. The default setting is 10.
<b>IAX Max Thread Count</b>	Configure the maximum number of IAX threads allowed. The default is 100.



<b>Auto Kill</b>	If enabled and no ACK is received for new messages after the specified wait time, the connection will be terminated.
<b>Authentication Debugging</b>	If enabled, authentication traffic in debugging will not show. The default is "No".
<b>Codec Priority</b>	<p>Configure codec negotiation priority. The default setting is "Reqonly".</p> <ul style="list-style-type: none"> <li>• <b>Caller</b> Consider the callers preferred order ahead of the host's.</li> <li>• <b>Host</b> Consider the host's preferred order ahead of the caller's.</li> <li>• <b>Disabled</b> Disable the consideration of codec preference all together.</li> <li>• <b>Reqonly</b> This is the same as "Disabled", except when the requested format is not available. The call will only be accepted if the requested format is available.</li> </ul>
<b>Type of Service</b>	Configure ToS bit for preferred IP routing.
<b>IAX Trunk Options</b>	
<b>Trunk Frequency</b>	Configure the frequency of trunk frames (in milliseconds). The default is 20.
<b>Trunk Time Stamps</b>	If enabled, time stamps will be attached to trunk frames. The default is "No".

## IAX Settings/Security


Table 114: IAX Settings/Static Defense

<b>Call Token Optional</b>	Enter a single IP address (e.g., 1.1.1.1) or a range of IP addresses (1.1.1.1/255.255.255.255) for which call token validation is not required.
<b>Max Call Numbers</b>	Configure the maximum number of calls allowed for a single IP address.
<b>Max Unvalidated Call Numbers</b>	Configure the maximum number of Unvalidated calls for all IP addresses.
<b>Max Call Numbers</b>	Configure to limit the number of calls for a give IP address of IP range.
<b>IP or IP Range</b>	Enter the IP address (1.1.1.1) or a range of IP addresses (1.1.1.1/255.255.255.255) to be considered for call number limits.



## INTERFACE SETTINGS

### Analog Hardware

The analog hardware (FXS port and FXO port) on the UCM630xA will be listed in this page. Click on  to edit signaling preference for FXS port or configure ACIM settings for FXO port.

Select "Loop Start" or "Kewl Start" for each FXS port. And then click on "Update" to save the change.

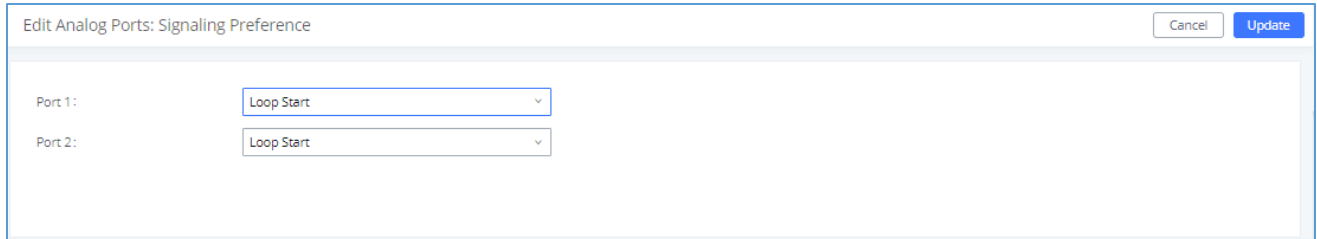


Figure 228 shows a dialog box titled "Edit Analog Ports: Signaling Preference". It contains two rows of settings:

- Port 1: Loop Start
- Port 2: Loop Start

Buttons for "Cancel" and "Update" are located in the top right corner.

Figure 228: FXS Ports Signaling Preference

For FXO port, users could manually enter the ACIM settings by selecting the value from dropdown list for each port. Or users could click on "Detect" and choose the detection algorithm, two algorithms exist (ERL, Pr) for the UCM630xA to automatically detect the ACIM value. The detecting value will be automatically filled into the settings.

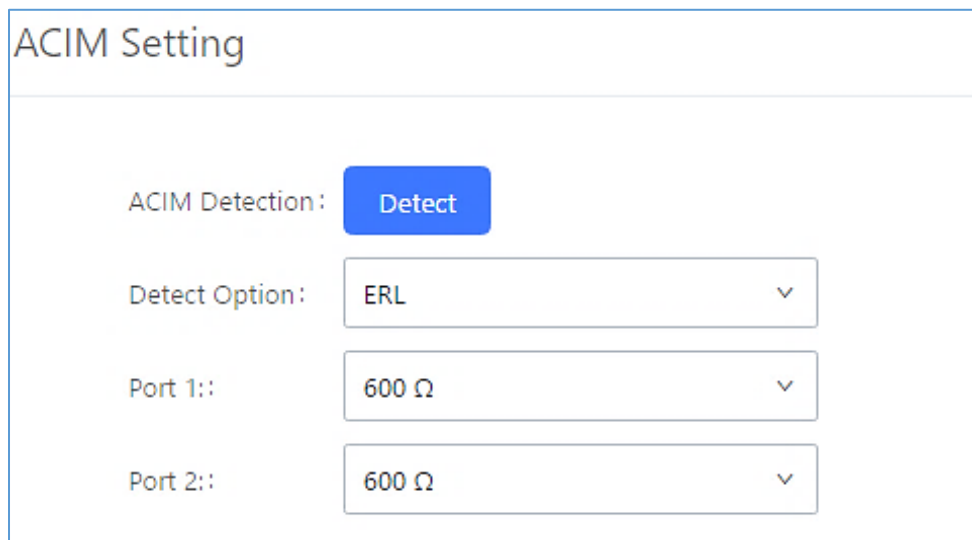


Figure 229 shows a dialog box titled "ACIM Setting". It contains the following settings:

- ACIM Detection: Detect
- Detect Option: ERL
- Port 1:: 600 Ω
- Port 2:: 600 Ω

Figure 229: FXO Ports ACIM Settings



**Table 115: PBX Interface Settings**

<b>Tone Region</b>	Select country to set the default tones for dial tone, busy tone, ring tone and etc. to be sent from the FXS port. The default setting is "United States of America (USA)".
<b>Advanced Settings</b>	
<b>FXO Opermode</b>	Select country to set the On-Hook Speed, Ringer Impedance, Ringer Threshold, Current Limiting, TIP/RING voltage adjustment, Minimum Operational Loop Current, and AC Impedance as predefined for your country's analog line characteristics. The default setting is "United States of America (USA)".
<b>FXS Opermode</b>	Select country to set the On-Hook Speed, Ringer Impedance, Ringer Threshold, Current Limiting, TIP/RING voltage adjustment, Minimum Operational Loop Current, and AC Impedance as predefined for your country's analog line characteristics. The default setting is "United States of America (USA)".
<b>FXS TISS Override</b>	<p>Configure to enable or disable override Two-Wire Impedance Synthesis (TISS). The default setting is No.</p> <p>If enabled, users can select the impedance value for Two-Wire Impedance Synthesis (TISS) override. The default setting is 600Ω.</p>
<b>PCMA Override</b>	<p>Select the codec to be used for analog lines. North American users should choose PCMU. All other countries, unless already known, should be assumed to be PCMA. The default setting is PCMU.</p> <p><b>Note:</b> This option requires system reboot to take effect.</p>
<b>Boost Ringer</b>	Configure whether normal ringing voltage (40V) or maximum ringing voltage (89V) for analog phones attached to the FXS port is required. The default setting is "Normal".
<b>Fast Ringer</b>	Configure to increase the ringing speed to 25HZ. This option can be used with "Low Power" option. The default setting is "Normal".
<b>Low Power</b>	Configure the peak voltage up to 50V during "Fast Ringer" operation. This option is used with "Fast Ringer". The default setting is "Normal".
<b>Ring Detect</b>	If set to "Full Wave", false ring detection will be prevented for lines where Caller ID is sent before the first ring and proceeded by a polarity reversal, as in UK. The default setting is "Standard".





<b>FXS MWI Mode</b>	Configure the type of Message Waiting Indicator on FXS lines. The default setting is "FSK". <ul style="list-style-type: none"> <li>• <b>FSK</b>: Frequency Shift Key Indicator</li> <li>• <b>NEON</b>: Light Neon Bulb Indicator.</li> </ul>
<b>FXO Frequency Tolerance</b>	Allows users to adjust the tolerance of the FXO ringing frequency. 63Hz is considered the standard value and is selected by default.

## DAHDI Settings

When users encounter issues such as audio delay in outbound calls using the analog trunk, they can adjust DAHDI settings on the UCM to attempt to lessen or resolve the issues.

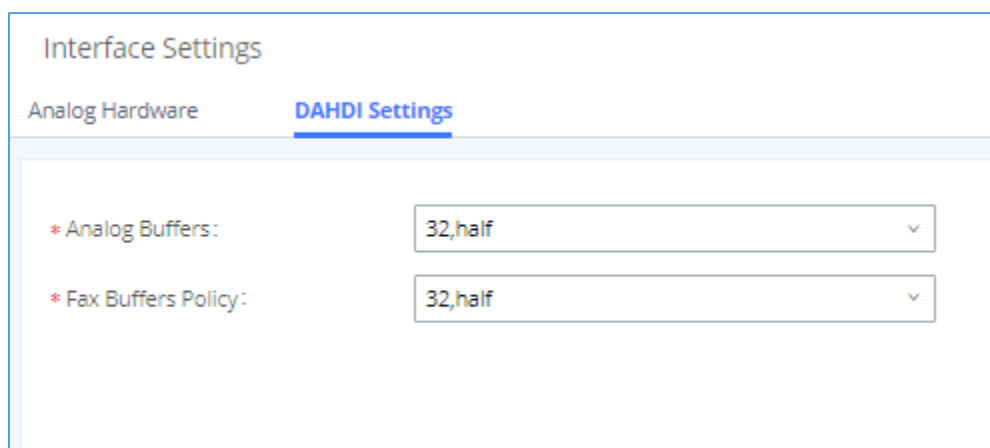


Figure 230: DAHDI Settings

For the value of the option such as “32, half”:

The number in the option indicates the number of read/write buffers for TDM (DAHDI).

The “Half”, “Immediate” or “Full” option indicates the strategy when reading/writing data from buffer.

- **“Half”**: Data will be read/written from buffer when half of the buffer is occupied with data.
- **“Immediate”**: Read/write from buffer whenever there is data occupying the buffer.
- **“Full”**: Data will be read/written from buffer when buffer is fully occupied with data.

Normally, DAHDI settings should be kept default and should be adjusted only when users encounter analog trunk/Fax-related issues.




## UCM RemoteConnect

An integrated & important part of Grandstream’s GDMS cloud-based device management service which runs on Amazon AWS with 99.999% reliability, the UCM RemoteConnect cloud service supports hassle-free Work-From-Home communications & collaborations using WebRTC-based license-free “Grandstream Wave” soft phones for desktop/Web/mobile devices (plus GUV series of USB headsets/Webcams), zero-touch out-of-box automated NAT firewall traversal for remote users & devices, IT-friendly remote management of UCM and attached endpoint devices, and more.

The RemoteConnect can be configured under **Value-added Features**→**RemoteConnect** After purchasing the RemoteConnect package.

**Remote Work Plan**



For improved call quality and service, please purchase a Remote Work Suite package.

- ✓ Intelligent NAT penetration service will allow for stable and clear remote audio/video calls.
- ✓ Cloud storage service
- ✓ Easily manage remote devices
- ...

[Learn more](#)

**Figure 231: RemoteConnect**

On GDMS platform, sign in and go to Device→PBX Device page, click on “Add Device” to add your UCM6300A device to GDMS system, once done an open beta plan will be assigned to the UCM.



**UCM RemoteConnect**

Plan
Plan Settings
Custom LOGO
Statistics

Subscription Tier:	Open Beta	<a href="#" style="color: #007bff;">Upgrade</a>
Subscription Period:	Valid until 4/1/2021	
Plan Status:	Active	
Remote Concurrent Users Number:	36	
Max Remote Users:	Unlimited	
GDMS Cloud Storage:	5 GB	
STUN Address:	nat-b.gdms.cloud	
UCM Public Address:	c074ad0a8c94-10671.b.gdms.cloud	
Public TLS Port:	1056	

Renew / Upgrade

**Figure 232: UCM RemoteConnect - Effective Plan**

**Note**

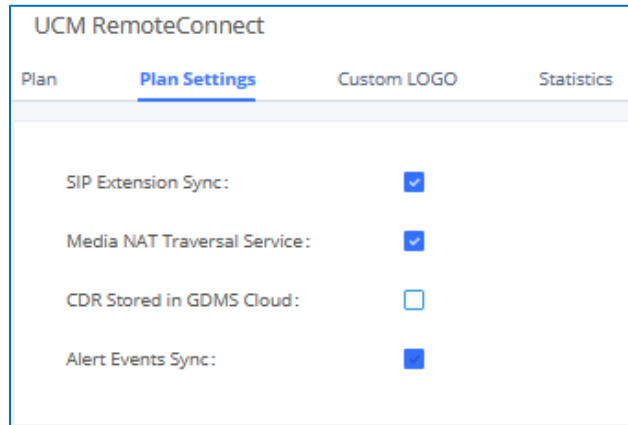
- After the UCM is added on GDMS, automated NAT traversal, SIP extension sync-up and basic statistics features are available without manual configuration required.

**Plan Settings**

After UCM is added into GDMS, all SIP extensions on the UCM will be synced up to GDMS automatically for users to allocate and manage SIP extension for their end devices. Also, the media NAT Traversal service, alert event sync configuration items are checked by default, the CDR data cloud storage in GDMS should be manually checked according to user needs.

The settings are under UCM **webGUI**→**Value-added Services**→**UCM RemoteConnect**→**Plan Settings**.



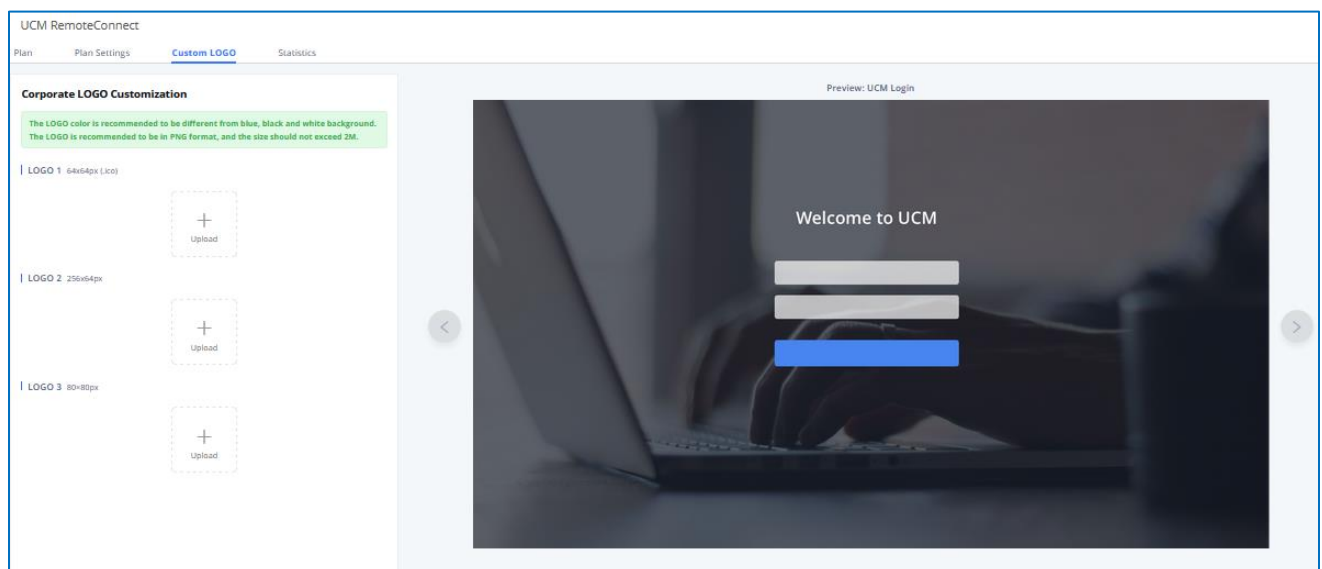


**Figure 233: UCM RemoteConnect Plan Settings**

## Custom logo

Custom logo feature allows users to select a local image file as the new logo. The pictures are in different formats and sizes according to the location of the logo. They are 64\*64px (only ico format is supported), 256\*256px, 80\*80px, which applies for "UCM Login", "Reset Password", "Email Template", "Wave/Login", "Browser Tab interface preview".

- LOGO 1: Replaces Browser tab icon
- LOGO 2: Replaces the Grandstream banner on the top left corner of the management login page and emails.
- LOGO 3: Replaces the Grandstream logo on the top left corner of the Wave Web interface and UCM management interface.



**Figure 234: Custom Logo**



## Statistics

After using UCM RemoteConnect, all remote calls will be logged and concurrent remote calls will be displayed on the UCM. The concurrent remote calls can be viewed under UCM web GUI → Value-Added Features → UCM RemoteConnect → Statistics page.

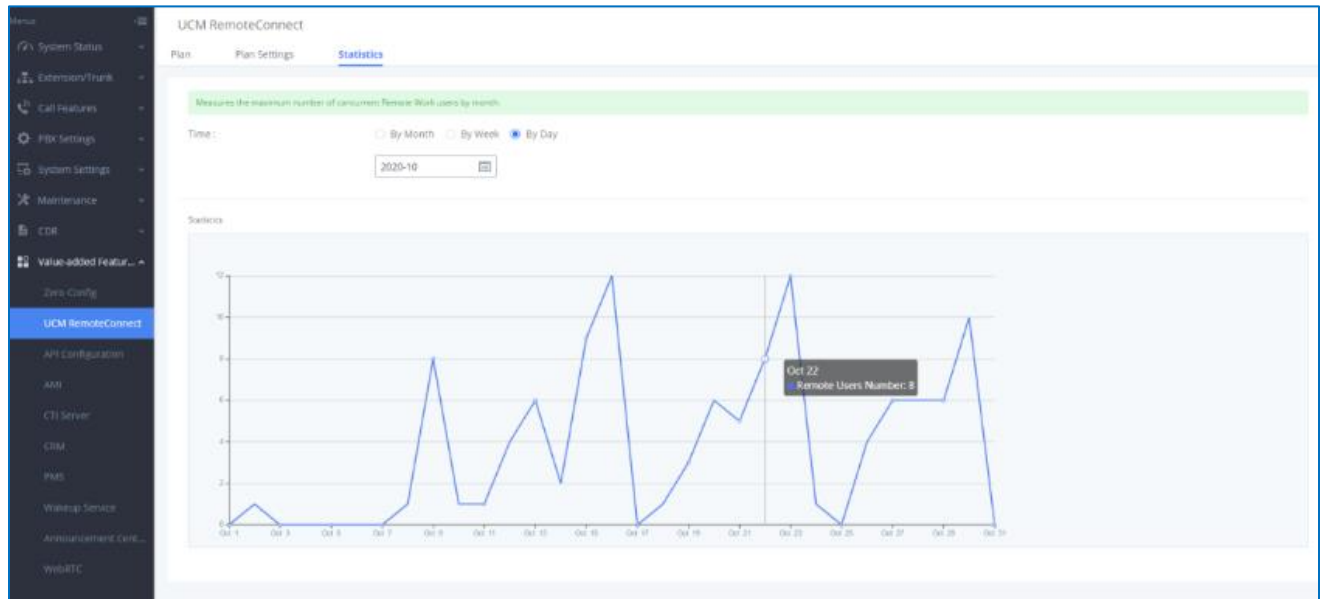


Figure 235: Concurrent Remote Calls

For more information, please visit <http://ucmrc.gdms.cloud/intro.html> and read our [UCM63XXA RemoteConnect guides](#)



## API CONFIGURATION

The UCM630xA supports third party billing interface API for external billing software to access CDR and call recordings on the PBX. The API uses HTTPS to request the CDR data and call recording data matching given parameters as configured on the third-party application.

### API Configuration Parameters

Before accessing the API, the administrators need enable API and configure the access/authentication information on the UCM630xA first under **Value-added Features**→**API Configuration**. The API configuration parameters are listed in the table below.

Table 116: Configuration Parameters (New)

HTTPS API Settings (New)	
<b>Enable</b>	Enable/Disable API. The default setting is enable.
<b>Username</b>	Configure the username for API Authentication.
<b>Password</b>	Configure the password for API Authentication.
<b>Call Control</b>	If enabled, 3 <sup>rd</sup> party applications will be able to manage inbound calls via API actions. <b>acceptCall</b> will accept incoming calls while <b>refuseCall</b> will reject them. If no actions are done within 10 seconds, calls will automatically be accepted.

### API Queries Supported

The new API supports now new queries listed below which will accomplish certain requests and get data about different modules on UCM630xA.

Table 117: New API Supported Queries

Queries Supported
getSystemStatus
getSystemGeneralStatus
listAccount
getSIPAccount
updateSIPAccount



listVoIPTrunk
addSIPTrunk
getSIPTrunk
updateSIPTrunk
deleteSIPTrunk
listOutboundRoute
addOutboundRoute
getOutboundRoute
updateOutboundRoute
deleteOutboundRoute
listInboundRoute
addInboundRoute
getInboundRoute
updateInboundRoute
deleteInboundRoute
playPromptByOrg
listBridgedChannels
listUnBridgedChannels
Hangup
callbargе
listQueue
getQueue
updateQueue
addQueue
deleteQueue
loginLogoffQueueAgent
pauseUnpauseQueueAgent
listPaginggroup
addPaginggroup
getPaginggroup
updatePaginggroup



deletePaginggroup
MulticastPaging
MulticastPagingHangup
listIVR
addIVR
getIVR
updateIVR
deleteIVR
cdrapi
recapi
pmsapi
queueapi
getPinSets
addPinSets
updatePinSets
deletePinSets

**Table 118: API Configuration Parameters**

CDR Real-time Output Settings	
<b>Enable</b>	Enables real-time CDR output module. This module connects to selected IP addresses and ports and posts CDR strings as soon as it is available.
<b>Server Address</b>	CDR server IP address
<b>Port</b>	CDR server IP port
Upload Prompts User Configuration	
<b>Username</b>	Username used to upload prompts.
<b>Password</b>	Password used to upload prompts.

## Upload Voice Prompt via API

Customers now can use the “Upload Prompts User Configuration” to upload/replace voice prompt files as an alternative method to the manual upload method on UCM **PBX Settings**→ **Voice Prompt**→ **Custom Prompt**.

The workflow of the prompt file upload goes as:

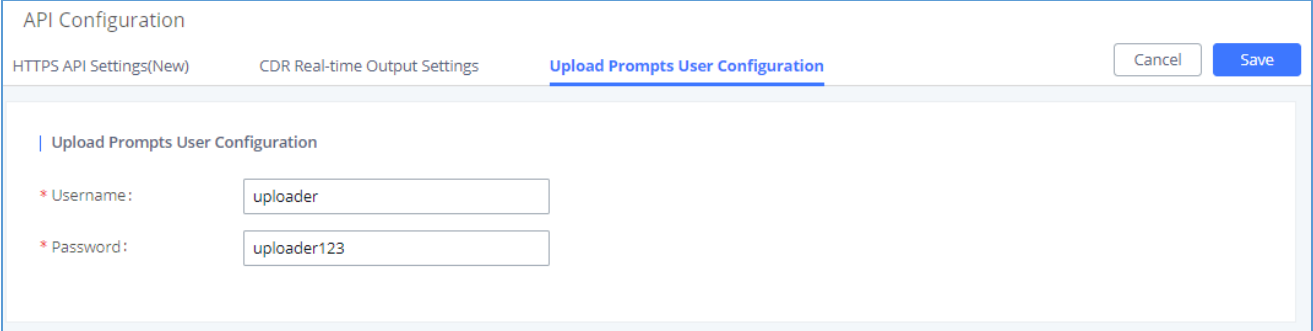




An HTTP/HTTPS request is sent to the UCM to upload/replace a voice prompt file, the request should include authentication details to the UCM and the name of the file to be uploaded. Then the UCM will contact an FTP server that should be hosted on the same IP address of the HTTP/HTTPS requester and download the prompt file from the FTP server.

The steps and conditions to upload the voice prompt via API are listed below:

1. Configure the prompt User under **value-added Features → API Configuration → Upload Prompts User Configuration**. By default, the username and password for voice prompt user are “Username: uploader; Password: uploader123”.



The screenshot shows a web browser window titled "API Configuration". Inside, there are three tabs: "HTTPS API Settings(New)", "CDR Real-time Output Settings", and "Upload Prompts User Configuration" (which is selected). Below the tabs, there is a section titled "Upload Prompts User Configuration" containing two form fields: "Username:" with the text "uploader" and "Password:" with the text "uploader123". At the top right of the form area, there are "Cancel" and "Save" buttons.

**Figure 236: Upload Prompt User Configuration**

2. Hash the password of the user configured to an MD5 Encryption format.
3. Set the permission on the FTP server to Anonymous on the local computer hosting the FTP server and make sure that the default FTP port 21 is used.
4. Send an HTTP/HTTPS command to trigger the Prompt file upload on the UCM. If UCM's HTTP server is set to HTTPS, the example of the request sent to the UCM is:

<https://192.168.124.89:8089/cgi?action=uploadprompt&username=uploader&password=9191a6394c21b3aabd779213c7179462&filename=test.mp3>

If UCM's HTTP server is set to HTTP, the example of the request sent to the UCM is:

<http://192.168.124.89:8089/cgi?action=uploadprompt&username=uploader&password=9191a6394c21b3aabd779213c7179462&filename=test.mp3>

**Note:** If the File name on the HTTP/HTTPS request exists already on the UCM's Custom voice prompts list the existing file will be overwritten by the new file downloaded from the FTP server.

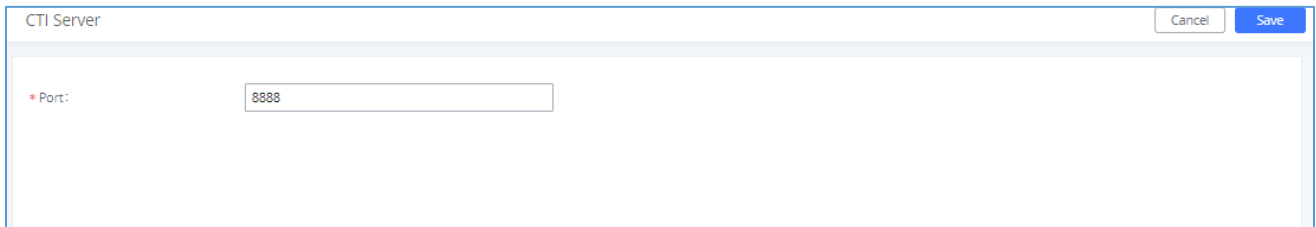


## CTI SERVER

UCM does support CTI server capabilities which are designed to be a part of the CTI solution suite provided by Grandstream, including GXP21XX and GXP17XX enterprise IP phones along with GS Affinity app.

Mainly the UCM will by default listening on port TCP 8888 for the connections from GS affinity application in order to interact, modify and serve data requests by the application which includes setting call features for the connected extension as call forward and DND.

Users can change the listening port under the menu page, Web GUI→**Value-added Features**→**CTI Server** as shown on below screenshot:



The screenshot shows a web interface titled "CTI Server". In the top right corner, there are "Cancel" and "Save" buttons. The main content area contains a label "Port:" followed by a text input field containing the value "8888".

**Figure 237: CTI Server Listening port**

More information about GS affinity and CTI Support on Grandstream products series please refer to the following link: [https://www.grandstream.com/hubfs/Product\\_Documentation/GS\\_Affinity\\_Guide.pdf](https://www.grandstream.com/hubfs/Product_Documentation/GS_Affinity_Guide.pdf)



## ASTERISK MANAGER INTERFACE (RESTRICTED ACCESS)

The UCM630xA supports Asterisk Manager Interface (AMI) with restricted access. AMI allows a client program to connect to an Asterisk instance commands or read events over a TCP/IP stream. It is particularly useful when the system admin tries to track the state of a telephony client inside Asterisk.

User could configure AMI parameters on UCM630xA Web GUI→**Value-added Features**→**AMI**. For details on how to use AMI on UCM630xA, please refer to the following AMI guide:

[https://www.grandstream.com/hubfs/Product\\_Documentation/UCM\\_series\\_AMI\\_guide.pdf](https://www.grandstream.com/hubfs/Product_Documentation/UCM_series_AMI_guide.pdf)

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 **Warning:**

Please do not enable AMI on the UCM630xA if it is placed on a public or untrusted network unless you have taken steps to protect the device from unauthorized access. It is crucial to understand that AMI access can allow AMI user to originate calls and the data exchanged via AMI is often very sensitive and private for your UCM630xA system. Please be cautious when enabling AMI access on the UCM630xA and restrict the permission granted to the AMI user. By using AMI on UCM630xA you agree you understand and acknowledge the risks associated with this.

---



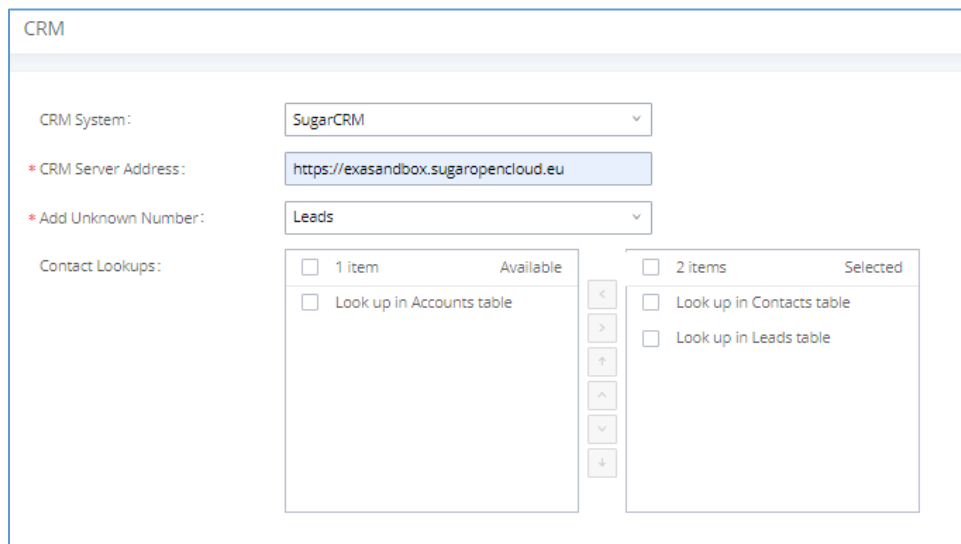
## CRM INTEGRATION

**Customer relationship management (CRM)** is a term that refers to practices, strategies and technologies that companies use to manage and analyze customer interactions and data throughout the customer lifecycle, with the goal of improving business relationships with customers.

The UCM630xA support the following CRMs: SugarCRM, vTigerCRM, ZohoCRM, Salesforce CRM and ACT! CRM, which allows users to look for contact information in the Contacts, Leads and / or Accounts tables, shows the contact record in CRM page, and saves the call information in the contact's history.

### SugarCRM

Configuration page of the SugarCRM can be accessed via admin login, on the UCM WebGUI → **Value-added Features** → **CRM**.



**Figure 238: SugarCRM Basic Settings**



1. Select “SugarCRM” from the CRM System Dropdown in order to use SugarCRM.

**Table 119: SugarCRM Settings**


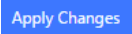
<b>CRM System</b>	Select a CRM system from the dropdown menu, four CRM systems are available: SugarCRM, vTigerCRM, ZohoCRM (v1&v2), Salesforce and ACT! CRM.
<b>CRM Server Address</b>	Enter the IP address of the CRM server.
<b>Add Unknown Number</b>	Add the new number to this module if it cannot be found in the selected module.


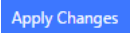


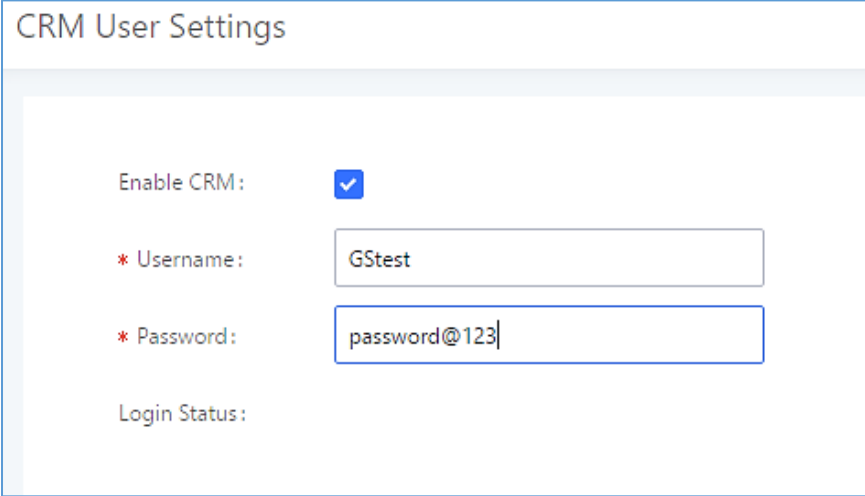
### Contact Lookups

Select from the **“Available”** list of lookups and press   to select where the UCM can perform the lookups on the CRM tables, Leads, Accounts, and Contacts.

Once settings on admin access are configured:

2. Click on  and .
3. Logout from admin access.
4. Login to the UCM as user and navigate under “User Portal→Value-added Feature→CRM User Settings”.

Click on **“Enable CRM”** and enter the username/password associated with the CRM account then click on  and . The status will change from “Logged Out” to “Logged In”. User can start then using SugarCRM features.



The screenshot shows the 'CRM User Settings' configuration page. It includes the following fields and controls:

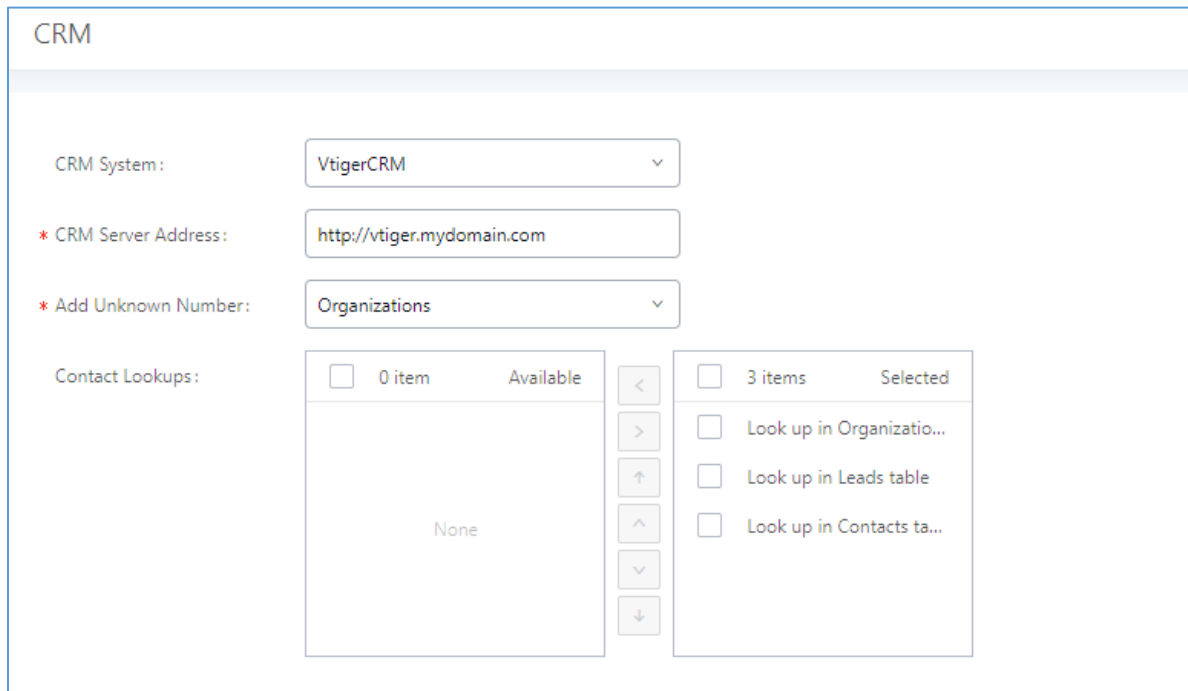
- Enable CRM:** A checkbox that is checked.
- \* Username:** A text input field containing the value 'GStest'.
- \* Password:** A text input field containing the value 'password@123'.
- Login Status:** A label indicating the current status, which is not explicitly filled in the image.

Figure 239: CRM User Settings

## VTigerCRM

Configuration page of the vTigerCRM can be accessed via admin login, on the UCM WebGUI→**Value-added Features**→**CRM**.








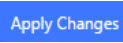
**Figure 240: vTigerCRM Basic Settings**

1. Select “vTigerCRM” from the CRM System Dropdown in order to use vTigerCRM.

**Table 120: vTigerCRM Settings**

<b>CRM System</b>	Select a CRM system from the dropdown menu, four CRM systems are available: SugarCRM, vTigerCRM, ZohoCRM (v1&v2), Salesforce and ACT! CRM.
<b>CRM Server Address</b>	Enter the IP address of the CRM server.
<b>Add Unknown Number</b>	Add the new number to this module if it cannot be found in the selected module.
<b>Contact Lookups</b>	Select from the “ <b>Available</b> ” list of lookups and press   to select where the UCM can perform the lookups on the CRM tables, Leads, Organizations, and Contacts.

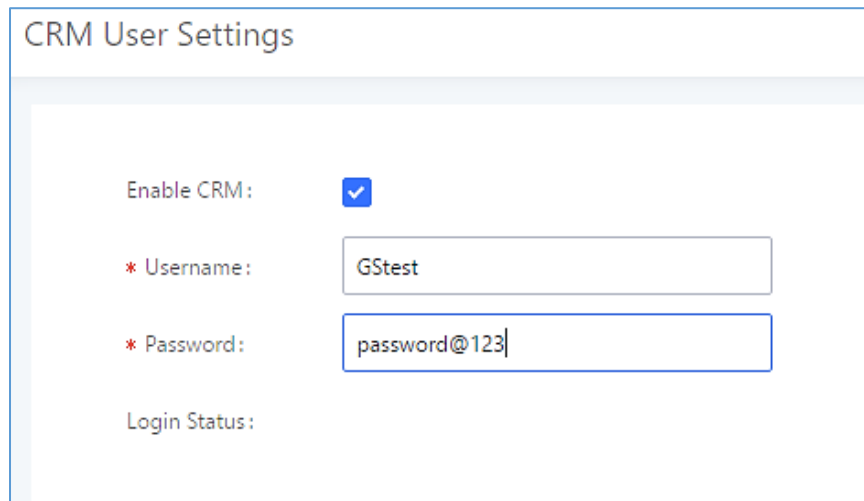
Once settings on admin access are configured:

2. Click on  and .
3. Logout from admin access.
4. Login to the UCM as user and navigate under “User Portal→Value-added Feature→CRM User Settings”.

Click on “**Enable CRM**” and enter the username/password associated with the CRM account then click on



**Save** and **Apply Changes**. The status will change from “Logged Out” to “Logged In”. User can start then using SugarCRM features.



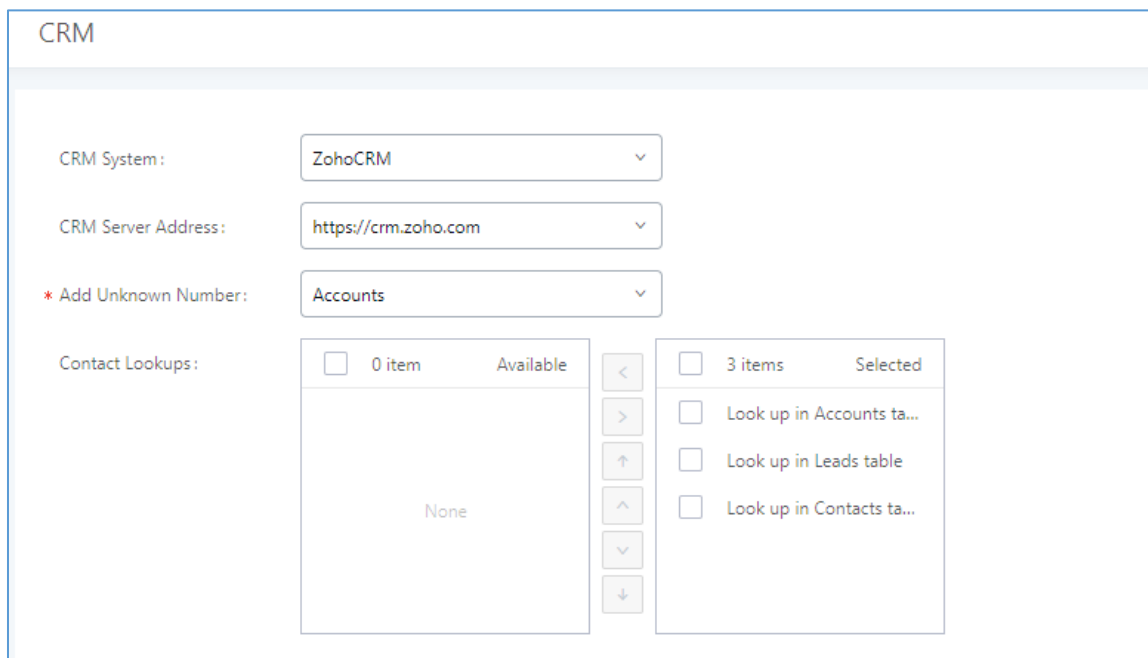
The screenshot shows a web form titled "CRM User Settings". It contains the following fields:

- Enable CRM:** A checkbox that is checked.
- \* Username:** A text input field containing "GStest".
- \* Password:** A text input field containing "password@123".
- Login Status:** A label with no associated input field.

**Figure 241: CRM User Settings**

## ZohoCRM

Configuration page of the ZohoCRM can be accessed via admin login, on the UCM WebGUI→**Value-added Features**→**CRM**.



The screenshot shows a web form titled "CRM". It contains the following fields:



- CRM System:** A dropdown menu with "ZohoCRM" selected.
- CRM Server Address:** A dropdown menu with "https://crm.zoho.com" selected.
- \* Add Unknown Number:** A dropdown menu with "Accounts" selected.
- Contact Lookups:** A complex field with two panes. The left pane is titled "0 item Available" and contains "None". The right pane is titled "3 items Selected" and contains three checkboxes:
  - Look up in Accounts ta...
  - Look up in Leads table
  - Look up in Contacts ta...

**Figure 242: ZohoCRM Basic Settings**

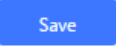
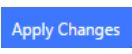



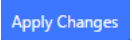
1. Select “ZohoCRM” from the CRM System Dropdown in order to use ZohoCRM.

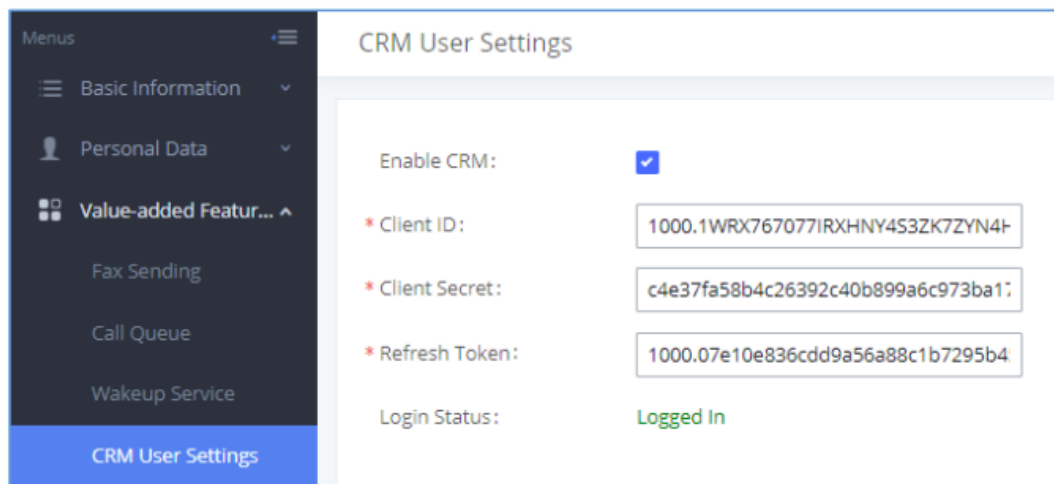
**Table 121: ZohoCRM Settings**

<b>CRM System</b>	Select a CRM system from the dropdown menu, four CRM systems are available: SugarCRM, vTigerCRM, ZohoCRM (v1&v2), Salesforce and ACT! CRM.
<b>CRM Server Address</b>	Enter the IP address of the CRM server.
<b>Add Unknown Number</b>	Add the new number to this module if it cannot be found in the selected module.
<b>Contact Lookups</b>	Select from the “ <b>Available</b> ” list of lookups and press   to select where the UCM can perform the lookups on the CRM tables, Leads, Accounts, and Contacts.

Once settings on admin access are configured:

2. Click on  and .
3. Logout from admin access.
4. Login to the UCM as user and navigate under “User Portal→Value-added Feature→CRM User Settings”.

Click on “**Enable CRM**” and enter the username/password associated with the CRM account then click on  and . The status will change from “Logged Out” to “Logged In”. User can start then using ZohoCRM features.



**Figure 243: CRM User Settings**

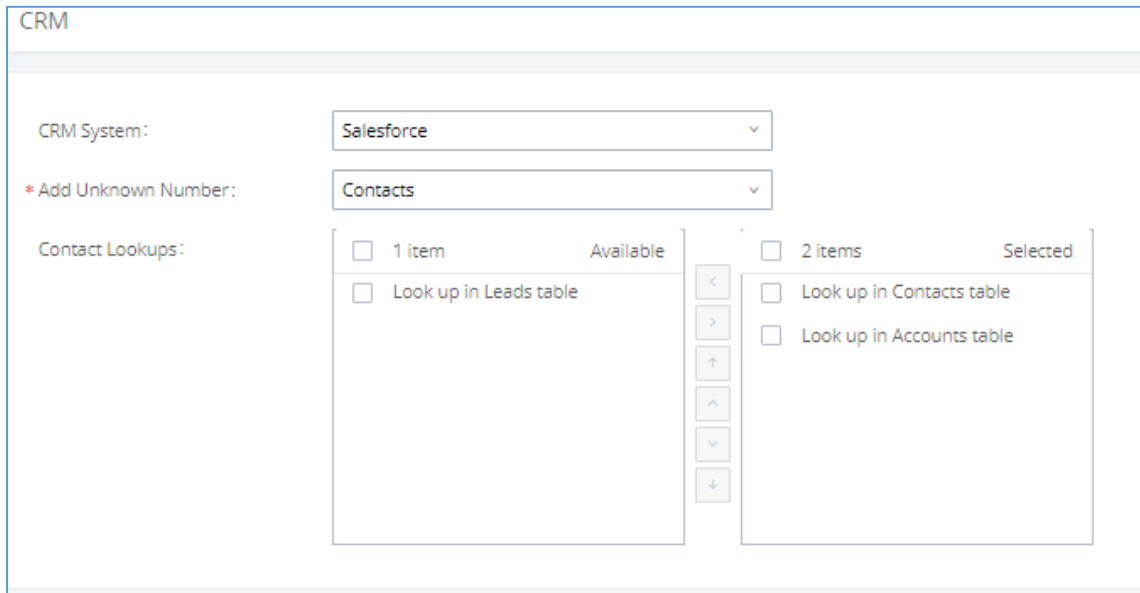
**Note:** ZohoV2CRM is supported as well while the CRM Server Address <https://www.zohozpis.com>





## Salesforce CRM



Configuration page of the Salesforce CRM can be accessed via admin login, on the UCM Web GUI → **Value-added Features** → **CRM**.



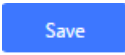
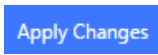
**Figure 244: Salesforce Basic Settings**

1. Select “Salesforce” from the CRM System Dropdown in order to use Salesforce CRM.

**Table 122: Salesforce Settings**

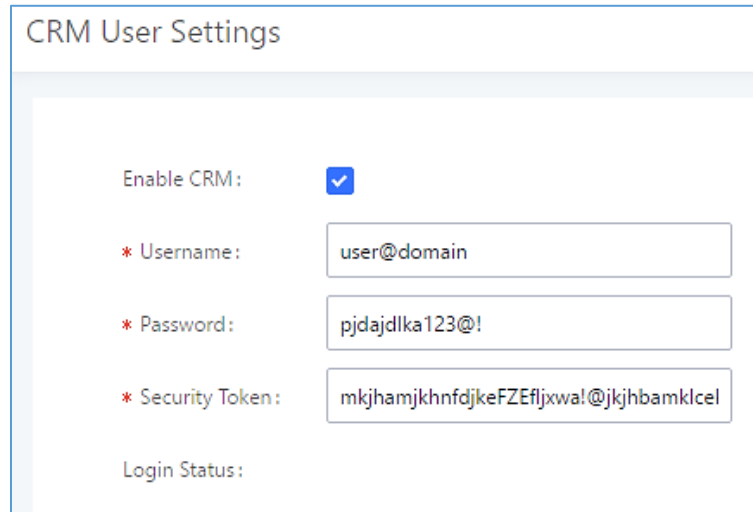
<b>CRM System</b>	Select a CRM system from the dropdown menu, four CRM systems are available: SugarCRM, vTigerCRM, ZohoCRM (v1&v2), Salesforce and ACT! CRM.
<b>Add Unknown Number</b>	Add the new number to this module if it cannot be found in the selected module.
<b>Contact Lookups</b>	Select from the “ <b>Available</b> ” list of lookups and press   to select where the UCM can perform the lookups on the CRM tables, Leads, Accounts, and Contacts.

Once settings on admin access are configured:

2. Click on  and .
3. Logout from admin access.
4. Login to the UCM as user and navigate under “User Portal → Value-added Feature → CRM User Settings”.



Click on “**Enable CRM**” and enter the **username**, **password** and **Security Token** associated with the CRM account then click on **Save** and **Apply Changes**. The status will change from “Logged Out” to “Logged In”. User can start then using Salesforce CRM features.



The screenshot shows the 'CRM User Settings' form. It includes a checkbox for 'Enable CRM' which is checked. Below it are three required fields: 'Username' with the value 'user@domain', 'Password' with the value 'pjdajdlka123@!', and 'Security Token' with the value 'mkjhamjkhndjkeFZEfljxwa!@jkjhbamklcel'. A 'Login Status' field is also present but empty.

**Figure 245: Salesforce User Settings**

## ACT! CRM

Configuration page of the ACT! CRM can be accessed via admin login, on the UCM Web GUI → **Value-added Features** → **CRM**.

The configuration steps of the ACT! CRM are as follows:

1. Navigate to **Value-Added Features** → **CRM** and select the “ACT! CRM” option.



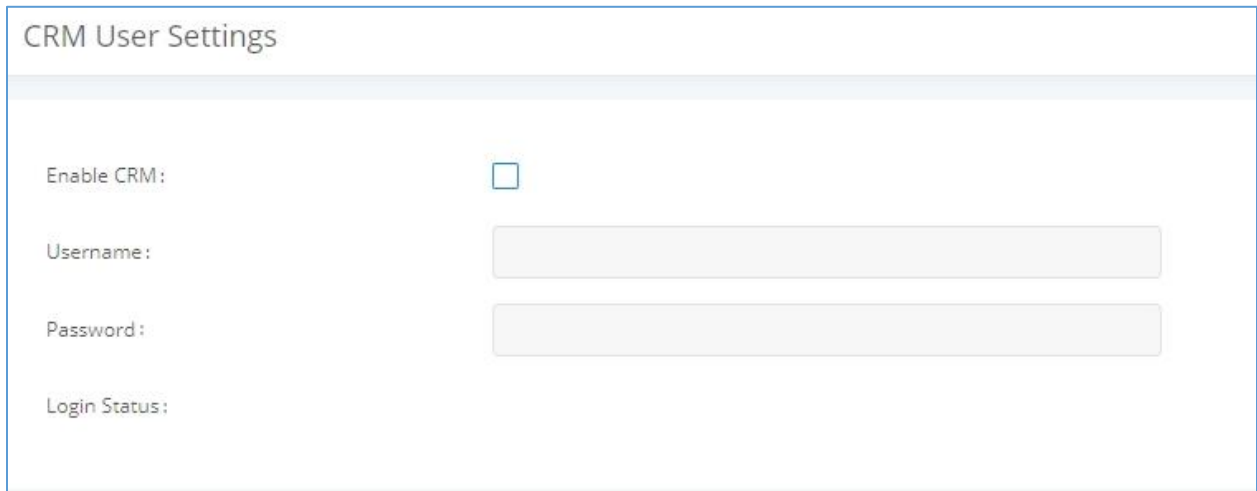
The screenshot shows the 'CRM' configuration page. It features a dropdown menu labeled 'CRM System:' with 'ACT! CRM' selected.

**Figure 246: Enabling ACT! CRM**

2. Log into the UCM as a regular user and navigate to **Value-Added Features** → **CRM User Settings** and check “Enable CRM” option and enter the username and password, which will be the ACT! CRM



account's **API Key** and **Developer Key**, respectively. To obtain these, please refer to the ACT! CRM API developer's guide here: <https://mycloud.act.com/act/Help>



The screenshot shows a web form titled "CRM User Settings". It contains four fields: "Enable CRM:" with an unchecked checkbox, "Username:" with a text input field, "Password:" with a text input field, and "Login Status:" with no visible input field.

**Figure 247: Enabling CRM on the User Portal**

**Note:** For more information on the ACT! CRM integration, please refer to the ACT! CRM documentation on our website.



## PMS INTEGRATION

UCM630xA supports Hotel Property Management System PMS, including check-in/check-out services, wakeup calls, room status, Do Not Disturb which provide an ease of management for hotel applications. This feature can be found on Web GUI→**Value-added Features**→**PMS**.

**Note:** The PMS integration on UCM is currently supported only with one of the three following solutions.

The PMS module built-in the UCM supports the following features based on each solution:

Table 123: PMS Supported Features

Feature	Mitel	HMobile	HSC	IDS
Check-In	✓	✓	X	✓
Check-out	✓	✓	X	✓
Wake-up Call	✓	✓	X	✓
Name Change	✓	X	✓	X
Update	X	✓	X	✓
Set Credit	✓	X	X	X
Set Station Restriction	✓	X	✓	X
Room Status	X	✓	X	✓
Room Move	X	✓	X	✓
Do Not Disturb	X	✓	✓	X
Mini Bar	X	✓	X	✓
MSG	X	✓	X	X
MWI	X	X	✓	X
Unconditional Call Forward	X	X	✓	X

### HMobile PMS Connector

In this mode, the system can be divided into three parts:

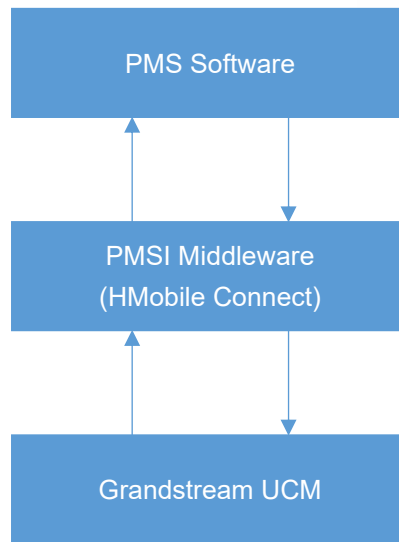
- PMS (Property Management System)
- PMSI (Property Management System Interface)
- PBX

Grandstream UCM6XXX series have integrated HMobile Connect PMSI which supports a large variety of PMS software providing following hospitality features: Check-in, Check-out, set Room Status, Wake-up call and more.

The following figure illustrates the communication flow between the UCM and PMS software, which is done



through a middleware system (HMobile Connect) acting as interface between both parties.



**Figure 248: UCM & PMS interaction**

## HSC PMS

In this mode, the system can be divided into two parts:

- PMS (Property Management System)
- PBX

Grandstream UCM6XXX series have integrated HSC PMS providing following features:

- Changing Display Name
- Set Station Restriction
- Call forwarding
- DND
- Name Change
- MWI

Note:

3. Added support for receiving HTTP GET keep-alive messages from HSC PMS. This will allow the PMS to be aware of its connection to the UCM and take the appropriate actions such as raising alarms, sending notifications, etc.
4. Added support for HTTP GET requests from HSC PMS to retrieve UCM extension information. UCM can provide the following information:
  - extension – UCM extension number

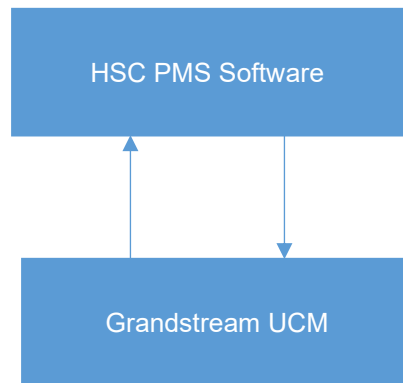


- name – extension display name / CID name
- mwi – MWI state
- permission – permission level of the extension
- cfwt – call forwarding always number
- dnd – DND state
- language – display language of the extension in ISO 639-1 format

The UCM should respond with either 200 OK or 404 responses.

#### 5. Added HTTPS support

The following figure illustrates the communication flow between the PBX (Grandstream UCM6xxx Series) and PMS software (HSC). The communication between both parties is direct with no middleware.



**Figure 249: UCM & HSC PMS interaction**

### Mitel PMS

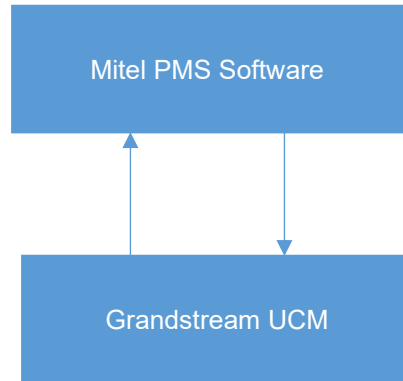
In this mode, the system can be divided into two parts:

- PMS (Property Management System)
- PBX

Grandstream UCM6XXX series have integrated Mitel PMS providing following hospitality features: Check-in, Check-out, set Room Status, Wake-up call and more.

The following figure illustrates the communication flow between the PBX (Grandstream UCM6xxx Series) and PMS software (Mitel). The communication between both parties is direct with no middleware.





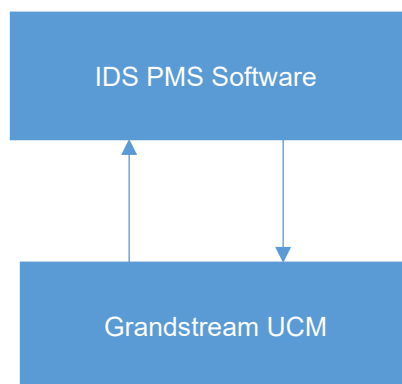
**Figure 250: UCM & Mitel PMS interaction**

## IDS PMS

In this mode, the system can be divided into two parts:

- PMS (Property Management System)
- PBX

The Grandstream UCM series integrates IDS PMS to set room status, Mini Bar, wake up calls, activate/deactivate dialing permissions, and more.



**Figure 251: UCM & IDS PMS interaction**

## PMS API

The PMS API allows users to use their own middleware to work with PMS systems instead of currently supported integrations.

Additionally, this API allows access to read and modify certain UCM parameters that current supported PMS integrations cannot. To use this, users must first enable and configure the HTTPS API settings.

For more details, please refer to online [HTTPS API](#), Pmsapi section.



## Connecting to PMS

On the UCM WebGUI → **Value-added Features** → **PMS** → **Basic Settings**” set the connection information for the PMS platform.

Table 124: PMS Basic Settings

Field	Description
<b>PMS Module</b>	Users can select the desired PMS module from the drop-down list. <ul style="list-style-type: none"> <li>• Hmobile.</li> <li>• Mitel.</li> <li>• HSC.</li> <li>• IDS.</li> <li>• PMS API.</li> </ul>
<b>Wakeup Prompt</b>	Prompt used when answering the wakeup calls it can be customized from “ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Custom Prompt</b> .”
<b>PMS URL</b>	Enter the PMS system URL
<b>UCM Port</b>	Enter the Port used by the PMS system
<b>Username</b>	Enter the Username to connect to the PMS system
<b>Password</b>	Enter the password to connect to the PMS system
<b>Site</b>	Enter PMS site
<b>Back Up Voicemail Recordings</b>	If enabled, this option allows backing up voicemail recordings to external storage after check-out.  <b>Note:</b> This option is available only when the PMS Module is set to PMS API.
<b>Email address</b>	Email address to send voicemail recordings to upon backup.

In order to use some PMS features please activate the feature code associated under “**Call Features** → **Feature Codes**”

- Update PMS Room Status
- PMS Wake Up Service

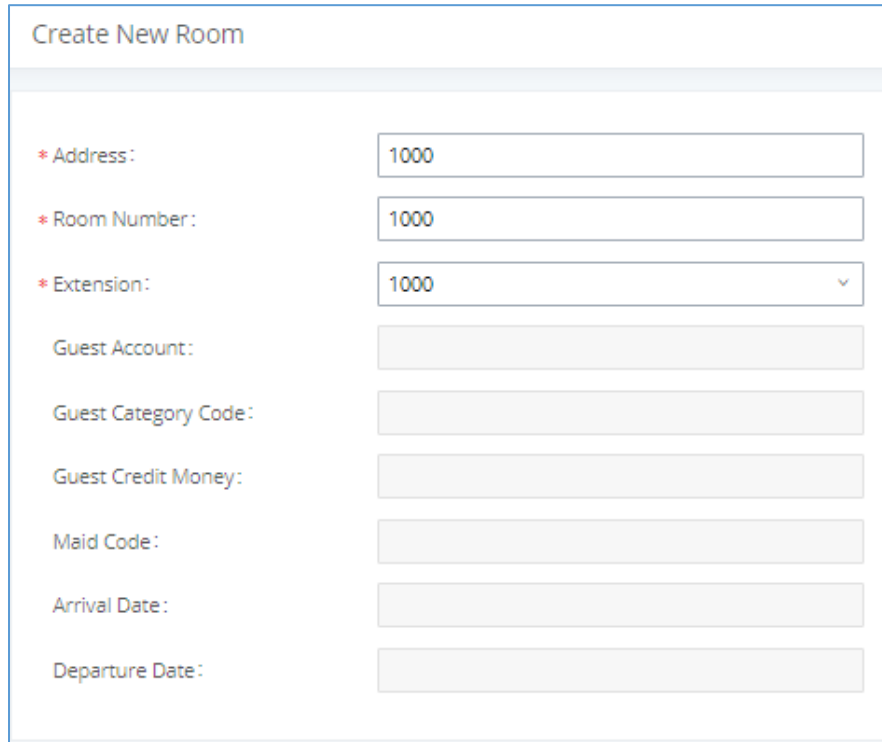




## PMS Features

### Room Status

User can create Rooms by clicking on "Add Room", the following Figure will be displayed then.



**Create New Room**

\* Address: 1000

\* Room Number: 1000

\* Extension: 1000

Guest Account:

Guest Category Code:

Guest Credit Money:

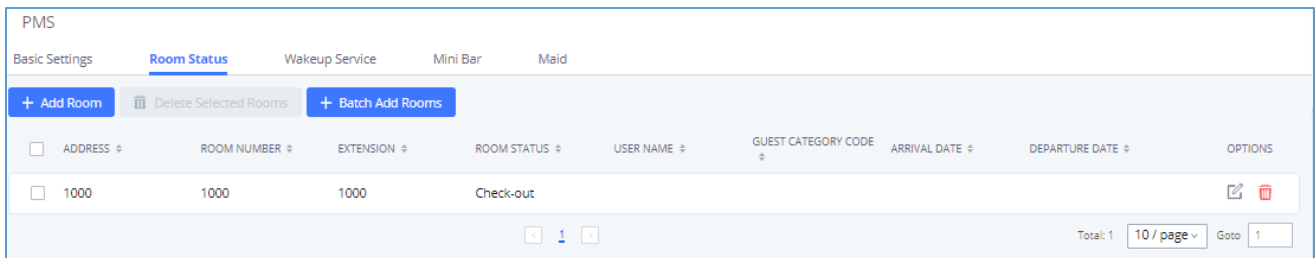
Maid Code:

Arrival Date:

Departure Date:

**Figure 252: Create New Room**



Click "Save" to create the new room, the fields above can be configured from the PMS platform, once set the following screen will be shown:



PMS

Basic Settings **Room Status** Wakeup Service Mini Bar Maid

+ Add Room Delete Selected Rooms + Batch Add Rooms

<input type="checkbox"/>	ADDRESS	ROOM NUMBER	EXTENSION	ROOM STATUS	USER NAME	GUEST CATEGORY CODE	ARRIVAL DATE	DEPARTURE DATE	OPTIONS
<input type="checkbox"/>	1000	1000	1000	Check-out					 

Total: 1 10 / page Goto 1

**Figure 253: Room Status**

User can create a batch of rooms as well by clicking on **+ Batch Add Rooms**, the following window will pop up:



### Batch Add Rooms

---

\* Start Address Number:

\* Start Room Number:

\* Start Extension: ▼

\* Create Number:

**Figure 254: Add batch rooms**

## Wake Up Service

In order to create a New Wake up service, user can click on "Add", the following window will pop up:

### Create New Wakeup Service

---

\* Room Number: ▼

\* Select Time:

\* Action Status: ▼

Type: ▼

**Figure 255: Create New Wake Up Service**



**Table 125: PMS Wake up Service**

Field	Description
<b>Room Number</b>	Select the room number where to call with a limitation of 63 characters.
<b>Select Time</b>	Set the time of the wakeup call
<b>Action Status</b>	Show the status of the call: <ul style="list-style-type: none"> <li><u>Programmed</u>: the call is scheduled for the time set</li> <li><u>Cancelled</u>: the call is canceled</li> <li><u>Executed</u>: the wakeup call is made</li> </ul> <p><b>Note:</b> Editing an already executed wakeup service will automatically change the service's status to "Programmed".</p>



<b>Type</b>	<ul style="list-style-type: none"> <li>• <u>Single</u>: The call will be made once on the specific time.</li> <li>• <u>Daily</u>: The call will be repeated every day on the specific time</li> </ul>
-------------	---

Once the call is made on the time specified, the following figure show the status of the wakeup call.

ROOM NUMBER #	ACTION STATUS #	TYPE #	ANSWER STATUS #	DATE	TIME	OPTIONS
1000	Programmed	Single	No action	2019-12-12	08:00	 

**Figure 256: Wakeup Call executed**

This call has been executed but has been rejected, that why we can see the “**Busy**” status.

## Mini Bar

In order to create a new mini bar, click on “Add Mini Bar” under UCM WebGUI→**Value-added Features**→**PMS**→**Mini Bar**, the following window will pop up:

Create New Mini Bar

---

\* Code:

\* Name:

\* Prompt:  [Upload Audio File](#)

Skip Maid and Password Authentication:

Enable Continuous Multi Goods Billing:

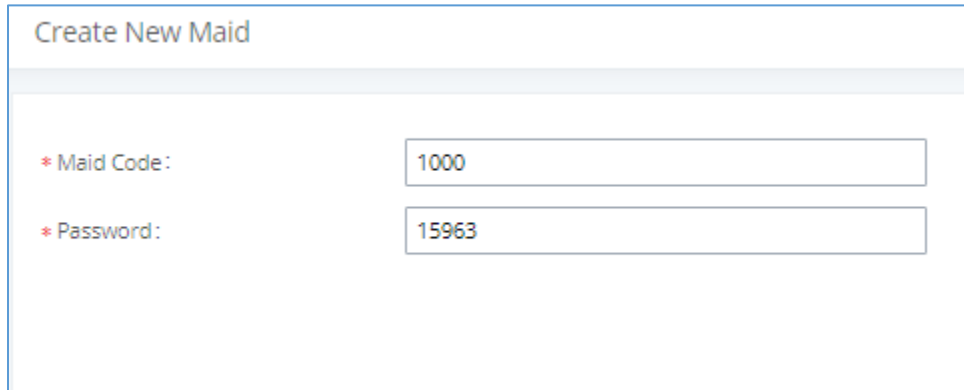
**Figure 257: Create New Mini Bar**

**Table 126: Create New Mini Bar**

<b>Code</b>	Enter a non-existing extension number to be dialed when using the mini bar feature.
<b>Name</b>	Enter a name for the mini bar.
<b>Prompt</b>	Select the Prompt to play once connected to the mini bar.
<b>Skip Maid and Password Authentication</b>	If enabled, the default maid code will be 0000, no authentication is required. (Enter 0000 followed by # to access the consumer goods)
<b>Enable Continuous Multi Goods Billing</b>	If enabled, please separate the goods' codes by*.



In order to create a new maid, click on [+ Create New Maid](#) under UCM WebGUI→**Value-added Features**→**PMS**→**Maid**.

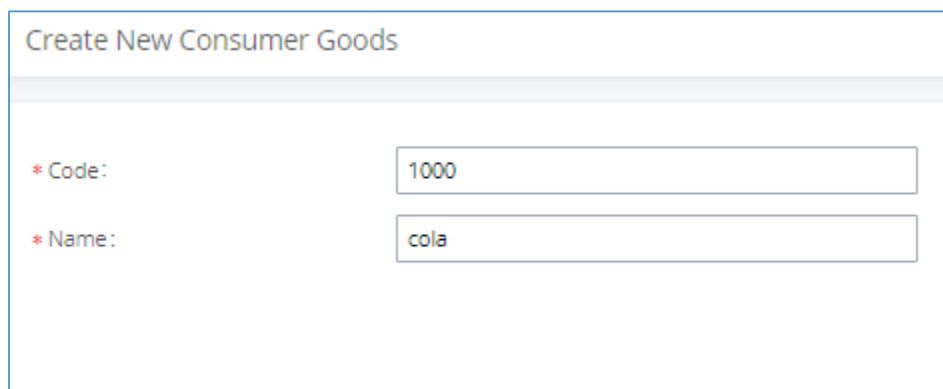


**Figure 258: Create New Maid**

**Table 127: Create New Maid**

<b>Maid Code</b>	Enter the Code to use when the maid wants to use the Mini Bar.
<b>Password</b>	Enter the password associated with the maid.

In order to create a new consumer goods, click on [+ Create New Consumer Goods](#) under UCM WebGUI→**Value-added Features**→**PMS**→**Mini Bar**, the following window will popup.







**Figure 259: Create New Consumer Goods**

<b>Code</b>	Enter the Goods Code.
<b>Name</b>	Enter the Name of the Goods



The Minibar page displays as:

PMS		
Basic Settings	Room Status	Wakeup Service
<b>Mini Bar</b>	Maid	
+ Add Mini Bar		
CODE	NAME	OPTIONS
4000	MiniBar	 
+ Add Consumer Goods		
CODE	NAME	OPTIONS
1000	cola	 

**Figure 260: Mini Bar**



## WAKEUP SERVICE

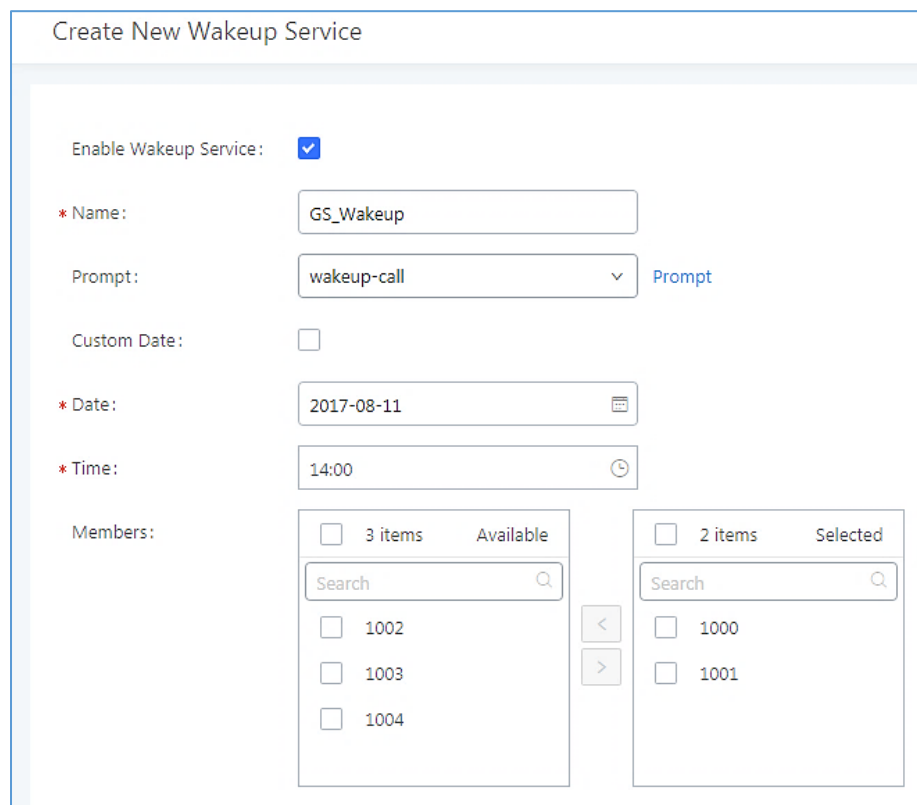
The Wake Up service can be used to schedule a reminder or wake up calls to any valid destination. This service is available on the UCM630xA as a separated module.

There are three ways to set up Wakeup Service:

- Using admin login
- Using user portal
- Using feature code

### Wake Up Service using Admin Login

1. Login to the UCM as admin.
2. Wake Up service can be found under Web GUI → **Value-added Features** → **Wakeup Service**, click on "Add" to create a new wakeup service. The following window will pop up.



**Create New Wakeup Service**

Enable Wakeup Service:

\* Name:

Prompt:  Prompt

Custom Date:

\* Date:

\* Time:

Members:

Available	Selected
3 items	2 items
<input type="checkbox"/> 1002 <input type="checkbox"/> 1003 <input type="checkbox"/> 1004	<input type="checkbox"/> 1000 <input type="checkbox"/> 1001

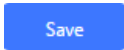
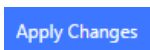
**Figure 261: Create New Wakeup Service**

3. Fill out the required fields and select the members to add to the wakeup group.



**Table 128: Wakeup Service**

<b>Enable Wakeup Service</b>	Enable Wakeup service.
<b>Name</b>	Enter a name (up to 64 characters) to identify the wakeup service.
<b>Prompt</b>	Select the prompt to play for that extension.
<b>Custom Date</b>	If disabled, users can select a specific date and time. If enabled users can select multiple days of the week to perform the wakeup.
<b>Date</b>	Select the date or dates when to performs the wakeup call.
<b>Time</b>	Select the time when to play the wakeup call.
<b>Members</b>	Select the members involved within the wakeup service group.

4. Click  and  to apply the changes.

A wakeup service entry is created. The UCM will send a wakeup call to every extension in the member list at the scheduled date and time.

**Note:** the wakeup service has the following limitation on how many members can be added depending on UCM model.

**Table 129: Max Wakeup Members**

UCM Model	Max members in a Wakeup Service
UCM6300A	50
UCM6302A	100
UCM6304A	150
UCM6308A	200

## Wake Up Service from User Portal

1. Login to the user portal on the UCM630xA.
2. Wake Up service can be found under “Value-added Features→Wakeup Service”, click on “Add” to create a new wakeup service.
3. Configures the Name, Prompt, Date and Time for the user to make the wakeup to.



4. Click **Save** and **Apply Changes** to apply the changes.

## Wake Up Service using Feature Code

1. Login to the UCM as admin.
2. Enable “Wakeup Service” from the WebGUI under “**Call Features**→**Feature Codes**”.

* Listen Spy:	<input type="text" value="*54"/>	* Whisper Spy:	<input type="text" value="*55"/>
* Barge Spy:	<input type="text" value="*56"/>	* Wakeup Service:	<input type="text" value="*36"/> <input checked="" type="checkbox"/>
* PMS Wakeup Servi...	<input type="text" value="*35"/> <input checked="" type="checkbox"/>	* Update PMS Room...	<input type="text" value="*23"/> <input checked="" type="checkbox"/>
* Presence Status:	<input type="text" value="*48"/> <input checked="" type="checkbox"/>		

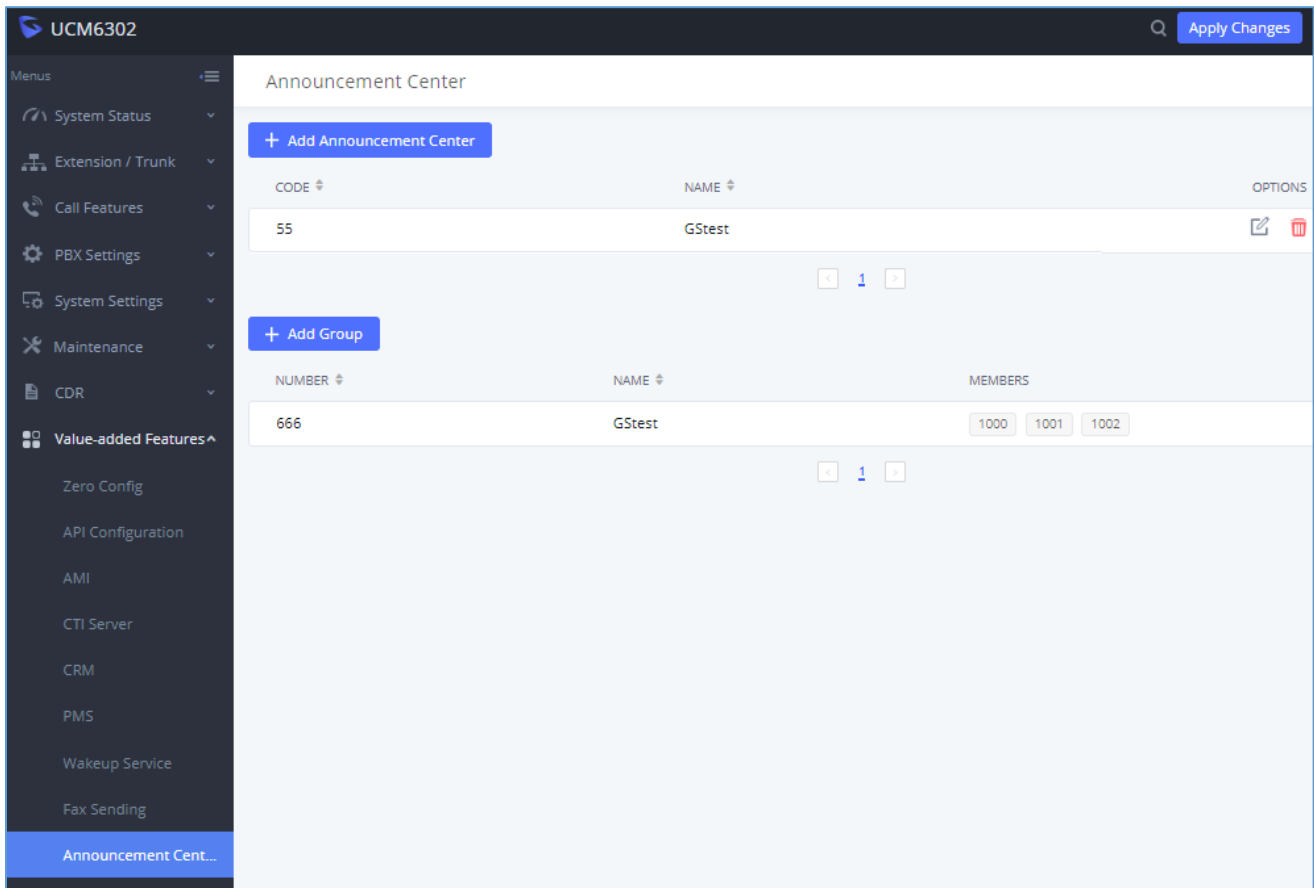
3. Click **Save** and **Apply Changes** to apply the changes.
4. Dial “\*36” which is the feature code by default to access to the UCM wakeup service to add, update, activate or deactivate UCM wakeup service.





## ANNOUNCEMENTS CENTER

The UCM630xA supports Announcements Center feature which allows users to pre-record and store voice message into UCM630xA with a specified code. The users can also create group with specified extensions. When the code and the group number are dialed together in the combination of **code + group number**, the specified voice message is sent to all group members and only extensions in the group will hear the voice message.



The screenshot shows the UCM6302 web interface for the Announcements Center. The sidebar on the left lists various system settings, with 'Announcement Cent...' selected. The main content area is titled 'Announcement Center' and includes an 'Apply Changes' button in the top right. Below the title is an '+ Add Announcement Center' button. A table lists existing announcement centers:

CODE	NAME	OPTIONS
55	GStest	[Edit] [Delete]

Below this table is an '+ Add Group' button. Another table lists existing groups:

NUMBER	NAME	MEMBERS
666	GStest	[1000] [1001] [1002]

Figure 262: Announcements Center



## Announcements Center Settings

Table 130: Announcements Center Settings


<b>Name</b>	Configure a name for the newly created Announcements Center to identify this announcement center.
<b>Code</b>	Enter a code number for the custom prompt. This code will be used in combination with the group number. For example, if the code is 55, and group number is 666. The user can dial 55666 to send prompt 55 to all members in group 666. <b>Note:</b> The combination number must not conflict with any number in the system such as extension number or meeting number.
<b>Custom Prompt</b>	This option is to set a custom prompt as an announcement to notify group members. The file can be uploaded from page 'Custom Prompt'. Click 'Prompt' to add additional record.
<b>Ring Timeout</b>	Configure the ring timeout for the group members. The default value is 30 seconds.
<b>Auto Answer</b>	If set to <b>Yes</b> the Auto answer will be enabled by the members.

## Group Settings

Table 131: Group Settings

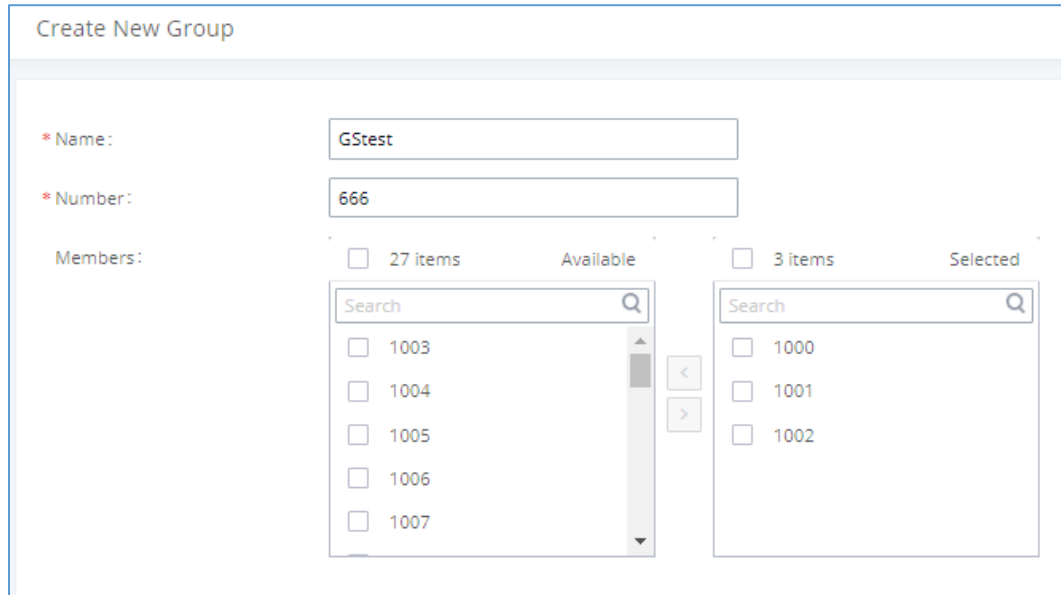
<b>Name</b>	Configure a name for the newly created group to identify the group. <b>Note:</b> Name cannot exceed 64 characters.
<b>Number</b>	Configure the group number. The group number is used in combination with the code. For example, if group number is 666, and code is 55. The user can dial 55666 to send prompt 55 to all members in group 666. <b>Note:</b> The combination number must not conflict with any number in the system such as extension number or meeting number and cannot exceed 64 characters.
<b>Members</b>	Select the group members from the available list.

Announcements Center feature can be found under Web GUI→**Value-added Features**→**Announcements Center**. The following example demonstrates the usage of this feature.

1. Click  to add new group.
2. Give a name to the newly created group.



3. Create a group number which is used with code to send voice message.
4. Select the extensions to be included in the group, who will receive the voice message.



**Figure 263: Announcements Center Group Configuration**

In this example, group “Test” has number 666. Extension 1000, 1001 and 1002 are in this group.

1. Click [+ Add Announcement Center](#) to create a new Announcement Center.
2. Give a name to the newly created Announcement Center.
3. Specify the code which will be used with group number to send the voice message to.
4. Select the message that will be used by the code from the Custom Prompt drop down menu. To create a new Prompt, please click “Prompt” link and follow the instructions in that page.



### Create New Announcement Center

<b>* Name:</b>	<input type="text" value="GStest"/>	
<b>* Code:</b>	<input type="text" value="55"/>	
<b>* Custom Prompt:</b>	<input type="text" value="Lee_Rosevere_-_12_-_Were_Almost_There-..."/>	<a href="#">Upload Audio File</a>
<b>* Ring Timeout:</b>	<input type="text" value="30"/>	
<b>* Auto Answer:</b>	<input type="text" value="No"/>	

**Figure 264: Announcements Center Code Configuration**

Code and Group number are used together to direct specified message to the target group. All extensions in the group will receive the message. For example, we can send code 55 to group 666 by dialing 55666 from any extension registered to the UCM630xA. All the members in group 666 which are extension 1000, 1001 and 1002 will receive this voice message after they pick up the call.

#### Announcement Center

[+ Add Announcement Center](#)

CODE	NAME	OPTIONS
55	GStest	<a href="#">✎</a> <a href="#">✖</a>

Total: 1

[+ Add Group](#)

NUMBER	NAME	MEMBERS	OPTIONS
666	GStest	<input type="text" value="1000"/> <input type="text" value="1001"/> <input type="text" value="1002"/>	<a href="#">✎</a> <a href="#">✖</a>

Total: 1

**Figure 265: Announcements Center Example**



# STATUS AND REPORTING

## PBX Status

The UCM630xA monitors the status for Trunks, Extensions, Queues, Meeting Rooms, Interfaces and Parking lot. It presents administrators the real-time status in different sections under Web GUI→**System Status**→**Dashboard**.

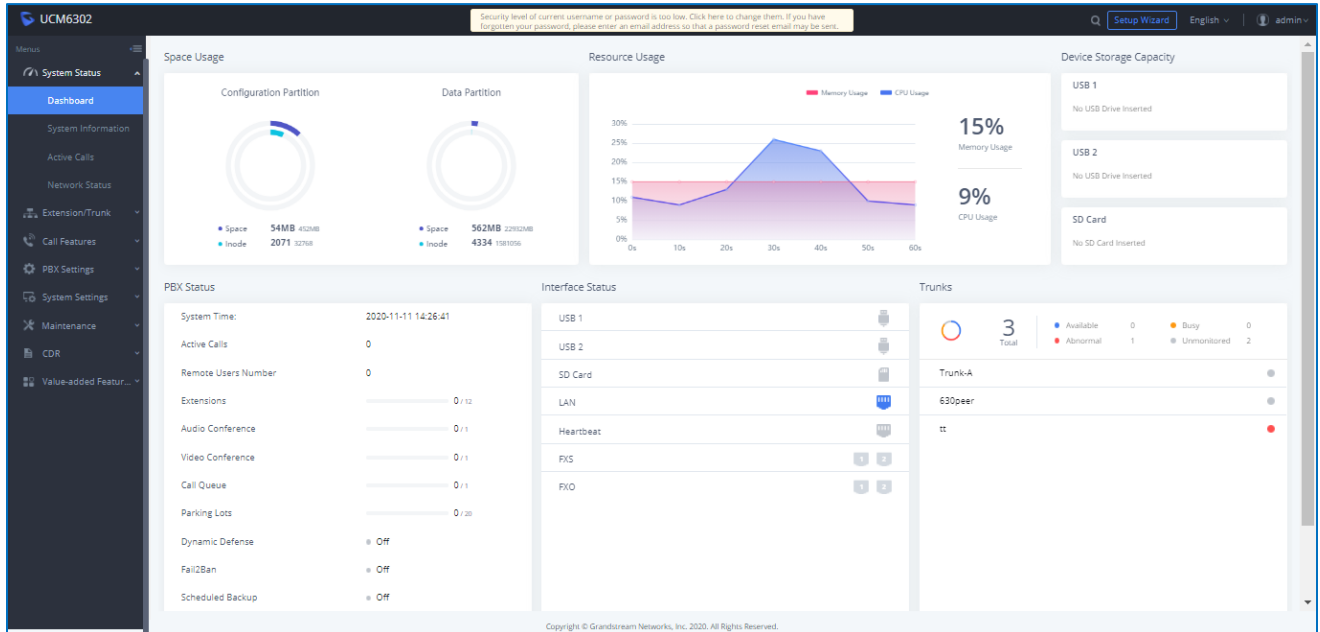


Figure 266: Status→PBX Status

## Trunks

Users could see all the configured trunk status in this section.

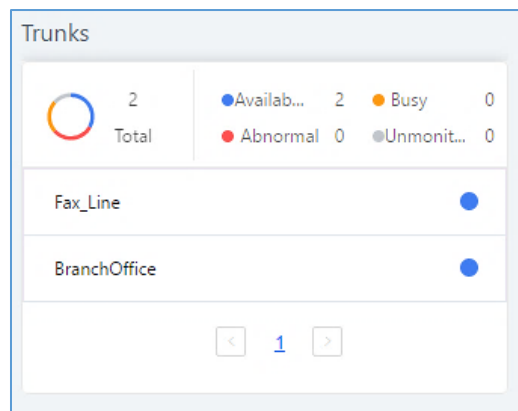


Figure 267: Trunk Status



**Table 132: Trunk Status**

<b>Status</b>	Display trunk status. <ul style="list-style-type: none"> <li><u>Analog trunk status:</u> <ul style="list-style-type: none"> <li><b>Available</b></li> <li><b>Busy</b></li> <li><b>Unavailable</b></li> <li><b>Unknown Error</b></li> </ul> </li> <li><u>SIP Peer trunk status:</u> <ul style="list-style-type: none"> <li><b>Unreachable:</b> The hostname cannot be reached.</li> <li><b>Unmonitored:</b> Heartbeat feature is not turned on to be monitored.</li> <li><b>Reachable:</b> The hostname can be reached.</li> </ul> </li> <li><u>SIP Register trunk status:</u> <ul style="list-style-type: none"> <li><b>Registered</b></li> <li><b>Unrecognized Trunk</b></li> </ul> </li> </ul>
<b>Trunks</b>	Display trunk name
<b>Type</b>	Display trunk Type: <ul style="list-style-type: none"> <li>Analog</li> <li>SIP</li> <li>IAX</li> </ul>
<b>Username</b>	Display username for this trunk.
<b>Port/Hostname/IP</b>	Display Port for analog trunk, or Hostname/IP for VoIP (SIP/IAX) trunk.



## Extensions

Extensions Status can be seen from the same configuration page, users can go under Web GUI → **Extension/Trunk** → **Extensions** and following page will be displayed listing the extensions and their status information.

<input type="checkbox"/>	STATUS	PRESENCE STA...	EXTENSION	NAME	TYPE	IP AND PORT	EMAIL...	OPTIONS
<input type="checkbox"/>	<span style="color: blue;">●</span> Ringing	Available	1000		SIP(WebRTC)	192.168.5.199:5070		
<input type="checkbox"/>	<span style="color: gray;">●</span> Unavailable	Available	1001		SIP(WebRTC)	--		
<input type="checkbox"/>	<span style="color: orange;">●</span> In Use	Available	5555		SIP(WebRTC)	192.168.5.199:63827		

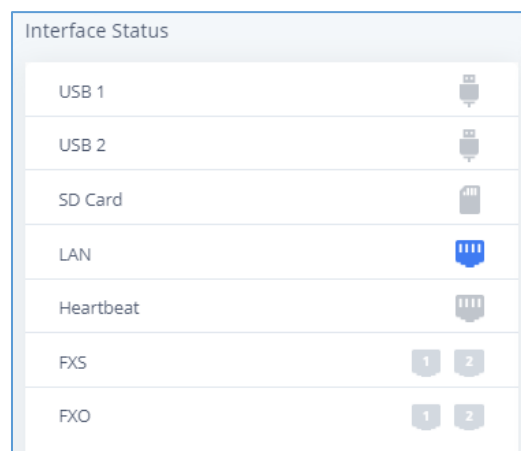
**Figure 268: Extension Status**


**Table 133: Extension Status**













<b>Status</b>	Display extension number (including feature code). The color indicator has the following definitions. <ul style="list-style-type: none"> <li>● <span style="color: green;">●</span> Green: Free</li> <li>● <span style="color: blue;">●</span> Blue: Ringing</li> <li>● <span style="color: yellow;">●</span> Yellow: In Use</li> <li>● <span style="color: grey;">●</span> Grey: Unavailable</li> </ul>
<b>Presence Status</b>	Display the presence status of the extension.
<b>Extension</b>	Display the extension number.
<b>Name</b>	First name and last name of the extension.
<b>IP and Port</b>	Display the IP and port number of the registered device.
<b>Email</b>	Display Email Notification status for the extension. When notification is waiting for be sent, shows  and once sent it will display 
<b>Terminal Type</b>	Displays extension type. <ul style="list-style-type: none"> <li>● SIP User</li> <li>● IAX User</li> <li>● Analog User</li> <li>● Ring Groups</li> <li>● Voicemail Groups</li> </ul>

## Interfaces Status

This section displays interface/port connection status on the UCM630xA. The following example shows the interface status for UCM6304A with USB, WAN port, FXS1, FXS2 and FXO1 connected.


**Figure 269: UCM6304A Interfaces Status**


**Table 134: Interface Status Indicators**

	USB connected.
	USB disconnected.
	SD Card connected.
	SD Card disconnected.
	LAN/WAN connected.
	LAN/WAN not configured.
	LAN/WAN disconnected.
	FXS/FXO connected.
	FXS/FXO waiting.
	FXS/FXO busy.
	FXS/FXO not configured.
	FXS/FXO disconnected.

## System Status

The UCM630xA system status can be accessed via Web GUI→**Status**→**System Status**, which displays the following system information.

### General

Under Web GUI→**System Status**→**System Information**→**General**, users could check the hardware and software information for the UCM630xA. Please see details in the following table.





Table 135: System Status→General

System Status →System Information→General	
<b>Model</b>	Product model.
<b>Part Number</b>	Product part number.
<b>System Time</b>	Current system time. The current system time is also available on the upper right of each web page.
<b>Up Time</b>	System up time since the last reboot.
<b>Boot</b>	Boot version.
<b>Core</b>	Core version.
<b>Base</b>	Base version.
<b>Program</b>	Program version. This is the main software release version.
<b>Recovery</b>	Recovery version.
<b>Lang</b>	Lang version
<b>GSWave</b>	GSWave version

## Network

Under Web GUI→**System Status**→**System Information**→**Network**, users could check the network information for the UCM630xA. Please see details in the following table.

Table 136: System Status→Network

System Status→System Status→Network	
<b>MAC Address</b>	Global unique ID of device, in HEX format. The MAC address can be found on the label coming with original box and on the label located on the bottom of the device.
<b>IPv4 Address</b>	IPv4 address.
<b>IPv6 Address Link</b>	IPv6 address
<b>Gateway</b>	Default gateway address.
<b>Subnet Mask</b>	Subnet mask address.
<b>DNS Server</b>	DNS Server address.
<b>Duplex Mode</b>	Duplex Mode
<b>Speed</b>	Speed



## Storage Usage

Users could access the storage usage information from Web GUI → **System Status** → **Dashboard** → **Storage Usage**. It shows the available and used space for Space Usage and Inode Usage.

Space Usage includes:

- **Configuration partition**  
This partition contains PBX system configuration files and service configuration files.
- **Data partition**  
Voicemail, recording files, IVR file, Music on Hold files etc.
- **USB disk**  
USB disk will display if connected.
- **SD Card**  
SD Card will display if connected.

Inode Usage includes:

- **Configuration partition**
- **Data partition**

**Note:**

Inode is the pointer used for file reference in the system. The system usually has limited resources of pointers



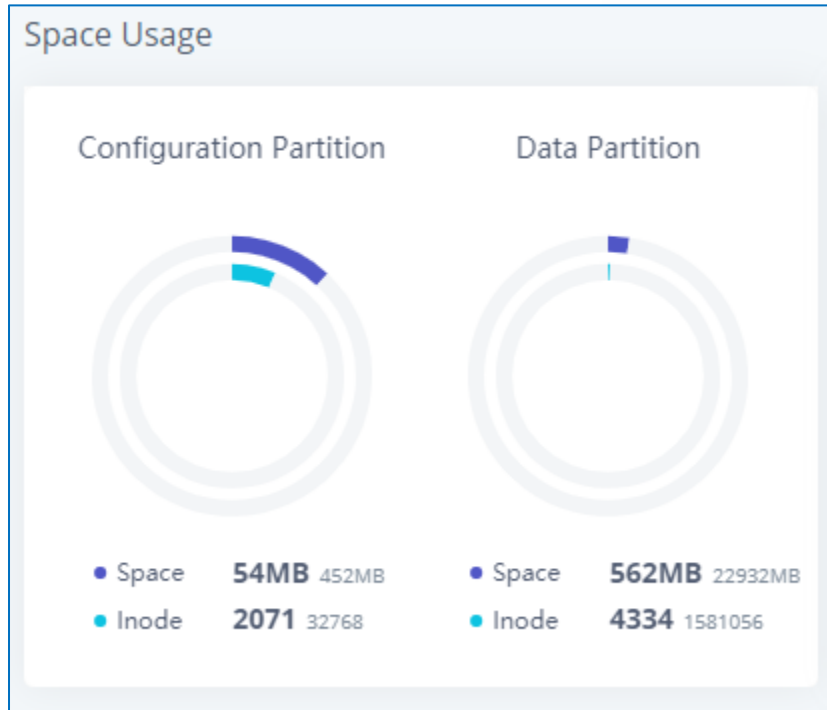


Figure 270: System Status→Storage Usage

## Resource Usage

When configuring and managing the UCM630xA, users could access resource usage information to estimate the current usage and allocate the resources accordingly. Under Web GUI → **System Status** → **Dashboard** → **Resource Usage**, the current CPU usage and Memory usage are shown in the pie chart.



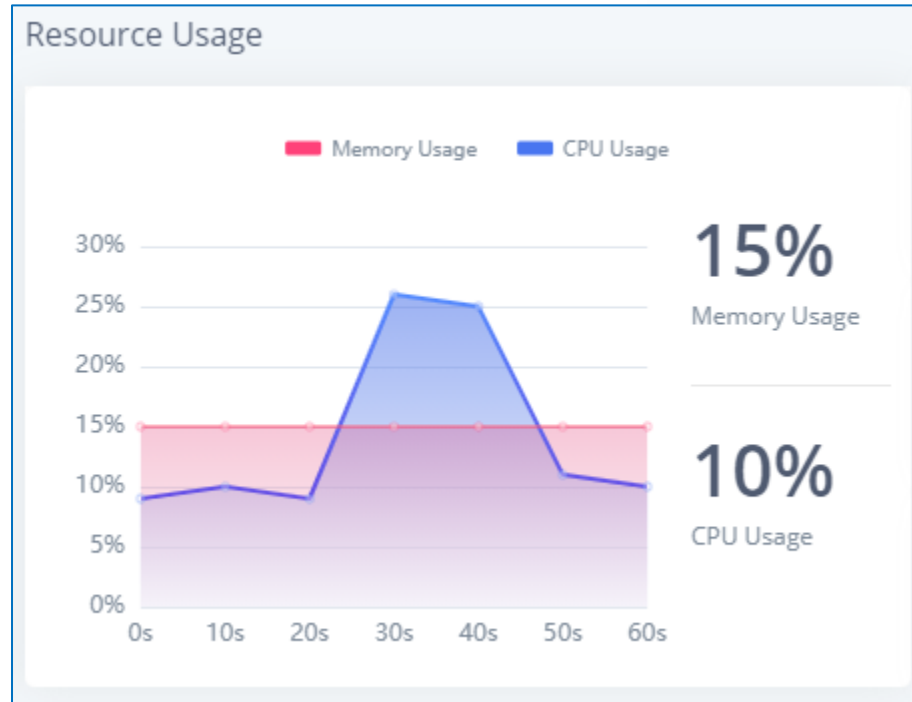


Figure 271: System Status→Resource Usage

## System Events

The UCM630xA can monitor important system events, log the alerts, and send Email notifications to the system administrator.

### Alert Events List

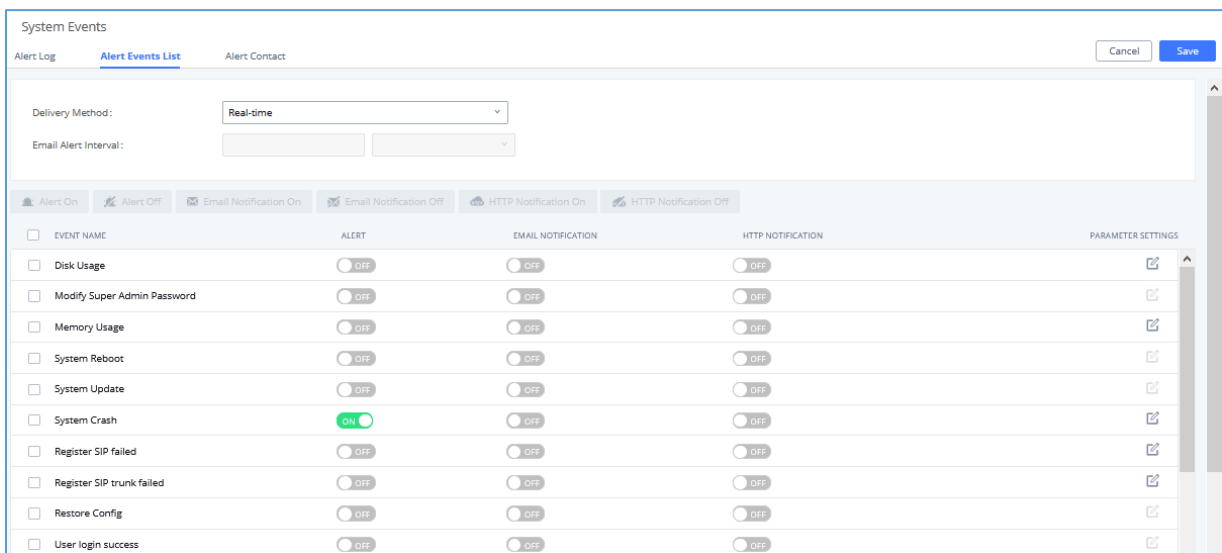
The system alert events list can be found under Web GUI → **Maintenance** → **System Events**. The following event and their actions are currently supported on the UCM630xA which will have alert and/or Email generated if occurred:

Table 137: Alert Events

Action index	Alert Events
1	Disk Usage
2	Modify Super Admin Password
3	Memory Usage
4	System Reboot
5	System Update
6	System Crash



7	Register SIP failed
8	Register SIP trunk failed
9	Restore Config
10	User login success
11	User login failed
12	SIP Internal Call Failure
13	SIP Outgoing Call through Trunk Failure
14	Fail2ban Blocking
15	SIP Lost Registration
16	SIP Peer Trunk Status
17	User Login Banned
18	HA failure warning
19	Emergency Calls
20	The CDR database is corrupted
21	NAS
22	Data Sync Backup
23	Remote Concurrent Calls
24	External Disk Status
25	CPU Usage Call Control



System Events

Alert Log | **Alert Events List** | Alert Contact

Delivery Method: Real-time

Email Alert Interval: [ ] [ ]

Alert On | Alert Off | Email Notification On | Email Notification Off | HTTP Notification On | HTTP Notification Off

EVENT NAME	ALERT	EMAIL NOTIFICATION	HTTP NOTIFICATION	PARAMETER SETTINGS
<input type="checkbox"/> Disk Usage	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	
<input type="checkbox"/> Modify Super Admin Password	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	
<input type="checkbox"/> Memory Usage	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	
<input type="checkbox"/> System Reboot	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	
<input type="checkbox"/> System Update	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	
<input type="checkbox"/> System Crash	<input checked="" type="radio"/> ON	<input type="radio"/> OFF	<input type="radio"/> OFF	
<input type="checkbox"/> Register SIP failed	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	
<input type="checkbox"/> Register SIP trunk failed	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	
<input type="checkbox"/> Restore Config	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	
<input type="checkbox"/> User login success	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	


**Figure 272: Alert Event List**



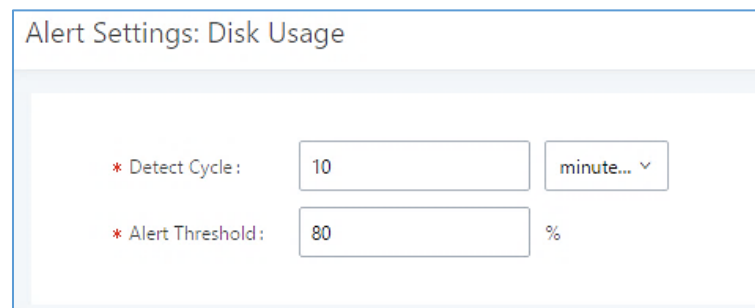
**Note:** For users who have purchased a GDMS package, once the option **Alert Events Sync** is enabled under **RemoteConnect**, the triggered events will be pushed to their GDMS platform.

For more information, please refer to:

[https://www.grandstream.com/hubfs/Product\\_Documentation/UCM630xA\\_RemoteConnect\\_User\\_Guide.pdf?sLang=en](https://www.grandstream.com/hubfs/Product_Documentation/UCM630xA_RemoteConnect_User_Guide.pdf?sLang=en)

Click on  to configure the parameters for each event. See examples below.

### 1. Disk Usage



Alert Settings: Disk Usage

\* Detect Cycle:  minute... ▾

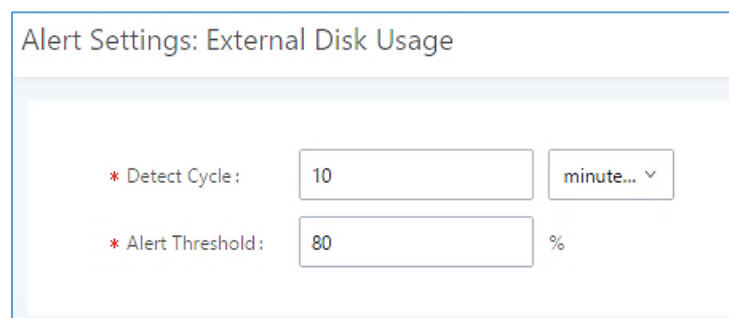
\* Alert Threshold:  %

**Figure 273: System Events→Alert Events Lists: Disk Usage**

- **Detect Cycle:** The UCM630xA will perform the internal disk usage detection based on this cycle. Users can enter the number and then select second(s)/minute(s)/hour(s)/day(s) to configure the cycle.
- **Alert Threshold:** If the detected value exceeds the threshold (in percentage), the UCM630xA system will send the alert.

**Note:** If the threshold is exceeded, any behavior of operating the disk will be rejected, including stopping file upload, IM writing, recording and CDR recording.

### 2. External Disk Usage



Alert Settings: External Disk Usage

\* Detect Cycle:  minute... ▾

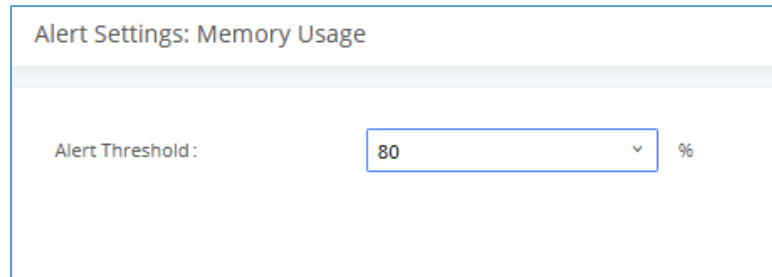
\* Alert Threshold:  %

**Figure 274: System Events→Alert Events Lists: External Disk Usage**



- **Detect Cycle:** The UCM630xA will perform the External disk usage detection based on this cycle. Users can enter the number and then select second(s)/minute(s)/hour(s)/day(s) to configure the cycle.
- **Alert Threshold:** If the detected value exceeds the threshold (in percentage), the UCM630xA system will send the alert.

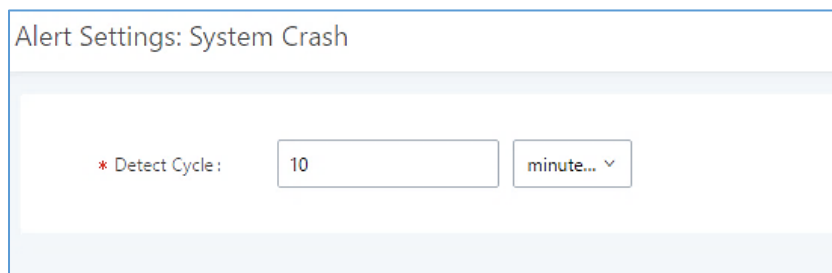
### 3. Memory Usage



**Figure 275: System Events→Alert Events Lists: Memory Usage**

- **Alert Threshold:** If the detected value exceeds the threshold (in percentage), the UCM630xA system will send the alert.

### 4. System Crash



**Figure 276: System Events→Alert Events Lists: System Crash**

- **Detect Cycle:** The UCM will detect the event at each cycle based on the specified time. Users can enter the number and then select second(s)/minute(s)/hour(s)/day(s) to configure the cycle.

Click on the switch  OFF  ON to turn on/off the alert and Email notification for the event. Users could also select the checkbox for each event and then click on button "Alert On", "Alert Off", "Email Notification On", "Email Notification Off" to control the alert and Email notification configuration.

### 5. Modify Super Admin Password

Once the super administrator password is modified, the system will record the password modification event in the alarm log.



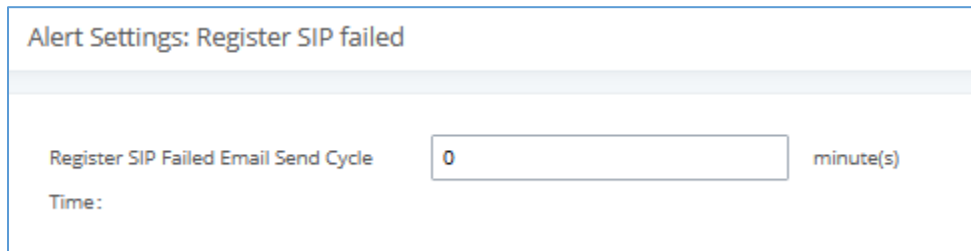
## 6. System Reboot

UCM will detect the system restart and will send an alert for it.

## 7. System Reboot

Once the system is upgraded, the system upgrade event will be recorded in the alarm log.

## 8. SIP registration failed

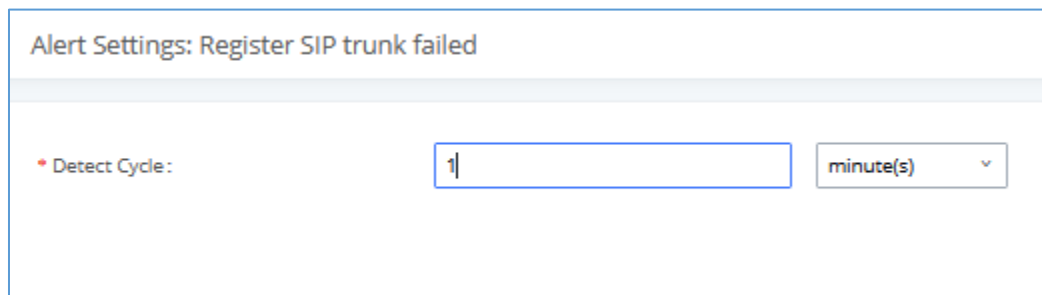


The screenshot shows a configuration window titled "Alert Settings: Register SIP failed". It contains a label "Register SIP Failed Email Send Cycle" followed by a text input field containing the number "0" and the unit "minute(s)". Below this is a label "Time:".

**Figure 277: System Events→Alert Events Lists: Register SIP Failed**

Configure the sending period of the SIP registration failure alert. The first registration failure alert of the same IP to the same SIP account will be sent immediately, and then no alerts will be sent for similar failure warnings in the cycle time. After the cycle time expires, an alert will be sent again to count the number of occurrences of similar SIP registration failure alerts during the cycle. When set to 0, alerts are always sent immediately.

## 9. Register SIP trunk failed



The screenshot shows a configuration window titled "Alert Settings: Register SIP trunk failed". It contains a label "Detect Cycle:" followed by a text input field containing the number "1" and a dropdown menu set to "minute(s)".

**Figure 278: System Events→Alert Events Lists: Register SIP Trunk Failed**

- **Detect Cycle:** The UCM will detect the failure of SIP trunk registration at a set interval. Users can enter the number and then select second(s)/minute(s)/hour(s)/day(s) to configure the cycle.





#### 10. Restore Config

Once the system configuration is restored, the configuration restoration event will be recorded in the alert log.

#### 11. User login success

Successful user login events will be recorded in the alert log.

#### 12. User login failed

User login failure events will be recorded in the alert log.

#### 13. SIP Internal Call Failure

If the system SIP extension call fails within the office, the event will be recorded in the alert log.

#### 14. SIP Outgoing Call through Trunk Failure

If the system SIP trunk outgoing call fails, the event will be recorded in the alert log.

#### 15. Fail2ban blocking

If the system Fail2ban is blocking, the event will be recorded in the alert log.

#### 16. SIP lost registration

If System SIP extension registration is lost, the event will be recorded in the alert log.

#### 17. SIP peer trunk status

If the SIP peer trunks status is abnormal, the event will be recorded in the alert log.

#### 18. User login banned

If user login is blocked, the event will be recorded in the alert log.

#### 19. Emergency Calls

If the system generates an emergency call, the event will be recorded in the alert log.

#### 20. NAS



If the system network disk is abnormal, the event will be recorded in the alarm log.

#### 21. Data Sync Backup

If the system performs data synchronization and backup abnormalities, the event will be recorded in the alert log.

#### 22. Remote concurrent calls

If the remote concurrent call fails, the event will be recorded in the alert log.

#### 23. External disk status

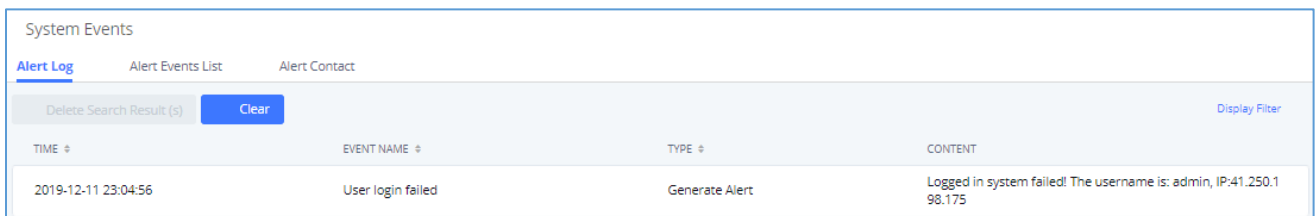
If the external disk of the system is Connected/Disconnected, the event will be recorded in the alarm log.

#### 24. CPU Flow control

The CPU flow control threshold is defined under **System Settings -> General Settings**, and the default value is 90%. When the traffic exceeds the predetermined value, the event will be recorded in the alert log and new calls will be prohibited.

### Alert Log

Under Web GUI→**Maintenance**→**System Events**→**Alert Log**, system messages from triggered system events are listed as alert logs. The following screenshot shows system crash alert logs.



TIME	EVENT NAME	TYPE	CONTENT
2019-12-11 23:04:56	User login failed	Generate Alert	Logged in system failed! The username is: admin, IP:41.250.198.175

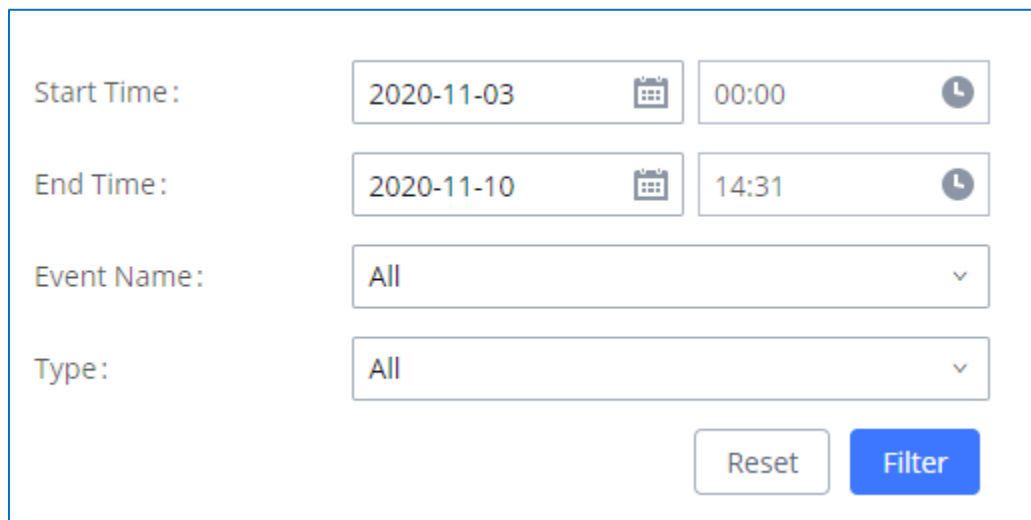
**Figure 279: System Events→Alert Log**

User could also filter alert logs by selecting a certain event category, type of alert log, and/or specifying a certain time period. The matching results will be displayed after clicking on **Filter**. Alert logs are classified into two types by the system:



1. **Generate Alert:** Generated when alert events happen, for example, alert logs for disk usage exceeding the alert threshold.
2. **Restore to Normal:** Generated when alert events being cleared, for example, logs for disk usage dropping back below the alert threshold.

User could filter out alert logs of “Generate Alert” or “Restore to Normal” by specifying the type according to need. The following figure shows an example of filtering out alert logs of type of “Restore to Normal”.



The screenshot shows a filter interface for alert logs. It includes the following fields and controls:

- Start Time:** A date field set to 2020-11-03 and a time field set to 00:00.
- End Time:** A date field set to 2020-11-10 and a time field set to 14:31.
- Event Name:** A dropdown menu currently set to 'All'.
- Type:** A dropdown menu currently set to 'All'.
- Buttons:** A 'Reset' button and a blue 'Filter' button.

**Figure 280: Filter for Alert Log**

## Alert Contact

This feature allows the administrator to be notified when one of the Alert events mentioned above happens. Users could add administrator's Email address under Web GUI→**Maintenance**→**System Events**→**Alert Contact** to send the alert notification to an email (Up to 10 Email addresses can be added) or also specify an HTTP server where to send this alert.

**Table 138: Alert Contact**

<b>Super Admin Email</b>	Configure the email addresses to send alert notifications to. Up to 10 email addresses can be added.
<b>Admin Email</b>	Configure the email addresses to send alert notifications to. Up to 10 email addresses can be added.
<b>Email Template</b>	Please refer to section <b>Email Templates</b>



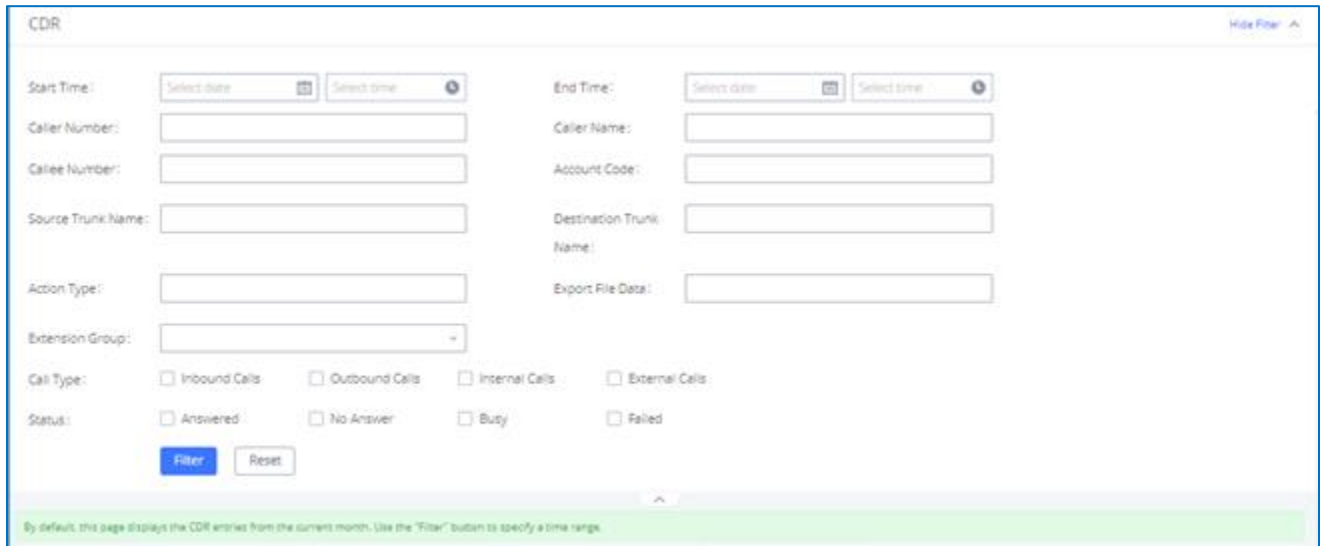
<b>Protocol</b>	Protocol used to communicate with the server. HTTP or HTTPS. Default one is <b>HTTP</b> .
<b>HTTP Server</b>	The IP address or FQDN of the HTTP/HTTPS server.
<b>HTTP Server Port</b>	HTTP/HTTPS port
<b>Warning Template</b>	Customize the template used for system warnings. By default: <code>{"action":"\${ACTION}","mac":"\${MAC}","content":"\${WARNING_MSG}"}</code>
<b>Notification Template</b>	Customize the notification template to receive relevant alert information. By default: <code>{"action":"\${ACTION}","cpu":"\${CPU_USED}","memory":"\${MEM_USED}","disk":"\${DISK_USED}","external_disk":"\${EXTERNAL_DISK_USED}"}</code> <b>Note:</b> The notification message with <b>"action:0"</b> will be sent periodically if Notification Interval is set.
<b>Notification Interval</b>	Modifies the frequency at which notifications are sent in seconds. No notifications will be sent if the value is "0". Default value: <b>20</b>
<b>Template Variables</b>	<code>\${MAC}</code> : MAC Address <code>\${WARNING_MSG}</code> : Warning message <code>\${TIME}</code> : Current System Time <code>\${CPU_USED}</code> : CPU Usage <code>\${MEM_USED}</code> : Memory Usage <code>\${ACTION}</code> : Message Type. Refer to [Table 137: Alert Events] <code>\${DISK_USED}</code> : Disk Usage <code>\${EXTERNAL_DISK_USED}</code> : Disk Usage

## CDR

CDR (Call Detail Record) is a data record generated by the PBX that contains attributes specific to a single instance of phone call handled by the PBX. It has several data fields to provide detailed description for the call, such as phone number of the calling party, phone number of the receiving party, start time, call duration, etc.

On the UCM630xA, the CDR can be accessed under Web GUI → **CDR** → **CDR**. Users could filter the call report by specifying the date range and criteria, depending on how the users would like to include the logs to the report. Click on "Filter" button to display the generated report.





The screenshot shows a 'CDR' filter interface with the following fields and options:

- Start Time:** Select date and Select time
- End Time:** Select date and Select time
- Caller Number:** Text input field
- Caller Name:** Text input field
- Callee Number:** Text input field
- Account Code:** Text input field
- Source Trunk Name:** Text input field
- Destination Trunk Name:** Text input field
- Action Type:** Text input field
- Export File Data:** Text input field
- Extension Group:** Dropdown menu
- Call Type:**
  - Inbound Calls
  - Outbound Calls
  - Internal Calls
  - External Calls
- Status:**
  - Answered
  - No Answer
  - Busy
  - Failed

Buttons: Filter, Reset

Footer note: By default, this page displays the CDR entries from the current month. Use the "Filter" button to specify a time range.

**Figure 281: CDR Filter**

**Table 139: CDR Filter Criteria**

<b>Call Type</b>	Groups the following: <ul style="list-style-type: none"> <li>• <b>Inbound calls:</b> Inbound calls are calls originated from a non-internal source (like a VoIP trunk) and sent to an internal extension.</li> <li>• <b>Outbound calls:</b> Outbound calls are calls sent to a non-internal source (like a VoIP trunk) from an internal extension.</li> <li>• <b>Internal calls:</b> Internal calls are calls from one internal extension to another extension, which are not sent over a trunk.</li> <li>• <b>External calls:</b> External calls are calls sent from one trunk to another trunk, which are not sent to any internal extension.</li> </ul>
<b>Status</b>	Filter with the call status, the available statuses are the following: <ul style="list-style-type: none"> <li>• Answered</li> <li>• No Answer</li> <li>• Busy</li> <li>• Failed</li> </ul>
<b>Source Trunk Name</b>	Select source trunk(s) and the CDR of calls going through inbound the trunk(s) will be filtered out.
<b>Destination Trunk Name</b>	Select destination trunk(s) and the CDR of calls going outbound through the trunk(s) will be filtered out.
<b>Action Type</b>	Filter calls using the Action Type, the following actions are available:



	<ul style="list-style-type: none"> <li>• Announce</li> <li>• Announcement page</li> <li>• Dial</li> <li>• Announcements</li> <li>• Callback</li> <li>• Call Forward</li> <li>• Meeting</li> <li>• DISA</li> <li>• Follow Me</li> <li>• IVR</li> <li>• Page</li> <li>• Parked Call</li> <li>• Queue</li> <li>• Ring Group</li> <li>• Transfer</li> <li>• VM</li> <li>• VMG</li> <li>• VQ_Callback</li> <li>• Wakeup</li> <li>• Emergency Call</li> <li>• Emergency Notify</li> <li>• SCA</li> </ul>
<b>Extension Group</b>	Specify the Extension Group name to filter with.
<b>Export File Data</b>	<p>Select the fields that will be exported, the following fields are available:</p> <ul style="list-style-type: none"> <li>• Account Code</li> <li>• Session</li> <li>• Premier caller</li> <li>• Action type</li> <li>• Source trunk name</li> <li>• Destination trunk name</li> <li>• Caller number</li> <li>• Caller ID</li> <li>• Caller name</li> <li>• Callee number</li> <li>• Answer by</li> <li>• Context</li> <li>• Start time</li> <li>• Answer time</li> </ul>



	<ul style="list-style-type: none"> <li>• End time</li> <li>• Call time</li> <li>• Talk time</li> <li>• Source channel</li> <li>• Dest channel</li> <li>• Call status</li> <li>• Dest channel extension</li> <li>• Last app</li> <li>• Last data</li> <li>• AMAFLAGS</li> <li>• UIQUEID</li> <li>• Call type</li> <li>• NAT</li> </ul>
<b>Account Code</b>	Select the account Code to filter with. If pin group CDR is enabled, the call with pin group information will be displayed as part of the CDR under Account Code Field.
<b>Start Time</b>	Specify the start time to filter the CDR report. Click on the calendar icon on the right and the calendar will show for users to select the exact date and time.
<b>End Time</b>	Specify the end time to filter the CDR report. Click on the calendar icon on the right and the calendar will show for users to select the exact date and time.
<b>Caller Number</b>	<p>Enter the caller number to filter the CDR report. CDR with the matching caller number will be filtered out.</p> <p>User could specify a particular caller number or enter a pattern. '.' matches zero or more characters, only appears in the end. 'X' matches any digit from 0 to 9, case-insensitive, repeatable, only appears in the end.</p> <p><u>For example:</u></p> <p><b>3XXX:</b> It will filter out CDR that having caller number with leading digit 3 and of 4 digits' length.</p> <p><b>3.:</b> It will filter out CDR that having caller number with leading digit 3 and of any length.</p>
<b>Caller Name</b>	Enter the caller name to filter the CDR report. CDR with the matching caller name will be filtered out.
<b>Callee Number</b>	<p>Enter the callee number to filter the CDR report. CDR with the matching callee number will be filtered out.</p> <p><b>Note:</b> The "Callee Number" filter field supports specifying Pattern (example: 3XXX) or using Leading digits (example: 3.) as filtering options.</p>

The call report will display as the following figure shows.



STATUS	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	TALK TIME	ACCOUNT CODE	RECORDING FILE OPTIONS
	5555	1000	DIAL	2019-12-11 09:53:03	0:00:11	0:00:06		-
STATUS	PREMIER CALLER	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	ACCOUNT CODE	RECORDING FILE OPTIONS
	5555	5555	1000	DIAL	2019-12-11 09:53:03	0:00:11		-

**Figure 282: Call Report**

The CDR report has the following data fields:

- **Start Time**  
Format: 2019-12-11 09:53:03
- **Action Type**  
Example:  
IVR  
DIAL  
WAKEUP
- **Call From**  
Example format: 5555
- **Call To**  
Example format: 1000
- **Call Time**  
Format: 0:00:11
- **Talk Time**  
Format: 0:00:06
- **Account Code**  
Example format:  
Grandstream/Test
- **Status**  
Answered, Busy, No answer or Failed.

Users could perform the following operations on the call report.

- **Sort by "Start Time"**  
Click on the header of the column to sort the report by "Start Time". Clicking on "Start Time" again will reverse the order.





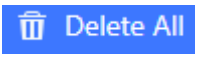
- **Download Searched Results**

Click on “Download Search Result(s)” to export the records filtered out to a .csv file.

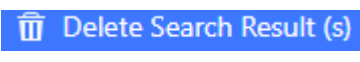
- **Download All Records**

Click on “Download All Records” to export all the records to a .csv file.

- **Delete All**

Click on  button to remove all the call report information.




- **Delete Search Result**

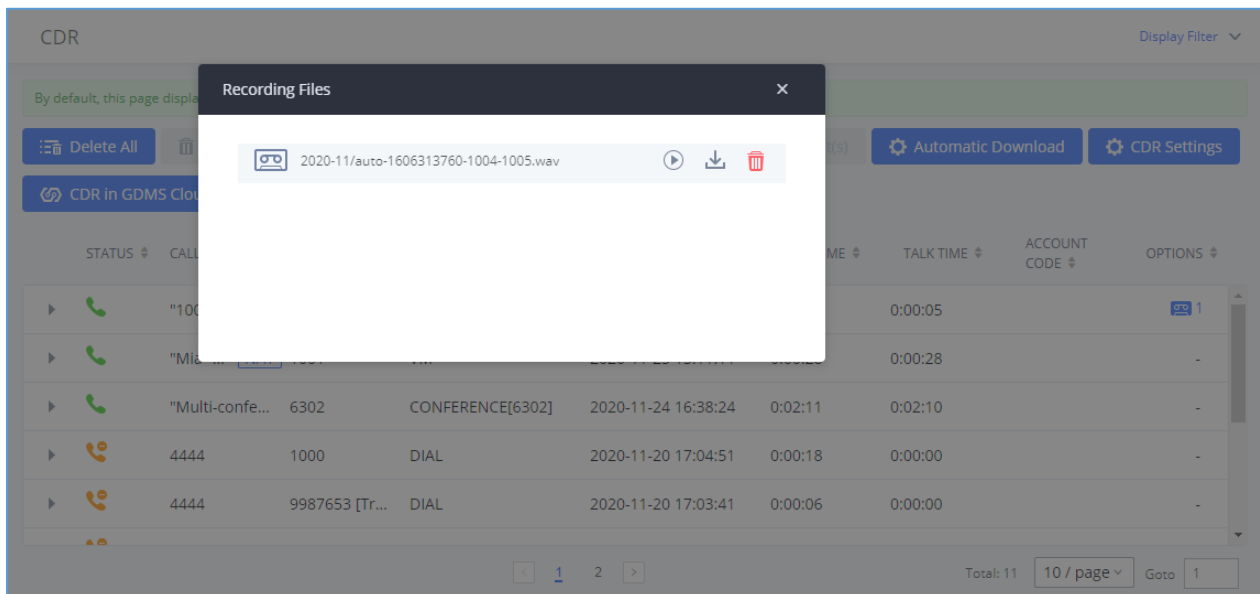
On the bottom of the page, click on  button to remove CDR records that appear on search results.

**Note:** When deleting CDR, a prompt will now appear asking whether to delete all recording files or not.

- **Play/Download/Delete Recording File (per entry)**

If the entry has audio recording file for the call, the three icons on the rightest column will be activated for users to select. In the following picture, the second entry has audio recording file for the call.

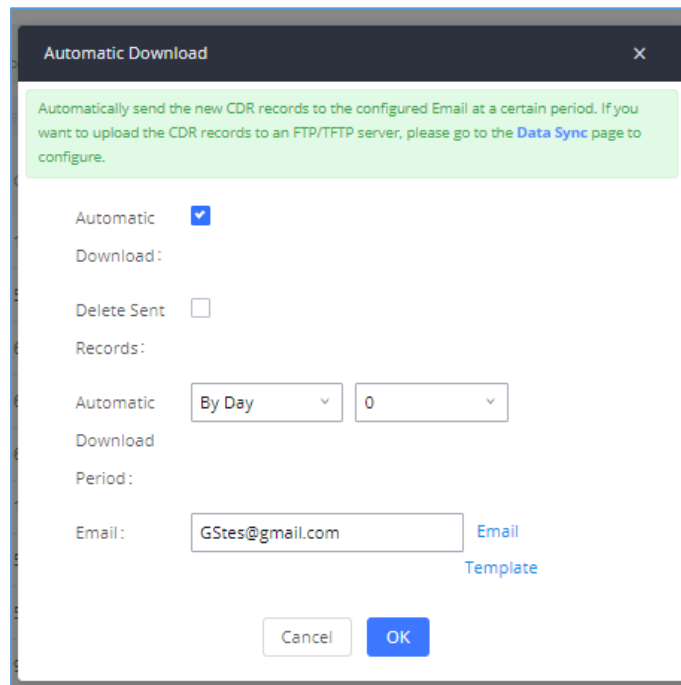
Click on  to play the recording file; click on  to download the recording file in .wav format; click on  to delete the recording file (the call record entry will not be deleted).



**Figure 283: Call Report Entry with Audio Recording File**

- **Automatic Download CDR Records**

User could configure the UCM630xA to automatically download the CDR records and send the records to multiple Email recipients in a specific hour. Click on “Automatic Download Settings” and configure the parameters in the dialog below.



**Figure 284: Automatic Download Settings**

To receive CDR record automatically from Email, check “Enable” and select a time period “By Day” “By Week” or “By Month”, select Hour of the day as well for the automatic download period. Make sure you have entered an Email or multiple email addresses where to receive the CDR records.

**Note:** users have the option to delete the sent records “Delete Sent Records”

Starting from UCM630xA firmware 1.0.10.x, transferred call will no longer be displayed as a separate call entry in CDR. It will display within call record in the same entry. CDR new features can be found under Web GUI→CDR→CDR. The user can click on the option icon for a specific call log entry to view details about this entry, such as premier caller and transferred call information.

STATUS	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	TALK TIME	ACCOUNT C ODE	RECORDING FILE OPTIONS
	5555	1000	DIAL	2019-12-11 09:53:03	0:00:11	0:00:06		-

**Figure 285: CDR Report**



STATUS	PREMIER CALLER	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	TALK TIME	ACCOUNT CODE	RECORDING FILE OPTIONS
	1000	"abllll lolo" 1000...	9985632	DIAL	2019-12-10 03:23:14	0:00:00	0:00:00		-
	1000	"abllll lolo" 1000...	6500	QUEUE[6500]	2019-12-10 03:23:14	0:00:00	0:00:00		1
	1000	"abllll lolo" 1000...	5555	QUEUE[6500]	2019-12-10 03:23:14	0:00:13	0:00:07		-

**Figure 286: Detailed CDR Information**

## Downloaded CDR File

The downloaded CDR (.csv file) has different format from the Web GUI CDR. Here are some descriptions.

- **Caller number, Callee number**

"Caller number": the caller ID.

"Callee number": the callee ID.

If the "Source Channel" contains "DAHDI", this means the call is from FXO/PSTN line.

caller number	callee number	context	calerid	source channel	dest channel	lastapp
	2009	from-internal	"Wake Up Call" <WakeUp>	Local/2009@from-internal-00000001;2	PJSIP/2009-00000013	Dial
2007	31100	from-internal	"" <2007>	PJSIP/2007-00000014	DAHDI/1-1	Dial
2009	1100	from-internal	"John Doe" <2009>	PJSIP/2009-00000015	PJSIP/trunk_1-00000016	Dial
1100	2014	from-did-direct	"1100" <1100>	DAHDI/1-1	PJSIP/2014-00000017	Dial

**Figure 287: Downloaded CDR File Sample**

- **Context**

There are different context values that might show up in the downloaded CDR file. The actual value can vary case by case. Here are some sample values and their descriptions.

**from-internal:** internal extension makes outbound calls.

**ext-did-XXXXX:** inbound calls. It starts with "ext-did", and "XXXXX" content varies case by case, which also relate to the order when the trunk is created.

**ext-local:** internal calls between local extensions.

- **Source Channel, Dest Channel**

**Sample 1:**

caller number	callee number	context	calerid	source channel	dest channel	disposition
2007	31100	from-internal	"" <2007>	PJSIP/2007-00000014	DAHDI/1-1	ANSWERED

**Figure 288: Downloaded CDR File Sample - Source Channel and Dest Channel 1**

DAHDI means it is an analog call, FXO or FXS.

For UCM6302A, DAHDI/(1-2) are FXO ports, and DAHDI(3-4) are FXS ports.

For UCM6304A, DAHDI/(1-4) are FXO ports, and DAHDI(5-6) are FXS ports.

For UCM6308A, DAHDI/(1-8) are FXO ports, and DAHDI(9-10) are FXS ports.



**Sample 2:**

caller number	callee number	context	calerid	source channel	dest channel	lastapp
2009	1100	from-internal	"John Doe" <2009>	PJSIP/2009-00000015	PJSIP/trunk_1-00000016	Dial

**Figure 289: Downloaded CDR File Sample - Source Channel and Dest Channel 2**

"SIP" means it is a SIP call. There are three format:

(a) **PJSIP/NUM-XXXXXX**, where NUM is the local SIP extension number. The last XXXXX is a random string and can be ignored.

(c) **PJSIP/trunk\_X/NUM**, where trunk\_X is the internal trunk name, and NUM is the number to dial out through the trunk.

(c) **PJSIP/trunk\_X-XXXXXX**, where trunk\_X is the internal trunk name and it is an inbound call from this trunk. The last XXXXX is a random string and can be ignored.

There are some other values, but these values are the application name which are used by the dialplan.

**IAX2/NUM-XXXXXXX**: it means this is an IAX call.

**Local/@from-internal-XXXXX**: it is used internally to do some special feature procedure. We can simply ignore it.

**Hangup**: the call is hung up from the dialplan. This indicates there are some errors or it has run into abnormal cases.

**Playback**: play some prompts to you, such as 183 response or run into an IVR.

**ReadExten**: collect numbers from user. It may occur when you input PIN codes or run into DISA

**Note**: The language of column titles in exported CDR reports and statistics reports will be based on the UCM's display language

### CDR Export Customization

Users can select the data they want to see in exported CDR reports by first clicking on the *Filter* button on the CDR page under **CDR→CDR** and selecting the desired information in the *Export File Data* field.



**Figure 290: CDR Export File data**

### CDR in GDMS Cloud

Cloud Storage for CDR Record which can be displayed under **CDR → CDR in GDMS Cloud**.

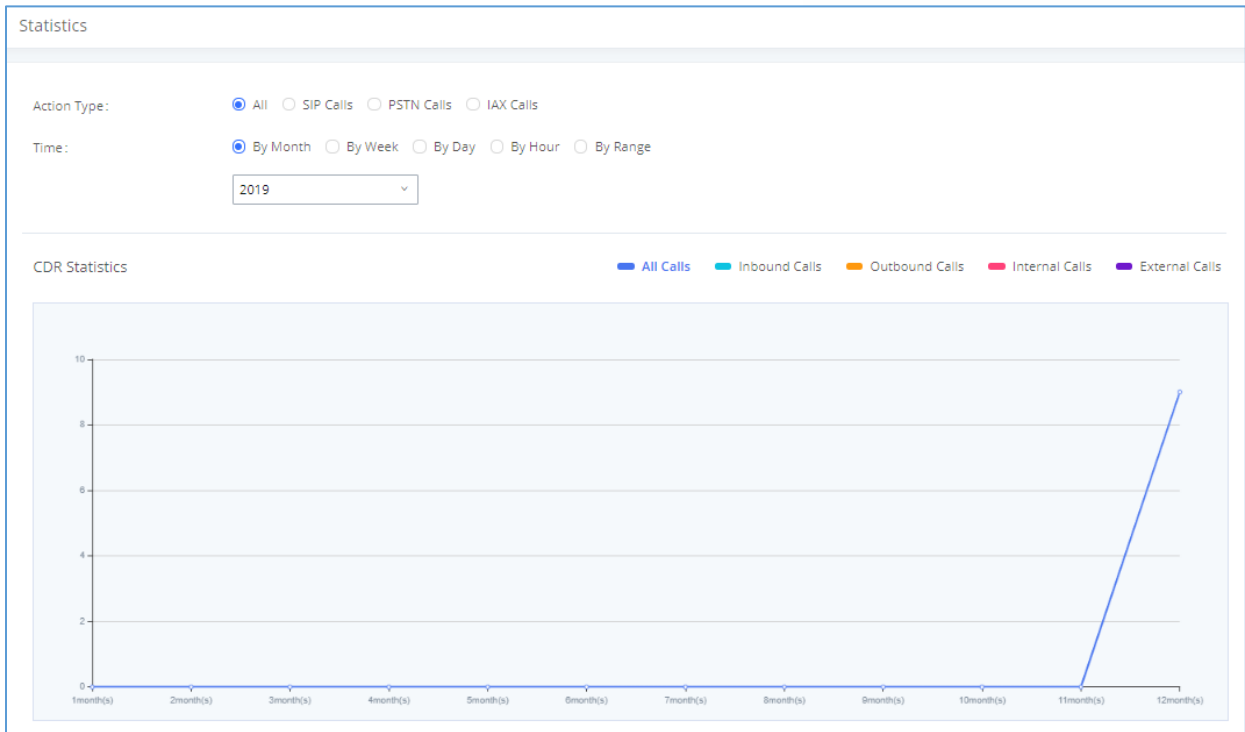
NAME	DATE	SIZE	OPTIONS
No Data			

**Figure 291: CDR in GDMS Cloud**

### Statistics

CDR Statistics is an additional feature on the UCM630xA which provides users a visual overview of the call report across the time frame. Users can filter with different criteria to generate the statistics chart.





**Figure 292: CDR Statistics**

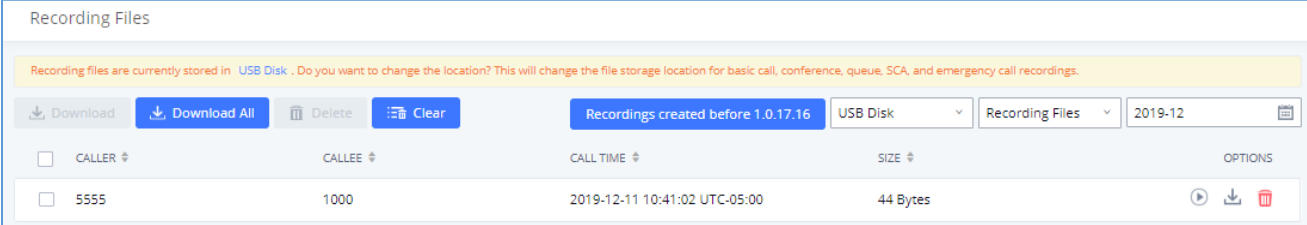
**Table 140: CDR Statistics Filter Criteria**

<b>Trunk Type</b>	Select one of the following trunk type. <ul style="list-style-type: none"> <li>All</li> <li>SIP Calls</li> <li>PSTN Calls</li> </ul>
<b>Call Type</b>	Select one or more in the following checkboxes. <ul style="list-style-type: none"> <li>Inbound calls</li> <li>Outbound calls</li> <li>Internal calls</li> <li>External calls</li> <li>All calls</li> </ul>
<b>Time Range</b>	<ul style="list-style-type: none"> <li>By month (of the selected year).</li> <li>By week (of the selected year).</li> <li>By day (of the specified month for the year).</li> <li>By hour (of the specified date).</li> <li>By range. For example, 2016-01 To 2016-03.</li> </ul>





## Recording Files

This page lists all the recording files recorded by "Auto Record" per extension/ring group/call queue/trunk, or via feature code "Audio Mix Record". If external storage device is plugged in, for example, SD card or USB drive, the files are stored on the external storage. Otherwise, internal storage will be used on the UCM630xA.



<input type="checkbox"/>	CALLER ↕	CALLEE ↕	CALL TIME ↕	SIZE ↕	OPTIONS
<input type="checkbox"/>	5555	1000	2019-12-11 10:41:02 UTC-05:00	44 Bytes	

Figure 293: CDR→Recording Files

- Click on **"Delete Selected Recording Files"** to delete the recording files.
- Click on **"Delete All Recording Files"** to delete all recording files.
- Click on **"Batch Download Recording Files"** in order to download the selected recording files.
- Click on **"Download All Recording Files"** to download all recordings files.
- Select Either **"USB Disk"** or **"Local"** to show recording files stored on external or internal storage, depending on selected storage space.
- Select whether to show call recordings, queue recordings or meeting recordings.
- Click on  to download the recording file in .wav format.
- Click on  to delete the recording file.
- To sort the recording file, click on the title "Caller", "Callee" or "Call Time" for the corresponding column. Click on the title again can switch the sorting mode between ascending order or descending order.

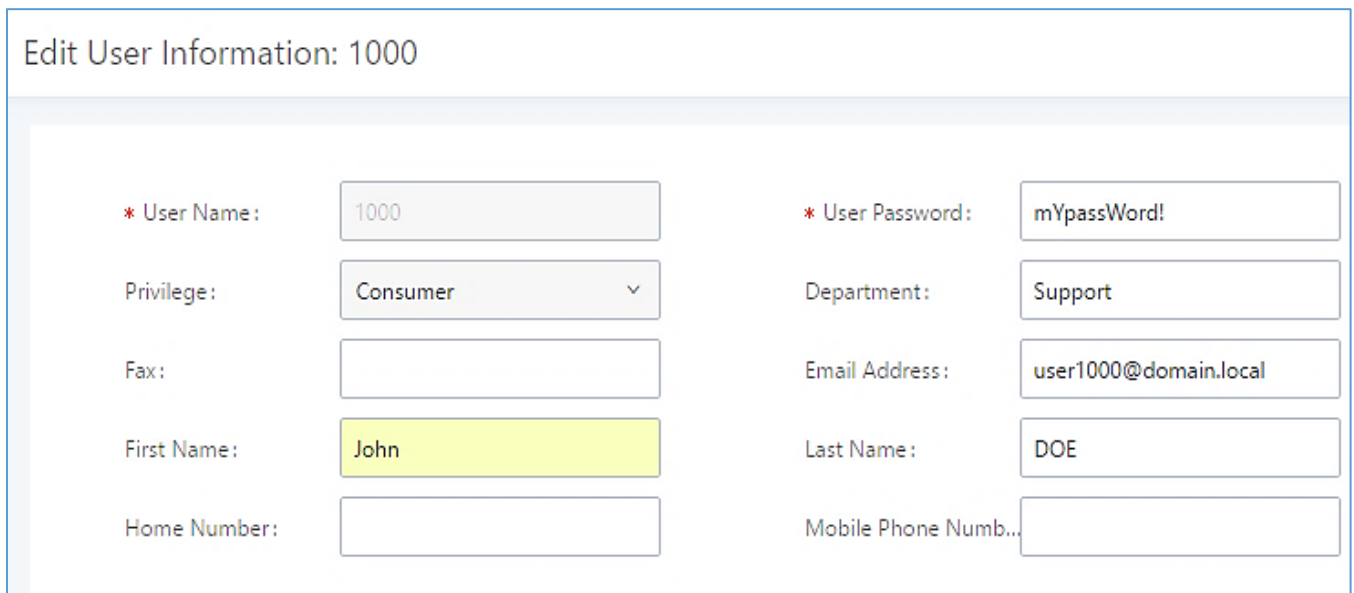


## USER PORTAL

Users could log into their web GUI portal using the extension number and user password. When an extension is created in the UCM630xA, the corresponding user account for the extension is automatically created. The user portal allows access to a variety of features which include user information, extension configuration and CDR as well as settings and managing value-added features like Call Queue, Wakeup Service and CRM.

Users also can access their personal data files (call recordings, Voicemail Prompts ...).

The login credentials are configured by Super Admin. The following figure shows the dialog of editing the account information by Super Admin. The Username must be the extension number and it is not configurable, and the password is set on "User Password" field and it should not be confused with the SIP extension password.



**Edit User Information: 1000**

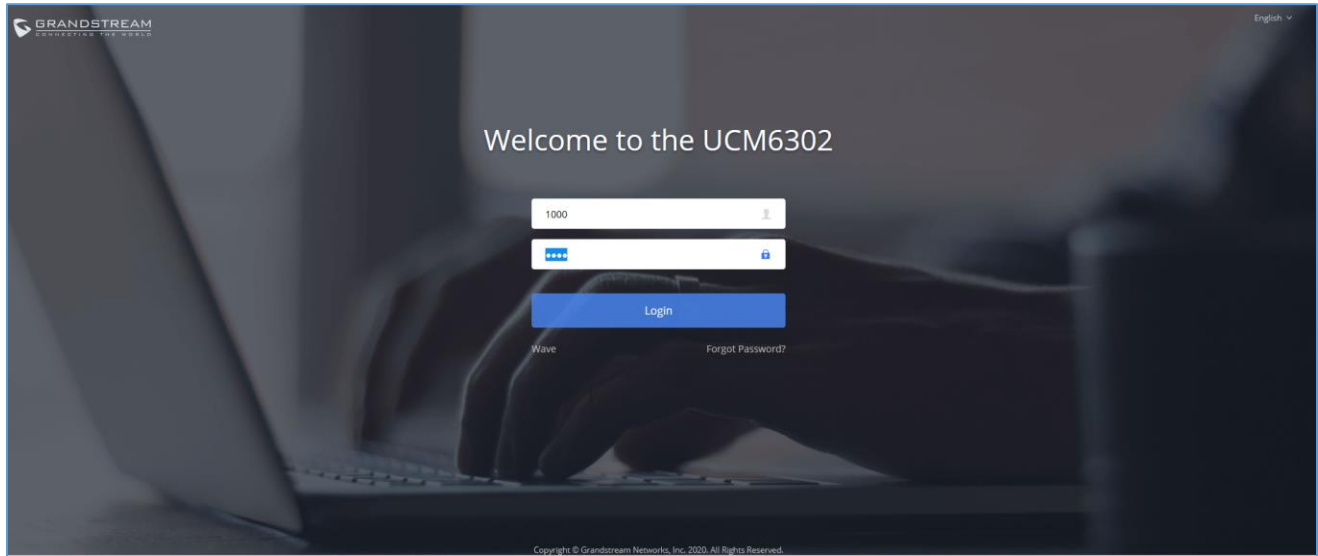
* User Name:	<input type="text" value="1000"/>	* User Password:	<input type="text" value="mYpassWord!"/>
Privilege:	<input type="text" value="Consumer"/>	Department:	<input type="text" value="Support"/>
Fax:	<input type="text"/>	Email Address:	<input type="text" value="user1000@domain.local"/>
First Name:	<input type="text" value="John"/>	Last Name:	<input type="text" value="DOE"/>
Home Number:	<input type="text"/>	Mobile Phone Numb...:	<input type="text"/>

**Figure 294: Edit User Information by Super Admin**

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

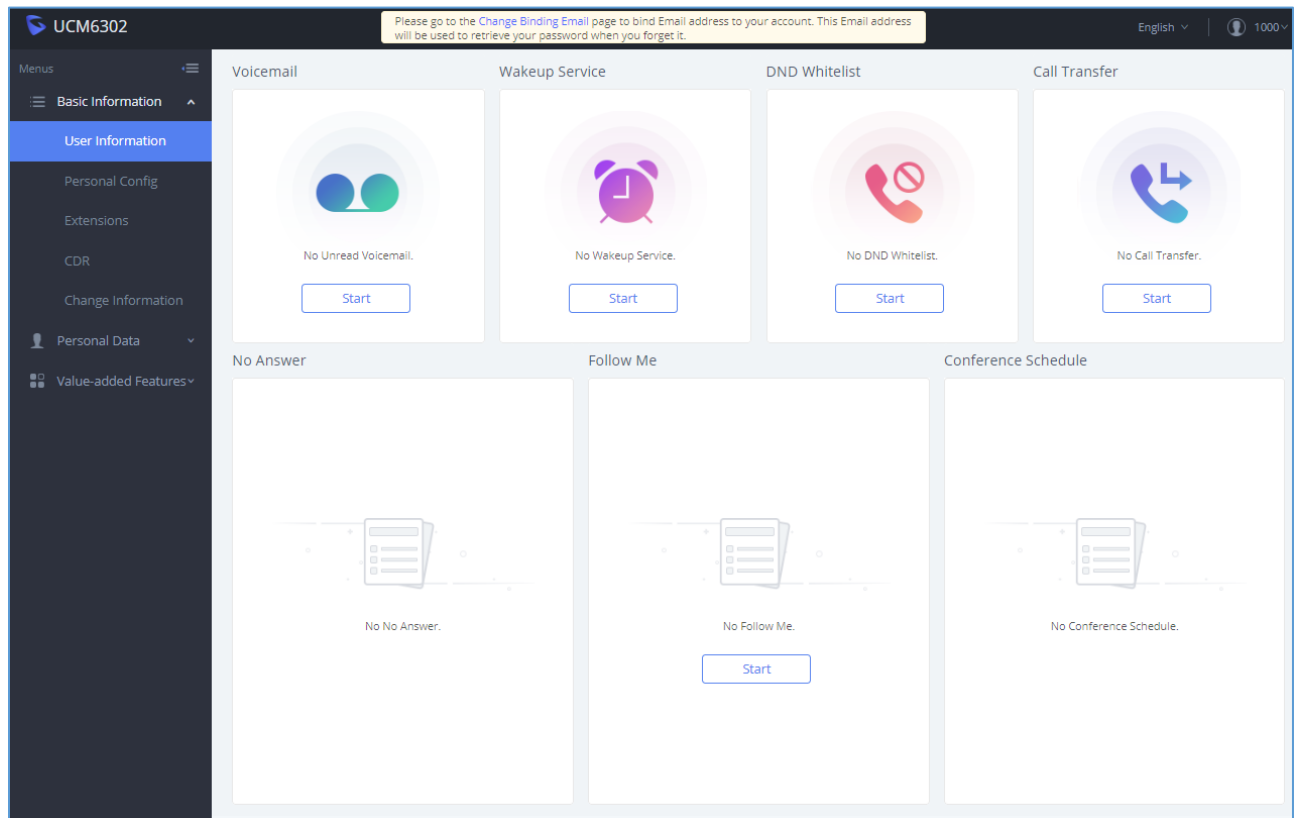






**Figure 295: User Portal Login**

After login, the Web GUI display is shown as below.



**Figure 296: User Portal Layout**

After successful login, the user has the following three configuration tabs:



## Basic Information

Under this menu, the user can configure and change his/her personal information including (first name, last name, password, email address, department...). And they can also set and activate their extension features (presence status, call forward, DND ....) to be reflected on the UCM.

Also, the user can see from this menu the Call Details Records and search for specific ones along with the possibility to download the records on CSV format for later usage.

## Personal Data

Under this section, the user can access and manage their personal data files which includes (voicemail files, call recordings ...) along with the possibility to set Follow me feature to without requesting the Super admin to set the feature from admin account.

## Value-added Features

On this section, the user has access to manage and use all rich value-added features which includes.

- + If user is a member of call queue, they can check the queue's activity from the "Call Queue" section.
- + Create and enable Wake Up service.
- + Enable and configure CRM connection to either SugarCRM or Salesforce.

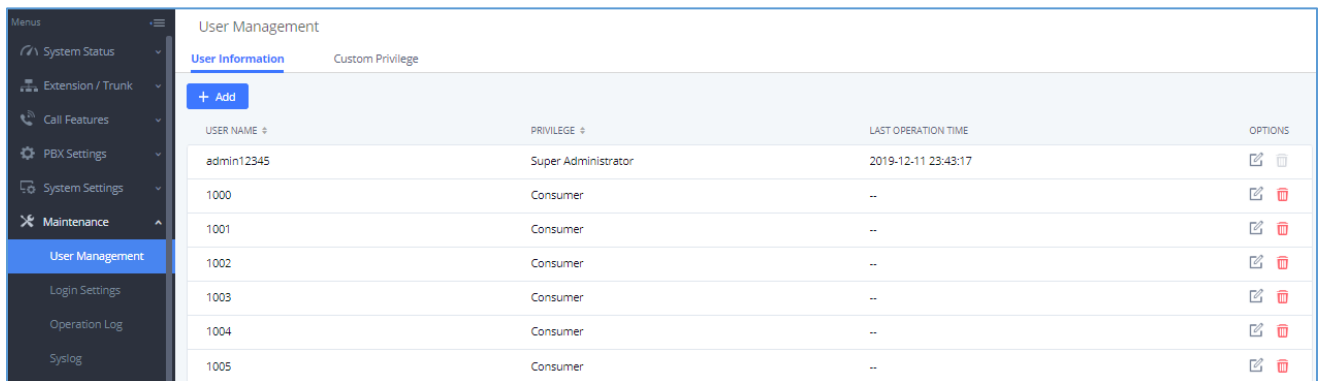
For the configuration parameter information in each page, please refer to **[Table 141: User Management→Create New User]** for options in **User Portal→Basic Information→User Information** page; please refer to **[EXTENSIONS]** for options in **User Portal→Basic Information→Extension** page; please refer to **[CDR]** for **User Portal→Basic Information→CDR** page.



# MAINTENANCE

## User Management

User management is on Web GUI → **Maintenance** → **User Management** page. User could create multiple accounts for different administrators to log in the UCM630xA Web GUI. Additionally, the system will automatically create user accounts along with creating new extensions for extension users to login to the Web GUI using their extension number and password. All existing user accounts for Web GUI login will be displayed on User Management page as shown in the following figure.

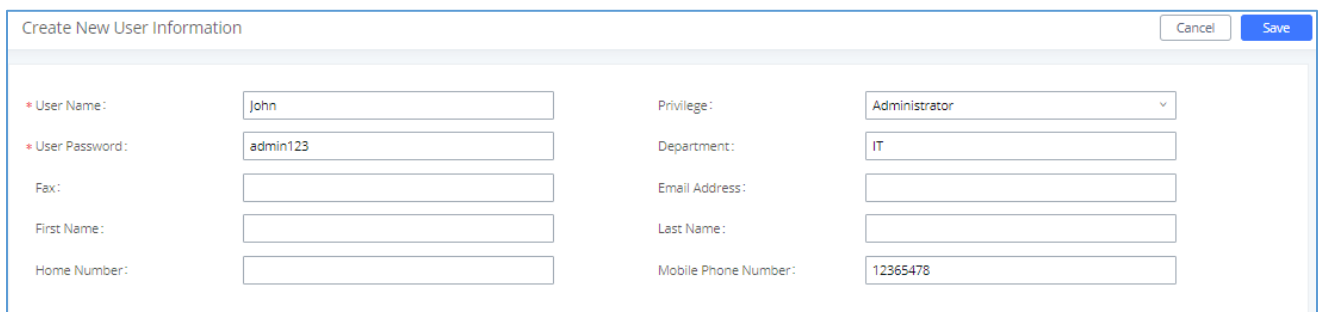


USER NAME	PRIVILEGE	LAST OPERATION TIME	OPTIONS
admin12345	Super Administrator	2019-12-11 23:43:17	[Edit] [Delete]
1000	Consumer	--	[Edit] [Delete]
1001	Consumer	--	[Edit] [Delete]
1002	Consumer	--	[Edit] [Delete]
1003	Consumer	--	[Edit] [Delete]
1004	Consumer	--	[Edit] [Delete]
1005	Consumer	--	[Edit] [Delete]

**Figure 297: User Management Page Display**

## User Information

When logged in as Super Admin, click on "Add" to create a new account for Web GUI user. The following dialog will prompt. Configure the parameters as shown in below table.





* User Name:	<input type="text" value="John"/>	Privilege:	<input type="text" value="Administrator"/>
* User Password:	<input type="text" value="admin123"/>	Department:	<input type="text" value="IT"/>
Fax:	<input type="text"/>	Email Address:	<input type="text"/>
First Name:	<input type="text"/>	Last Name:	<input type="text"/>
Home Number:	<input type="text"/>	Mobile Phone Number:	<input type="text" value="12365478"/>

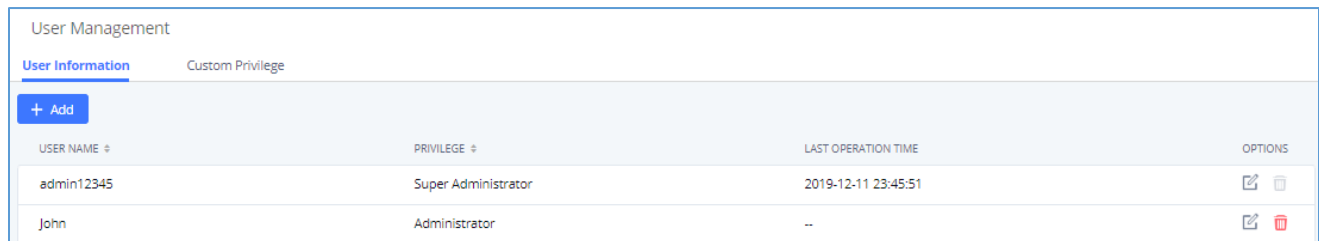
**Figure 298: Create New User**







**Table 141: User Management→Create New User**

<b>Username</b>	Configure a username to identify the user which will be required in Web GUI login. Letters, digits, and underscore are allowed in the username.
<b>User Password</b>	Configure a password for this user which will be required in Web GUI login. English input is allowed without space, ' and " .
<b>Privilege</b>	This is the role of the Web GUI user. Currently only "Admin" is supported when Super Admin creates a new user.
<b>Department</b>	Enter the necessary information to keep a record for this user.
<b>Email Address</b>	
<b>First Name</b>	
<b>Last Name</b>	
<b>Home Number</b>	
<b>Phone Number</b>	

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



The screenshot shows the 'User Management' interface with two tabs: 'User Information' (selected) and 'Custom Privilege'. There is a '+ Add' button. Below is a table with columns: USER NAME, PRIVILEGE, LAST OPERATION TIME, and OPTIONS.

USER NAME	PRIVILEGE	LAST OPERATION TIME	OPTIONS
admin12345	Super Administrator	2019-12-11 23:45:51	 
John	Administrator	--	 

**Figure 299: User Management – New Users**

## Custom Privilege

Four privilege levels are supported:

- **Super Administrator**

- This is the highest privilege. Super Admin can access all pages on UCM630xA 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 is the only user with "Super Admin" privilege. The Username and Privilege level cannot be changed or deleted.
- Super Admin could change its own login password on Web GUI→**Maintenance**→**Login Settings** page.



- Super Admin could view operations done by all the users in Web GUI → **Maintenance** → **User Management** → **Operation Log**

- **Administrator**

- 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**

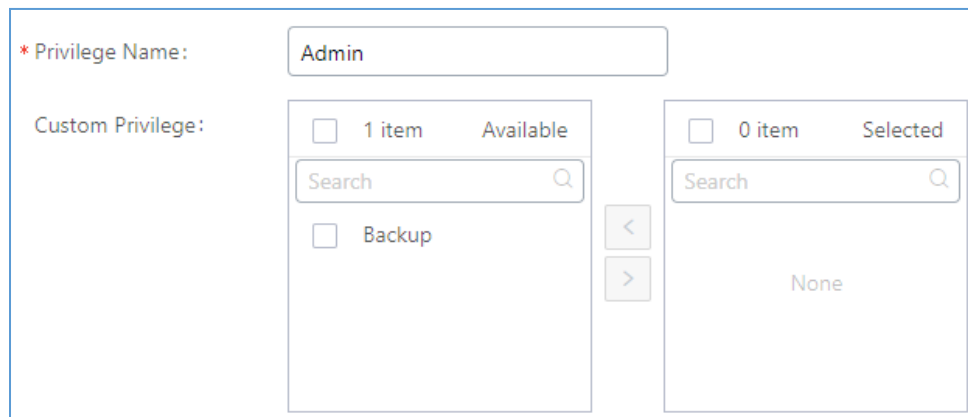
**Maintenance** → **Cleaner**

**Maintenance** → **Reset/Reboot**

**Settings** → **User Management** → **Operation Log**


- “Admin” privilege users cannot create new users for login.

**Note:** By default, administrator accounts are not allowed to access backup menu, but this can be assigned to them by editing the option “**Maintenance** → **User Management** → **Custom Privilege**” then press  to edit the “Admin” account and include backup operation permission for these types of users.



**Figure 300: Assign Backup permission to “Admin” users**

- **Consumer**

- A user account for Web GUI login is created automatically by the system when a new extension is created.
- The user could log in the Web GUI with the extension number and password to access user information, extension configuration, CDR of that extension, personal data, and value-added features. For more details; please refer to [User Portal Guide](#).
- The SuperAdmin user can click on  on the “General\_User” in order to enable/disable the custom privilege from deleting their own recording files, changing SIP credentials, and disabling voicemail service in their user portal account.



Edit Custom Privilege: General\_User

---

\* Privilege Name:

Enable Delete Recording

Files:

Allowed to change Auth ID   
and SIP password:

Allow to Enable Voicemail:

**Figure 301: General User**

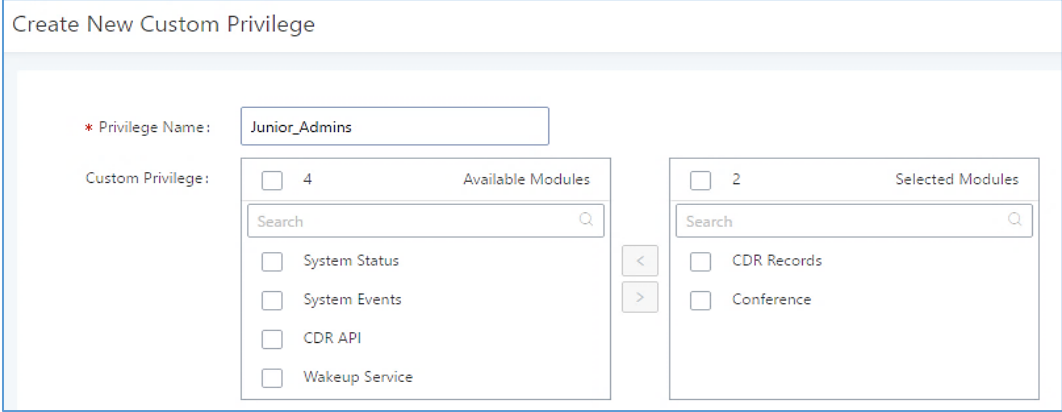
- **Custom Privilege**

The Super Admin user can create users with different privileges. 33 items are available for privilege customization.

- API Configuration
- API Configuration
- Backup
- Callback
- Call Queue
- CDR Recording Files
- CDR Records
- CDR Statistics
- Dial By Name
- DISA
- Emergency Calls
- Event List
- Extensions
- Outbound Routes
- Inbound Routes
- Feature Codes
- IVR
- Paging/Intercom
- Parking Lot
- Pickup Groups
- PMS - Wakeup Service



- Ring Groups
- SCA
- Speed Dial
- System Status
- System Events
- Time Settings
- Meeting
- Voicemail
- Voice Prompt
- Wakeup Service
- Zero Config
- LDAP Server
- UCM RemoteConnect
- Announcement.



**Figure 302: Create New Custom Privilege**

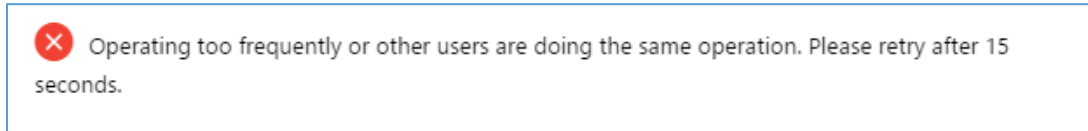
Log in UCM630xA as super admin and go to **Maintenance**→**User Management**→**Custom Privilege**, create privilege with customized available modules.

To assign custom privilege to a sub-admin, navigate to UCM Web GUI→**Maintenance**→**User Management**→**User Information**→Create New User/Edit Users, select the custom privilege from “Privilege” option.

### **Concurrent Multi-User Login**

When there are multiple Web GUI users created, concurrent multi-user login is supported on the UCM630xA. 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.



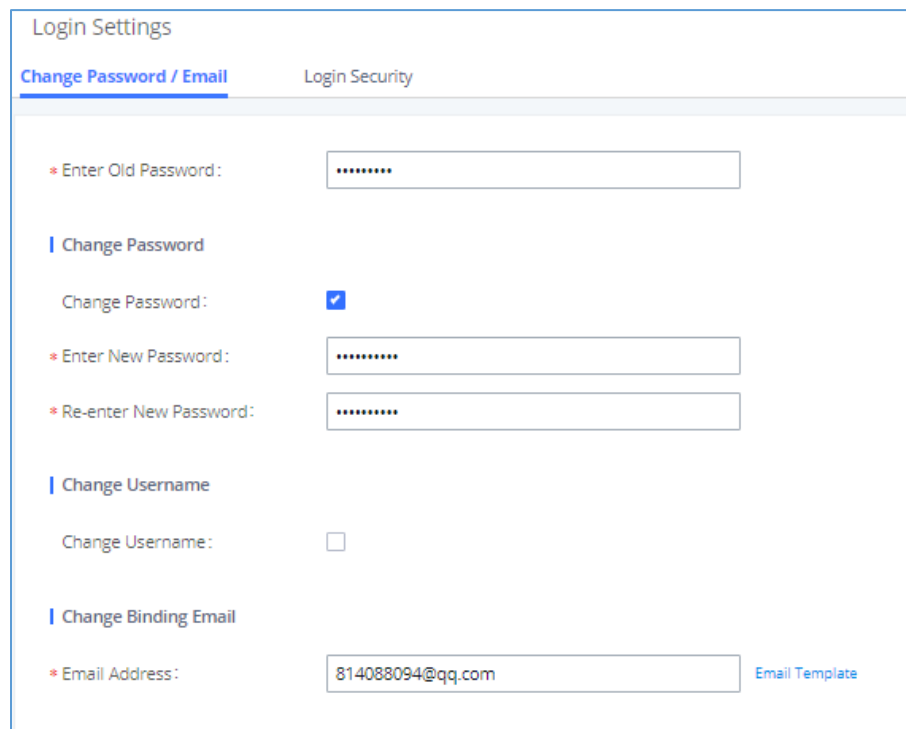


**Figure 303: Multiple User Operation Error Prompt**

## Change Password

After logging in the UCM630xA Web GUI for the first time, it is highly recommended for users to change the default password to a more complicated password for security purpose. Follow the steps below to change the Web GUI access password.

1. Go to Web GUI→**Maintenance**→**Login Settings**→**Change Password / Email** 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.



Login Settings

**Change Password / Email**      Login Security

\* Enter Old Password:

**Change Password**

Change Password:

\* Enter New Password:

\* Re-enter New Password:

**Change Username**

Change Username:

**Change Binding Email**

\* Email Address:  [Email Template](#)

**Figure 304: Change Password**





<b>Enter Old Password</b>	Enter the Old Password for UCM630xA
<b>Change Password</b>	Enable Change Password
<b>Enter New Password</b>	Enter the New Password for UCM630xA
<b>Re-enter New Password</b>	Retype the New Password for UCM630xA
<b>Change Username</b>	Enable Change Username
<b>Please enter the username</b>	Enter the Username
<b>Email Address</b>	The Email address is the User Email Address. It is used for receiving password information if the user forgets his password.

## Change Username

UCM630xA allows users now to change Super Administrator username.

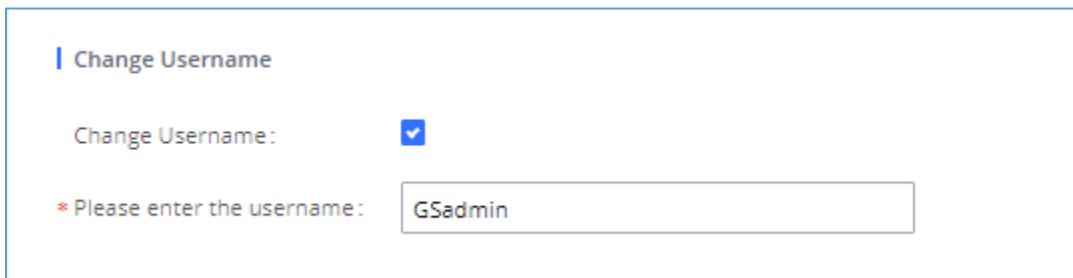


Figure 305: Change Username

## Change binding Email

UCM630xA allows user to configure binding email in case login password is lost. UCM630xA login credential will be sent to the designated email address. The feature can be found under Web GUI → **Maintenance** → **Login Settings** → **Change Password / Email**

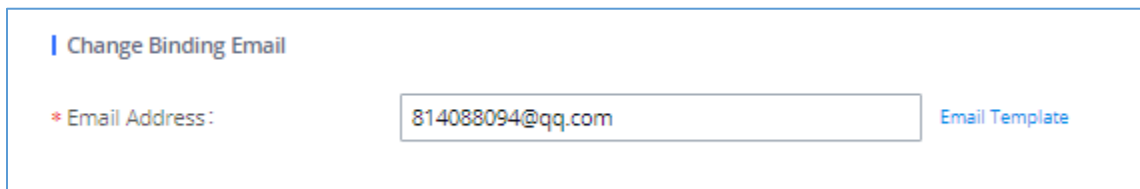


Figure 306: Change Binding Email

Table 142: Change Binding Email option

<b>Email Address</b>	Email Address is used to retrieve password when password is lost
----------------------	--



## Login Security

After the user logs in the UCM630xA Web GUI, the user will be automatically logged out after certain timeout, or he/she can be banned for a specific period if the login timeout is exceeded. Those values can be specified under UCM630xA web GUI→**Maintenance**→**Login Settings**→**Login Security** page.

The “**User Login Timeout**” value is in minute and the default setting is 10 minutes. If the user does not make any operation on Web GUI within the timeout, the user will be logged out automatically. After that, the Web GUI 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 GUI login session and the user will not be automatically logged out.

“**Maximum number of login attempts**” can prevent the UCM630xA from brutal force decryption, if this number is exceeded user IP address will be banned from accessing the UCM for a period of time based on user configuration, the default value is 5.

“**User ban period**” specify the period of time in minutes an IP will be banned from accessing the UCM if the User max number of try login is exceeded, the default value is 5.

“**Login Banned User List**” show the list of IPs’ banned from the UCM.

“**Login Whitelist**” User can add a list of IPs’ to avoid the above restriction, thus, they can exceed the User max number of try login.



Login Settings
Cancel Save

Change Password / Email
Login Security

- \* User Login Timeout:
- \* Maximum number of login attempts:
- \* User ban period:

Login Banned User List

IP ADDRESS	USER NAME	BANNED TIME	OPTIONS
No Data			

Login Whitelist

The IP addresses in the Login Whitelist will not be restricted. This option doesn't support network segment format.

+ Add

IP ADDRESS	OPTIONS
No Data	

**Figure 307: Login Timeout Settings**

## Operation Log





Super Admin has the authority to view operation logs on UCM630xA Web GUI→**Settings**→**User Management**→**Operation Log** page. Operation logs list operations done by all the Web GUI users, for example, Web GUI login, creating trunk, creating outbound rule and etc. There are 7 columns to record the operation details “Date”, “Username”, “IP Address”, “Results”, “Page Operation”, “Specific Operation” and “Remark”.




Operation Log <span style="float: right;">Display Filter ▾</span>						
<span>🗑️ Delete Search Result (s)</span> <span style="margin-left: 20px;"><a href="#">🗑️ Clear</a></span> <span style="margin-left: 20px;"><a href="#">📄 Download Search Result (s)</a></span> <span style="margin-left: 20px;"><a href="#">⬇️ Download All Log</a></span>						
DATE ▾	USER NAME ▾	IP ADDRESS ▾	RESULTS ▾	PAGE OPERATION ▾	SPECIFIC OPERATION ▾	REMARK ▾
2019-12-11 23:52:56	admin12345	41.250.0.39	Operation successful	Login Settings: updateLoginBanned	OLD_ip: 41.250.198.175.	<a href="#">Click to modify notes</a>
2019-12-11 23:52:41	admin12345	41.250.0.39	Operation successful	Extensions: Login	User Name: admin12345.	<a href="#">Click to modify notes</a>
2019-12-11 23:45:49	admin12345	41.250.198.175	Operation successful	addUser	user_name: John.	<a href="#">Click to modify notes</a>
2019-12-11 23:43:07	admin12345	41.250.198.175	Operation successful	Extensions: Login	User Name: admin12345.	<a href="#">Click to modify notes</a>
2019-12-11 23:35:11	admin12345	41.250.198.175	Operation successful	Extensions: Login	User Name: admin12345.	<a href="#">Click to modify notes</a>
2019-12-11 23:35:07	admin12345	41.250.198.175	Operation successful	Extensions: Login	User Name: admin12345.	<a href="#">Click to modify notes</a>
2019-12-11 23:22:27	admin12345	41.250.198.175	Operation successful	Extensions: Login	User Name: admin12345.	<a href="#">Click to modify notes</a>
2019-12-11 23:05:15	admin12345	41.250.198.175	Operation successful	Extensions: Login	User Name: admin12345.	<a href="#">Click to modify notes</a>
2019-12-11 23:04:55	admin	41.250.198.175	Wrong account or password!	Login	User Name: admin.	<a href="#">Click to modify notes</a>
2019-12-11 23:04:31	admin12345	41.250.198.175	Operation successful	Logout: Logout	User Name: admin12345.	<a href="#">Click to modify notes</a>

Total: 124  / page

**Figure 308: Operation Logs**

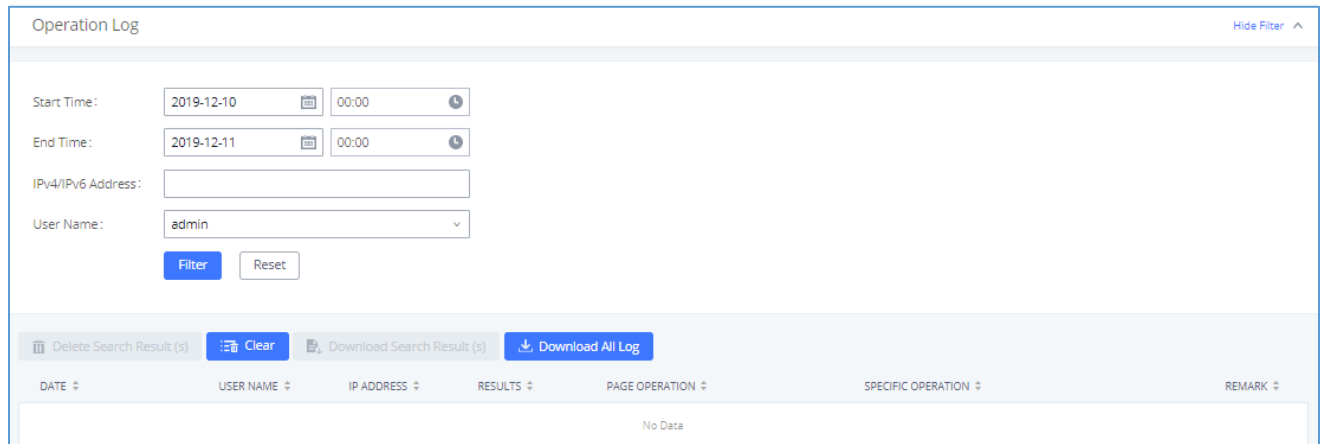
The operation log can be sorted and filtered for easy access. Click on  or  at the top of each column to sort. For example, clicking on  for "Date" will sort the logs according to newer operation date and time. Clicking on  for "Date" will reverse the order.

**Table 143: Operation Log Column Header**

<b>Date</b>	The date and time when the operation is executed.
<b>Username</b>	The username of the user who performed the operation.
<b>IP Address</b>	The IP address from which the operation is made.
<b>Results</b>	The result of the operation.
<b>Page Operation</b>	The page where the operation is made. For example, login, logout, delete user, create trunk and etc.
<b>Specific Operation</b>	Click on  to view the options and values configured by this operation.
<b>Remark</b>	Allows users to add notes and remarks to each operation

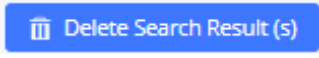

User could also filter the operation logs by time condition, IP address and/or username. Configure these conditions and then click on "Display Filter".





**Figure 309: Operation Logs Filter**

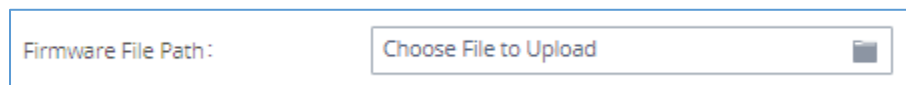
The above figure shows an example that operations made by user “support” on device with IP 192.168.40.173 from 2014-11-01 00:00 to 2014-11-06 15:38 are filtered out and displayed.

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

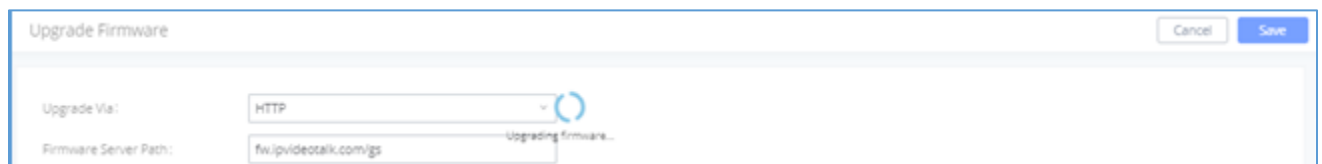
## Upgrading

The UCM630xA can be upgraded to a new firmware version locally. And in order to do that, please follow the below steps:

1. Download the latest UCM630xA firmware file from the following link and save it in your PC.  
<https://www.grandstream.com/support/firmware>
2. Log in the Web GUI as administrator in the PC.
3. Go to Web GUI→**Maintenance**→**Upgrade**, upload the firmware file by clicking on “choose file to upload” and select the firmware file from your PC. The default firmware file name is UCM630xAfw.bin



**Figure 310: Local Upgrade**



**Figure 311: Upgrading Firmware Files**



- Wait until the upgrading process is successful and a window will be popped up in the Web GUI.



**Figure 312: Reboot UCM630xA**

- Click on "OK" to reboot the UCM630xA and check the firmware version after it boots up.



**Notes:**

- Please do not interrupt or power cycle the UCM630xA during upgrading process.
  - The firmware file name allows the use of the special characters besides the following restricted characters: # \$ ^ & \* + ( ) [ ] / ; ' | , < > ?
- 

### No Local Firmware Servers

Service providers should maintain their own firmware upgrade servers. For users who do not have TFTP/HTTP/HTTPS server, some free windows version TFTP servers are available for download from [http://www.solarwinds.com/products/freetools/free\\_tftp\\_server.aspx](http://www.solarwinds.com/products/freetools/free_tftp_server.aspx)  
<http://tftpd32.jounin.net>

Please check our website at <https://www.grandstream.com/support/firmware> for latest firmware.

Instructions for local firmware upgrade via TFTP:

- Unzip the firmware files and put all of them in the root directory of the TFTP server;
- Connect the PC running the TFTP server and the UCM630xA to the same LAN segment;
- Launch the TFTP server and go to the File menu→Configure→Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade;
- Start the TFTP server and configure the TFTP server in the UCM630xA web configuration interface;
- Configure the Firmware Server Path to the IP address of the PC;
- Update the changes and reboot the UCM630xA.

End users can also choose to download a free HTTP server from <http://httpd.apache.org/> or use Microsoft IIS web server.



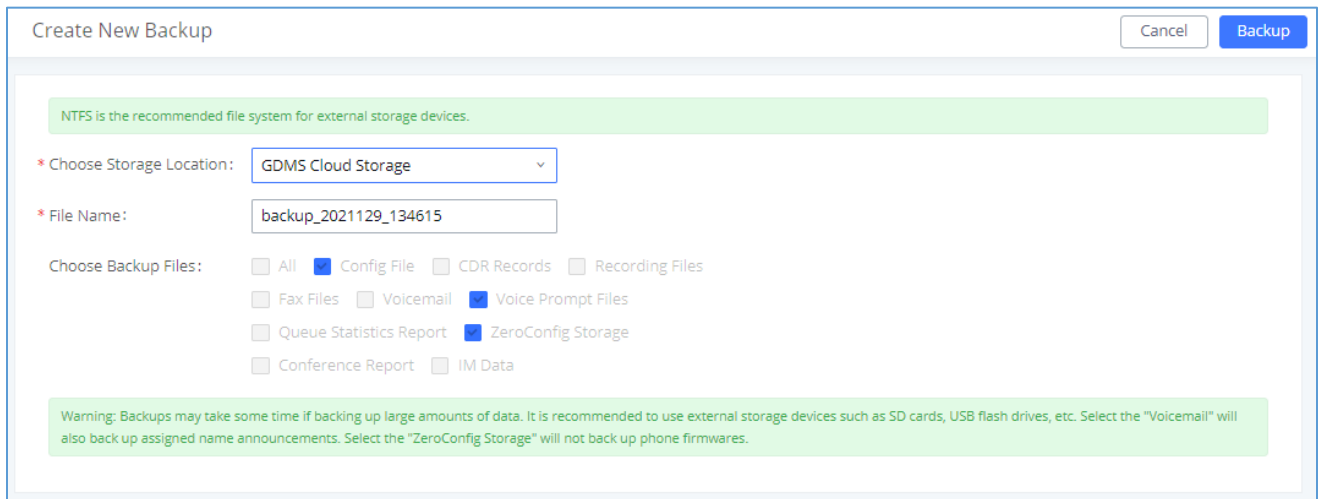
## Backup

The UCM630xA configuration can be backed up locally or via network. The backup file will be used to restore the configuration on UCM630xA when necessary.

### Backup/Restore




Users could backup the UCM630xA configurations for restore purpose under Web GUI → **Maintenance** → **Backup** → **Backup/Restore**.

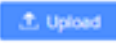
Click on "Backup" to create a new backup file. Then the following dialog will show.



**Figure 313: Create New Backup**

1. Choose the type(s) of files to be included in the backup.
2. Choose where to store the backup file: USB Disk, SD Card, Local, NAS or GDMS.
3. Name the backup file.
4. Click on "Backup" to start backup.

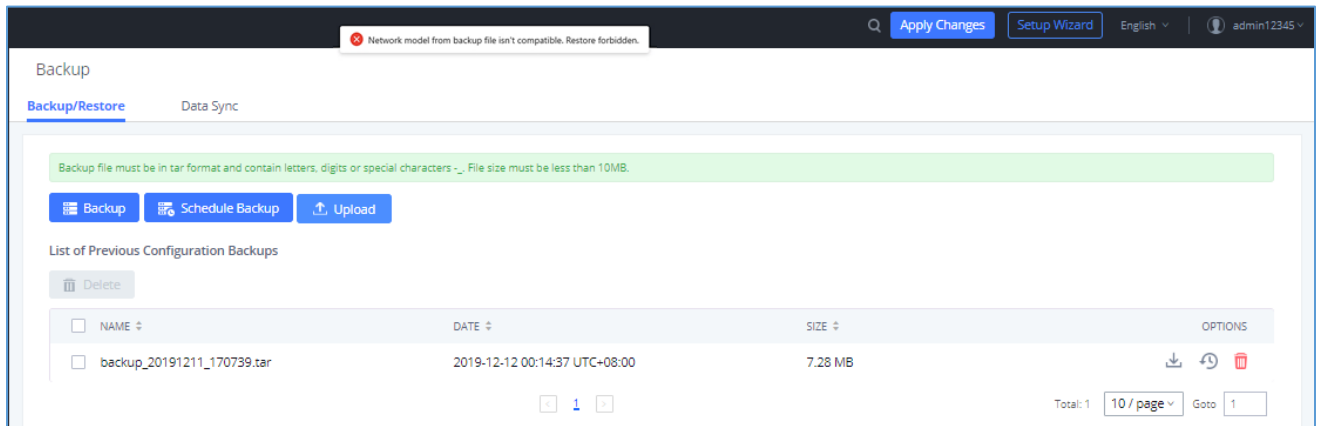
Once the backup is done, the list of the backups will be displayed with date and time in the web page. Users can download , restore , or delete  it from the UCM630xA internal storage or the external device.

Click on  to upload backup file from the local device to UCM630xA. The uploaded backup file will also be displayed in the web page and can be used to restore the UCM630xA.

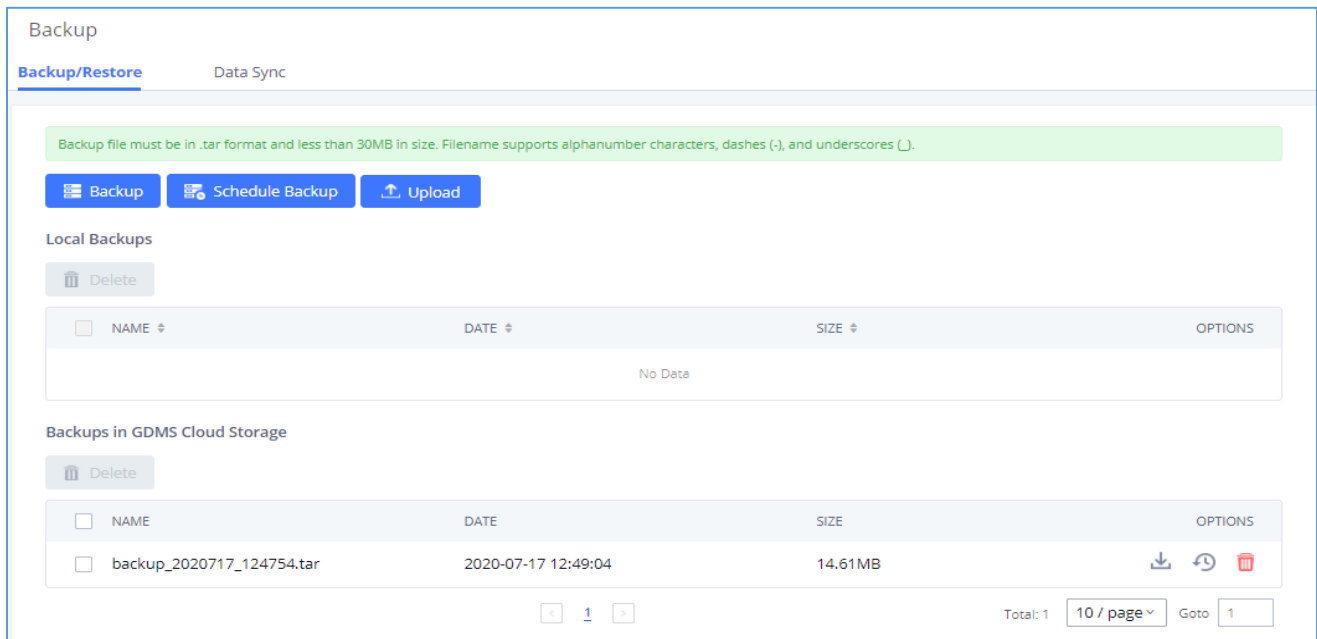
**Note:** users can restore backups of models with more FXO ports to models with less FXO ports as long as the configurations related to the extra FXO ports are removed.



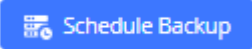
Please make sure the FXO port settings, total number of extensions and total number of meeting rooms are compactable before restoring to another UCM model. Otherwise it will prompt a warning and stop the restore process as shown below:



**Figure 314: Restore Warning**



**Figure 315: Backup / Restore**

The  option allows UCM to perform automatically backup on the user specified time. Regular backup file can only be stored in USB / SD card / SFTP server. User is allowed to set backup time from 0-23 and how frequent the backup will be performed.





### Schedule Backup

NTFS is the recommended file system for external storage devices.

Enable Scheduled Backup:

Choose Storage Location:

\* Account:

Password:

\* Server Address:

Destination Directory:

\* Backup Time:

\* Backup Frequency:

Choose Backup Files:

All
  Config File
  CDR Records
  Recording Files

Voicemail
  Voice Prompt Files
  Queue Statistics Report

ZeroConfig Storage
  Conference Report

**Figure 316: Local Backup**

## Data Sync

Besides local backup, users could backup the voice records/voice mails/CDR in a daily basis to a remote server via SFTP protocol automatically under Web GUI→**Maintenance**→**Backup**→**Data Sync**.

The client account supports special characters such as @ or ".". Allowing the use email address as SFTP accounts. It allows users as well to specify the destination directory on SFTP server for backup file. If the directory does not exist on the destination, UCM630xA will create the directory automatically



Backup

Backup/Restore Data Sync

---

Use SFTP to automatically sync CDR, recordings, voicemail, CDR, and fax every day.

**Data Sync Configuration**

Enable Data Sync:

Choose Data Sync Files:

CDR Records  
 Voice Mail

Recording Files  
 Fax

\* Account:

Password:

\* Server Address:

Destination Directory:

\* Sync Time:

+ Test Connection
+ Synchronize All Data

**Data Sync Log**

Clean


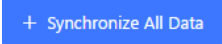
No record to view

**Figure 317: Data Sync**

**Table 144: Data Sync Configuration**




<b>Enable Data Sync</b>	Enable the auto data sync function. The default setting is "No".
<b>Account</b>	Enter the Account name on the SFTP backup server.
<b>Password</b>	Enter the Password associate with the Account on the SFTP backup server.
<b>Server Address</b>	Enter the SFTP server address.
<b>Destination Directory</b>	Specify the directory in SFTP server to keep the backup file. Format: 'xxx/xxx/xxx', If this directory does not exist, UCM will create this directory automatically.
<b>Sync Time</b>	Enter 0-23 to specify the backup hour of the day.



Before saving the configuration, users could click on . The UCM630xA will then try connecting the server to make sure the server is up and accessible for the UCM630xA. Save the changes and all the backup logs will be listed on the web page. After data sync is configured, users could also manually synchronize all data by clicking on  instead of waiting for the backup time interval to come.

## Restore Configuration from Backup File

To restore the configuration on the UCM630xA from a backup file, users could go to Web GUI → **Maintenance** → **Backup** → **Backup/Restore**.

- A list of previous configuration backups is displayed on the web page. Users could click on  of the desired backup file and it will be restored to the UCM630xA.
- If the backup was stored on GDMS, it will be displayed under Backups GDMS Cloud Storage, that can be restored by clicking on .
- If users have other backup files on PC to restore on the UCM630xA, click on "Upload Backup File" first and select it from local PC to upload on the UCM630xA. Once the uploading is done, this backup file will be displayed in the list of previous configuration backups for restore purpose. Click on  to restore from the backup file.

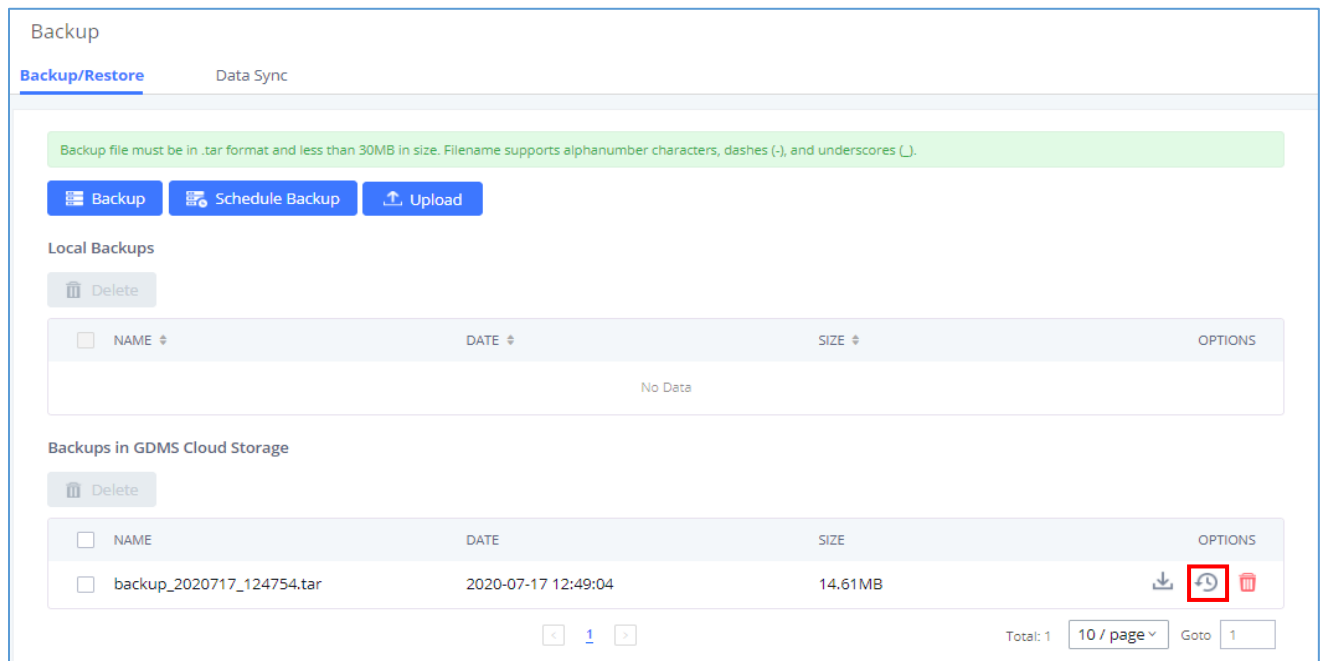


Figure 318: Restore UCM630xA from Backup File




**Note:**

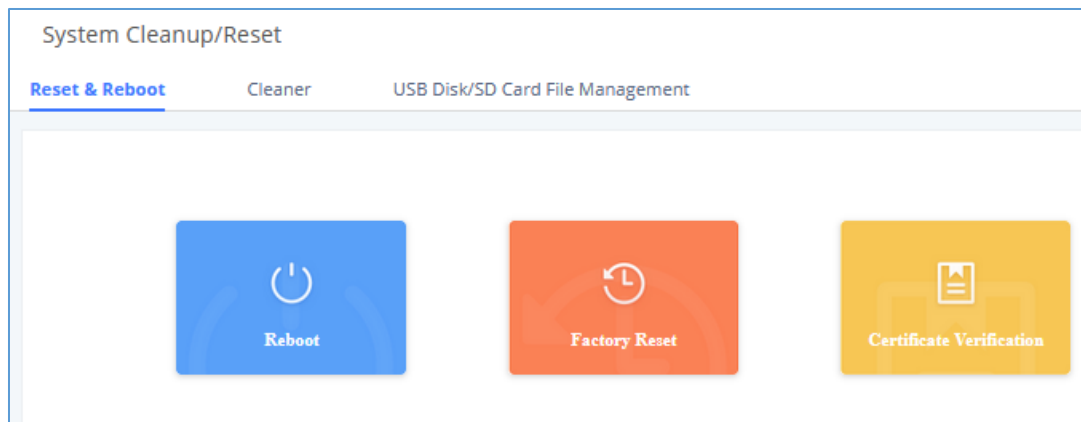
- The uploaded backup file must be a tar file with no special characters like \*,!,#,@,&,\$,%^,(,),/, \,space in the file name.
- The uploaded back file size must be under 10MB.

## System Cleanup/Reset

### Reset and Reboot

Users could perform reset and reboot under Web GUI→**Maintenance**→**System Cleanup/Reset**→**Reset and Reboot**.

- To reboot the device, click on reboot icon.
- To factory reset the device, click on reset icon, then all the configurations and data will be reset to factory default.



**Figure 319: Reset and Reboot**

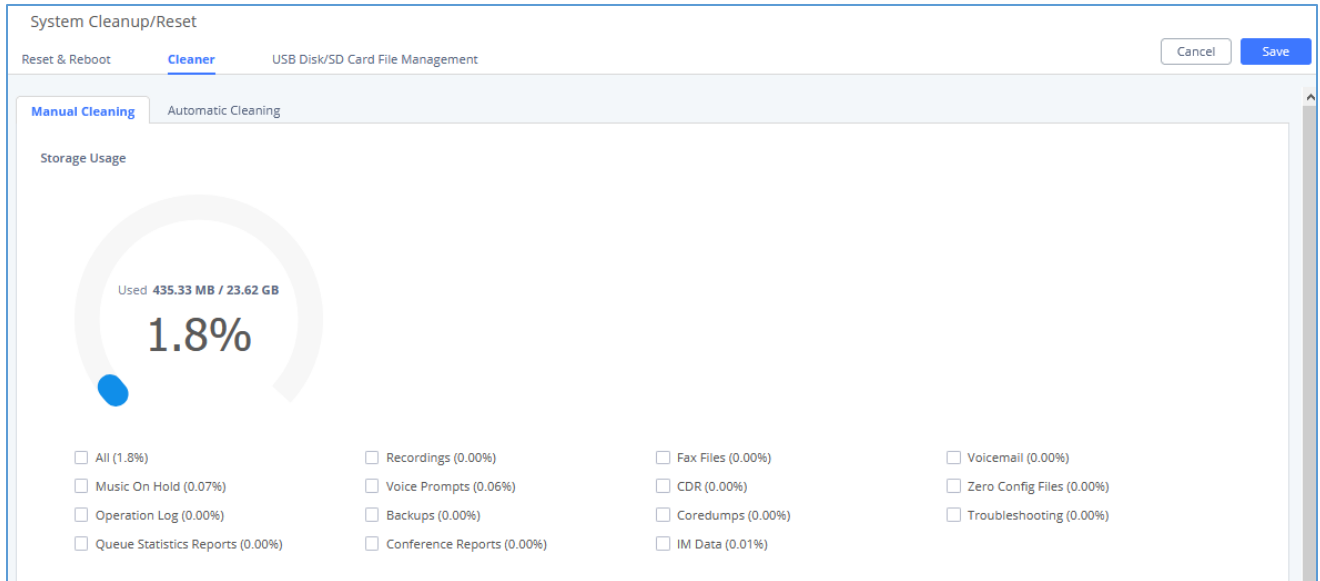
- User can also verify UCM certificate under the same path.

### Cleaner

Users could configure to clean the Call Detail Report/Voice Records/Voice Mails etc... manually and automatically under Web GUI→**Maintenance**→**System Cleanup/Reset**→**Cleaner**.

The following screenshot show the settings and parameters to configure the manual cleaner feature on UCM630xA.





**Figure 320: Manual Cleaning**

Users can either clean all the data on the UCM or specify the modules to clean such as: Recordings, Fax Files, Voicemail, Music on Hold, Voice Prompts, CDR, ZeroConfig Files, Operation Log, Backups, Coredumps, Troubleshooting, Queue Statistics Reports, Meeting Reports, IM Data.

User can also set an automatic cleaning under **Cleaner→Automatic Cleaning**. The following screenshot show the settings and parameters to configure the cleaner feature on UCM630xA.



Manual Cleaning
Automatic Cleaning

Clean CDR, recordings, voicemail, and fax automatically.

**CDR Cleaner**

Enable Cleaner:

Clean Time:

Cleaning Conditions:

Clean Interval (d):

**Report Cleaner**

Enable Cleaner:

Cleanup Type:  Queue Statistics Report  Conference Call Statistics Report

Clean Time:

Cleaning Conditions:

Clean Interval (d):

**IM Data Cleaner**

Enable Cleaner:

Cleanup Type:  IM Message  IM Share Files

Clean Time:

Cleaning Conditions:

File Clean Interval (d):

**File Cleaner**

Enable Cleaner:

Clean Files in External Storage:

Choose Cleaner Files:  Basic Call Recording Files  Audio Conference Recording Files  
 Call Queue Recording Files  Voicemail Files  
 Emergency Calls Recording Files  Fax  
 Backup Files  SCA Recording Files

Clean Time:

Cleaning Conditions:

File Clean Threshold:

Keep Last X Days:

**Cleaner Log**

**Figure 321: Automatic Cleaning**

**Table 145: Automatic Cleaning Configuration**

<b>Enable CDR Cleaner</b>	Enable the CDR Cleaner function.
<b>CDR Clean Time</b>	Enter 0-23 to specify the hour of the day to clean up CDR.
<b>Cleaning Conditions</b>	<b>By Schedule:</b> If the clean interval is 3, cleaning will be performed every 3 days to remove all records that were generated 3 days ago.



	<p><b>Keep Last X Records:</b> If the max number of CDR has been reached, CDR will be deleted starting with the oldest entry at the configured cleaning time.(Note: The amount of records displayed on the page of call queue statistics is not one-to-one with the actual amount of records in the database.)</p> <p><b>Keep Last X Days:</b> Delete all entries older than X days.</p>
<b>Clean Interval</b>	Enter 1-30 to specify the day of the month to clean up CDR when <b>By Schedule</b> is selected as <b>Cleaning Conditions</b> .
<b>Max Entries</b>	Set the maximum number of CDR entries to keep when <b>Keep Last X Records</b> is selected as <b>Cleaning Conditions</b> .
<b>Keep Last X Day</b>	Enter the number of days of call log entries to keep when <b>Keep Last X days</b> is selected as <b>Cleaning Conditions</b> .
<b>Enable Queue Statistics Report Cleaner</b>	Enable scheduled queue log cleaning. By default, is disabled.
<b>Queue Statistics Report Cleaner Clean Time</b>	Enter the hour of the day to start the cleaning. The valid range is 0-23.
<b>Cleaning Conditions</b>	<p><b>By Schedule:</b> If the clean interval is 3, cleaning will be performed every 3 days to remove all records that were generated 3 days ago.</p> <p><b>Keep Last X Records:</b> If the max number of Queue Statistics has been reached, Queue Statistics will be deleted starting with the oldest entry at the configured cleaning time.(Note: The amount of records displayed on the page of call queue statistics is not one-to-one with the actual amount of records in the database.)</p> <p><b>Keep Last X Days:</b> Delete all entries older than X days.</p>
<b>Clean Interval</b>	Enter how often (in days) to clean queue logs when <b>By Schedule</b> is selected as <b>Cleaning Conditions</b> . The valid range is 1-30.
<b>Max Entries</b>	Set the maximum number of Queue Statistics entries to keep when <b>Keep Last X Records</b> is selected as <b>Cleaning Conditions</b> .
<b>Keep Last X Day</b>	Enter the number of days of call log entries to keep when <b>Keep Last X days</b> is selected as <b>Cleaning Conditions</b> .
<b>Enable Meeting Statistics Report Cleaner</b>	Enable scheduled Meeting log cleaning. By default, is disabled.



<b>Cleaning Conditions</b>	<ul style="list-style-type: none"> <li>• <b>By Schedule:</b> If the clean interval is 3, cleaning will be performed every 3 days to remove all records that were generated 3 days ago.</li> <li>• <b>Keep Last X Records:</b> If the max number of Meeting Statistics Report has been reached, Meeting Statistics Report will be deleted starting with the oldest entry at the configured cleaning time. (Note: The amount of records displayed on the page of call queue statistics is not one-to-one with the actual amount of records in the database.)</li> <li>• <b>Keep Last X Days:</b> Delete all entries older than X days.</li> </ul>
<b>Clean Interval</b>	Enter how often (in days) to clean queue logs when <b>By Schedule</b> is selected as <b>Cleaning Conditions</b> . The valid range is 1-30.
<b>Max Entries</b>	Set the maximum number of Meeting Statistics Report entries to keep when <b>Keep Last X Records</b> is selected as <b>Cleaning Conditions</b> .
<b>Keep Last X Day</b>	Enter the number of days of call log entries to keep when <b>Keep Last X days</b> is selected as <b>Cleaning Conditions</b> .
<b>Enable File Cleaner</b>	Enter the Voice Records Cleaner function.
<b>Clean Files in External Device</b>	If enabled the files in external device (USB/SD card) will be atomically cleaned up as configured.
<b>Choose Cleaner File</b>	Select the files for system automatic clean. <ul style="list-style-type: none"> <li>• Basic Call Recording Files.</li> <li>• Meeting Recording Files.</li> <li>• Call Queue Recording Files.</li> <li>• Voicemail Files.</li> <li>• Backup Files.</li> </ul>
<b>Clean time</b>	Enter the hour of the day to start the cleaning. The valid range is 0-23.
<b>Cleaning Conditions</b>	<ul style="list-style-type: none"> <li>• <b>By Schedule:</b> If the clean interval is 3, cleaning will be performed every 3 days to delete all files.</li> <li>• <b>By Threshold:</b> Check at the configured cleaning time every day to see if the storage threshold has been exceeded and perform cleaning of all files if it has.</li> <li>• <b>Keep Last X Days:</b> Delete all files older than X days.</li> </ul>
<b>File Clean Interval</b>	Enter 1-30 to specify the day of the month to clean up the files.
<b>File Clean Threshold</b>	Enter the internal storage disk usage threshold (in percent). Once this threshold is exceeded, the file cleanup will proceed as scheduled. Valid



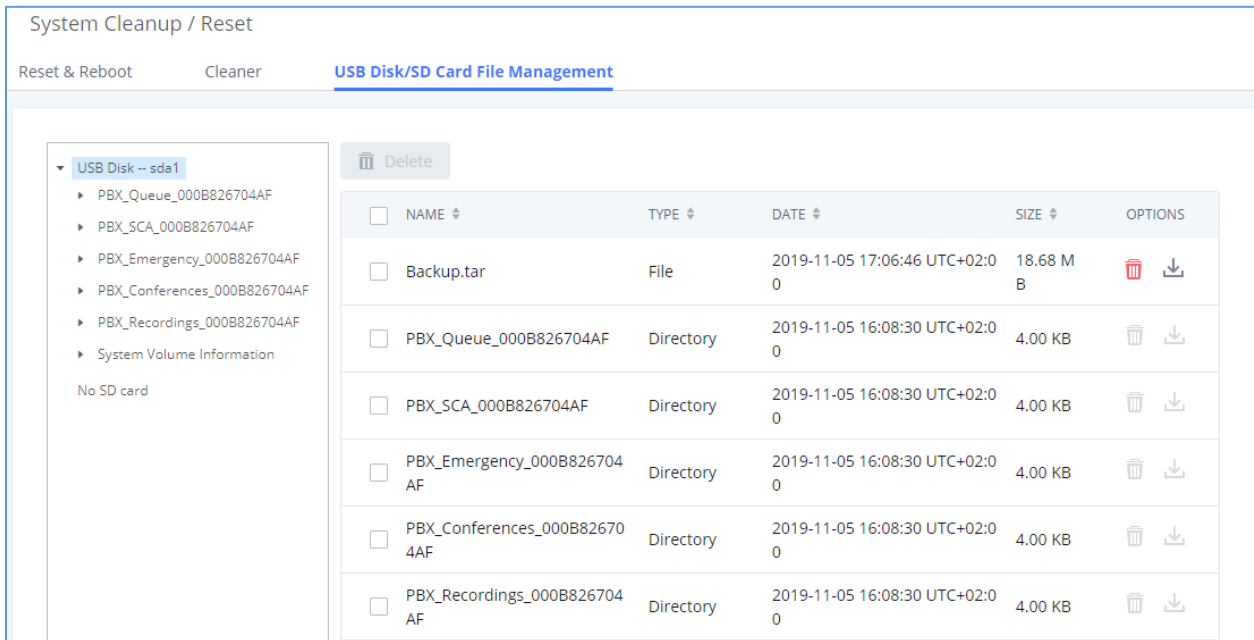


	range is 0-99.
<b>Keep Last X Days</b>	Automatically delete all recordings older than this x days when the threshold is reached. If not set, all data is cleared
<b>Cleaner Log</b>	Press Clean “button” to clean cleaner log.

All the cleaner logs will be listed on the bottom of the page.

## USB/SD Card Files Cleanup

Users could configure to clean or download the Call Detail Report/Voice Records/Voice Mails automatically under Web GUI→**Maintenance**→**System Cleanup/Reset**→**USB / SD Card Files Cleanup**.



The screenshot shows the 'System Cleanup / Reset' page with the 'USB Disk/SD Card File Management' tab selected. On the left, a tree view shows the contents of 'USB Disk -- sda1', including directories for PBX\_Queue, PBX\_SCA, PBX\_Emergency, PBX\_Conferences, and PBX\_Recordings, along with 'System Volume Information' and 'No SD card'. On the right, a table lists the files and directories with their respective types, dates, and sizes. Each row has checkboxes for selection and icons for deleting or downloading the file.

NAME	TYPE	DATE	SIZE	OPTIONS
Backup.tar	File	2019-11-05 17:06:46 UTC+02:00	18.68 MB	[Delete] [Download]
PBX_Queue_000B826704AF	Directory	2019-11-05 16:08:30 UTC+02:00	4.00 KB	[Delete] [Download]
PBX_SCA_000B826704AF	Directory	2019-11-05 16:08:30 UTC+02:00	4.00 KB	[Delete] [Download]
PBX_Emergency_000B826704AF	Directory	2019-11-05 16:08:30 UTC+02:00	4.00 KB	[Delete] [Download]
PBX_Conferences_000B826704AF	Directory	2019-11-05 16:08:30 UTC+02:00	4.00 KB	[Delete] [Download]
PBX_Recordings_000B826704AF	Directory	2019-11-05 16:08:30 UTC+02:00	4.00 KB	[Delete] [Download]

**Figure 322: USB/SD Card Files Cleanup**

**Table 146: USB/SD Card Files Cleanup**

<b>Current Path</b>	Displays the current path.
<b>Directory</b>	Select the directory user want to clean.
<b>Delete Selected File</b>	Select multiple entries to delete from USB or SD card.

## System Recovery

In some cases (for example after wrong upgrading procedure where the user doesn't follow the correct steps to perform an upgrade) the system may go into some hardware/software issues where the web UI access is lost



as well as SSH, in this case the only solution would be to perform a full system recovery in order to reset or update the software version of the device in order to use it again.

1. To access recovery mode on UCM, please follow below steps:
2. Remove the power from the unit and keep the network cable connected.
3. Press using a PIN the reset button and keep holding.
4. Plug back the power supply while maintaining the reset button pressed.
5. Wait for couple of seconds until you hear a click sound.
6. Release the reset button, and the system should display on the LCD a message “Recovery Mode” along with an IP address.

Once at this stage, the administrator can access the recovery mode web portal by typing in either the IP0 address (typically WAN) or IP1 address (typically LAN) into a browser address bar. The following page should appear:



**Figure 323: UCM6302A Recovery Web Page**

Make sure to enter the correct admin password, and press login to access the recovery mode page :



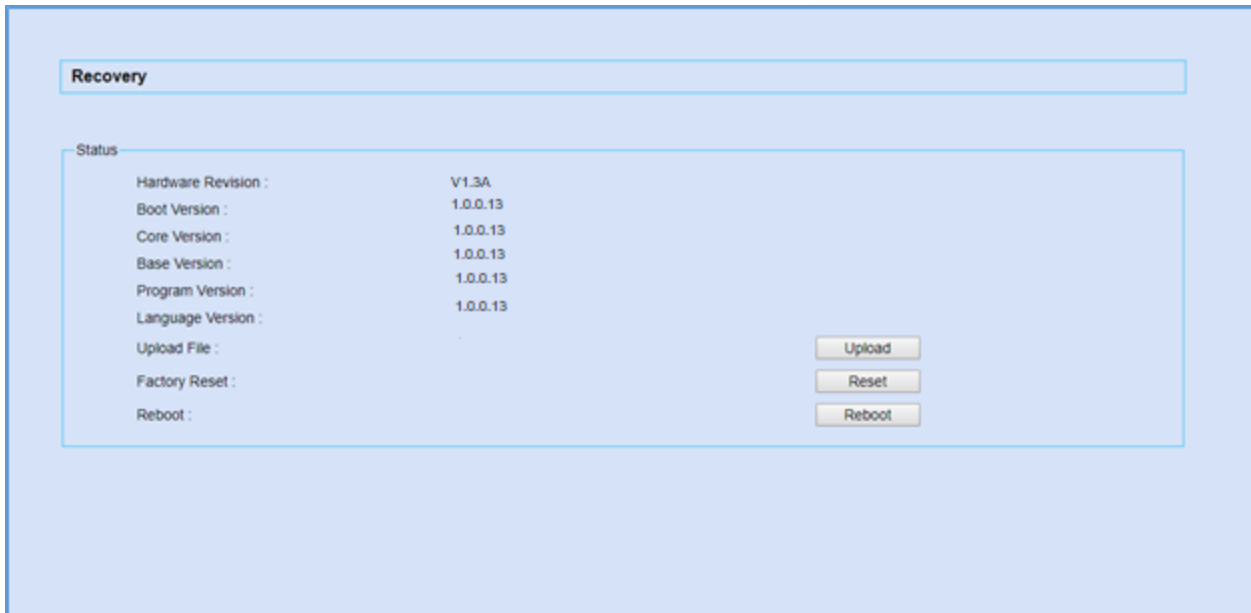


Figure 324: Recovery Mode

From here, the user can either upload a firmware file, factory reset or just reboot the device.

## Syslog

On the UCM630xA, users could dump the syslog information to a remote server under Web GUI → **Maintenance** → **Syslog**. Enter the syslog server hostname or IP address and select the module/level for the syslog information as well as Process Log Level.

The default syslog level for all modules is "error", which is recommended in your UCM630xA settings because it can be helpful to locate the issues when errors happen.

Some typical modules for UCM630xA functions are as follows and users can turn on "NOTICE" and "VERBOSE" levels besides "error" level.

- **pbx**: This module is related to general PBX functions.
- **pjsip**: This module is related to SIP calls.
- **chan\_dahdi**: This module is related to analog calls (FXO/FXS).

---

 **Note:**

Syslog is usually for debugging and troubleshooting purpose. Turning on all levels for all syslog modules is not recommended for daily usage. Too many syslog prints might cause traffic and affect system performance.

The reserved size for Syslog entries on the cache memory of the UCM is 50M, once this sized is reached the



UCM will clean up 2M of the oldest Syslog entries to allow to save new logs.

---

## Network Troubleshooting

On the UCM630xA, users could capture traces, ping remote host and traceroute remote host for troubleshooting purpose under Web GUI→**Maintenance**→**Network Troubleshooting**.

The following sections shows the steps to capture different types of traffic traces for analysis purposes.

### Ethernet Capture

The captured trace can be downloaded for analysis. The instructions or result will be displayed in the Web GUI output result.



### Network Troubleshooting

Ethernet Capture
IP Ping
Traceroute

NTFS is the recommended file system for external storage devices.

**Regular Debugging**

Capture Type: Ethernet Capture

Interface Type: LAN

Enable SFTP Data Sync:

Capture Filter:

Save to external storage:

▶ Start
⬇ Download

**Output Result**

The file has been deleted or does not exist.

**S RTP Debugging**

Enable S RTP Debugging:

⬇ Download

**Figure 325: Ethernet Capture**

**Table 147: Ethernet Capture**

<b>Capture Type</b>	<p>Ethernet Capture: Gets a packet capture of all network traffic going through the device.</p> <p>WebSocket Capture: Gets a packet capture of WebSocket protocol. Mainly used for troubleshooting GS Wave Web calling and conferencing issues.</p>
<b>Interface Type</b>	Select the network interface to monitor.
<b>Enable SFTP Data Sync</b>	Check this box to save the capture files in the SFTP server. Please make sure the configuration of data synchronization works before.



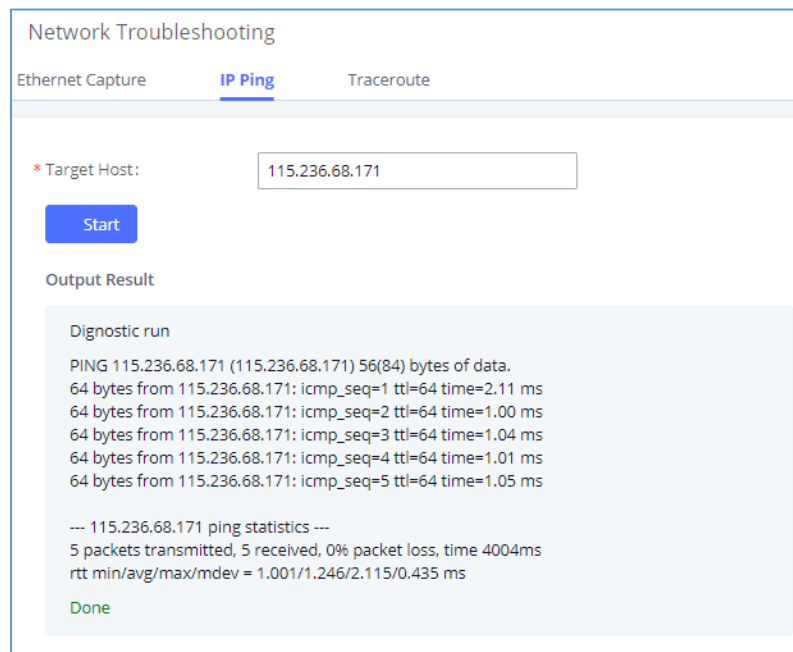
<b>Storage to External Device</b>	Check this box to activate storage of the capture either on the USB or SD Card.
<b>Capture Filter</b>	Enter the filter to obtain the specific types of traffic, such as (host, src, dst, net, proto...).
<b>Save to external storage</b>	Save to external storage
<b>Start</b>	Click to start the trace.
<b>Stop</b>	Click to stop the trace.
<b>Download</b>	Click to download the trace if trace is stored locally.
<b>Enable SRTP Debugging</b>	Check this box to troubleshoot calls encrypted with TLS/SRTP.

The output result is in .pcap format. Therefore, users could specify the capture filter as used in general network traffic capture tool (host, src, dst, net, protocol, port, port range) before starting to capture the trace.

**Note:** Capture files saved on external devices will now have “capture” prepended to file names.

## IP Ping

Enter the target host in host name or IP address. Then press "Start" button. The output result will dynamically display in the window below.



**Figure 326: Ping**



## Traceroute

Enter the target host in host name or IP address. Then press "Start" button. The output result will dynamically display in the window below.

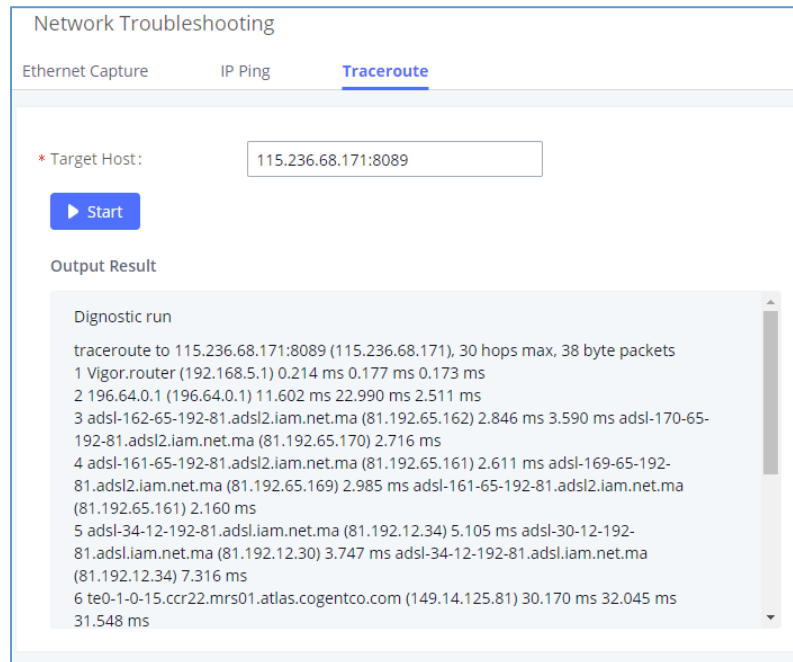


Figure 327: Traceroute

## Signaling Troubleshooting

### Analog Record Trace

- **Analog Record Trace**

Analog record trace can be used to troubleshoot analog trunk issue, for example, the UCM630xA user has caller ID issue for incoming call from Analog trunk. Users can access analog record trace under Web GUI → **Maintenance** → **Signal Troubleshooting**.

Here is the step to capture trace:

1. Select FXO or FXS for "Record Ports". If the issue happens on FXO 1, select FXO port 1 to record the trace.
2. Click on "Start".
3. Make a call via the analog port that has the issue.
4. Once done, click on "Stop".



- Click on "Download" to download the analog record trace.

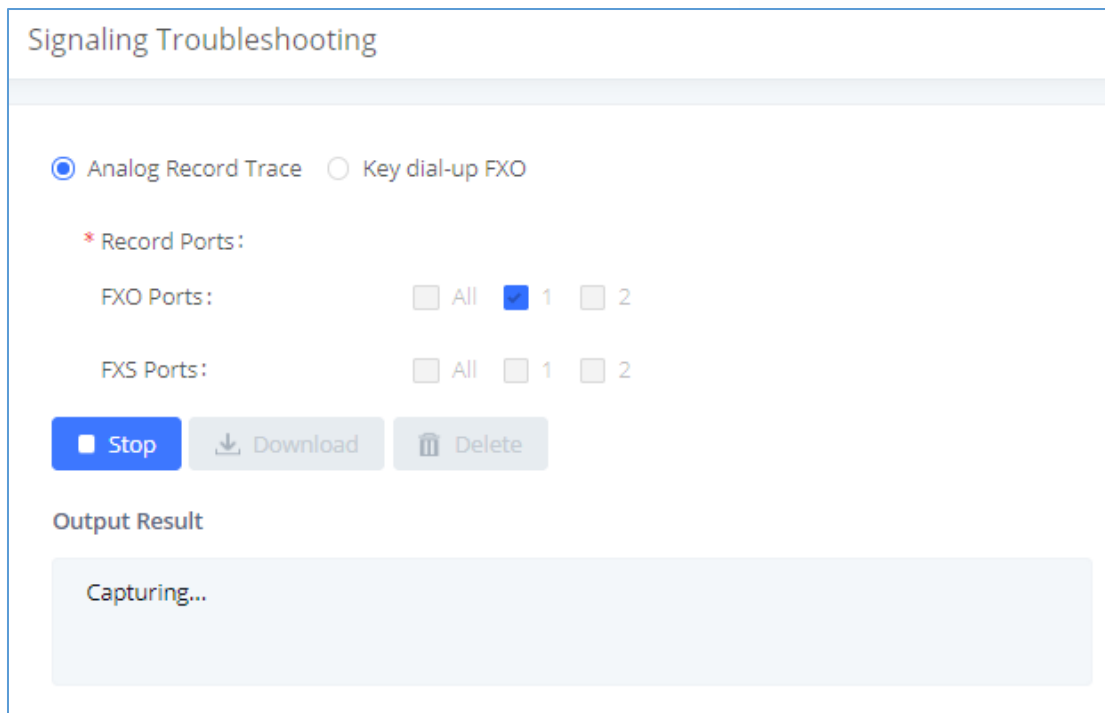


Figure 328: Troubleshooting Analog Trunks

- **A key Dial-up FXO**

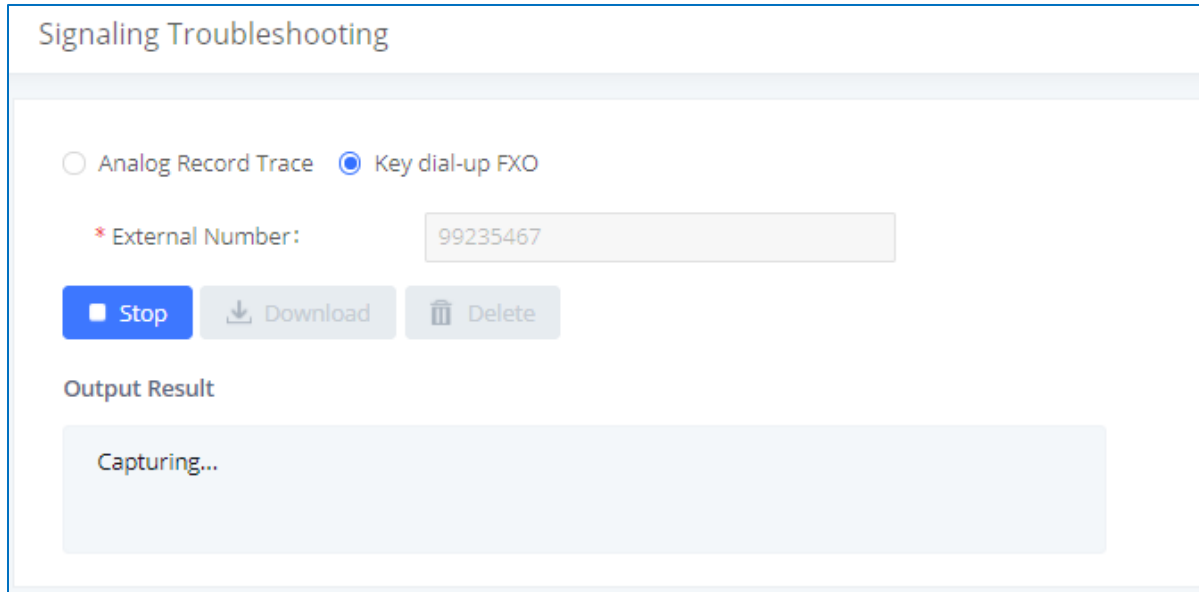
Users can directly set a PSTN number on the “**External Extension**” text box to troubleshoot issues related to the analog trunk easily, the following steps shows how to use this feature:

1. Configure analog trunk on UCM, including outbound route.
2. Enter a reachable external number in “**External Extension**”.
3. Press “**Start**” button. The call will be initiated to the external number.
4. Answer and finish the call before pressing “**Stop**” button.

The trace will be available for analysis to download after output result shows “Done! Click on Download to download the captured packets”.







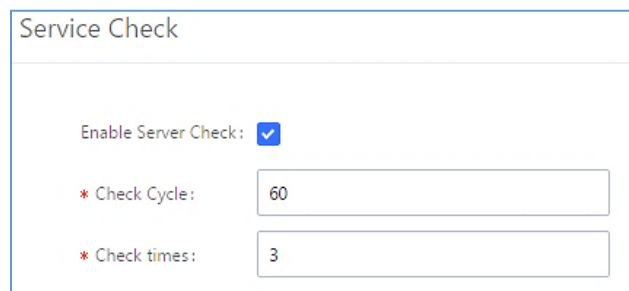
**Figure 329: A Key Dial-up FXO**

**Note:** When using a Key Dial-up FXO feature the outbound trunk for the analog trunk need to have internal permission. As well as it should be the trunk with the highest outbound route priority.

After capturing the trace, users can download it for basic analysis. Or you can contact Grandstream Technical support in the following link for further assistance if the issue is not resolved.  
<https://www.grandstream.com/support>

## Service Check

Enable Service Check to periodically check UCM630xA. Check Cycle is configurable in seconds and the default setting is 60 sec. Check Times is the maximum number of failed checks before restart the UCM630xA. The default setting is 3. If there is no response from UCM630xA after 3 attempts (default) to check, current status will be stored and the internal service in UCM630xA will be restarted.



**Figure 330: Service Check**



## Network Status

In UCM630xA Web GUI→**System Status**→**Network Status**, the users can view active Internet connections. This information can be used to troubleshoot connection issue between UCM630xA and other services.

Network Status						
<b>Active Connections</b> Active Unix Domain Sockets						
PROTO	RCV-Q	SEND-Q	LOCAL-ADDRESS	FOREIGN-ADDRESS	STATE	
tcp	0	0	0.0.0.0:7681	0.0.0.0:*	LISTEN	
tcp	0	0	0.0.0.0:7777	0.0.0.0:*	LISTEN	
tcp	0	0	0.0.0.0:389	0.0.0.0:*	LISTEN	
tcp	0	0	0.0.0.0:3306	0.0.0.0:*	LISTEN	
tcp	0	0	127.0.0.1:45678	0.0.0.0:*	LISTEN	
tcp	0	0	127.0.0.1:8439	0.0.0.0:*	LISTEN	
tcp	0	0	0.0.0.0:8888	0.0.0.0:*	LISTEN	
tcp	0	0	0.0.0.0:8088	0.0.0.0:*	LISTEN	
tcp	0	0	127.0.0.1:25	0.0.0.0:*	LISTEN	
tcp	0	0	0.0.0.0:636	0.0.0.0:*	LISTEN	

Total: 53    10 / page    Goto 1

**Figure 331: Network Status**



## EXPERIENCING THE UCM630xA SERIES IP PBX

Please visit our website: <https://www.grandstream.com/> to receive the most up- to-date updates on firmware releases, additional features, FAQs, documentation and news on new products.

We encourage you to browse our [product related documentation](#), [FAQs](#) and [User and Developer Forum](#) for answers to your general questions. If you have purchased our products through a Grandstream Certified Partner or Reseller, please contact them directly for immediate support.

Our technical support staff is trained and ready to answer all of your questions. Contact a technical support member or [submit a trouble ticket online](#) to receive in-depth support.

Thank you again for purchasing Grandstream UCM630xA series IP PBX appliance, it will be sure to bring convenience and color to both your business and personal life.

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