

UCM630x Audio Series – User Manual

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 TableAPI 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.

Alert

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.

Caution

Please do not use a different power adaptor with the UCM630xA as it may cause damage to the product and void the manufacturer warranty.

Note

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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.

Interfaces	
Analog Telephone FXS Ports	<ul style="list-style-type: none">● UCM6300A: None● UCM6302A: 2 RJ11 ports● UCM6304A: 4 RJ11 ports● UCM6308A: 8 RJ11 ports All ports have lifeline capability in case of power outage.
PSTN Line FXO Ports	<ul style="list-style-type: none">● UCM6300A: None● UCM6302A: 2 RJ11 ports● UCM6304A: 4 RJ11 ports● UCM6308A: 8 RJ11 ports All ports have lifeline capability in case of power outage
Network Interfaces	Three self-adaptive Gigabit ports (switched, routed or dual card mode) with PoE+

NAT Router	Yes (supports router mode and switch mode)
Peripheral Ports	<ul style="list-style-type: none"> ● UCM6300A: 1x USB 3.0, and 1x SD card interface ● UCM6302A: 1x USB 2.0, 1x USB 3.0, and 1x SD card interface ● UCM6304A/6308A: 2x USB 3.0 and 1x SD card interface
LED Indicators	<ul style="list-style-type: none"> ● UCM6300A/6302A/UCM6304A: None ● UCM6308A: Power 1/2, FXS, FXO, LAN, WAN, Heartbeat
LCD Display	<ul style="list-style-type: none"> ● UCM6300A/6302A/6304A: 320*240 LCD with touch screen for Shortcut Keys and Scroll Bar. ● UCM6308A: 128x32 dot matrix graphic LCD with DOWN and OK buttons
Reset Switch	Yes, long press for factory reset and short press for reboot
Voice Capabilities	
Voice-over-Packet Capabilities	LEC with NLP Packetized Voice Protocol Unit, 128ms-tail length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711, NetEQ, FEC 2.0, jitter resilience up to 50% audio packet loss
Voice and Fax Codecs	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
QoS	Layer 2 QoS (802.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS
Signalling and Control	
API	Full API available for third-party platform and application integration.
DTMF Methods	Inband, RFC4733, and SIP INFO
Provisioning Protocol and Plug-and-Play	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
Network Protocols	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®
Disconnect Methods	Busy/Congestion/Howl Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect
Security	
Media Encryption	SRTP, TLS1.2, HTTPS, SSH, 802.1x
Physical	
Universal Power Supply	<ul style="list-style-type: none"> ● UCM6300A/6302A: Input: 100 ~ 240VAC, 50/60Hz; Output: DC 12V, 1.5A ● UCM6304A: 1x DC 12V Power Jack Input: 100~240VAC, 50/60Hz; Output: DC 12V, 2A

	<ul style="list-style-type: none"> ● UCM6308A: 2x DC 12V Power Jack Input: 100~240VAC,50/60Hz; Output: DC12V, 2A
Dimensions	<ul style="list-style-type: none"> ● UCM6300A/6302A/6304A: 270mm(L) x 175mm(W) x 36mm(H) ● UCM6308A: 485mm(L) x 187.2mm(W) x 46.2mm(H)
Weight	<ul style="list-style-type: none"> ● UCM6300A: Unit weight 705g, Package weight 1131g ● UCM6302A: Unit weight 725g, Package weight 1221g ● UCM6304A: Unit weight 775g, Package weight 1621g ● UCM6308A: Unit weight 2538g, Package weight 3463g
Temperature & Humidity	<ul style="list-style-type: none"> ● Operating: 32 - 113°F / 0 ~ 45°C, Humidity 10 - 90% (non-condensing) ● Storage: 14 - 140°F / -10 ~ 60°C, Humidity 10 - 90% (non-condensing)
Mounting	<ul style="list-style-type: none"> ● UCM6300A/6302A/6304A: Wall mount and desktop. ● UCM6308A: Rack mount and desktop.
Additional Features	
Multi-language Support	<ul style="list-style-type: none"> ● Web UI: English, Simplified Chinese, Traditional Chinese, Spanish, French, Portuguese, German, Russian, Italian, Polish, Czech, Turkish ● Customizable IVR/voice prompts: English, Chinese, British English, German, Spanish, Greek, French, Italian, Polish, Portuguese, Russian, Swedish, Turkish, Ukrainian, Hebrew, Arabic, Netherlands ● Customizable language pack to support any other languages
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT, NTT
Polarity Reversal/ Wink	Yes, with enable/disable option upon call establishment and termination
Call Center	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/workload, in-queue announcement
Customizable Auto Attendant	Up to 5 layers of IVR (Interactive Voice Response) in multiple languages
Telephony Operating System	Based on Asterisk version 16
Maximum Call Capacity	<p>UCM6300A:</p> <ul style="list-style-type: none"> ● Users: 250 ● Concurrent calls (G.711): 50 ● Max concurrent SRTP calls: 50 <p>UCM6302A:</p> <ul style="list-style-type: none"> ● Users: 500 ● Concurrent calls (G.711): 75 ● Max concurrent SRTP calls: 75 <p>UCM6304A:</p> <ul style="list-style-type: none"> ● Users: 1000 ● Concurrent calls (G.711): 150 ● Max concurrent SRTP calls: 120 <p>UCM6308A:</p>

	<ul style="list-style-type: none"> • Users: 1500 • Concurrent calls (G.711): 200 • Max concurrent SRTP calls: 150
Maximum Attendees of Meeting Bridges	<ul style="list-style-type: none"> • UCM6300A: 3 Meeting rooms and up to 50 parties • UCM6302A: 5 Meeting rooms and up to 75 parties • UCM6304A: 7 Meeting rooms and up to 120 parties • UCM6308A: 9 Meeting rooms and up to 150 parties
Wave App	Free; Available for desktop (Windows 10+, Mac OS 10+), web (Firefox and Chrome Browsers) and mobile (Android & iOS), allows users to join UCM-hosted meetings, communicate with other users/solutions and make/receive calls using SIP accounts registered to a UCM6300 Audio series IP PBX
Call Features	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, fax to email, speed dial, call back, dial by name, emergency call, call follow me, blacklist/whitelist, voice conference, video conference, eventlist, feature codes, busy camp-on/ call completion, voice control, post-meeting reports, virtual fax sending/receiving, email to fax
Firmware Upgrade	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
Internet Protocol Standards	RFC 3261, RFC 3262, RFC 3263, RFC 3264, RFC 3515, RFC 3311, RFC 4028, RFC 2976, RFC 3842, RFC 3892, RFC 3428, RFC 4733, RFC 4566, RFC 2617, RFC 3856, RFC 3711, RFC 5245, RFC 5389, RFC 5766, RFC 6347, RFC 6455, RFC 8860, RFC 4734, RFC 3665, RFC 3323, RFC 3550
Compliance	<ul style="list-style-type: none"> • FCC: Part 15 (CFR 47) Class B, Part 68 • CE: EN 55032, EN 55035, EN61000-3-2, EN61000-3-3, EN 62368.1, ES 203 021, ITU-T K.21 • IC: ICES-003, CS-03 Part I Issue 9 • RCM: AS/NZS CISPR 32, AS/NZS 62368.1, AS/CA S002, AS/CA S003.1/.2 • Power adapter: UL 60950-1 or UL 62368-1

Technical Specifications

i 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.

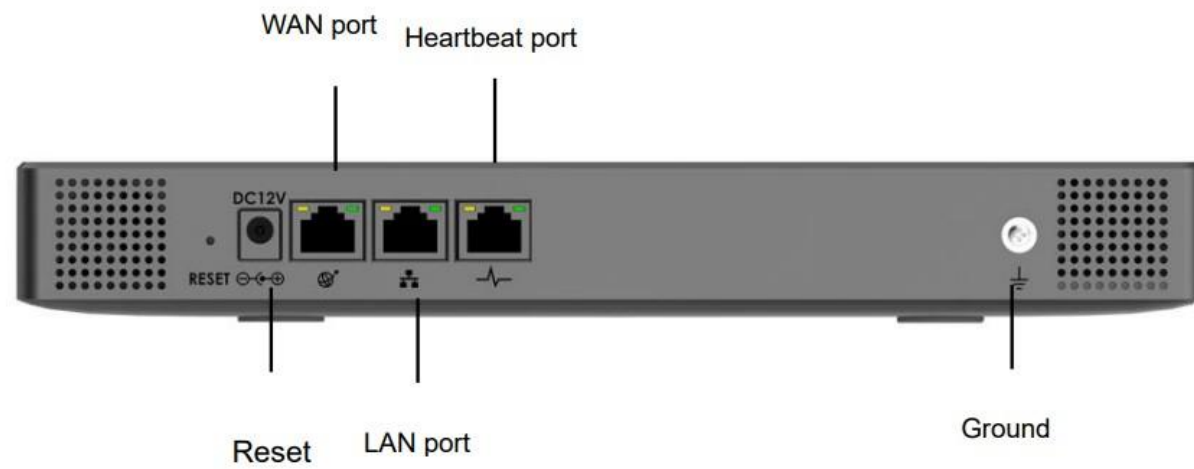
Equipment Packaging

Main Case	1
Power Adaptor	1

Ethernet Cable	1
Quick Installation Guide	1

UCM630xA Equipment Packaging

UCM6300A front and back view

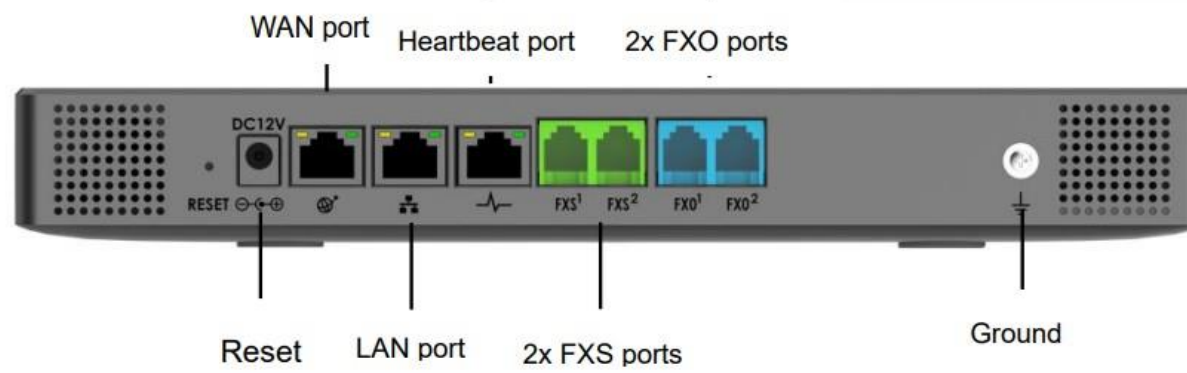


UCM6300A Back View



UCM6300A Front View

UCM6302A front and back view

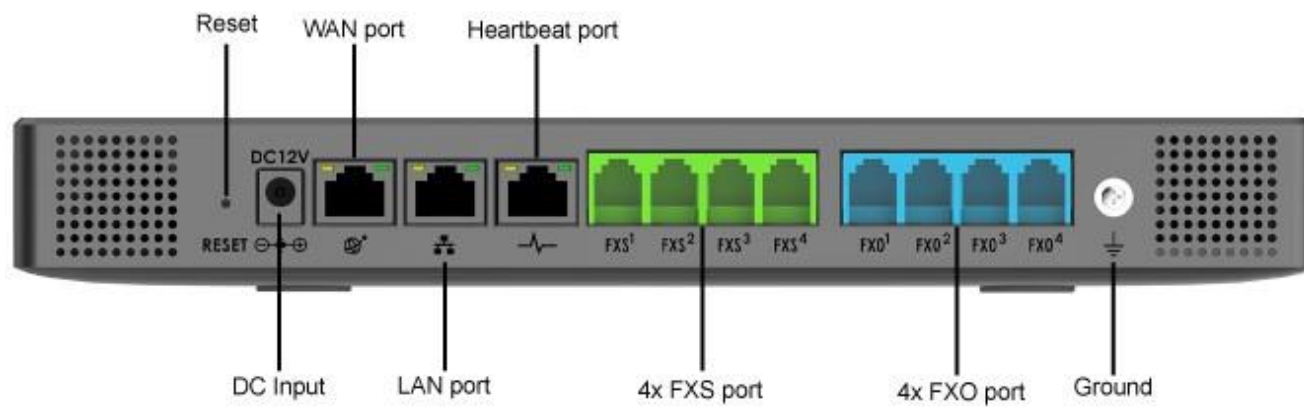


UCM6302A Back View



UCM6302A Front View

UCM6304A front and back view

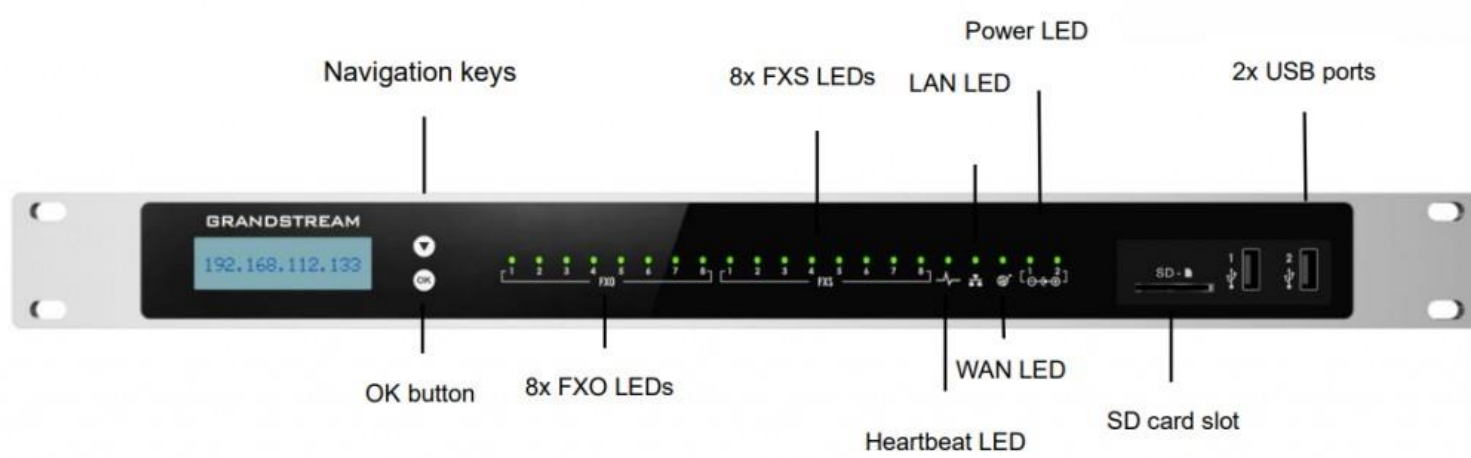


UCM6304A Front View

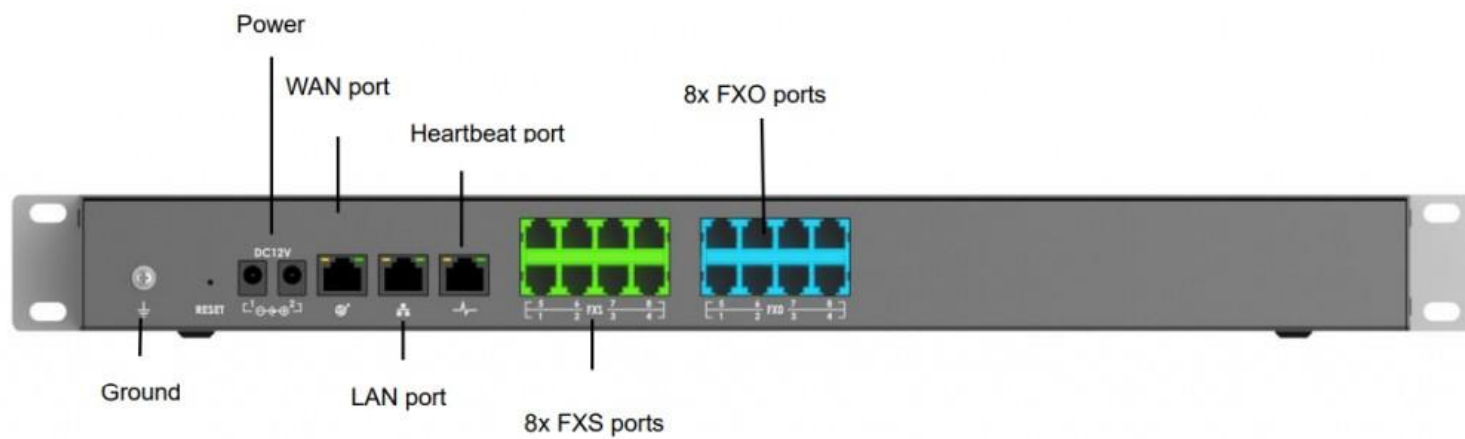


UCM6304A Back View

UCM6308A front and back view



UCM6308A Front View



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.

ℹ 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.

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.

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 the table [LCD Menu Options].

- **Menu Navigation**

Scrolling down 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.

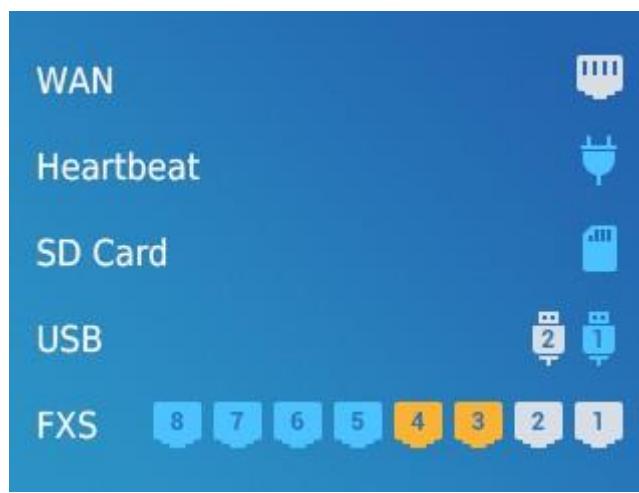
View Events	<ul style="list-style-type: none">○ Critical Events○ Other Events
Device Info	<ul style="list-style-type: none">○ Hardware: Hardware version number○ Software: Software version number○ P/N: Part number○ WAN MAC: WAN side MAC address○ LAN MAC: LAN side MAC address○ Uptime: System uptime

Network Info	<ul style="list-style-type: none"> ○ WAN Mode: DHCP, Static IP, or PPPoE ○ WAN IP: IP address ○ WAN Subnet Mask ○ LAN IP: IP address ○ LAN Subnet Mask
Network Menu	<ul style="list-style-type: none"> ○ WAN Mode: Select WAN mode as DHCP, Static IP or PPPoE ○ Static Route Reset: Select this to reset static route settings.
Factory Menu	<ul style="list-style-type: none"> ○ Reboot ○ Factory Reset ○ LCD Test Patterns <p>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.</p> <ul style="list-style-type: none"> ○ Fan Mode <p>Select Auto or On.</p> <ul style="list-style-type: none"> ○ LED Test Patterns <p>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.</p> <ul style="list-style-type: none"> ○ RTC Test Patterns <p>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.</p> <ul style="list-style-type: none"> ○ Hardware Testing <p>Select Test SVIP to verify hardware connections within the device. The result will display on the LCD when the test is complete.</p>
Web Info	<ul style="list-style-type: none"> ○ Protocol: Web access protocol (HTTP/ HTTPS). HTTPS is used by default. ○ Port: Web access port number, which is 8089 by default.
SSH Switch	<ul style="list-style-type: none"> ○ Enable SSH ○ Disable SSH <p>SSH access is disabled by default</p>

LCD Menu Options

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.



Ports Status

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

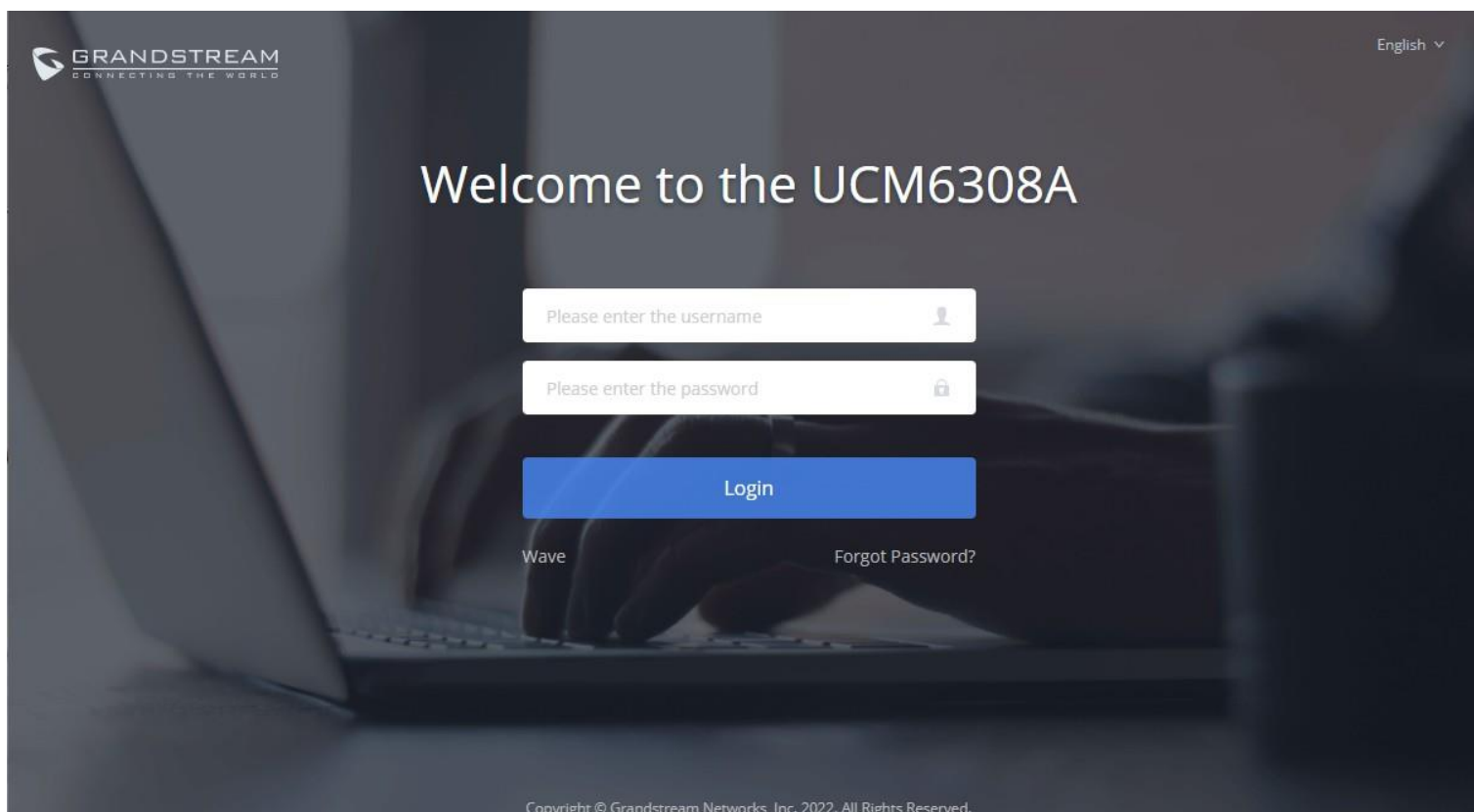
LED Indicator	LED Status
Power 1/Power 2	
PoE	
LAN	<ul style="list-style-type: none"> ○ Solid: Connected
WAN	<ul style="list-style-type: none"> ○ Fast Blinking: Data Transferring
USB	<ul style="list-style-type: none"> ○ Slow Blinking: Trying to connect
SD	<ul style="list-style-type: none"> ○ OFF: Not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

UCM6304A/6308A LED Indicators

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:



UCM630xA 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.

i 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.

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.

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

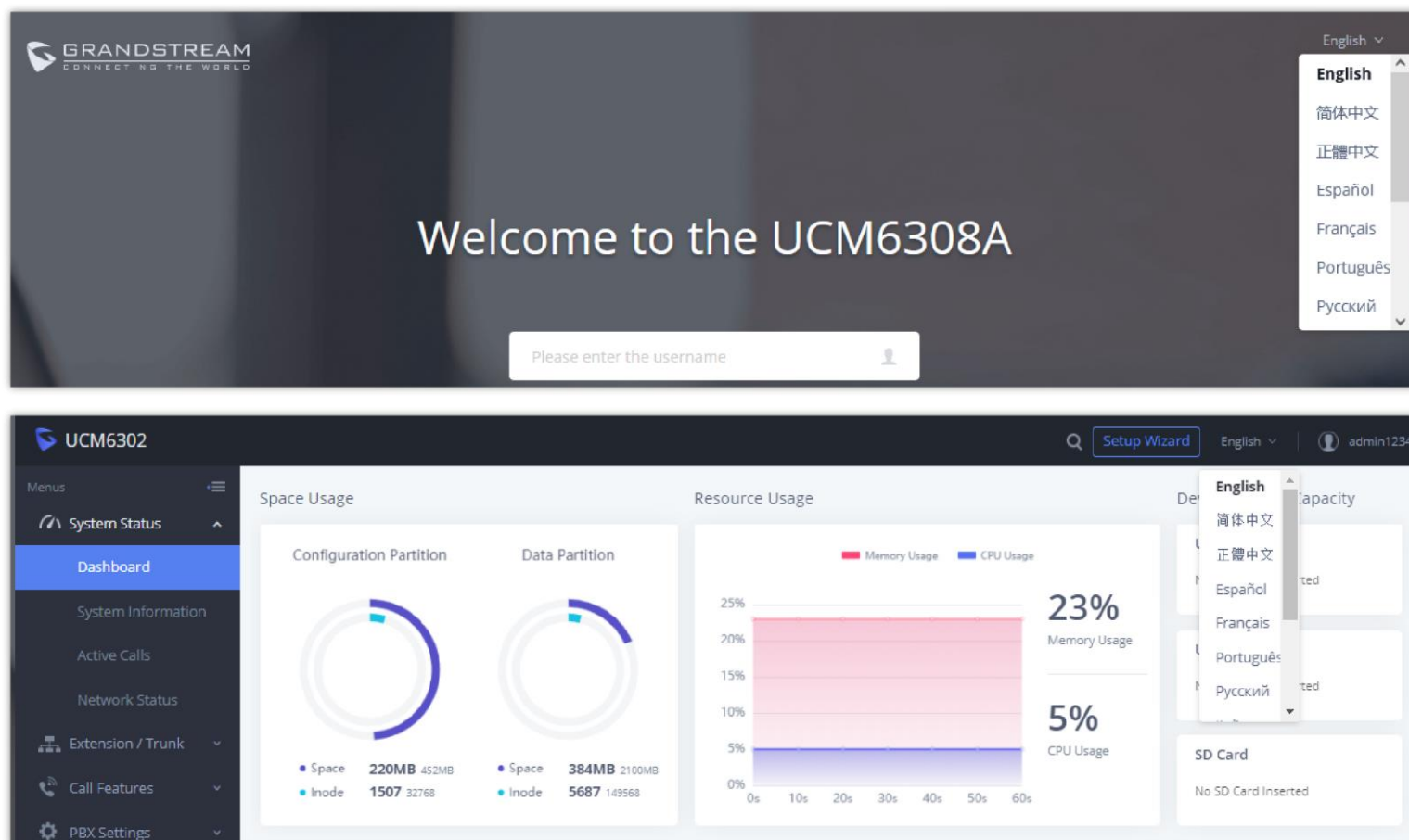
There are 8 main sections in the web portal to manage various features of the UCM.

- **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.
- **Other 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.**

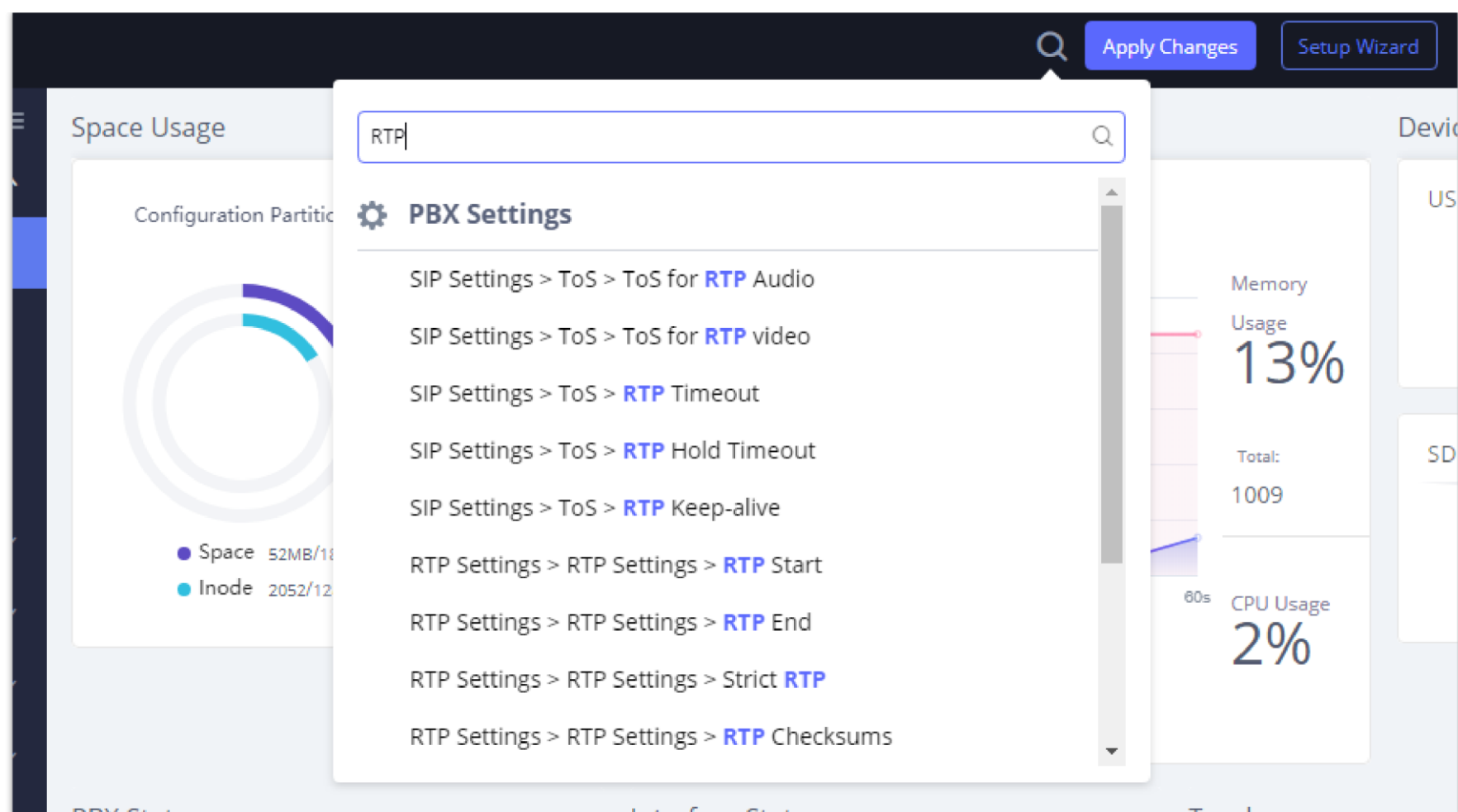
Users can select the UCM's web UI display language in the top-right corner of the page.



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.



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.

General Settings

Device Name:

Enable CPU Flow Control:

CPU Flow Control Threshold:

Data Partition Write Threshold:

General Settings Interface

General Settings	
Device Name	Configure the name of the UCM.
Enable CPU Flow Control	Enables the CPU flow control.
CPU Flow Control Threshold	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 90% .
Data Partition Write Threshold	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 90% .

General Settings Parameters

IM Settings

IM Settings

In IM Settings tab, the user can choose to enable or disable read receipts when exchanging messaging using Wave.

IM Settings

IM Settings
Cloud IM Service
IM Server

Read Receipts

New Message Email Notification [Email Template](#)

* Maximum chat file size(M)

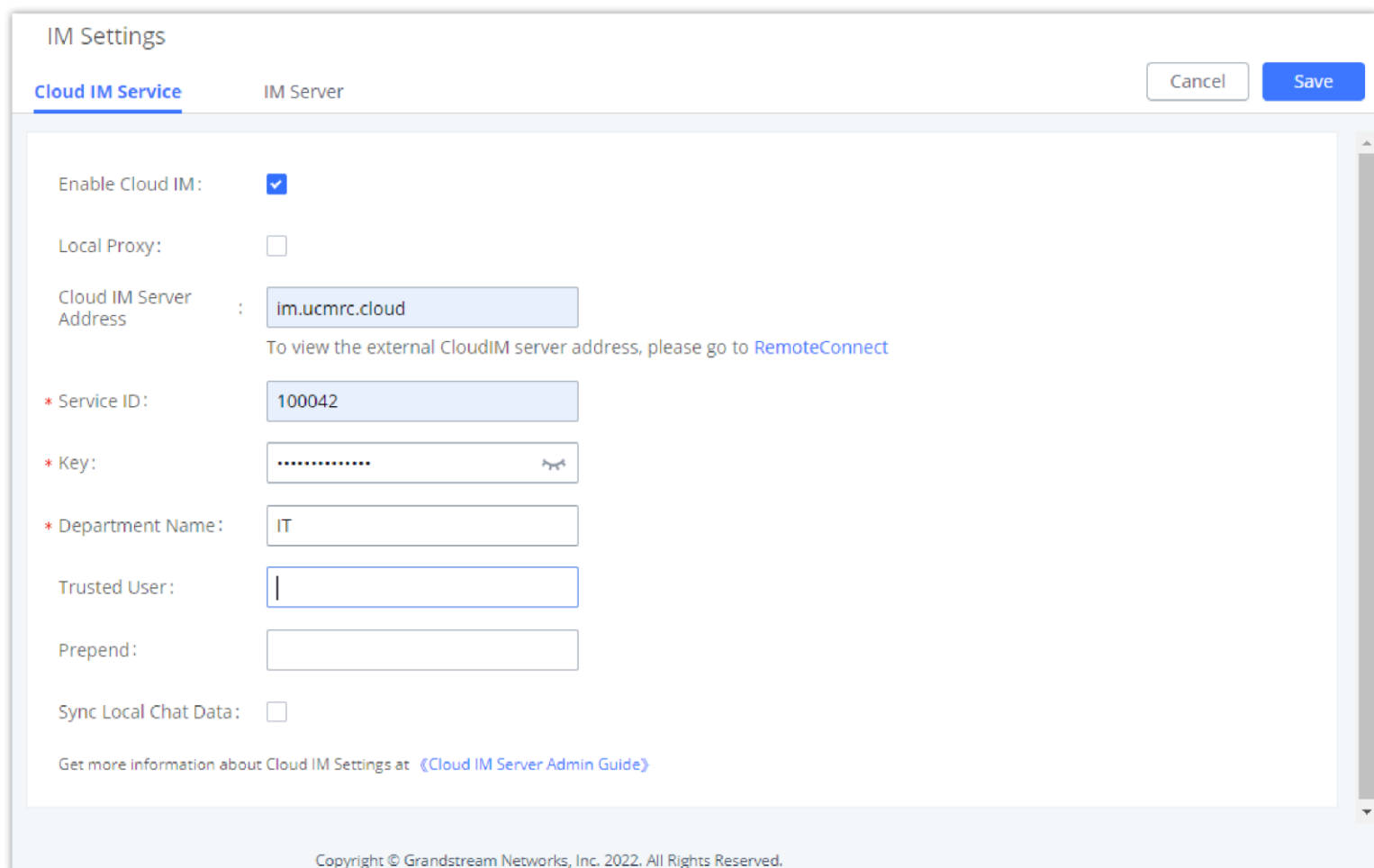
IM Settings

Read Receipts	<ol style="list-style-type: none"> 1. Configures whether Wave users can see the read status of sent messages when using local IM. 2. If using Cloud IM, read receipts must be configured on the IM server (GDMS or custom IM server) being used. To configure this on GDMS, navigate to the top right corner of the GDMS page Plan & Services->My Plans->Edit Cloud IM page.
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New Message Email Notification	Regardless of whether you are currently using local IM or Cloud IM, when Wave is offline under this domain for more than 7 days after enabling it, an email notification of new messages will be sent when a new message is received.
Maximum Chat File Size (MB)	<ol style="list-style-type: none"> 1. Configures whether Wave users can see the maximum chat file size when using local IM. 2. If using Cloud IM, maximum chat file size must be configured on the IM server (GDMS or custom IM server) being used. To configure this on GDMS, navigate to the top right corner of the GDMS page Plan & Services->My Plans->Edit Cloud IM page

Cloud IM Service


After enabling Cloud IM, it means that all IM data in Grandstream Wave is stored in the external server Cloud IM, and is no longer stored locally in UCM.GDMS can configure Cloud IM service for UCM devices. At this time, the UCM device synchronizes the configuration item information.



Cloud IM

Cloud IM Service	
Enable Cloud IM	If you have purchased the Cloud IM package or purchased the Grandstream IM server, you can configure it. If you have not purchased it, the configuration will not take effect, but PBX local IM service is allowed. Please note that after enabling this feature, local chat data will not be visible.
Local Proxy	If enabled, the local proxy will be used to forward files and text messages if the IM server cannot be connected to upon Wave login due to certificate issues.
Cloud IM Server Address	The address of the server that provides IM service, you can fill in the address of the Cloud IM server provided by the RemoteConnect package or the IM server address of the GDMS.
Service ID	The service ID of the Cloud IM server.
Key	The Key to the Cloud IM server.



Site Name	Enter the name of the site.
Trusted User	The trusted user of the cloud IM. Only letters, numbers, and special characters are allowed.
Prepend	As the extension prefix, it is added before the extension number.
Sync Local Chat Data	<p>Syncing existing local chat data to Cloud IM server. The Wave chat feature will not be available during the syncing process. It is recommended to avoid syncing during active working hours.</p> <p>- Time Range</p> <ul style="list-style-type: none"> • All • Last 12 Months • Last 6 Months • Last 3 Months • Last Month <p>- Data Type</p> <ul style="list-style-type: none"> • IM Data • Images • Files

 Only account details and department information will be synced on local IM and cloud IM. Other configurations such as profile picture, work status and favorite contacts will not be synced, and these are stored in local IM or cloud IM respectively. Therefore, please be aware that when switching between local IM and cloud IM, part of the data cannot be synced and the previously stored data on local IM or cloud IM (depending on which one is switched to) will be retrieved.

IM Server




If Enable IM Server Mode is toggled on, UCM will function only as an IM server. The UCM management portal will remove PBX related services and supports the binding of multiple cross-region UCM devices. The UCM device that wants to bind the IM server address is also bound by turning on the Cloud IM mode, and the IM data in his Grandstream Wave is stored in this IM server.

IM Server

Company Name:	GStest	Trusted User:	C074AD0A8C94
Server Address:	c074ad0a8c94-10671.b.gdms.cloud 	Service ID:	100001
Key:	2322458e635c4b98871b37b5079087c6 		

[Close IM Server Mode](#)

Bound Device Information

DEPARTMENT 	MAC ADDRESS 	DIAL PREFIX 
No Data		

IM server configuration interface

Company name	The entered company name
Server Address	The domain name or IP address of the Cloud IM server.
Service ID	The service ID of the Cloud IM server.
Key	The Key of the Cloud IM server.
Trusted User	The trusted user of the cloud IM. Only letter, number, and special characters are allowed.
Bound device information	
Department	The department represented by the bound UCM.

MAC Address	MAC address of the bound UCM device.
Dial prefix	Extension prefix

IM Server parameters

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**.

Redirect From Port 80	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".
Port	Specifies the port number used to access the UCM HTTP server. Default is "8089".
Enable IP Address Whitelist	If enabled, only the server addresses in the 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".
Permitted IP(s)	List of addresses that can access the UCM web portal.Ex: 192.168.6.233 / 255.255.255.255
External Host	Configure a URL and port (optional) used to access the UCM web portal or a public link to the conference room if the UCM is behind NAT.
Wave Settings	
External Host	Configure a URL and port (optional) used to access the UCM web portal or a public link to the conference room if the UCM is behind NAT.
Port	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	
Default Certificate Auto Renewal	If enabled, the default browser certificate will be automatically renewed after 398 days (the max certificate validity period of Chrome, Firefox, and Safari browsers). User-defined certificates are not affected.
Options	Selects the method of acquiring SSL certificates for the UCM web server. Two methods are currently available: Upload Certificate: Upload the appropriate files from one's own PC. Request Certificate: Enter the domain for which to request a certificate for from "Let's Encrypt".
TLS Private Key	Uploads the private key for the HTTP server. Note: Key file must be under 2MB in file size and *.pem format. The file name will automatically be changed to "private.pem".
TLS Cert	Uploads the certificate for the HTTP server. Note: Certificate must be under 2MB in file size and *.pem format. This will be used for TLS connections and contains a private key for the client and a signed certificate for the server.
Domain	Enter the domain to request the certificate for and click on "Request Certificate" button.

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.

Method	<p>Select "Route", "Switch" or "Dual" mode on the network interface of UCM630X Audio Series. The default setting is "Switch".</p> <ul style="list-style-type: none"> ● Route: WAN port will be used for the uplink connection. LAN port will function similarly to a regular router port. ● Switch: WAN port will be used for the uplink connection. LAN port will be used as a bridge for connections. ● Dual: Both WAN and LAN ports will be used for uplink connections labeled as LAN2 and 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.
MTU	Specifies the maximum transmission unit value. Default is 1492.
IPv4 Address	
Preferred DNS Server	If configured, this will be used as the Primary DNS server.
WAN (when "Method" is set to "Route")	
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
Subnet Mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
Gateway IP	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
DNS Server 1	Enter the DNS server 1 address for static IP settings.
DNS Server 2	Enter the DNS server 2 address for static IP settings.
Username	Enter the username to connect via PPPoE.
Password	Enter the password to connect via PPPoE.
Layer 2 QoS 802.1Q/VLAN Tag	Assign the VLAN tag of the layer 2 QoS packets for the WAN port. The default value is 0.
Layer 2 QoS 802.1p Priority Value	Assign the priority value of the layer 2 QoS packets for the WAN port. The default value is 0.
LAN (when Method is set to "Route")	
IP Address	Enter the IP address assigned to the LAN port. The default setting is 192.168.2.1.
Subnet Mask	Enter the subnet mask. The default setting is 255.255.255.0.
DHCP Server Enable	Enable or disable DHCP server capability. The default setting is "Yes".

DNS Server 1	Enter DNS server address 1. The default setting is 8.8.8.8.
DNS Server 2	Enter DNS server address 2. The default setting is 208.67.222.222.
Allow IP Address From	Enter the DHCP IP Pool starting address. The default setting is 192.168.2.100.
Allow IP Address To	Enter the DHCP IP Pool ending address. The default setting is 192.168.2.254.
Default IP Lease Time	Enter the IP lease time (in seconds). The default setting is 43200.
LAN (when Method is set to "Switch")	
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
Subnet Mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
Gateway IP	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
DNS Server 1	Enter the DNS server 1 address for static IP settings.
DNS Server 2	Enter the DNS server 2 address for static IP settings.
Username	Enter the username to connect via PPPoE.
Password	Enter the password to connect via PPPoE.
Layer 2 QoS 802.1Q/VLAN Tag	Assign the VLAN tag of the layer 2 QoS packets for the LAN port. The default value is 0.
Layer 2 QoS 802.1p Priority Value	Assign the priority value of the layer 2 QoS packets for the LAN port. The default value is 0.
LAN 1 / LAN 2 (when Method is set to "Dual")	
Default Interface	If "Dual" is selected as "Method", users will need to assign the default interface to be LAN 1 (mapped to UCM6302 Audio Series WAN port) or LAN 2 (mapped to UCM6302 Audio Series LAN port) and then configure network settings for LAN 1/LAN 2. The default interface is LAN 2.
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
Subnet Mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
Gateway IP	Enter the gateway IP address for static IP settings when the port is assigned as the default interface. The default setting is 0.0.0.0.
DNS Server 1	Enter the DNS server 1 address for static IP settings.
DNS Server 2	Enter the DNS server 2 address for static IP settings.

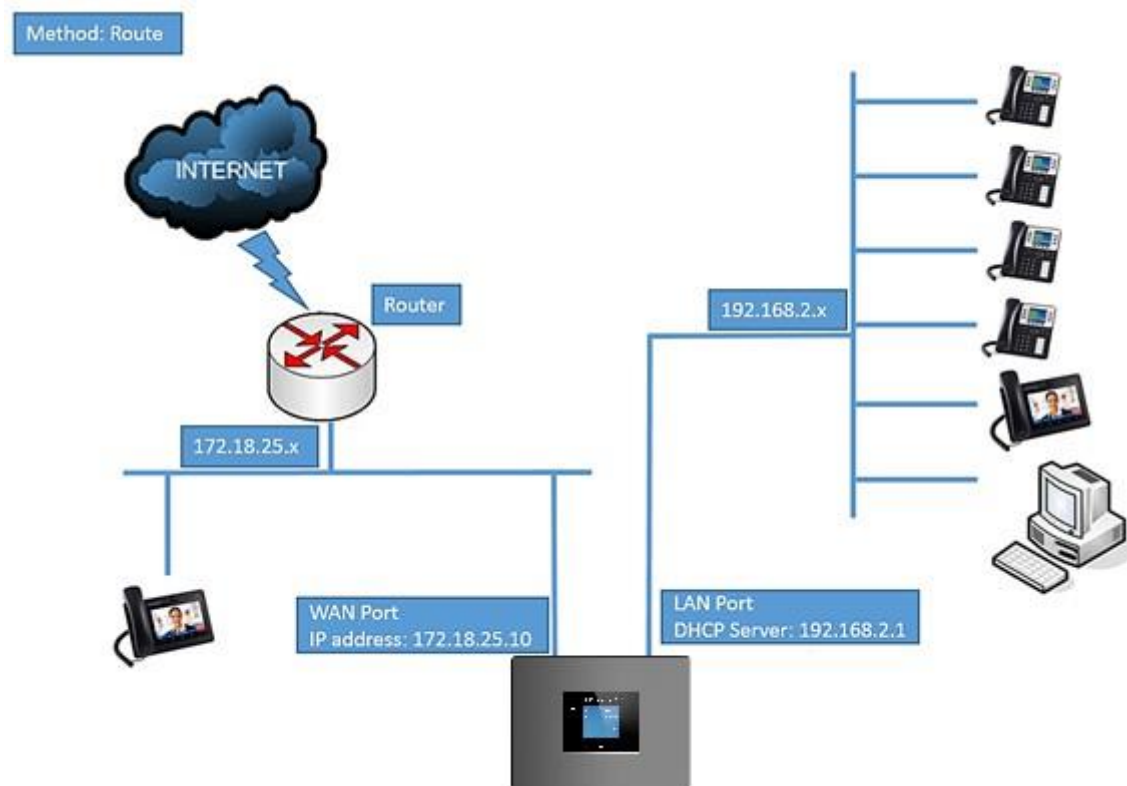
Username	Enter the username to connect via PPPoE.
Password	Enter the password to connect via PPPoE.
Layer 2 QoS 802.1Q/VLAN Tag	Assign the VLAN tag of the layer 2 QoS packets for the LAN port. The default value is 0.
Layer 2 QoS 802.1p Priority Value	Assign the priority value of the layer 2 QoS packets for the LAN port. The default value is 0.
IPv6 Address	
WAN (when "Method" is set to "Route")	
IP Method	Select Auto or Static. The default setting is Auto
IP Address	Enter the IP address for static IP settings.
IP Prefixlen	Enter the Prefix length for static settings. Default is 64
DNS Server 1	Enter the DNS server 1 address for static settings.
DNS Server 2	Enter the DNS server 2 address for static settings.
LAN (when Method is set to "Route")	
DHCP Server	Select Disable, Auto, or DHCPv6. <ul style="list-style-type: none"> ● Disable: the DHCPv6 server is disabled. ● Auto: Stateless address auto configuration using NDP protocol. ● DHCPv6: Stateful address auto configuration using DHCPv6 protocol. The default setting is Disabled.
DHCP Prefix	Enter DHCP prefix. (Default is 2001:db8:2:2::)
DHCP prefixlen	Enter the Prefix length for static settings. Default is 64
DNS Server 1	Enter the DNS server 1 address for static settings. Default is (2001:4860:4860::8888)
DNS Server 2	Enter the DNS server 2 address for static settings. Default is (2001:4860:4860::8844)
Allow IP Address From	Configure starting IP address assigned by the DHCP prefix and DHCP prefixlen.
Allow IP Address To	Configure the ending IP address assigned by the DHCP Prefix and DHCP prefixlen.
Default IP Lease Time	Configure the lease time (in second) of the IP address.
LAN (when Method is set to "Switch")	
IP Method	Select Auto or Static. The default setting is Auto
IP Address	Enter the IP address for static IP settings.
IP Prefixlen	Enter the Prefix length for static settings. Default is 64

DNS Server 1	Enter the DNS server 1 address for static settings.
DNS Server 2	Enter the DNS server 2 address for static settings.
LAN 1 / LAN 2 (when Method is set to "Dual")	
Default Interface	Users will need to assign the default interface to be LAN 1 (mapped to UCM630X Audio Series WAN port) or LAN 2 (mapped to UCM630X Audio Series LAN port) and then configure network settings for LAN 1/LAN 2. The default interface is LAN 1.
IP Method	Select Auto or Static. The default setting is Auto
IP Address	Enter the IP address for static IP settings.
IP Prefixlen	Enter the Prefix length for static settings. Default is 64
DNS Server 1	Enter the DNS server 1 address for static settings.
DNS Server 2	Enter the DNS server 2 address for static settings.
Network Port Traffic Control	
LAN (when Method is set to "Switch")	
Enable Network Port Traffic Storm Alert	The UCM will send a an alert notification/email when there is an excessive number of packets in the LAN that impacts the overall performance of the network. Note: To enable this feature email or HTTP notification should be set up correctly In Maintenance → System Events.
Ignore Safe Operational Flow	When enabled, it will ignore traffic storm alarms triggered by users' own operations after logging in, including security operations such as firmware upgrades, uploading backup files, beeping, zero-configuration templates/firmware, record files, and downloading files from network disks.
Network Port Receiving Traffic Control	You can monitor the traffic in the RX direction on each network port and generate an alarm when the corresponding alarm event is turned on and the set threshold value is exceeded. The threshold range is 1 - 1024000 in kbps and 1 - 1000 in mbps.
LAN 1 & LAN 2 (when Method is set to "Dual")	
Enable Network Port Traffic Storm Alert	The UCM will send a an alert notification/email when there is an excessive number of packets in the LAN that impacts the overall performance of the network. Note: To enable this feature email or HTTP notification should be set up correctly In Maintenance → System Events.
Ignore Safe Operational Flow	When enabled, it will ignore traffic storm alarms triggered by users' own operations after logging in, including security operations such as firmware upgrades, uploading backup files, beeping, zero-configuration templates/firmware, recording files, and downloading files from network disks
LAN1 & LAN2 - Network Port Receiving Traffic Control	You can monitor the traffic in the RX direction on each network port and generate an alarm when the corresponding alarm event is turned on and the set threshold value is exceeded. The threshold range is 1 - 1024000 in kbps and 1 - 1000 in mbps.

LAN & WAN (When Method is set to Route Mode)	
Enable Network Port Traffic Storm Alert	The UCM will send a an alert notification/email when there is an excessive number of packets in the LAN that impacts the overall performance of the network. Note: To enable this feature email or HTTP notification should be set up correctly In Maintenance → System Events .
Ignore Safe Operational Flow	When enabled, it will ignore traffic storm alarms triggered by users' own operations after logging in, including security operations such as firmware upgrades, uploading backup files, beeping, zero-configuration templates/firmware, recording files, and downloading files from network disks.
WAN: Network Port Receiving Traffic Control	You can monitor the traffic in the RX direction on each network port and generate an alarm when the corresponding alarm event is turned on and the set threshold value is exceeded. The threshold range is 1 - 1024000 in kbps and 1 - 1000 in mbps.
LAN: Network Port Receiving Traffic Control	You can monitor the traffic in the RX direction on each network port and generate an alarm when the corresponding alarm event is turned on and the set threshold value is exceeded. The threshold range is 1 - 1024000 in kbps and 1 - 1000 in mbps.

o **Method: Route**

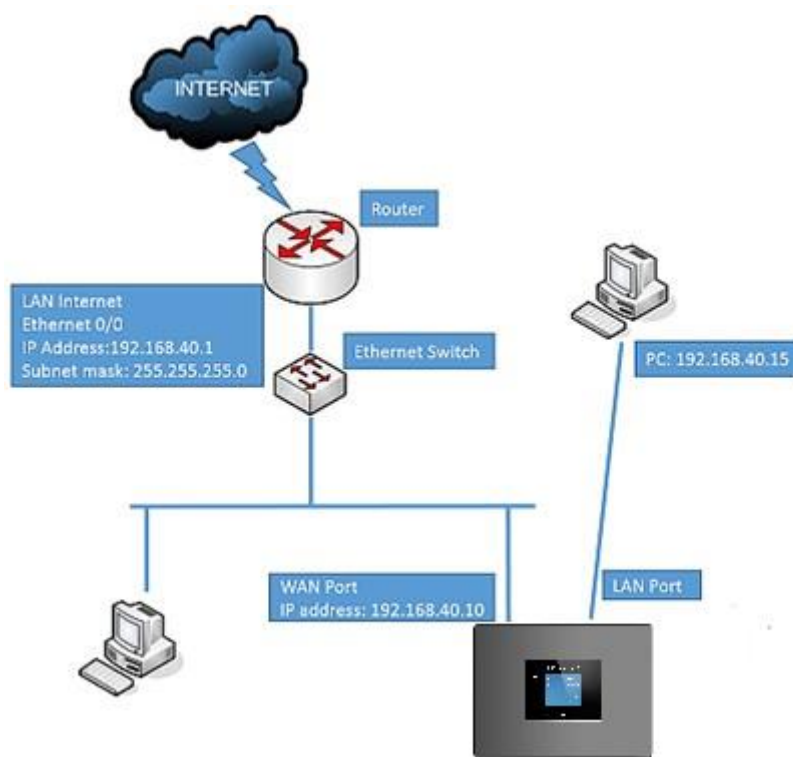
When the UCM630xA has, method set to Route in network settings, WAN port interface is used for uplink connection and LAN port interface is used as a router. Please see a sample diagram below.



UCM6302A Network Interface Method: Route

o **Method: Switch**

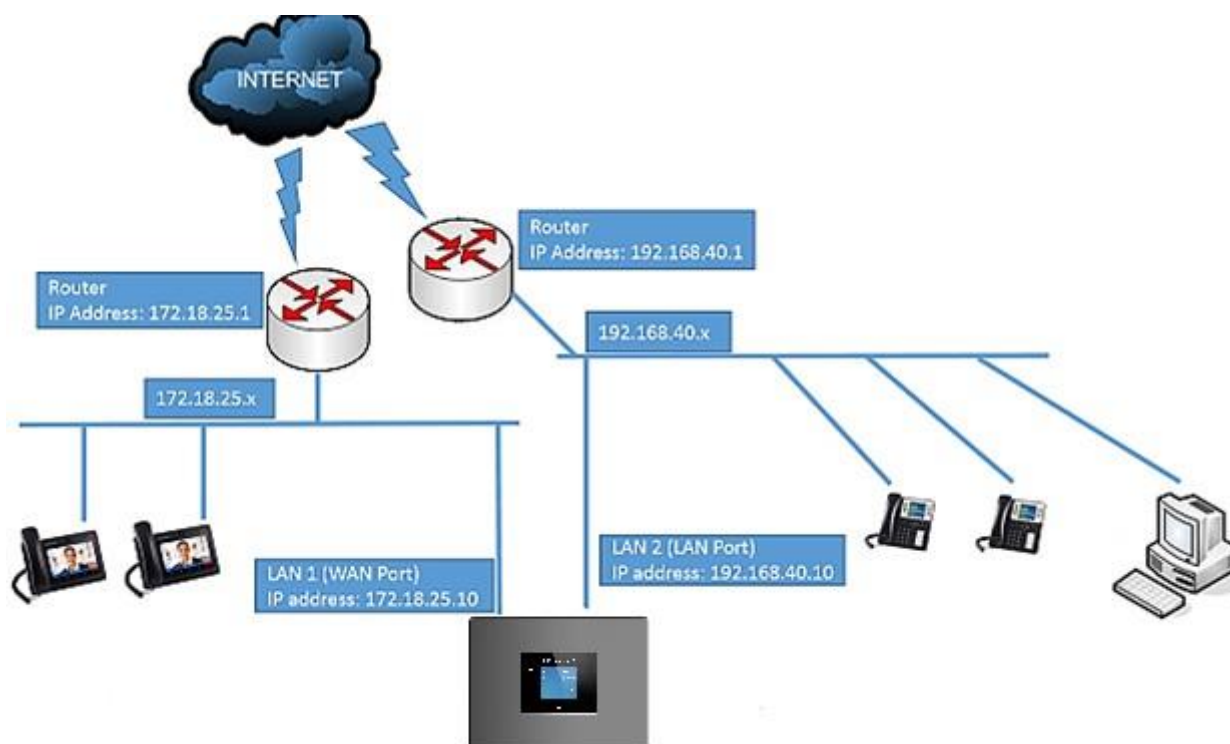
WAN port interface is used for uplink connection; LAN port interface is used as room for PC connection.



UCM6302A Network Interface Method: Switch

o **Method: Dual**

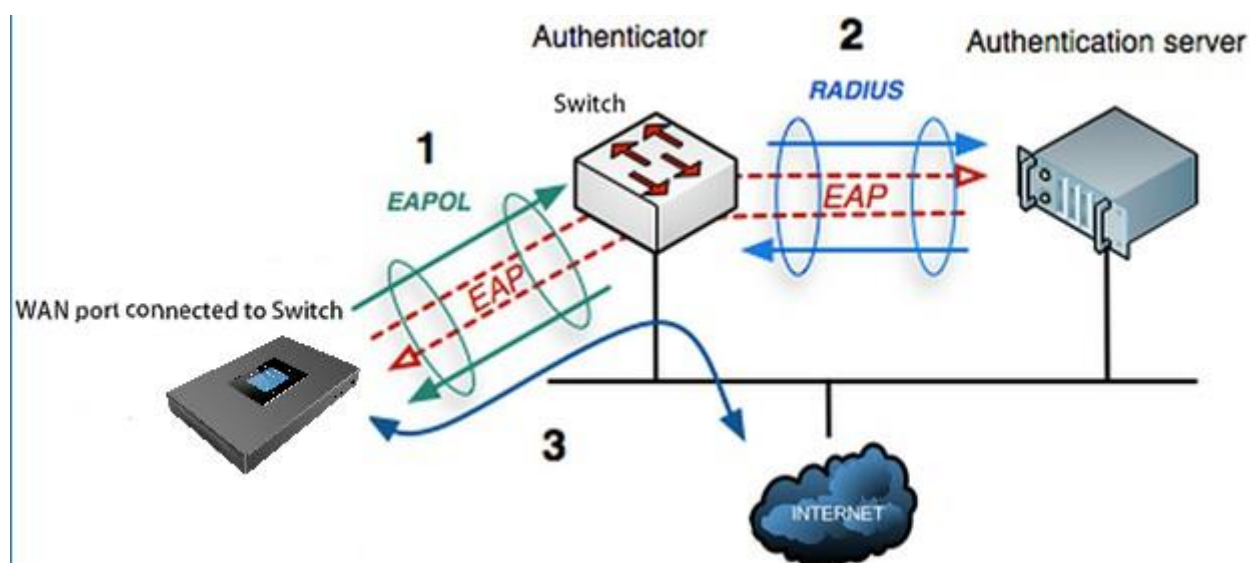
Both WAN port and LAN port are used for uplink connection. Users will need assign LAN 1 or LAN 2 as the default interface in option "Default Interface" and configure "Gateway IP" if static IP is used for this interface.



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.



The screenshot shows the 'Network Settings' interface with the '802.1X Settings' tab selected. The configuration includes:

- 802.1X Mode:** A dropdown menu set to 'EAP-MD5'.
- * Identity:** A text input field containing '8021xxUCM6302'.
- * MD5 Password:** A text input field with masked characters (dots) and a small eye icon to toggle visibility.

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.

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

UCM630xA Network Settings → 802.1X

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.

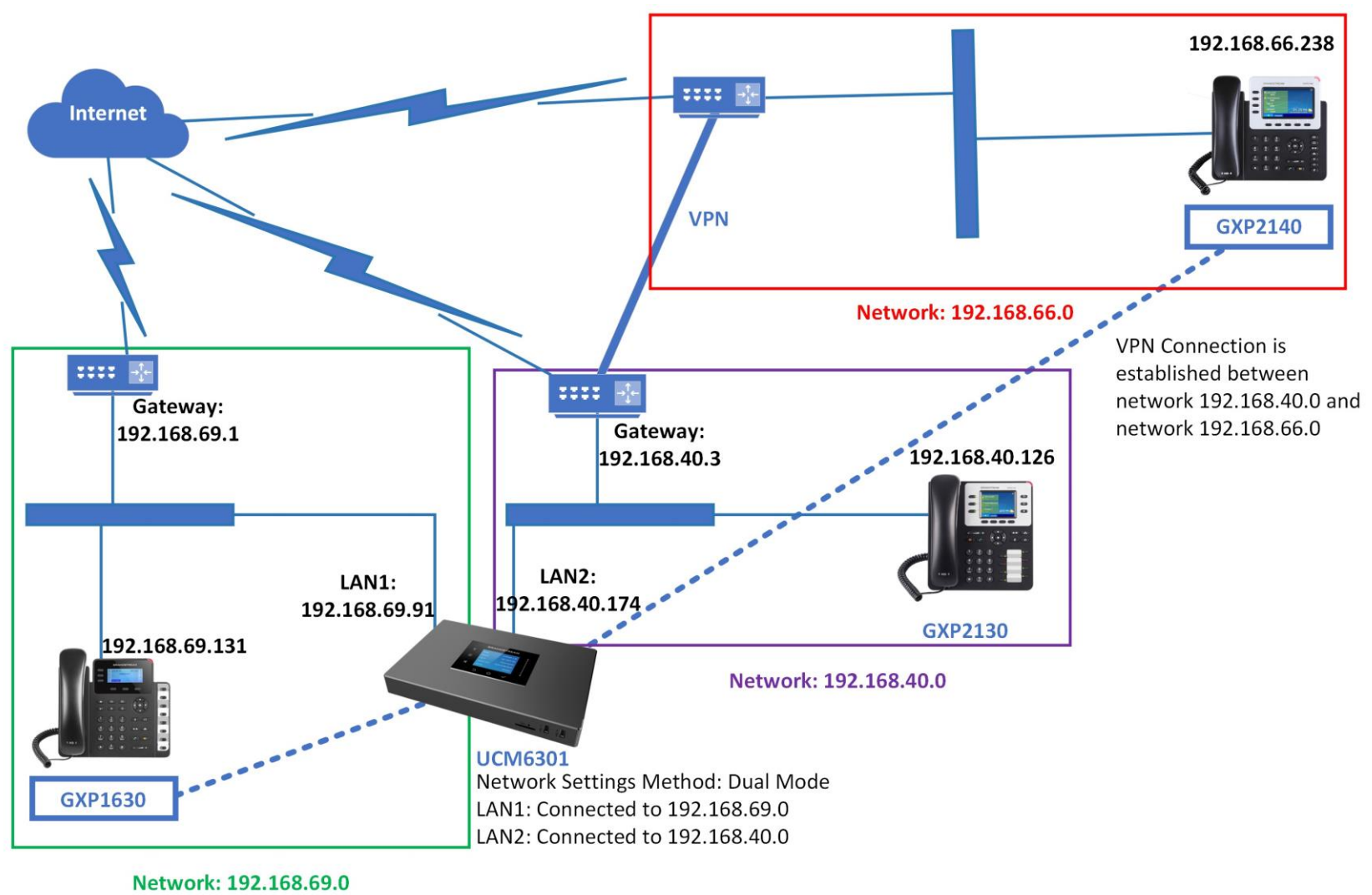
- Click on **“Add IPv4 Static Route”** to create a new IPv4 static route or click on **“Add IPv6 Static Route”** to create a new IPv6 static route. The configuration parameters are listed in the table below.
- Once added, users can select  to edit the static route.
- Select  to delete the static route.

Destination	<p>Configure the destination IPv4 address or the destination IPv6 subnet for the UCM630xA to reach using the static route.</p> <p>Example:</p> <p>IPv4 address – 192.168.66.4</p> <p>IPv6 subnet – 2001:740:D::1/64</p>
Subnet Mask	<p>Configure the subnet mask for the above destination address. If left blank, the default value is 255.255.255.255.</p> <p>Example:</p> <p>255.255.255.0</p>
Gateway	<p>Configure the IPv4 or IPv6 gateway address so that the UCM630xA can reach the destination via this gateway. Gateway address is optional.</p> <p>Example:</p> <p>192.168.40.5 or 2001:740:D::1</p>
Interface	<p>Specify the network interface on the UCM630xA to reach the destination using the static route.</p> <p>LAN interface is eth0; WAN interface is eth1.</p>

UCM630xA Network Settings → Static Routes

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

The following diagram shows a sample application of static route usage on UCM6304A.



--- GXP1630 and GXP2140 call each other

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:	<input type="text" value="192.168.66.0"/>
Subnet Mask:	<input type="text" value="255.255.255.0"/>
Gateway:	<input type="text" value="192.168.40.3"/>
* Protocol Type:	<input type="text" value="WAN"/>

UCM6304A Static Route Configuration

Port Forwarding

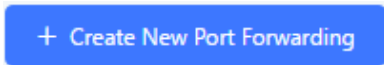
The UCM network interface supports router function which provides users the ability to do port forwarding. If LAN mode is set to "Route" under Web GUI→**System Settings**→**Network Settings**→**Basic Settings** page, port forwarding is available for configuration.

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

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

UCM630xA Network Settings →Port Forwarding

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.

Create New Port Forwarding

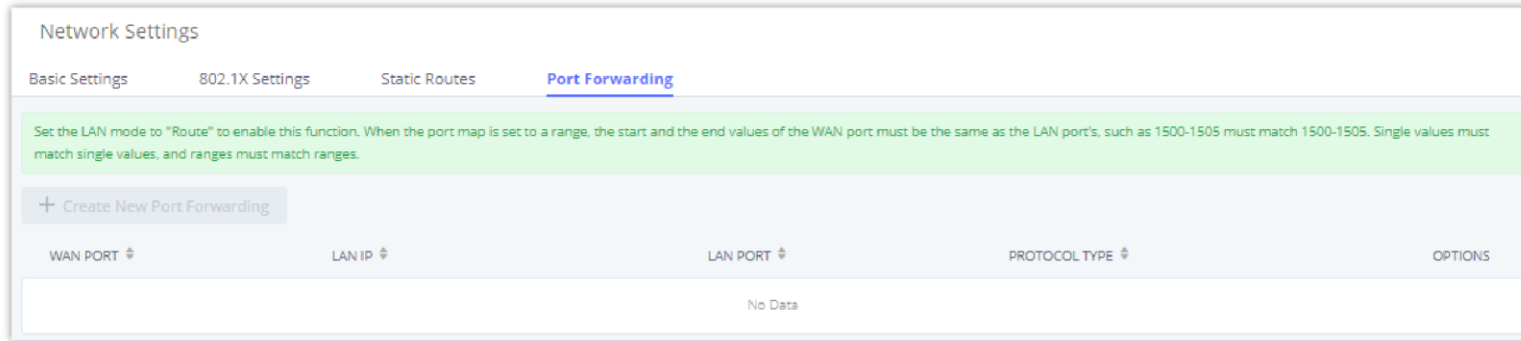
* WAN Port:

* LAN IP:

* LAN Port:

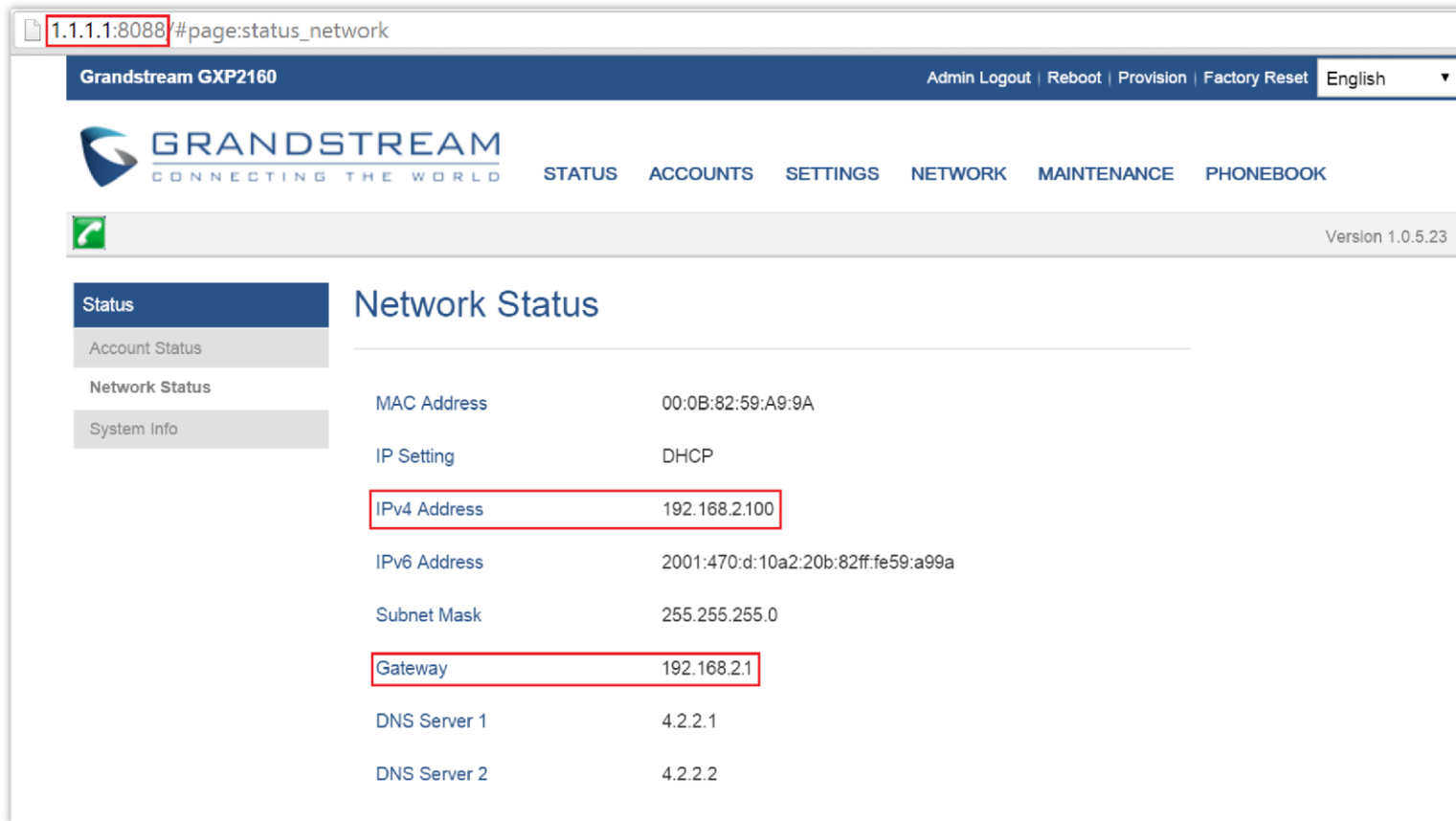
* Protocol Type: ▼

Create New Port Forwarding



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).



GXP2160 Web Access using UCM6302A Port Forwarding

ARP Settings

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

ARP GC Threshold 1	Minimum number of entries to keep. Garbage collector will not purge entries if there are fewer than this number. The default value is 128.
ARP GC Threshold 2	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.
ARP GC Threshold 3	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.

ARP Settings

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.

• OpenVPN uses TLS version 1.2. Please make sure that the OpenVPN server has the same TLS version, otherwise the connection will fail.

OpenVPN® Enable:

Configuration Method:

OpenVPN® Server Address:

OpenVPN® Server Protocol:

OpenVPN® Device Mode:

OpenVPN® Use Compression:

Allow Weak SSL Ciphers:

OpenVPN® Encryption Algorithm:

OpenVPN® CA Cert:

OpenVPN® Client Cert:

OpenVPN® Client Key:

User Authentication:

Username:

Password:

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OpenVPN® Feature on the UCM630xA

OpenVPN® Enable	Enable / Disable the OpenVPN® feature.
Configuration Method	Select the OpenVPN® configuration method. Manual Configuration: Allows to configure OpenVPN® settings manually. Upload Configuration File: Allows to upload .ovpn and .conf files to the UCM and to automatically configure OpenVPN® settings.
OpenVPN® Server Address	Configures the hostname/IP and port of the server. For example 192.168.1.2:22
OpenVPN® Server Protocol	Specify the protocol user, user should use the same settings as used on the server
OpenVPN® Device mode	Use the same setting as used on the server. <ul style="list-style-type: none"> • Dev TUN: Create a routed IP tunnel. • Dev TAP: Create an Ethernet tunnel.
OpenVPN® Use Compression	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.
Enable Weak SSL Ciphers	Either to enable the Weak SSL ciphers or not.
OpenVPN® Encryption Algorithm	Specify the cryptographic cipher. Users should make sure to use the same setting that they are using on the OpenVPN server.
OpenVPN® CA Cert	Upload as SSL/TLS root certificate. This file will be renamed as 'ca.crt' automatically.
OpenVPN® Client Cert	Upload a client certificate. This file will be renamed as 'client.crt' automatically.
OpenVPN® Client Key	Upload a client private key. This file will be renamed as 'client.key' automatically.

Username	Username used to authenticate into the server.
Password	Password used to authenticate into the server.

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 -"/> ✔ Configure Groups
Enable Wildcard:	Wildcards are a Plus / Enhanced feature. Upgrade Now! ✔
Advanced Records:	TXT, SPF, and SRV records and the use of some special clients are Plus / Enhanced features. Upgrade now to use them. ✔

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.

UCM6302

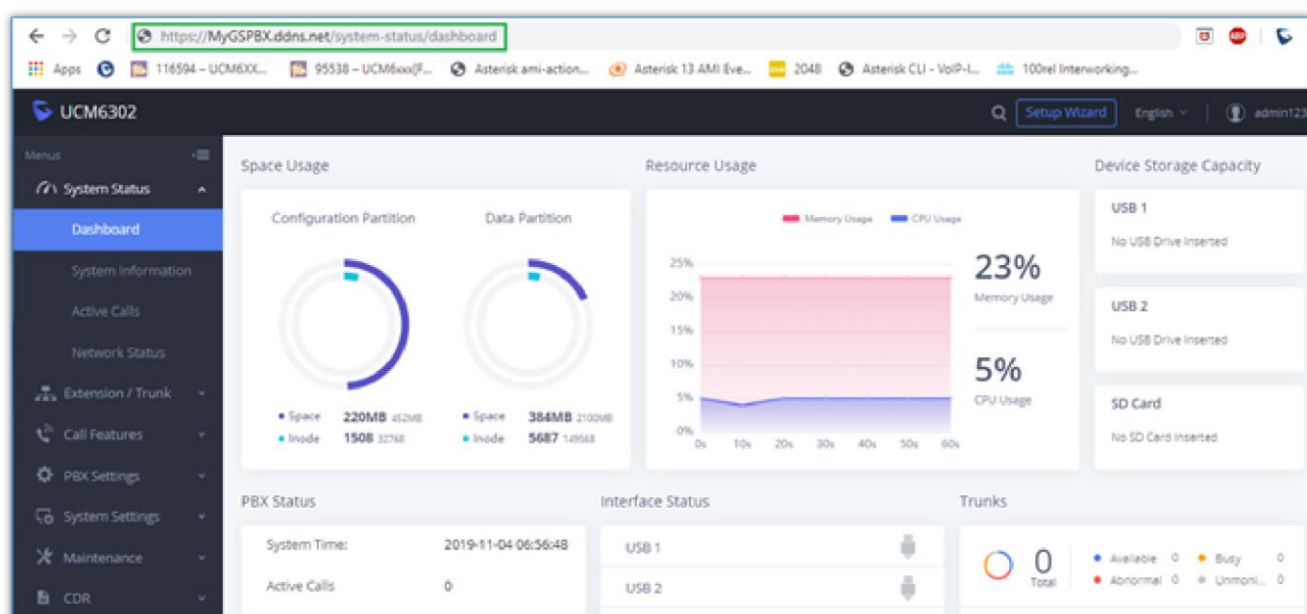
Menus

- System Status
- Extension / Trunk
- Call Features
- PBX Settings
- System Settings
- HTTP Server
- Network Settings
- OpenVPN®
- DDNS Settings

DDNS Settings

DDNS Server:	<input type="text" value="no-ip.com"/>
Enable DDNS:	<input checked="" type="checkbox"/>
* Username:	<input type="text" value="user_no_ip"/>
* Password:	<input type="password" value="....."/>
* Host Name:	<input type="text" value="MyGSPBX.ddns.net"/>

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



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.

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

Port	Process	Type	Protocol or Service
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

UCM630xA Firewall → Static Defense → Current Service

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

Ping Defense Enable	If enabled, ICMP response will not be allowed for Ping request. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (UCM630xA) interface.
SYN-Flood Defense Enable	<p>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.</p> <p>eth(0)LAN defends against attacks directed to the LAN IP address of the UCM630xA.</p> <p>eth(1)WAN defends against attacks directed to the WAN IP address of the UCM630xA.</p> <p>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.</p>
Ping-of-Death Defense Enable	Enable to prevent Ping-of-Death attack to the device. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (UCM630xA) interface.

Typical Firewall Settings

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)

- o Protocol is not UDP

Create New Firewall Rule

* Rule Name:

* Action:

* Type:



* Interface:

* Service:

Create New Firewall Rule

Rule Name	Specify the Firewall rule name to identify the firewall rule.
Action	<p>Select the action for the Firewall to perform.</p> <ul style="list-style-type: none"> <input type="radio"/> ACCEPT <input type="radio"/> REJECT <input type="radio"/> DROP
Type	<p>Select the traffic type.</p> <ul style="list-style-type: none"> <input type="radio"/> IN <p>If selected, users will need specify the network interface "LAN" or "WAN" (for UCM630xA) for the incoming traffic.</p> <ul style="list-style-type: none"> <input type="radio"/> OUT
Interface	Select the interface to use the Firewall rule.
Service	<p>Select the service type.</p> <ul style="list-style-type: none"> <input type="radio"/> FTP <input type="radio"/> SSH <input type="radio"/> Telnet <input type="radio"/> HTTP <input type="radio"/> LDAP <input type="radio"/> Custom <p>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".</p>
Source IP Address and Port	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).
Destination IP Address and Port	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).
Protocol	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.

Dynamic Defense Enable	Enable dynamic defense. The default setting is disabled.
Blacklist Update Interval	Configure the blacklist update time interval (in seconds). The default setting is 120.
Connection Threshold	Configure the connection threshold. Once the number of connections from the same host reaches the threshold, it will be added into the blacklist. The default setting is 100.
Dynamic Defense Whitelist	<p>Allowed IPs and ports range, multiple IP addresses and port range.</p> <p>For example:</p> <p>192.168.2.100-192.168.2.105, 1000:9999</p>

UCM630xA Firewall Dynamic Defense

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.

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.

Fail2ban Settings

Global Settings	
Enable Fail2Ban	Enable Fail2Ban. The default setting is disabled. Please make sure both "Enable Fail2Ban" and "Asterisk Service" are turned on to use Fail2Ban for SIP authentication on the UCM630X.
Banned Duration	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.

Max Retry Duration	Within this duration (in seconds), if a host exceeds the max times of retry as defined in "MaxRetry", the host will be banned. The default setting is 600.
MaxRetry	Configure the number of authentication failures during "Max Retry Duration" before the host is banned. The default setting is 5.
Fail2Ban Whitelist	Configure IP address, CIDR mask, or DNS host in the whitelist. Fail2Ban will not ban the host with a matching address in this list. Up to 50 addresses can be added to the list descriptions/comments can be added for each whitelist entry for admin to log what's the whitelist IP address is for.
Local Settings	
Asterisk Service	Enable Asterisk service for Fail2Ban. The default setting is disabled. Please make sure both "Enable Fail2Ban" and "Asterisk Service" are turned on to use Fail2Ban for SIP authentication on the UCM630X.
Listening Port Number	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.
MaxRetry	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".
Login Attack Defense	Enables defense against excessive login attacks to the UCM's web GUI. The default setting is disabled.
Listening Port Number	This is the Web GUI listening port number which is configured under System Settings→ HTTP Server→ Port. The default is 8089.
MaxRetry	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.
Customer Service System Call Defense	Enable call defense in the customer service system. Off by default.
Listening Port Number	The current service listening port. Default UDP port: 5060, TCP port: 5060, 5061, WebSocket communication port: 8088.
MaxRetry	Set the maximum number of calls allowed in the "time span". The local matching threshold has a higher priority than the global matching threshold. The default setting is 5.
Blacklist	
Blacklist	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.

Tick "Enable remote SSH" 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.

The screenshot shows the 'Security Settings' page with four tabs: 'Static Defense', 'Dynamic Defense', 'Fail2Ban', and 'SSH Access'. The 'SSH Access' tab is selected. Under this tab, there are three settings: 'Enable SSH Access' with a checked checkbox, 'Enable Remote SSH' with a checked checkbox, and 'Access Token' with a text input field containing seven asterisks (*****).

SSH Access

Enable SSH Access	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.
Enable Remote SSH	If this option is enabled, remote SSH access will be allowed through the Feedback platform. It is strongly recommended to keep this disabled unless remote troubleshooting is necessary.
Access Token	Please enter the token to request SSH data.

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.

Field	Value	Additional Info
* Base DN:	dc=pbx,dc=com	
PBX DN:	ou=pbx	,dc=pbx,dc=com
Root DN:	cn=admin	,dc=pbx,dc=com
* Root Password:	Eye icon
* Confirm Root Password:	Eye icon
LDAP Cert:	server.crt	Reset Certificates
LDAP Private Key:	private.key	Reset Certificates
LDAP CA Cert:	server.ca	Reset Certificates

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.

PHONEBOOK DN	Count
ou=pbx,dc=pbx,dc=com	30

Default LDAP Phonebook DN

< Edit Phonebook: pbx

+ Add Contact

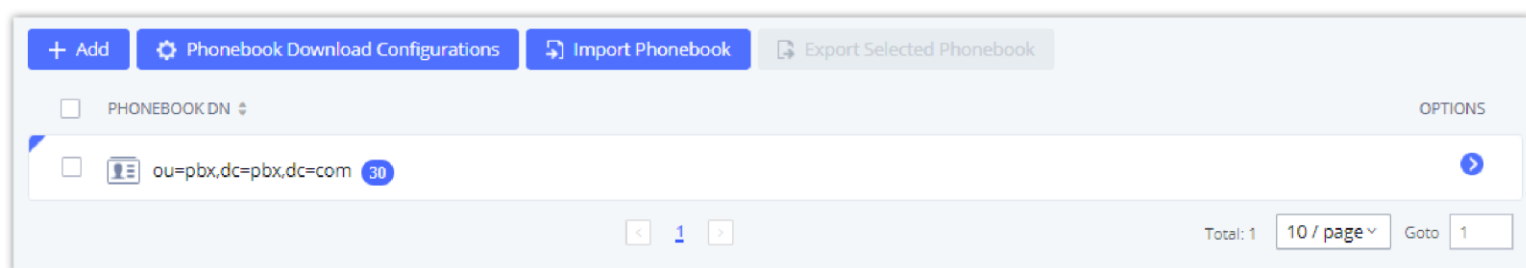
EXTENSION	CALLERID NAME	OPTIONS
1000		
1001		
1002		
1003		
1004		
1005		
1006		
1007		
1008		
1009		

Total: 30 10 / page Goto 1

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.



LDAP Server → LDAP Phonebook

o Add new phonebook

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

Add Phonebook ✕

* Phonebook Prefix:

Phonebook DN:

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

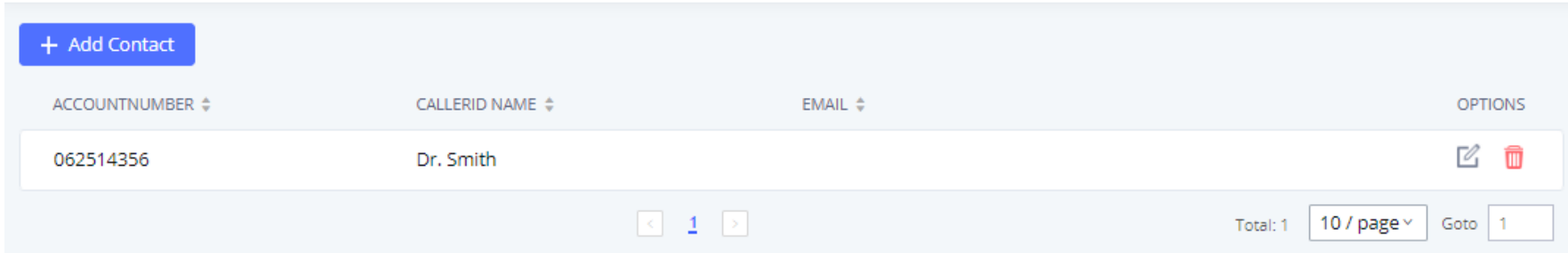


to edit the phonebook attributes and contact list (see figure below) or select



to delete the phonebook.

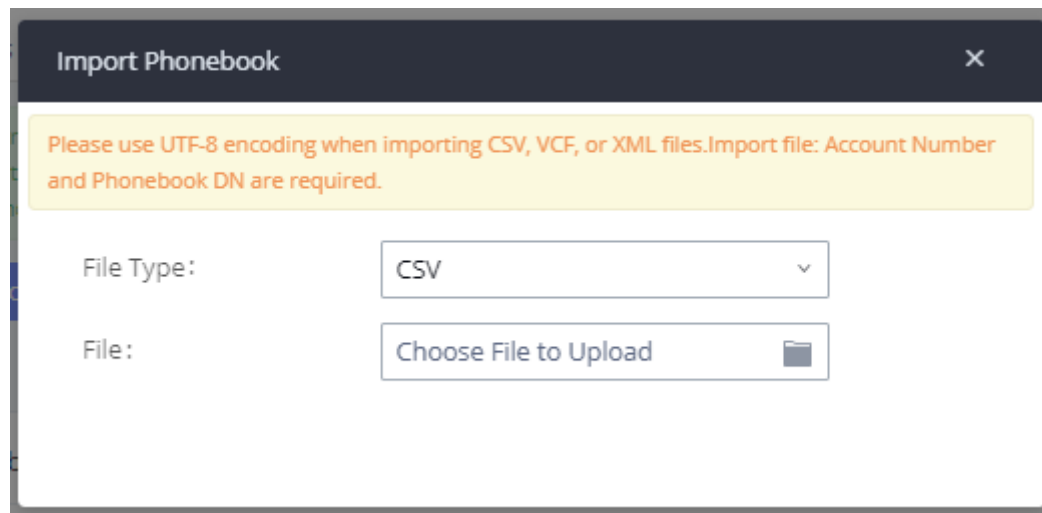
< Edit Phonebook: GStest



Edit LDAP Phonebook

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

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



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.

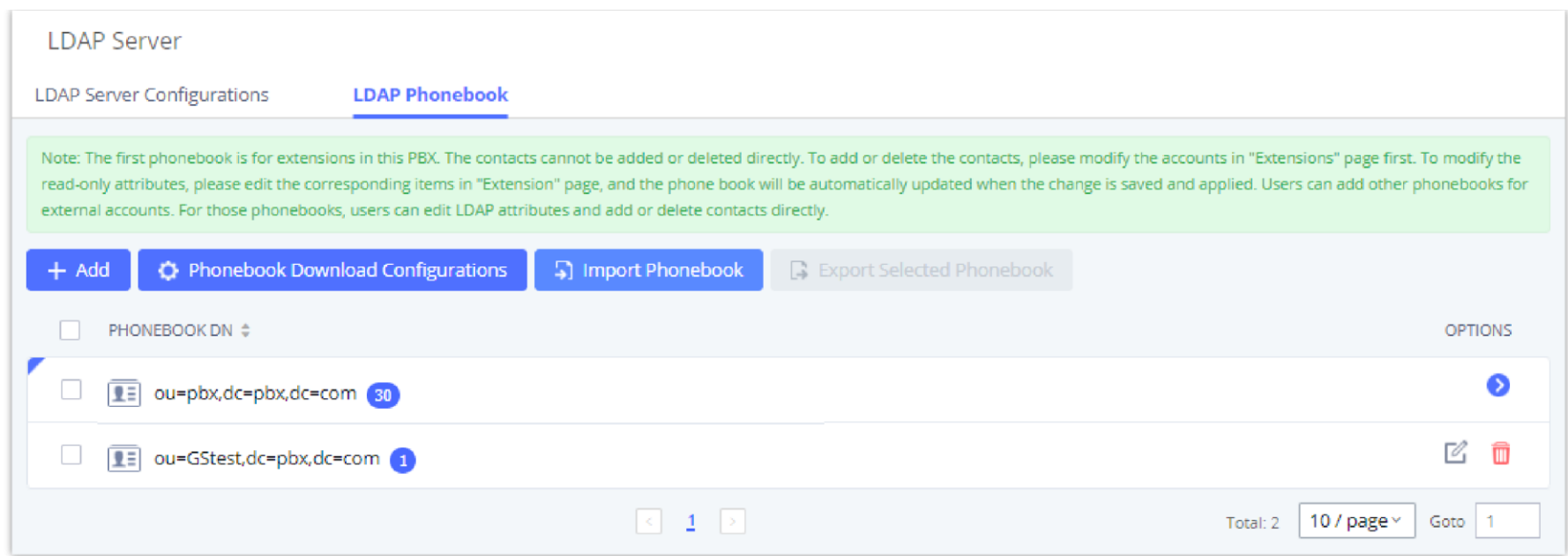
	A	B	C	D	E	F	G	H	I	J
1	First Name	Last Name	Account Number	CallerID Name	Email	Department	Mobile Number	Home Number	Fax	Phonebook DN
2	John	Doe	1001	1001		IT	1001000000			phonebook
3	Jane	Doe	1002	1002		Sales	1002000000			phonebook
4	William	Chung	1003	1003		Marketing	1003000000			phonebook
5	Linda	Kuo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

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.

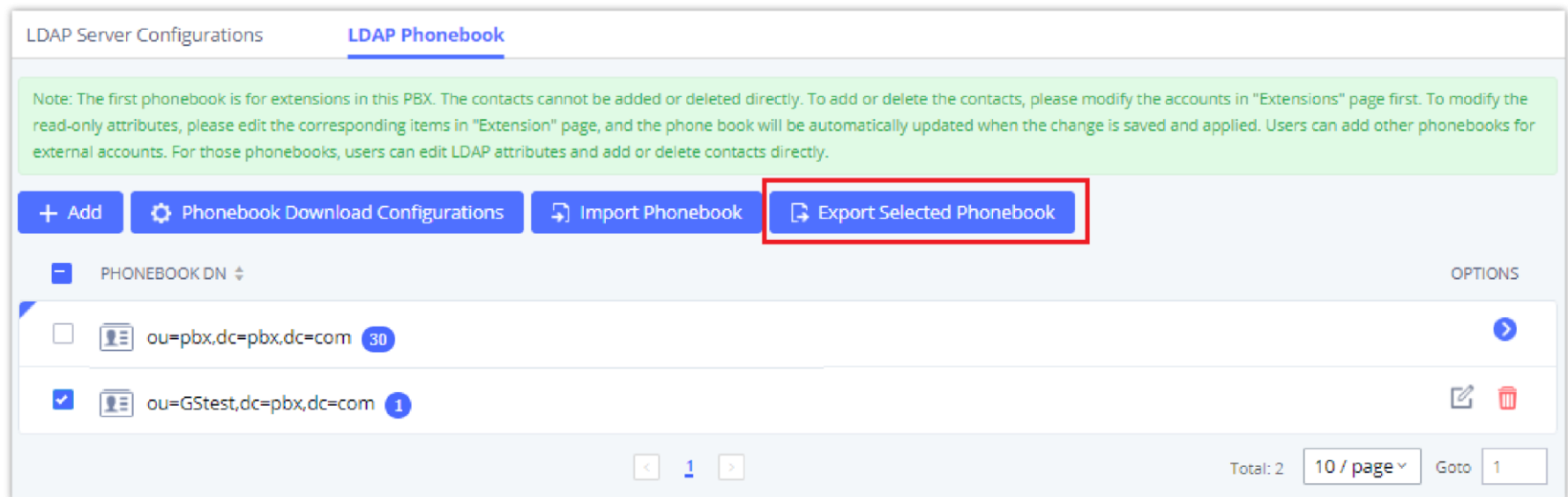


LDAP Phonebook After Import

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.



Export Selected LDAP Phonebook

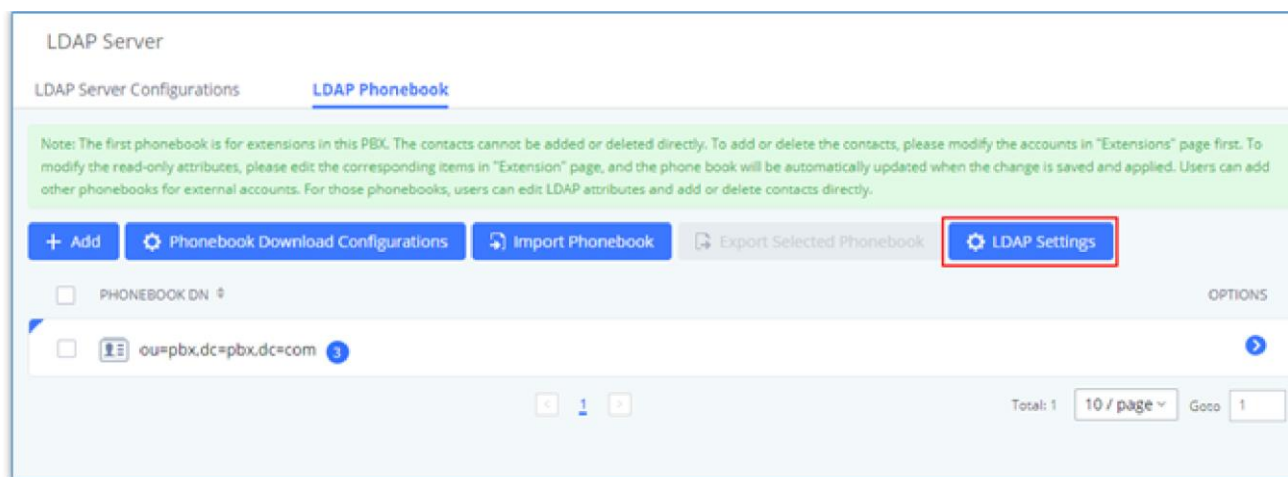
LDAP Settings

Prerequisites to support contacts sync-up to IP Phones, UCM needs to support the following:

1. If Cloud IM is enabled, UCM can send remote UCM's contacts to each end device.
2. Contacts from remote UCM can be synced by Cloud IM or LDAP sync via trunk. The contacts data must be complete and consistent.
3. If Cloud IM is enabled, the contacts sent from UCM to end device should integrate Cloud IM contacts.
4. If Cloud IM is disabled, the contacts sent from UCM to end device should only contain contacts on the UCM.

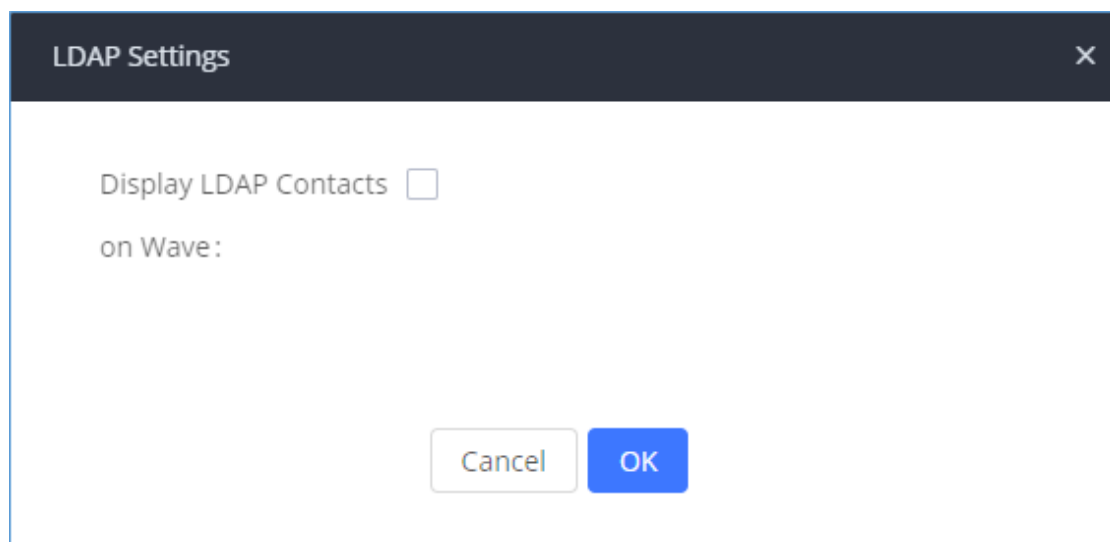
To support contacts sync-up to Wave, it allows Wave to obtain enterprise contacts from Cloud IM or LDAP. On UCM SIP peer trunk, if LDAP sync is enabled, end point can obtain remote UCM extensions' info via LDAP. Also, it will allow configuring whether to sync up LDAP contacts on Wave so that Wave doesn't receive duplicate contacts info.

Under UCM **webUI** → **System Settings** → **LDAP Server**, click on "LDAP Settings", option "Wave enable LDAP phonebook" is available for configuration. If enabled, all Wave users on this UCM will display LDAP contacts. Otherwise, it will not display.



LDAP Settings

Please note the LDAP contacts displayed on Wave will exclude the duplicate contacts from Cloud IM.



Display LDAP Contacts on Wave

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:

Phonebook Name	Enter a name for the phonebook
Server Address	The IP address of the LDAP server
Base DN	Enter the base domain name.
Username	Enter the username used to authenticate into the LDAP server, if authentication is required.
Password	Enter the password used to authenticate into the LDAP server, if authentication is required.
Filter	Enter the filter. Ex: ((CallerIDName=%)(AccountNumber=%))
Port	Enter the port number. Default port is 389
LDAP Number Attributes	Enter the number attributes for the remote server.
Automatic Update Cycle	If "None" is selected, LDAP phonebooks will not automatically update. Otherwise, LDAP phonebooks will automatically update at 00:00 / 12:00 AM with the selected frequency.
LDAP Name Attributes	Enter the name attributes for the remote server.
Client Type	Choose the client type. For encrypted data transfer please choose LDAPS.

LDAP Client CA Cert	LDAP Client Public Certification
LDAP Client Private Key	LDAP Client Private Certification

The UCM can automatically update the phonebook, by configuring the 'LDAP Automatic Update Cycle'. Available options are: 1 day/2days/7 days. It is set to 'None' by default.

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

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: LDAP Server Configurations]**.

LDAP

LDAP protocol	LDAP ▾
Server Address	192.168.40.134
Port	389
Base	dc=pbx,dc=com
User Name	
Password	
LDAP Number Filter	(AccountNumber=%)
LDAP Name Filter	(CallerIDName=%)
LDAP Version	<input type="radio"/> Version 2 <input checked="" type="radio"/> Version 3
LDAP Name Attributes	CallerIDName
LDAP Number Attributes	AccountNumber
LDAP Display Name	AccountNumber CallerIDName
Max. Hits	50
Search Timeout	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="button" value="Save"/> <input type="button" value="Save and Apply"/> <input type="button" value="Reset"/>

GXP2170 LDAP Phonebook Configuration

AD Client Type

Phonebook Name	Enter a name for the phonebook
Server Address	The IP address of the AD server
Base DN	Enter the base domain name.
Username	Enter the username used to authenticate into the LDAP server, if authentication is required.
Password	Enter the password used to authenticate into the LDAP server, if authentication is required.
Filter	Enter the filter. Ex: ((CallerIDName=%)(AccountNumber=%%))
Port	Enter the port number. Default port is 389
AD Attributes	AccountNumber must be included if the default configuration is used.
Automatic Update Cycle	If "None" is selected, LDAP phonebooks will not automatically update. Otherwise, LDAP phonebooks will automatically update at 00:00 / 12:00 AM with the selected frequency.
Host Name	Enter the host name of the remote AD server.

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**.

- !** 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.

Remote NTP Server	Configure the NTP server address to sync time from.
Enable DHCP Option 2	If enabled, DHCP Option 2 will override the Time Zone setting on the PBX.
Enable DHCP Option 42	If enabled, DHCP Option 42 will override the NTP server configured on the PBX.
Time Zone	Select your timezone. To update your list of timezones, please click on " Update Time Zone List ". Note: Updating the time zone list requires rebooting the UCM.

Remote NTP Server: pool.ntp.org

Enable DHCP Option 2:

Enable DHCP Option 42:

* Time Zone: (UTC+00:00) Etc/GMT-0

Update Time Zone List

Cancel Save

Automatic Date and Time Settings

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.

Time Settings

Automatic Date and Time **Set Date and Time** NTP Server Office Time Holiday

Cancel Save

Current Date and Time: Select date Select time

Date Format: yyyy-mm-dd

Time Format: Use 24-hour Format

Set Time Manually

i 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.

Create New Office Time

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

Create New Office Time



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 **Office Time** Holiday

<input type="checkbox"/>	INDEX	TIME	WEEK	MONTH	DAY	OPTIONS
<input type="checkbox"/>	1	09:00-18:00	Mon Tue Wed Thu Fri	Default	Default	<input type="button" value="Edit"/> <input type="button" value="Delete"/>

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.

Create New Holiday

* Name:

Holiday Memo:

Year:

Month:

Jan	Feb	Mar	Apr
May	Jun	Jul	Aug
Sept	Oct	Nov	Dec

Day:

1	2	3	4	5	6	7
8	9	10	11	12	13	14
15	16	17	18	19	20	21
22	23	24	25	26	27	28
29	30	31				

Show Advanced

Options:

Week:

Sun	Mon	Tue	Wed
Thu	Fri	Sat	

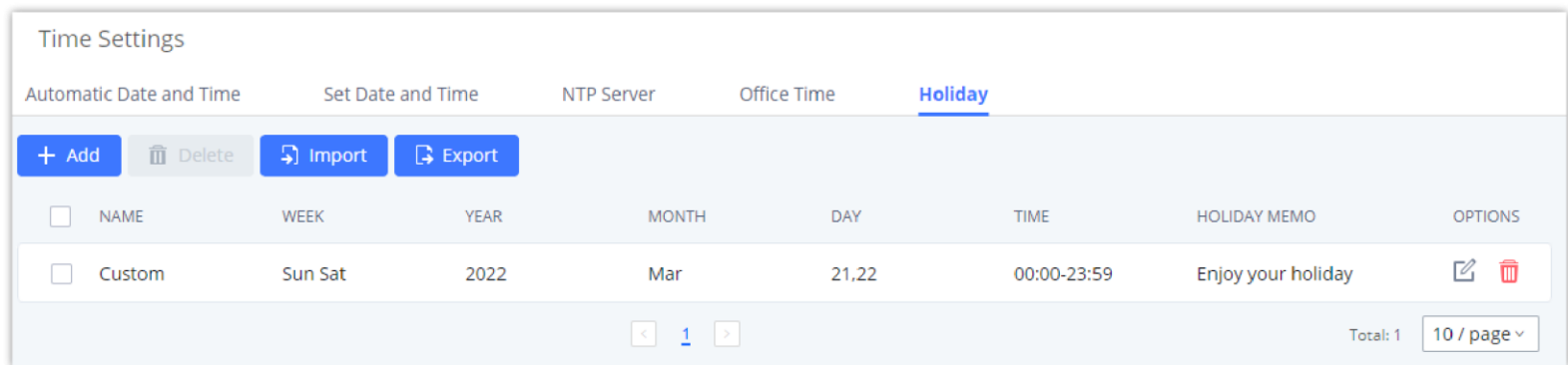
Time: -

Create New Holiday



Name	Specify the holiday name to identify this holiday.
-------------	--

Holiday Memo	Create a note for the holiday.
Month	Select the month for the holiday.
Year	Select the Year for the holiday. Note: In the "Year" option, select "All" to set annual fixed holiday information.
Day	Select the day for the holiday.
Show Advanced Options	Check this option to show advanced options. If selected, please specify the days as holiday in one week below.
Week	Select the days as holiday in one week.
Time	Select the time on which the holiday starts.

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.



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.

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**.

TLS Enable	Enable or disable TLS during transferring/submitted your Email to another SMTP server. The default setting is "Yes".
Type	Select Email type. <p>MTA: Mail Transfer Agent. The Email will be sent from the configured domain. When MTA is selected, there is no need to set up SMTP server for it or no user login is required. However, the Emails sent from MTA might be considered as spam by the target SMTP server.</p> <p>Client: Submit Emails to the SMTP server. A SMTP server is required, and users need login with correct credentials.</p>

Domain	Specify the domain name to be used in the Email when using type "MTA".
SMTP Server	Specify the SMTP server when using type "Client".
Enable SASL Authentication	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.
Username	Username is required when using type "Client". Normally it is the Email address.
Password	Password to login for the above Username (Email address) is required when using type "Client".
Enable Email-to-Fax	Monitors the inbox of the configured email address for the specified subject. If enabled, the UCM will get a copy of the attachment from the email and send it to the XXX extension by fax. The attachment must be in PDF/TIF/TIFF format.
Email-to-Fax Blacklist/Whitelist	The user can enable the Email-to-Fax Blacklist or Email-to-Fax Whitelist.
Email-to-Fax Subject Format	Select the email subject format to use for emails to fax. XXX refers to the extension that the fax will be sent to. This extension can only contain numbers.
Internal Black/Whitelist	Email address blacklist/whitelist for local extensions.
External Blacklist/Whitelist	Email address blacklist/whitelist for non-local contacts. Separate multiple addresses with semicolon (;) (i.e. "xxx;yyy").
Fax Sending Success/Failure Confirmation	If enabled, the UCM will send an email notification to the sender about the fax sending result.
POP/POP3 Server Address	Configure the POP/POP3 server address for the configured username Example: pop.gmail.com
POP/POP3 Server Port	Configure the POP/POP3 server port for the configured username Example: 995
Display Name	Specify the display name in the FROM header in the Email.
Sender	Specify the sender's Email address. For example: pbx@example.mycompany.com.

Email Settings

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 Template	Email footer hyperlink	Email Send Log
TLS Enable:	<input checked="" type="checkbox"/>		
Type:	Client		
Email Template Sending Format:	HTML		
* SMTP Server:	192.168.6.202:25		
* Enable SASL Authentication:	<input checked="" type="checkbox"/>		
* Username:	GStest@gmail.com		
* Password:	*****		
Enable Email-to-Fax:	<input type="checkbox"/>		
POP/POP3 Server Address:			
POP/POP3 Server Port:			
* Display Name:	PBX		
* Sender:	GStest@gmail.com		
Test			

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.

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 Templates

Users can customize email templates for password reset, voicemail, meeting scheduling, extensions, fax, meeting report, PMS, CDR, emergency call, missed calls, alert events, call queue statistics and etc.

- Click on



icon to edit the template.

Email Settings				
Email Settings <u>Email Template</u> Email Footer Hyperlink Email Send Log				
TYPE	NAME	TIME	OPTIONS	
Multimedia Meeting Schedule	mcm_template.html	2022-04-19 17:29:59 UTC+01:00		
SLA Alert	callqueuesla_template.html	2022-03-31 13:07:37 UTC+01:00		
Wave Welcome	welcome_template.html	2022-03-31 13:07:37 UTC+01:00		
Remote Registration	register_template.html	2022-03-31 13:07:37 UTC+01:00		
Extension	account_template.html	2022-03-31 13:07:37 UTC+01:00		
CDR	cdr_template.html	2011-12-03 11:30:03 UTC+01:00		
Fax	fax_template.html	2011-12-03 11:30:03 UTC+01:00		
Missed Calls	missedcall_template.html	2011-12-03 11:30:03 UTC+01:00		
Voicemail	voicemail_template.html	2011-12-03 11:30:03 UTC+01:00		
Call Queue Statistics	callqueuestatistics_template.html	2011-12-03 11:30:03 UTC+01:00		
Fax Sending	sendfax_template.html	2011-12-03 11:30:03 UTC+01:00		
Emergency Calls	emergency_template.html	2011-12-03 11:30:03 UTC+01:00		
Meeting Report	conferencereport_template.html	2011-12-03 11:30:03 UTC+01:00		

Email Template

Note

The "Multimedia Meeting Schedule" template is improved. Click on "Edit" for this template to view the improved default template.

- o Added "Edge" and "Safari" as supported browser.
- o Added "Download Wave" button for user to download Wave app from: <https://fw.gdms.cloud/wave/download/>
- o Improved descriptions

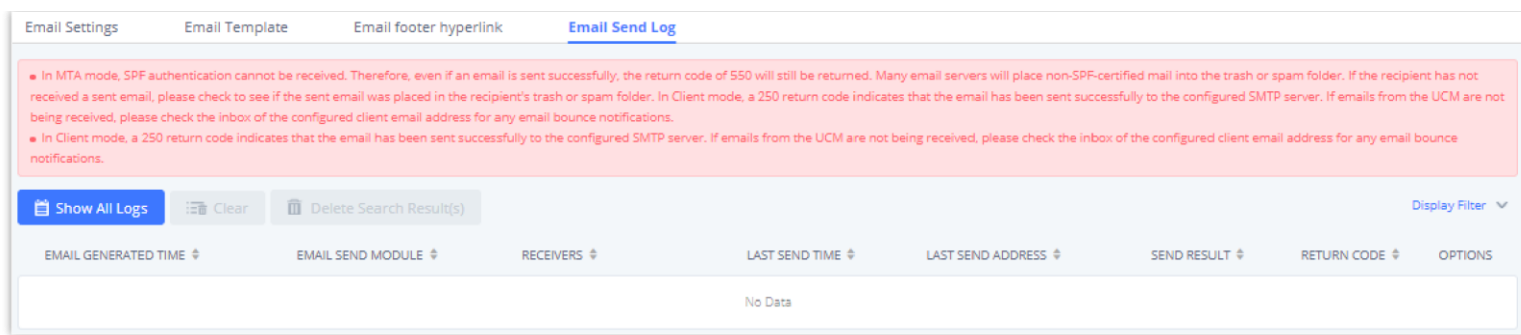
Email Footer Hyperlink

Under UCM Web GUI → System Settings → Email Settings → Email Footer Hyperlink, users could edit the text and URL to modify the email footer hyperlink.

Email Footer Hyperlink

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.



Email Send Log

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 them 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: <ul style="list-style-type: none"> ○ All Modules ○ Extension ○ Voicemail ○ Meeting Schedule ○ User Password ○ Alert Events ○ CDR ○ Test

Email Log – Display Filter

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	(i)
2020-12-21 18:00:04	Video Conference Schedule	m.g@gmail.com	12-21 18:00:06	m.g@gmail.com	sent	250	(i)
2020-12-20 18:00:04	Video Conference Schedule	m.g@gmail.com	12-20 18:00:07	m.g@gmail.com	sent	250	(i)
2020-12-19 18:00:03	Video Conference Schedule	m.g@gmail.com	12-19 18:00:06	m.g@gmail.com	sent	250	(i)
2020-12-18 18:00:03	Video Conference Schedule	m.g@gmail.com	12-18 18:00:06	m.g@gmail.com	sent	250	(i)
2020-12-17 18:00:03	Video Conference Schedule	m.p@gmail.com	12-17 18:00:06	m.p@gmail.com	sent	250	(i)

Total: 32 | 10 / page | Goto 1

Email Logs

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

Code	Description
250	Mail sent successfully
501	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.
535	The user name and password verification in the client mode is incorrect. Please check whether the user name and password are configured correctly.

Code	Description
550	<p>Possible reasons:</p> <ol style="list-style-type: none"> 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.
552	The sent email is too large or the email attachment type is prohibited
553	The sender and the email account are inconsistent, please configure the sender as your email account correctly.
554	The email was identified as spam. Please reduce the sending frequency or try again the next day
none	<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>

Email Codes

SMS Settings

SMS Configuration

Configuring the SMS feature on the UCM6300 series allows the administrators to enable two-factor authentication and to send alerts and meeting notices.

SMS Settings > SMS Settings

SMS Settings SMS Template SMS Delivery Log Cancel Save

Enable SMS

* SMS Carrier

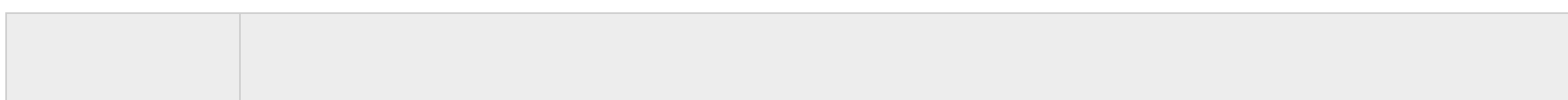
Region

* Account ID

* Secret

* From

SMS Settings



SMS Carrier	Choose the SMS carrier: <ul style="list-style-type: none"> • Amazon • Twilio
Region	Choose the region.
Account ID	Enter the ID of the account created at the carrier.
Secret	Enter the secret code.
From	Enter the number phone allocated for the UCM.

SMS Template

The template of the SMS can be modified in "SMS Template" tab. Please note that carriers may require to pre-register the templates for SMS that the UCM will send. Refer to the [Amazon](#) and [Twilio](#) documentation for more information.

SMS Settings > SMS Settings

SMS Settings SMS Template SMS Delivery Log

SMS templates are subject to carrier specifications, and carriers may require senders to pre-register templates for each type of message they plan to send. Please refer to the operator's requirements for details. More details can be found here: [Amazon](#), [Twilio](#)

TYPE	TEMPLATE CONTENT	OPTIONS
Verification Code	[UCM] Your verification code is \${code} . It will expire in 10 minutes.	
Alarm Notification	[UCM] \${hostName}(\${macAddr}) system event: \${content}	

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SMS Template

SMS Delivery Log

All the SMS messages sent will be logged in the following tab.

SMS Settings

SMS Settings SMS Template SMS Delivery Log

Show All Logs
Clear
Delete Search Result(s)
Display Filter ▼

SEND RESULT ↕	RECIPIENT ↕	SMS CATEGORY ↕	SEND TIME ↕
No data			

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SMS Delivery Log

HA

Dual-system hot standby provides a highly reliable and fault-tolerant solution for enterprises using the UCM6300 series/UCM6300A series. Based on two UCM devices of the same product model and software version, one of them is in the "Active" working state in real-time, and the other is in the "Standby" working state. The daily data on the host server will be synchronized to the standby machine in real-time, and the standby machine always monitors the running status of the host. When the host fails, including hardware failures and severe software failures, the standby machine will immediately take over the business and enter the "Active" working state, and Upgrade to a host to ensure that the business is not interrupted, and the call will automatically resume.

The HA function provides two modes of operating. The first mode of function is **Local Hot Standby**, which offers a deployment that switches dynamically when the primary UCM encounters an issue. The second mode is **Remote Disaster Recover**, it offers deploying a back up UCM remotely, this offers an architecture that would not be affected by any disaster that might occur on the geolocation of the primary UCM.

! Important Note

In order to set HA, both UCMs should have static IP addresses.

HA settings

The users can configure the HA under **System Settings → HA settings** page.

When setting up an HA dual-system, please keep the models and versions of the two devices forming the HA consistent, otherwise database compatibility problems may occur.
 If you need to connect an external USB device, it is recommended to use USB3.0 for both computers. Please ensure that the specifications are consistent, otherwise it will cause an exception.
 When enabling HA IPv6 support, it is highly recommended to reboot both UCMs at the same time after completing the HA configuration.

High Available Enable

HA Mode Local Hot Standby Remote Disaster Recovery

Force Switch

* Hot Standby Station Type

* Hot Standby Cluster IP

* Hot Standby Peer IP

* Hot Standby Peer MAC Address

* Heartbeat Port

* Heartbeat Timeout Period (s)

Software Fault Switch

Hardware Fault Switch FXO1

Enable IPv6

Scan External Storage Files

Get more information about HA Settings at [HA User Guide](#)

HA Settings

Parameter	Description
Force Switch	Enables/disables the HA functionality. By default, is Disabled.
Force Switch	Clicking the button will immediately force a switchover to the standby UCM.
Hot Standby Station Type	Used for the initial assignment of the HA active/standby role of the UCM system. If set to Primary, the current UCM system will be assigned as the initial active device. If set to Secondary, the peer UCM system will be assigned as the initial standby device. The roles of the UCM systems may change as HA switchovers occur.
Hot Standby Cluster IP	To use this service, the active and standby UCM systems need to use the same static IP address.
Hot Standby Peer IP	Local IP address of Hot Standby peer device.
Hot Standby Peer MAC Address	The MAC address of Hot Standby peer device.
Heartbeat Port	The number of the heartbeat port should be consistent with the peer heartbeat port.

Heartbeat Timeout Period (s)	Upon timeout, the Standby UCM will take over services.
Software Fault Switch	Enable Software Fault Switch
Hardware Fault Switch	If issues are detected with the selected connection interfaces, the backup UCM6510 will take over services after the master/slave handover. If not checked, UCM will send only a fault alarm.
Enable IPv6	If enabled, HA on UCM can be used with IPv6 while compatible with IPv4.
Scan External Storage Files	<p>Only applicable if there are more than 5000 UCM files in external storage such as SD card, USB, or NAS. Users can click this button to scan those paths in order to display all available files on the UCM web UI. Configured file storage paths can be viewed on the File Manager page.</p> <p>It is recommended to configure external storage data synchronization when forming HA for the first time, If HA is configured, files created after HA setup will be automatically displayed on the UCM web UI and do not need to be scanned for.</p>

Local Hot Standby

Parameter	Description
Network Port Domain Name	<ul style="list-style-type: none"> • WAN • LAN
Force Switch	Clicking the button will immediately force a switchover to the standby UCM.
Remote Disaster Recovery Station Type	Used for the initial assignment of the HA active/standby role of the UCM system. If set to Primary, the current UCM system will be assigned as the initial active device. If set to Secondary, the peer UCM system will be assigned as the initial standby device. The roles of the UCM systems may change as HA switchovers occur.
Remote Disaster Recovery Peer MAC Address	The MAC address of Remote Disaster Recovery peer device
Heartbeat Port	The heartbeat port should be the same as the peer device's heartbeat port.
Heartbeat Timeout Period (s)	Upon timeout, the Standby UCM will take over services.
Local Heartbeat IP	Fill in the IP address of the heartbeat port of the local site, in the format: xxx.xxx.xxx.xxx, which is used for the peer end to detect the local machine status, heartbeat negotiation, and communication address for data synchronization.
Local Heartbeat Gateway IP	Fill in the IP address of the local heartbeat gateway for remote disaster recovery, in the format: xxx.xxx.xxx.xxx
Local Heartbeat Address Subnet Mask	Fill in the subnet mask of the heartbeat address of the remote disaster recovery local end, in the format: xxx.xxx.xxx.xxx, such as: 255.255.255.0.
Peer Heartbeat IP	Fill in the IP address of the heartbeat port of the peer site, in the format: xxx.xxx.xxx.xxx, which is used to detect peer status, heartbeat negotiation, and

	communication address for data synchronization.
Scan External Storage Files	<p>Only applicable if there are more than 5000 UCM files in external storage such as SD card, USB, or NAS. Users can click this button to scan those paths in order to display all available files on the UCM web UI. Configured file storage paths can be viewed on the File Manager page.</p> <p>It is recommended to configure external storage data synchronization when forming HA for the first time, If HA is configured, files created after HA setup will be automatically displayed on the UCM web UI and do not need to be scanned for.</p>

Remote Disaster Recovery

HA Status

Once the HA is configured, the user can view its status under **System Settings → HA → HA Status** as shown below

HA	
HA Settings	HA Status
HA Mode	Local Hot Standby
Hot Standby Status	Dual
Hot Standby Full Backup Status	No backup
MAC Address of Current UCM	C0:74:AD:7D:B3:7A
Role of Current UCM	Active
GDMS HA Abnormal Status	No Primary UCM has been added to GDMS

HA Status

HA Log

The user can view the HA log through the **system settings → HA → HA log** page. The HA log effectively records the execution results of past full backup actions, as well as the historical records that triggered the active/standby switchover.

Note

The UCM63xx series supports SNMP to be able to use 3rd party monitoring tools to monitor both UCMs in HA setup.

SNMP

UCM63xx supports SNMP in case the system administrator chooses to use third party monitoring tools. These are the options available when setting up SNMP.

SNMP Settings

SNMP Settings

Enable	Tick this box to enable SNMP.
Device Name	Enter the device name.
Location	Enter the location.
Contact Email Address	Enter the email address used to send the SNMP alerts to.
Enable SNMP Trap Proxy	Tick this box to enable a proxy for SNMP Trap.
SNMP Trap Proxy Listening Port	The port number on which the SNMP Trap Proxy is listening on.

SNMP Community

You can also create SNMP communities and affect a certain level of access. An SNMP community is a group created to aggregate many management stations. The community name is used to authenticate and identify these machines in the NMS (Network Management System).

SNMP Community

Name	Enter a name for the community
Access Level	<p>Select an access level:</p> <ul style="list-style-type: none"> • Read Only: The SNMP community will be able only to read SNMP messages.

SNMP Trap Destinations

SNMP Traps is a very useful feature when there are many network components to manage. Instead of sending requests to all the machines in the network in order to view their SNMP logs risking slowing down or bringing the network to a complete halt, SNMP Traps can be configured so these machines can send unrequested messages to the manager to notify it about critical events and general failures.

SNMP Trap Destinations

Name	Enter a name of your SNMP Trap destination.
IP Address	Enter the SNMP Trap destination's IP address.
Port	Enter the port of the SNMP Trap destination.
Community	Select the community that you want
Type	<p>Select the type of SNMP:</p> <ul style="list-style-type: none"> • Trapsink: Select this option if you want to send SNMP v1 traps. • Trap2sink: Select this option if you want to send SNMP v2 traps. • Informsink: Select this option if you want to send "Inform" notifications only.

SNMP Version 3

UCM 63xx also supports SNMP v3 in case the system administrator decides to add more security to the monitoring process. SNMP v3 is a very good solution to monitor devices that interface directly with Internet. SNMP v3 offers more security than its predecessors by hashing the authentication information, encrypting the SNMP messages exchanged between the managed devices and the network management system which prevent eavesdropping. Also, it prevents any data tampering which protects the integrity of the data exchanged.

SNMP v3

Name	Set the user's name
-------------	---------------------

Authentication Protocol	Select the authentication protocol: <ul style="list-style-type: none"> • MD5 • SHA
Authentication Password	Set the authentication password.
Privacy Protocol	Select the protocol to use to encrypt the data <ul style="list-style-type: none"> • DES • AES-128 • AES-192 • AES-256
Privacy Password	Set the privacy password.
Group Level	Set the group level: <ul style="list-style-type: none"> • Read Only. • Read/Write.

SNMP Trap Proxy

Create New SNMP Trap Proxy
Cancel Save

* Name:

* IP Address:

* Port:

SNMP Trap Proxy

Name	Enter a name for the proxy server.
IP Address	Enter the proxy server's IP address.
Port	Enter the proxy server's port.

RADIUS

The UCM6300 offers Radius-based authentication for the super administrator and other administrators. This requires configuring a Radius server then enabling Radius client on the UCM6300 which can be found under **System Settings** → **RADIUS**

Radius

Supports configuring two types of account privileges on the Radius server: Super Administrator and Administrator.

Enable Radius Web Access Control

As Default Login Method

* Radius Auth Server Address

* Radius Auth Server Port

* Radius Shared Secret 👁

* Maximum Number of Retransmission

* Radius Timeout (s)

RADIUS

Enable Radius Web Access Control	Enable or disable Radius.
As Default Login Method	Enable Radius as the default login method to the web UI of the UCM
Radius Auth Server Address	Enter the IP address/hostname of Radius server.
Radius Auth Server Port	Enter the port of radius server Default port number is: 1812
Radius Shared Secret	Enter Radius Shared Secret
Maximum Number of Retransmission	Enter the number of retransmissions. The interval is 1 to 5.
Radius Timeout (s)	The maximum seconds before a session expires if there is no response from the server. The interval is between 1 to 30 seconds.

TR-069

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

Parameter	Description
Enable TR-069	Toggle it on to enable TR-069. It is enabled by default
ACS URL	URL for TR-069 Auto Configuration Servers (ACS), e.g., http://myacs.grandstream.com
TR-069 Username	ACS username for TR-069, must be the same as in the ACS configuration.
TR-069 Password	ACS password for TR-069, must be the same as in the ACS configuration.
Periodic Inform Enable	Enables periodic inform. If set to 'Yes', device will send inform packets to the ACS.

Parameter	Description
Periodic Inform Interval	Periodic time when UCM630xA will send inform packets to TR-069 ACS server. This option is specified in seconds.
ACS Connection Request Username	The username for the ACS to connect to UCM.
ACS Connection Request Password	The password for the ACS to connect to UCM.
Connection Request Port	Port for incoming connection requests. The default value is 7547 .
CPE Cert File	The Cert file for UCM to connect to the ACS via SSL.
CPE Cert Key	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 > **Device Management** > **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.

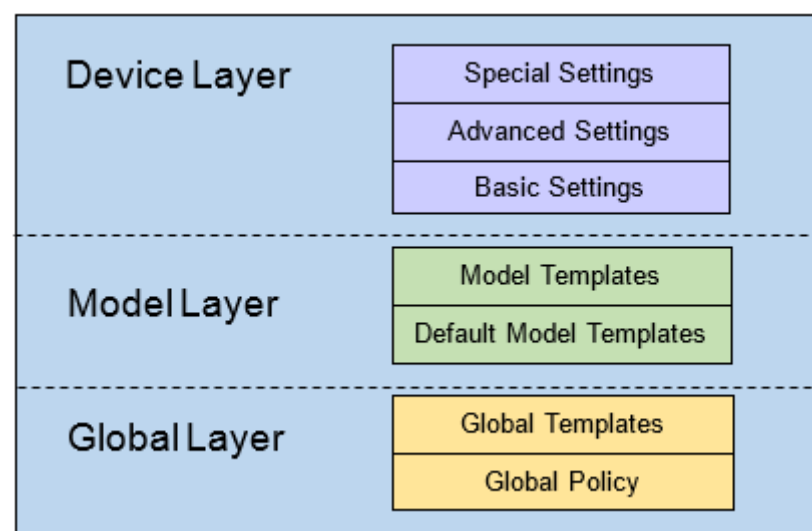
- **Model**

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]**.



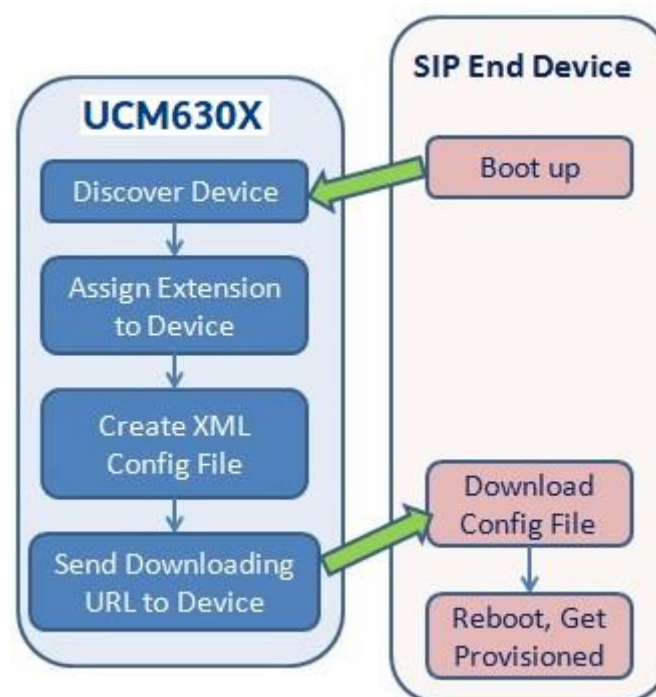
Zero Config Configuration Architecture for End Point Device

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.



UCM630xA Zero Config

- o **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.

- o **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.

- o **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>Device Management>**Zero Config**>**Zero Config Settings**, fill in the auto provision information.

Auto Provision Settings

Enable Zero Config	Enable or disable the zero-config feature on the PBX. The default setting is enabled.
Enable Automatic Configuration Assignment	By default, this is disabled. If disabled, when SIP device boots up, the 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. Note: 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.
Auto Assign Extension	If enabled, when the device is discovered, the PBX will automatically assign an extension within the range defined in "Zero Config Extension Segment" to the device. The default setting is disabled.
Zero Config Extension Segment	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→ PBX Settings → General Settings → General page→Extension Preference section: "Auto Provision Extensions".
Enable Pick Extension	If enabled, the extension list will be sent out to the device after receiving the device's request. This feature is for the GXP series phones that support selecting extension to be provisioned via phone's LCD. The default setting is disabled.
Pick Extension Segment	Click on the link "Pick Extension Segment" to specify the extension list to be sent to the device. The default range is 4000 to 4999. Pick Extension Segment range can be defined in Web GUI → PBX Settings → General Settings → General page → Extension Preference section: "Pick Extensions".
Pick Extension Period (hour)	Specify the number of minutes to allow the phones being provisioned to pick extensions.

Subnet Whitelist	<p>This feature allows the UCM to provision devices in different subnets other than UCM network.</p> <p>Enter subnets IP addresses to allow devices within these subnets to be provisioned. The syntax is <IP>/<CIDR>.</p> <p>Examples:</p> <p>10.0.0.1/8</p> <p>192.168.6.0/24</p> <p>Note: Only private IP ranges (10.0.0.0 172.16.0.0 192.168.0.0) are supported.</p>
-------------------------	--

Auto Provision Settings

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>Device Management>**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>Device Management>**Zero Config>Auto Provisioning Settings**.

Auto Discover
×

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: ▾

Subnet Whitelist: ▾

Scan IP: . . .

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 Web GUI) are displayed in the list.

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

Discovered Devices

When the UCM is set to "Dual" network method, the user will be able to choose which LAN interface to use for Auto-Discovery.

Auto Discover

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

Interface: LAN 1

PBX Network Interface IP Address : LAN 1

Network Segment: 192.168.50.0 - 192.168.50.255

Auto Discover LAN1/LAN2

Firmware

In Firmware tab, users can upload to and manage firmware for endpoints. Additionally, firmware upload size limit has been increased from 300MB to 1GB.

Zero Config
Model Templates Model Update **Firmware** Zero Config Settings
Cancel Save

Firmware Storage Path : Local

Firmware List

Upload

NAME ↕	MODEL ↕	FIRMWARE VERSION ↕	DATE ↕	SIZE ↕	STATUS ↕	OPTIONS
No Data						

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Firmware Storage

Upload New Firmware

- **Enable:** toggles whether the UCM will provision this firmware to endpoints if they are using the UCM as the firmware server. If not enabled, the UCM will reject requests from endpoints for this firmware.
- **Model:** The device model for which this firmware is intended for. Only for self-reference and has no effect on provisioning.
- **Firmware:** The firmware version of the file being uploaded. Only for self-reference and has no effect on provisioning.
- **Remark:** Add a comment about the uploaded firmware. Only for self-reference and has no effect on provisioning.
- **Choose File to Upload:** Select the firmware file to upload from the user's PC. The file name must match the firmware file name requested by the endpoint.

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	1	192.168.5.172		https://192.168.5.147:80	#####	000B8249	1.0.7.10	0	GXW4248	no	5060	
4	===== Device Start =====													
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	1	192.168.5.114		https://192.168.5.147:80	#####	000B826B	1.0.3.227	0	GXV3240	no	5060	
8	===== Device Start =====													
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	1	192.168.5.201		https://192.168.5.147:80	#####	000B826F	1.0.1.106	0	--	no	5080	

Device List – CSV file Sample

Note

Please ensure that the .csv file is encoded in UTF-8 to be able to import the devices correctly into the UCM.

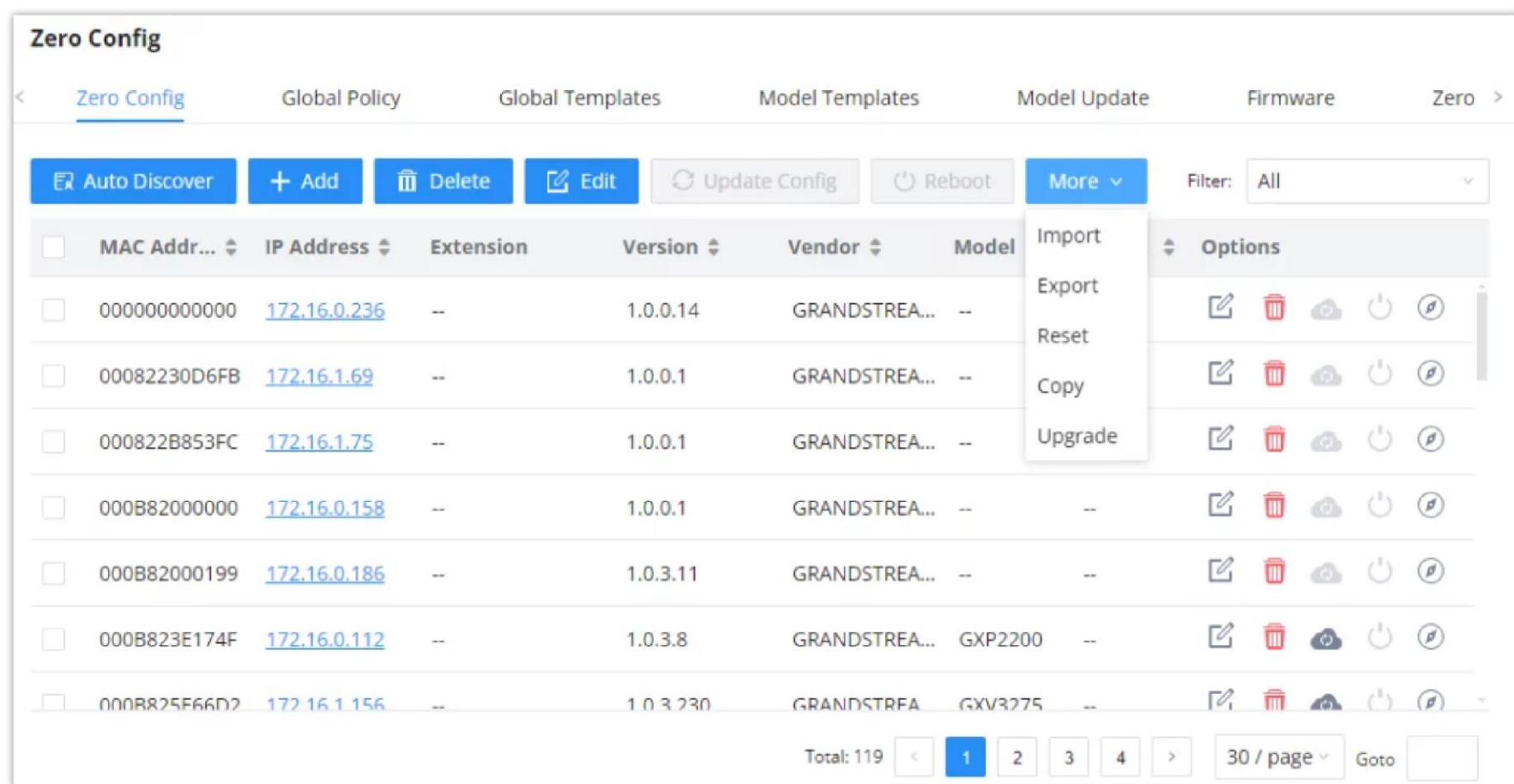
Managing Discovered Devices

- **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:** Select a filter

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.




Managing Discovered Devices

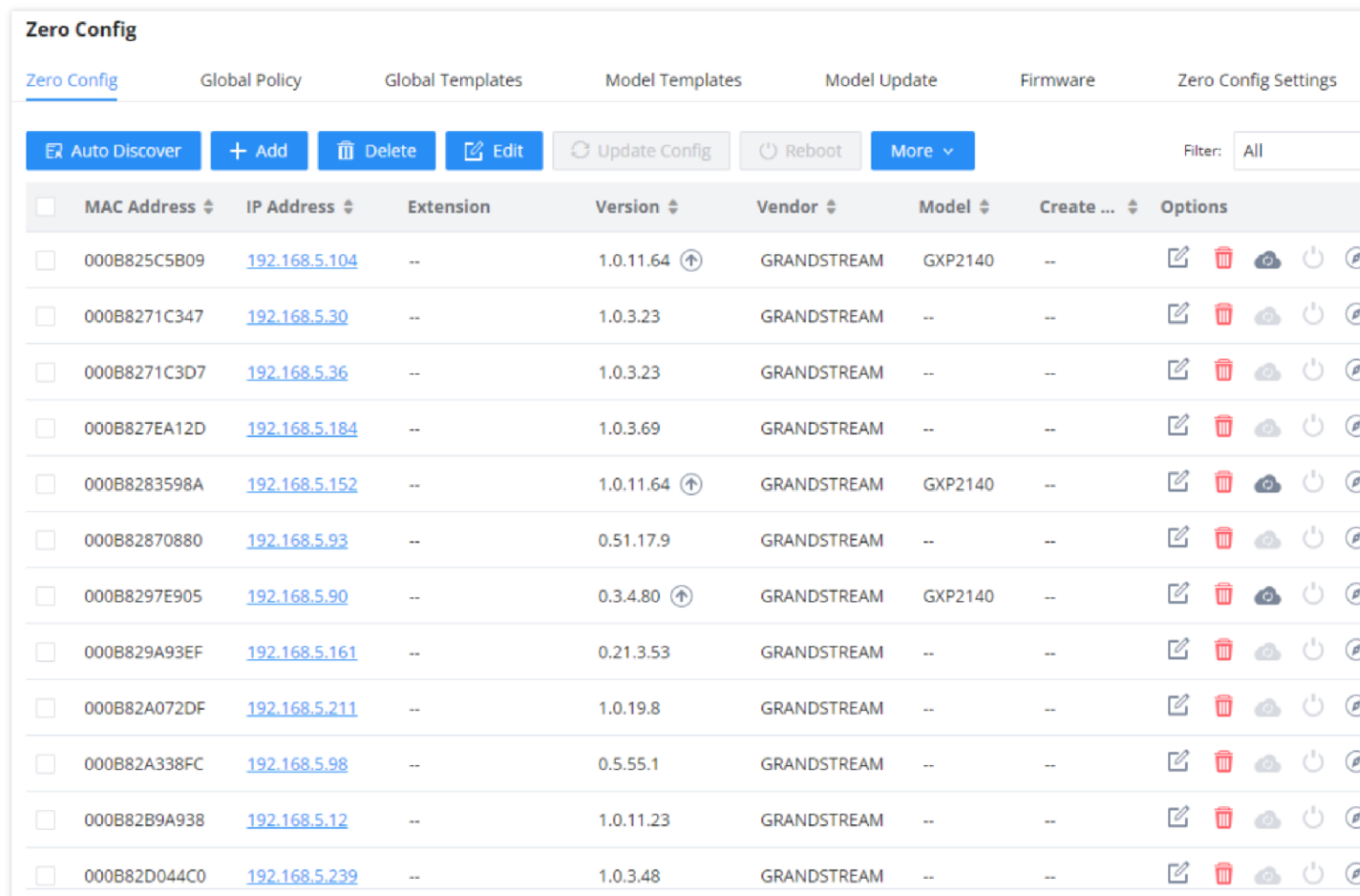
From the main menu of zero config, users can perform the following operations:
































































- Click on **Auto Discover** in order to access to the discovery menu as shown on [Discovery] section.
- Click on **Add** to add a new device to zero config database using its MAC address.
- Click on **Delete** to delete selected devices from the zero-config database.
- Click on **Edit** to modify selected devices.
- Click on **Update Config** 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 **Reboot** 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 **More** to perform the following additional actions:
 - **Import:** Import list of devices.
 - **Export:** Export the list of the discovered devices. Select the devices to export before clicking this button. In case no devices were selected, all the discovered devices will be exported.
 - **Reset:** Reset the discovered devices. Select the devices to reset before clicking this button. In case no devices were selected all the discovered devices will be reset to their initial configuration.
 - **Copy:** Copy the configuration across devices.

- **Upgrade:** Upgrade the software of the discovered devices to the latest version. Select the devices which you want to upgrade. In case when no devices were selected, all the discovered devices will be upgraded to the latest firmware version.

All these operations will be detailed on the next sections.

When a new firmware is detected for the discovered devices, an upgrade icon  will appear next to the firmware version installed on the device.



MAC Address	IP Address	Extension	Version	Vendor	Model	Create ...	Options
000B825C5B09	192.168.5.104	--	1.0.11.64 	GRANDSTREAM	GXP2140	--	    
000B8271C347	192.168.5.30	--	1.0.3.23	GRANDSTREAM	--	--	    
000B8271C3D7	192.168.5.36	--	1.0.3.23	GRANDSTREAM	--	--	    
000B827EA12D	192.168.5.184	--	1.0.3.69	GRANDSTREAM	--	--	    
000B8283598A	192.168.5.152	--	1.0.11.64 	GRANDSTREAM	GXP2140	--	    
000B82870880	192.168.5.93	--	0.51.17.9	GRANDSTREAM	--	--	    
000B8297E905	192.168.5.90	--	0.3.4.80 	GRANDSTREAM	GXP2140	--	    
000B829A93EF	192.168.5.161	--	0.21.3.53	GRANDSTREAM	--	--	    
000B82A072DF	192.168.5.211	--	1.0.19.8	GRANDSTREAM	--	--	    
000B82A338FC	192.168.5.98	--	0.5.55.1	GRANDSTREAM	--	--	    
000B82B9A938	192.168.5.12	--	1.0.11.23	GRANDSTREAM	--	--	    
000B82D044C0	192.168.5.239	--	1.0.3.48	GRANDSTREAM	--	--	    

Zero Config

Global Configuration

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**

 Global Templates configuration has higher priority to Global Policy configuration.

Global Policy

Global Policy can be accessed in Web GUI>Device Management>**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.



Global Policy Categories

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

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

Global Policy Parameters – Localization

Default Call Settings	
Dial Plan	Configure the default dial plan rule. For syntax and examples, please refer to user manual of the SIP devices to be provisioned for more details.
Enable Call Features	When enabled, "Do Not Disturb", "Call Forward" and other call features can be used via the local feature code on the phone. Otherwise, the ITSP feature code will be used.
Use # as Dial Key	If set to "Yes", pressing the number key "#" will immediately dial out the input digits.
Auto Answer by Call-info	If set to "Yes", the phone will automatically turn on the speaker phone to answer incoming calls after a short reminding beep, based on the SIP Call-Info header sent from the server/proxy. The default setting is enabled.
NAT Traversal	Configure if NAT traversal mechanism is activated.
User Random Port	If set to "Yes", this parameter will force random generation of both the local SIP and RTP ports.
General Settings	

Call Progress Tones	<p>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:</p> <p>f1=val, f2=val[, c=on1/ off1[- on2/ off2[- on3/ off3]]];</p> <ul style="list-style-type: none"> ○ Frequencies are in Hz and cadence on and off are in 10ms). ○ "on" is the period (in ms) of ringing while "off" is the period of silence. Up to three cadences are supported. ○ Please refer to user manual of the SIP devices to be provisioned for more details
HEADSET Key Mode	<p>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.</p>

Global Policy Parameters – Phone Settings

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

Number Attribute	<p>Specify the “number” attributes of each record which are returned in the LDAP search result.</p> <p>Example:</p> <p>telephoneNumber</p> <p>telephoneNumber Mobile</p>
Display Name	<p>Configure the entry information to be shown on phone’s LCD. Up to 3 fields can be displayed.</p> <p>Example:</p> <p>%cn %sn %telephoneNumber</p>
Max Hits	<p>Specify the maximum number of results to be returned by the LDAP server. Valid range is 1 to 3000. The default value is 50.</p>
Search Timeout	<p>Specify the interval (in seconds) for the server to process the request and client waits for server to return.</p> <p>Valid range is 0 to 180. Default value is 30.</p>
Sort Results	<p>Specify whether the searching result is sorted or not. Default setting is No.</p>
Incoming Calls	<p>Configure to enable LDAP number searching when receiving calls. The default setting is No.</p>
Outgoing Calls	<p>Configure to enable LDAP number searching when making calls. The default setting is No.</p>
Lookup Display Name	<p>Configures the display name when LDAP looks up the name for incoming call or outgoing call.</p> <p>It must be a subset of the LDAP Name Attributes.</p>
XML Phonebook	
Phonebook XML Server	<p>Select the source of the phonebook XML server.</p> <ul style="list-style-type: none"> ○ Disable <p>Disable phonebook XML downloading.</p> <ul style="list-style-type: none"> ○ Manual <p>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.</p> <ul style="list-style-type: none"> ○ Local UCM Server <p>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.</p>
Phonebook Download Interval	<p>Configure the phonebook download interval (in Minute). If set to 0, automatic download will be disabled. Valid range is 5 to 720.</p>
Remove manually edited entries on download	<p>If set to “Yes”, when XML phonebook is downloaded, the entries added manually will be automatically removed.</p>

Upgrade and Provision	
Firmware Source	<p>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:</p> <ul style="list-style-type: none"> ○ URL <p>If select to use URL to upgrade, complete the configuration for the following four parameters: "Upgrade Via", "Server Path", "File Prefix" and "File Postfix".</p> <ul style="list-style-type: none"> ○ Local UCM Server <p>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.</p> <ul style="list-style-type: none"> ○ Local USB Media <p>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.</p> <ul style="list-style-type: none"> ○ Local SD Card Media <p>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.</p>
Upgrade via	When URL is selected as firmware source, configure upgrade via TFTP, HTTP or HTTPS.
Server Path	When URL is selected as firmware source, configure the firmware upgrading server path.
File Prefix	Configure the Config Server Path.
Config Server Path	When URL is selected as firmware source, configure the firmware file postfix. If configured, only the configuration file with the matching encrypted postfix will be downloaded and flashed into the phone.
Allow DHCP Option 43/66	If DHCP option 43 or 66 is enabled on the LAN side, the TFTP server can be redirected.
Automatic Upgrade	<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"> ○ By week <p>Once selected, specify the day of the week to check HTTP/TFTP server for firmware upgrades or configuration files changes.</p> <ul style="list-style-type: none"> ○ By day <p>Once selected, specify the hour of the day to check the HTTP/TFTP server for firmware upgrades or configuration files changes.</p> <ul style="list-style-type: none"> ○ By minute <p>Once selected, specify the interval X that the SIP end device will request for new firmware every X minutes.</p>
Firmware Upgrade Rule	Specify how firmware upgrading and provisioning request to be sent.
Zero Config	Select either to enable or disable zero config.
Web Access	
Admin Password	Configure the administrator password for admin level login.
End-User Password	Configure the end-user password for the end user level login.
Web Access Mode	Select HTTP or HTTPS as the web access protocol.

Web Server Port	Configure the port for web access. The valid range is 1 to 65535.
RTSP Port	Configure the RTSP Port.
Enable UPnP Discovery	Select either to enable or disable Enable UPnP Discovery
Login Settings	Configure the login settings.
User Login Timeout	Configure User Login Timeout.
Maximum Consecutive Failed Login Attempts	Configure Maximum Consecutive Failed Login Attempts.
Login Error Lock Time	Configure Login Error Lock Time.
Security	
Disable Telnet/SSH	Enable Telnet/SSH access for the SIP end device. If the SIP end device supports Telnet access, this option controls the Telnet access of the device; if the SIP end device supports SSH access, this option controls the SSH access of the device.
Syslog	
Syslog Server	Configure the URL/IP address for the syslog server.
Syslog Level	Select the level of logging for syslog.
Send SIP Log	Configure whether the SIP log will be included in the syslog message.

Global Policy Parameters – Maintenance

Basic Settings	
IP Address	Configure how the SIP end device shall obtain the IP address. DHCP or PPPoE can be selected. DHCP Once selected, users can specify the Host Name (option 12) of the SIP end device as DHCP client, and Vendor Class ID (option 60) used by the client and server to exchange vendor class ID information. PPPoE Once selected, users need specify the Account ID, Password and Service Name for PPPoE.
Host Name	Specifies the name of the client. This field is optional but may be required by Internet Service Providers.
Vendor Class ID	Used by clients and servers to exchange vendor class ID.
Account ID	Enter the PPPoE account ID.
Password	Enter the PPPoE Password.
Service Name	Enter the PPPoE Service Name.
Advanced Setting	
Layer 3 QoS	Define the Layer 3 QoS parameter. This value is used for IP Precedence, Diff-Serv or MPLS. Valid range is 0-63.
Layer 3 QoS For RTP	Assign the priority value of the Layer 3 QoS for RTP packets. Valid range is 0 -63.

Layer 3 QoS For SIP	Assign the priority value of the Layer 3 QoS for SIP packets. Valid range is 0 -63.
Layer 2 QoS Tag	Assign the VLAN Tag of the Layer 2 QoS packets. Valid range is 0 -4095.
Layer 2 QoS Priority Value	Assign the priority value of the Layer 2 QoS packets. Valid range is 0-7.
STUN Server	Configure the IP address or Domain name of the STUN server. Only non-symmetric NAT routers work with STUN.
Keep Alive	Select either to enable or disable Keep Alive.
Keep Alive Interval	Specify how often the phone will send a blank UDP packet to the SIP server in order to keep the "ping hole" on the NAT router to open. Valid range is 10-160.
Register Expiration	Specify the Register Expiration.
Local SIP Port	Configure Local SIP Port.
Local RTP Port	Configure Local RTP Port.
Auto On-Hook Timer(s)	Configure Auto On-Hook Timer(s).
Ring Timeout	Configure Ring Timeout.
SIP Transport	Select either UDP, TCP or TLS/TCP as SIP transport protocol.
Direct IP Call	Select either to disable or enable Direct IP Call support.
SIP Proxy Compatibility Mode	Select either to disable or enable SIP Proxy Compatibility Mode.
Unregister On Reboot	Select either to disable or enable Unregister On Reboot.
Whitelist	
Whitelist	Select either to enable or disable Whitelist
SIP Phone Number Whitelist	Configure the SIP Phone Number Whitelist.

Global Policy Parameters – Network Settings

Wallpaper	
Screen Resolution 1024 x 600	<p>Check this option if the SIP end device shall use 1024 x 600 resolution for the LCD screen wallpaper.</p> <p>Source</p> <p>Configure the location where wallpapers are stored.</p> <p>File</p> <p>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.</p>

<p>Screen Resolution 800 x 400</p>	<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"> ○ Source <p>Configure the location where wallpapers are stored.</p> <ul style="list-style-type: none"> ○ File <p>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.</p>
<p>Screen Resolution 480 x 272</p>	<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"> ○ Source <p>Configure the location where wallpapers are stored.</p> <ul style="list-style-type: none"> ○ File <p>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.</p>
<p>Screen Resolution 320 x 240</p>	<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"> ○ Source <p>Configure the location where wallpapers are stored.</p> <ul style="list-style-type: none"> ○ File <p>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.</p>

Global Policy Parameters – Customization

Email Settings

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


Global Templates can be accessed in Web GUI>Device Management>**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.

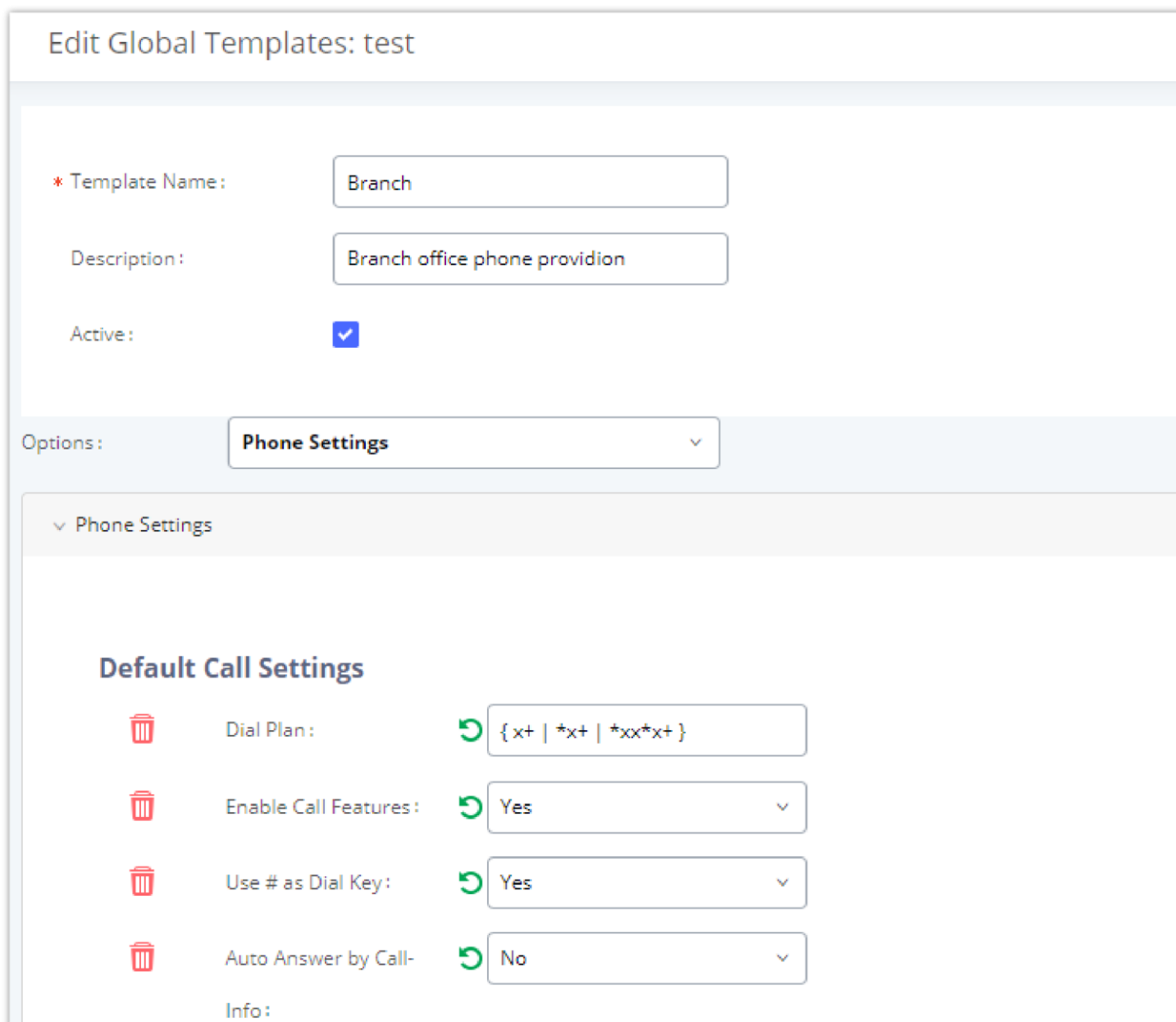
Click on “Add” to add a global template. Users will see the following configurations.

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

Create New Template

- o 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 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>Device Management>**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 Device Management>**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.



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

Create New 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>

Edit Model Templates: GRP2613_template



* Model:

* Template Name:

Description:

Default Model Template:

Active:

Options:

▼ Custom Parameters

Custom Parameters

Please enter P-values into the Name fields. Example: To configure Language to English, enter "P1362" into the Name field and "en_US" into the Value field.

	P1362	en	Description	Possible Match Exists
--	-------	----	-------------	-----------------------

+ Add New Field

Edit Model Template


- Click on Save when done. The model template will be displayed on Web GUI > **Device Management** > **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.
- Click the "Copy Template" button to copy the configuration items of the selected model template to another template, thereby reducing template editing work. Note: The model template only supports copying between devices of the same model.
- Click the "Import/Export" button to upload/export the model template list in .CSV format.

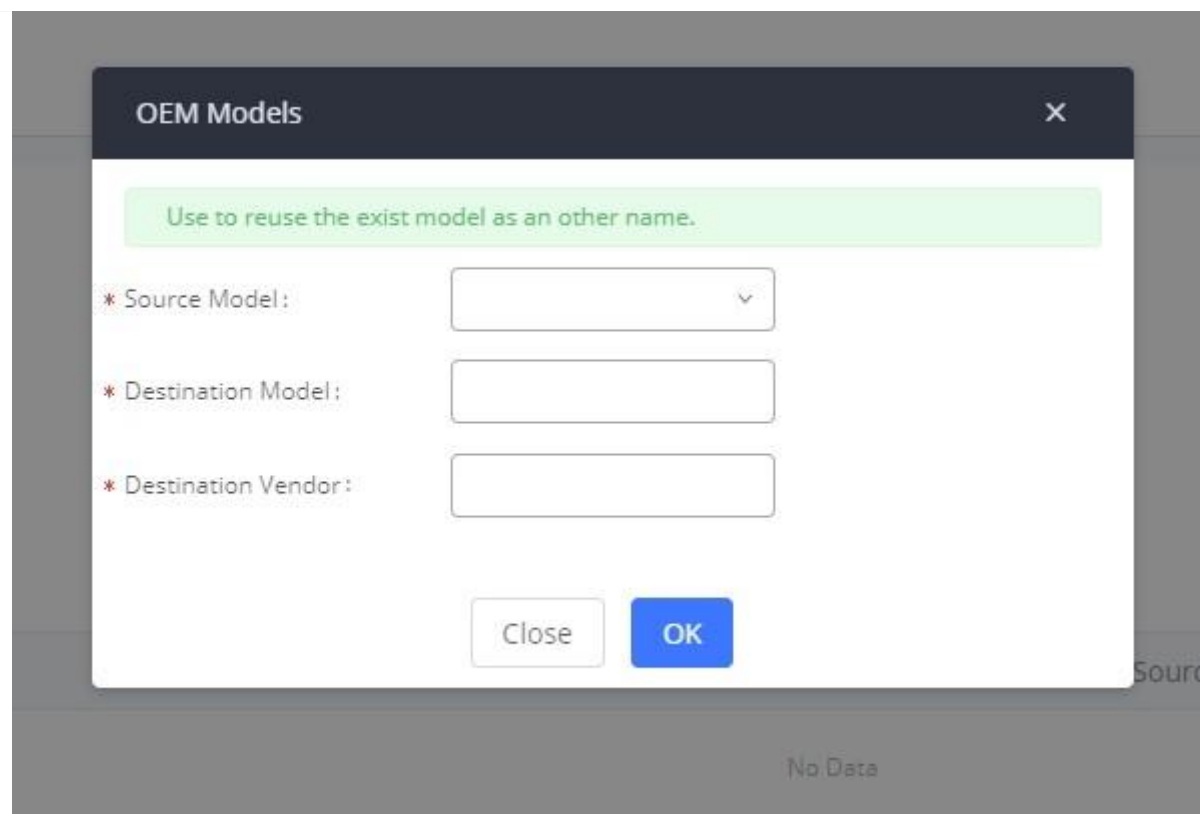
Model Update

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.













OEM Models

Model Template Package List

Templates for most of the Grandstream models are built in with the UCM630xA already. Templates for Grandstream Wave and Grandstream surveillance products require users to download and install under Web GUI → Device Management > **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

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:

Upload Model Template Manually

Device Configuration

On Web GUI, page **Device Management>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.

Cancel Update Save

Edit Device: C074AD112233

* Model: GRANDSTREAM GXV3350

* MAC Address: c074ad112233

IP Address:

Version:

Basic Settings Advanced Settings

Accounts

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

Create New Device

Manage Devices

The device manually created or discovered from Auto Discover will be listed in the Web GUI>**Device Management>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 checked="" type="checkbox"/>	000B82000001	192.168.2.111	1000	unknown	GRANDSTREAM	GXV3275	--	
<input type="checkbox"/>	000B8227FB15	192.168.2.108	--	1.0.3.208	GRANDSTREAM	GXV3275	--	
<input type="checkbox"/>	000B82A46ACE	192.168.2.106	--	1.0.0.36	GRANDSTREAM	--	--	
<input type="checkbox"/>	000B82D33AC4	192.168.2.105	--	20.19.10.30	GRANDSTREAM	--	--	
<input type="checkbox"/>	000B82F66470	192.168.2.107	--	10.19.9.26	GRANDSTREAM	--	--	


Manage Devices

- o Click on to access the Web GUI of the phone.

- o 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

3 Default Model Template

Preview

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 → **Device Management > Zero Config > Global Policy** will be applied here. Clicking on "Modify Global Policy" to redirect to page **Device Management > 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>Device Management>**Zero Config**>**Model Templates** page. Please see default model template option in **[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.

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

Custom Fields

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

	P1362	en	Description	Possible Match Exists
--	-------	----	-------------	-----------------------

+ Add New Field

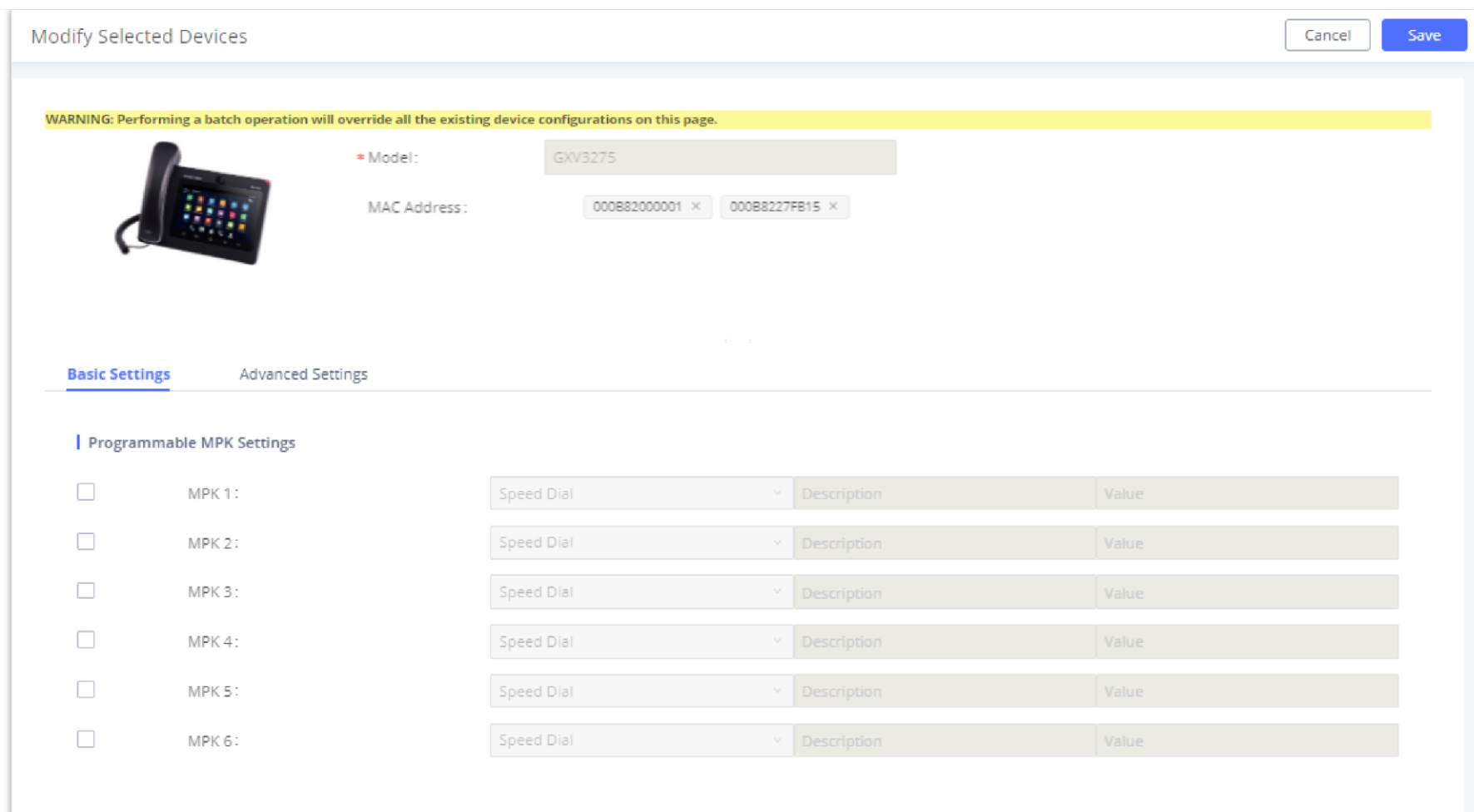
Edit Customize Device Settings

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

- o 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.

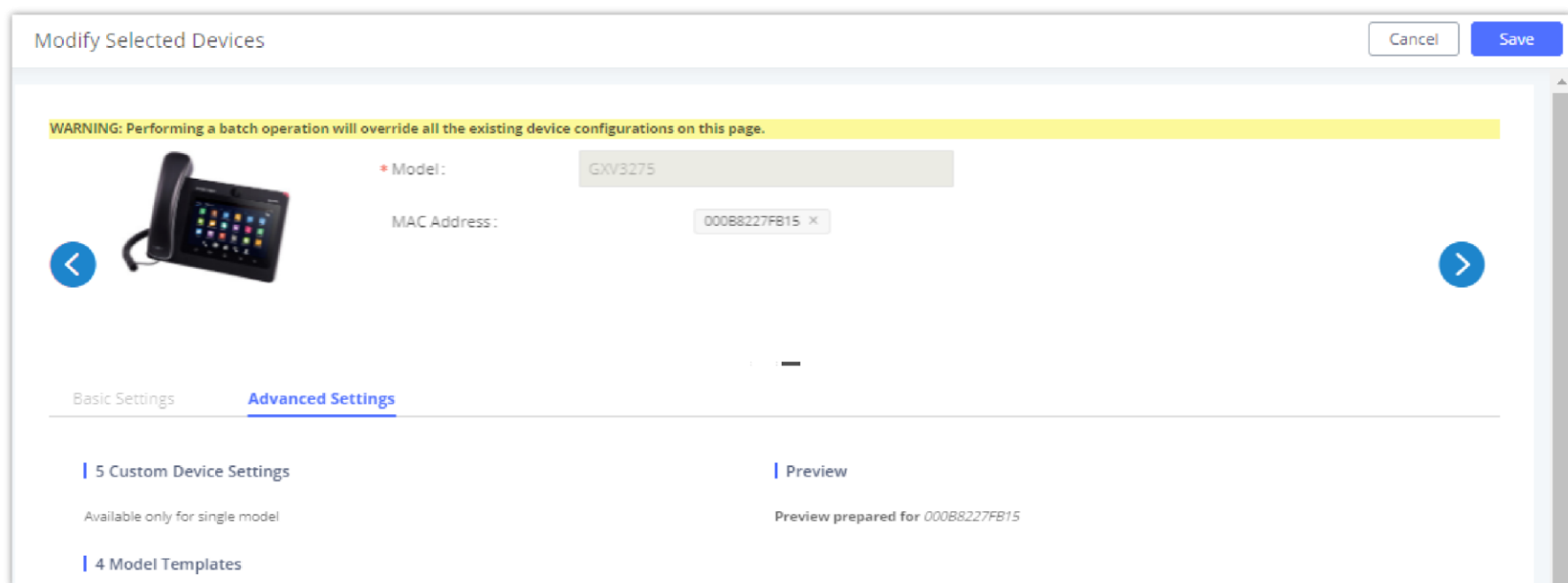


Modify Selected Devices – Same Model


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.

































Modify Selected Devices – Different Models

 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>Device Management>**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.

Zero Config									
Zero Config		Global Policy	Global Templates	Model Templates	Model Update	Zero Config Settings			
<input type="button" value="Auto Discover"/> <input type="button" value="Add"/> <input type="button" value="Delete"/> <input type="button" value="Edit"/> <input type="button" value="Update Config"/> <input type="button" value="Reboot"/> <input type="button" value="Reset"/> <input type="button" value="Upload"/> <input type="button" value="Copy"/>									
Filter:	All								
<input type="checkbox"/>	MAC ADDRESS	IP ADDRESS	EXTENSION	VERSION	VENDOR	MODEL	CREATE CONFIG	OPTIONS	
<input type="checkbox"/>	000B82000001	192.168.2.111	1000	unknown	GRANDSTREAM	GXV3275	--	    	
<input type="checkbox"/>	000B8227FB15	192.168.2.108	--	1.0.3.208	GRANDSTREAM	GXV3275	--	    	
<input type="checkbox"/>	000B82A46ACE	192.168.2.106	--	1.0.0.36	GRANDSTREAM	--	--	    	
<input type="checkbox"/>	000B82A95F94	192.168.5.115	--	1.0.9.132	GRANDSTREAM	GXP2170	--	    	
<input type="checkbox"/>	000B82D33AC4	192.168.2.105	--	20.19.10.30	GRANDSTREAM	--	--	    	
<input type="checkbox"/>	000B82F66470	192.168.2.107	--	10.19.9.26	GRANDSTREAM	--	--	    	
							Total: 6	30 / page	Goto 1

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>Device Management>**Zero Config**>**Zero Config Settings**, select "Enable Zero Config".
2. Go to Web GUI>Device Management>**Zero Config**>**Global Policy**, configure Date Format, Time Format and Firmware Source as follows.

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

> Maintenance

Upgrade and Provision

Firmware Source:

Source: URL


Upgrade via: TFTP

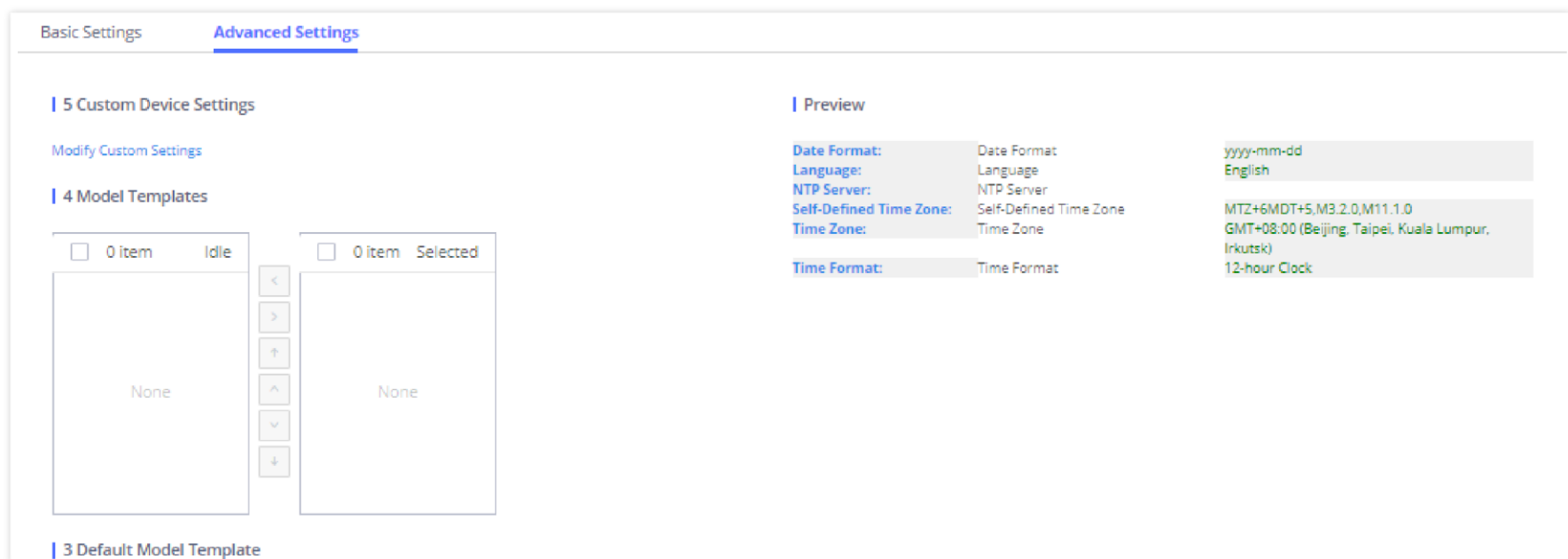
Server Path: fm.grandstream.com/gs

File Prefix:

File Postfix:

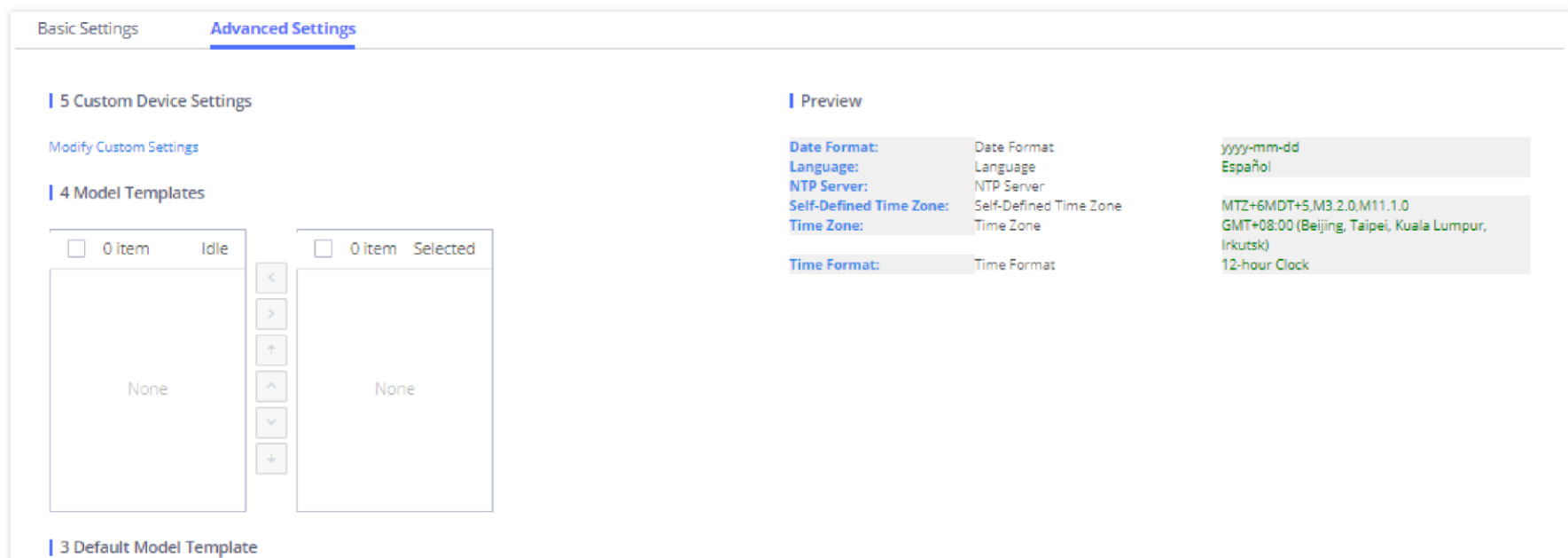
Zero Config Sample – Global Policy

- Go to Web GUI>Device Management>**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.
- Go to Web GUI>Device Management>**Zero Config>Model Templates**, create another model template “Spanish Support Template” for GXP2170. Add option “Language” and set it to “Español”.
- 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>Device Management>**Zero Config>Zero Config**.
- On Web GUI>Device Management>**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.
- 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.



Zero Config Sample – Device Preview 1

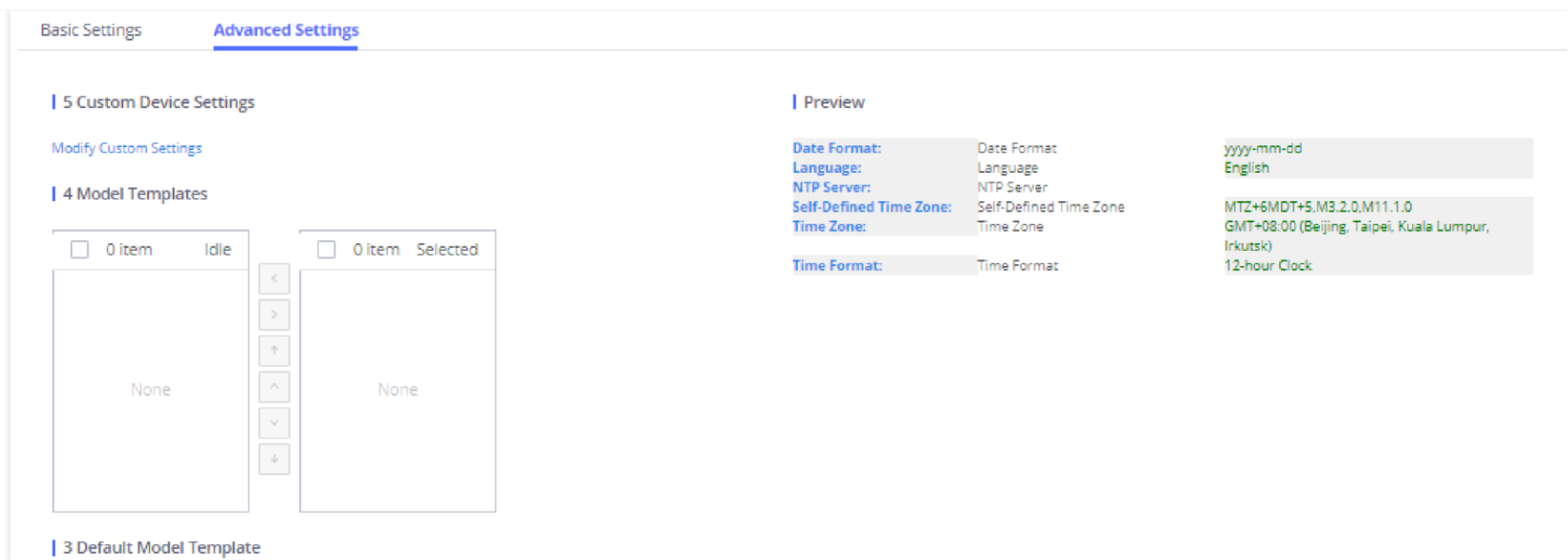
- For the 3 phones used by Spanish support, in “Basic settings” select an available extension for account 1 and click on “Save”. Then click on “Advanced settings” tab to bring up the following dialog.



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.



Zero Config Sample – Device Preview 3

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

11. On the Web GUI>Device Management>**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.

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.

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

Keep Voicemail after Emailing	Whether to keep the local voicemail recording after sending them. If set to "Default", the global settings will be used. Note: When set to "Default", the global settings in Call Features → Voicemail → Voicemail Email Settings will be used.
Enable Keep-alive	If enabled, an empty SDP packet will be sent to the SIP server periodically to keep the NAT port open. The default setting is "No".
Keep-alive Frequency	Configure the Keep-alive interval (in seconds) to check if the host is up. The default setting is 60 seconds.
Enable SCA	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.
Emergency CID Name	CallerID name that will be used for emergency calls and callbacks.
Disable This Extension	If selected, this extension will be disabled on the UCM630X. Note: The disabled extension still exists on the PBX but cannot be used on the end device.
Sync Contact	If enabled, this extension number will be displayed in the Wave contact, otherwise, it will not be displayed, and it cannot be found in the chat, but the user can still dial this number.
User Settings	
First Name	Configure the first name of the user. The first name can contain characters, letters, digits, and _.
Last Name	Configure the last name of the user. The last name can contain characters, letters, digits, and _.
Email Address	Fill in the Email address for the user. Voicemail will be sent to this Email address.
User Password	Configure the password for user portal access. A random numeric password is automatically generated. It is recommended to use the randomly generated password for security purposes.
Language	Select the voice prompt language to be used for this extension. The default setting is "Default" which is the selected voice prompt language under Web GUI→PBX Settings→Voice Prompt→Language Settings . The dropdown list shows all the currently available voice prompt languages on the UCM630X. To add more languages to the list, please download the voice prompt package by selecting "Check Prompt List" under Web GUI→PBX Settings→Voice Prompt→Language Settings .
Concurrent Registrations	The maximum endpoints which can be registered into this extension. For security concerns, the default value is 3. Note:
Mobile Phone Number	Configure the phone number for the extension, user can type the related star code for the phone number followed by the extension number to directly call this number. For example, the user can type *881000 to call the mobile number associated with extension 1000.
Department	Configure the user's department. The department can be configured in User Management->Address Book Management->Department Management. Job Title: The user's department position.
Contact Privileges	
Same as Department Contact Privileges	When enabled, The extension will inherit the same privilege attributed to the department it belongs to.

Contact View Privileges	Select the privileges regarding the contact view in SIP endpoints and Wave.
--------------------------------	---

SIP Settings	
NAT	Use NAT when the UCM630X is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is a one-way audio issue, usually it is related to NAT configuration or the Firewall's support of SIP and RTP ports. The default setting is enabled.
Enable Direct Media	By default, the UCM630X 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 UCM630X to negotiate endpoint-to-endpoint media routing. The default setting is "No".
DTMF Mode	Select DTMF mode for the user to send DTMF. The default setting is "RFC4733". If "Info" is selected, the 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.
TEL URI	If the phone has an assigned PSTN telephone number, this field should be set to "User=Phone". The "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.
Alert-Info	When present in an INVITE request, the alert-Info header field specifies an alternative ring tone to the UAS.
Enable T.38 UDPTL	Enable or disable T.38 UDPTL support.
TURN Relay	<p>Enable this option if the following are true:</p> <ol style="list-style-type: none"> 1. UCM is deployed on a private network. 2. There are remote endpoints outside the UCM's network registering to it via its public IP address. 3. The network's firewall is not configured for media port forwarding. 4. Media NAT penetration is required. <p>Once a TURN server is configured, media will be forwarded to it. This configuration does not affect endpoints that are registered via the UCM's RemoteConnect address.</p>
SRTP	Enable SRTP for the call. The default setting is disabled.
Jitter Buffer	<p>Select the jitter buffer method.</p> <ul style="list-style-type: none"> ● Disable: Jitter buffer will not be used. ● Fixed: Jitter buffer with a fixed size (equal to the value of "jitter buffer size") ● Adaptive: Jitter buffer with an adaptive size (no more than the value of "max jitter buffer"). ● NetEQ: Dynamic jitter buffer via NetEQ.
Packet Loss Retransmission	<p>Configure to enable Packet Loss Retransmission.</p> <ul style="list-style-type: none"> ● NACK ● NACK+RTX(SSRC-GROUP) ● OFF
Video FEC	Check to enable Forward Error Correction (FEC) for Video.
Audio FEC	Check to enable Forward Error Correction (FEC) for Audio.
Silence Suppression	If enabled, the UCM will send CN packets for silence suppression after a successful CN negotiation in the SIP SDP. If the client endpoint's OPUS codec supports the reception of DTX packets, the UCM

	will send DTX packets instead.
FECC	Configure to enable Remote Camera Management.
ACL Policy	<p>Access Control List manages the IP addresses that can register to this extension.</p> <ul style="list-style-type: none"> ● Allow All: Any IP address can register to this extension. ● Local Network: Only IP addresses in the configured network segments can register to this extension. Press “Add Local Network Address” to add more IP segments.
SRTP Crypto Suite	<p>The following encryption protocols can be used to encrypt an RTP stream.</p> <ul style="list-style-type: none"> ● AES_CM_128_HMAC_SHA1_80 (This is the default used protocol) ● AES_256_CM_HMAC_SHA1_80 ● AEAD_AES_128_GCM ● AEAD_AES_256_GCM
Codec Preference	Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p, RTX and VP8.

Call Transfer	
Presence Status	Select which presence status to set for the extension and configure call forward conditions for each status. Six possible options are possible: “Available”, “Away”, “Chat”, “Custom”, “DND” and “Unavailable”. More details at [PRESENCE].
Internal Calls & External Calls	
Call Forward Unconditional	<p>Enable and configure the Call Forward Unconditional target number. Available options for target number are:</p> <ul style="list-style-type: none"> ● “None”: Call forward deactivated. ● “Extension”: Select an extension from the dropdown list as CFU target. ● “Custom Number”: Enter a customer number as a target. For example: *97. ● “Voicemail”: Select an extension from the dropdown list. Incoming calls will be forwarded to the voicemail of the selected extension. ● “Ring Group”: Select a ring group from the dropdown list as CFU target. ● “Queues”: Select a queue from the dropdown list as CFU target. ● “Voicemail Group”: Select a voicemail group from the dropdown list as CFU target. ● Custom Prompt: The call will be forwarded to a custom prompt. <p>The default setting is “None”.</p>
CFU Time Condition	<p>Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time conditions are ‘All’, ‘Office Time’, ‘Out of Office Time’, ‘Holiday’, ‘Out of Holiday’, ‘Out of Office Time or Holiday’, ‘Office Time and Out of Holiday’, ‘Specific Time’, ‘Out of Specific Time’, ‘Out of Specific Time or Holiday’, ‘Specific Time and Out of Holiday’.</p> <p>Notes:</p> <ul style="list-style-type: none"> ● “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. ● Specific time can be configured under the Specific Time section. Scroll down the add Time Condition for a specific time. ● Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.
Call Forward No Answer	<p>Configure the Call Forward No Answer target number. Available options for target number are:</p> <ul style="list-style-type: none"> ● “None”: Call forward deactivated. ● “Extension”: Select an extension from the dropdown list as CFN target. ● “Custom Number”: Enter a customer number as a target. For example: *97. ● “Voicemail”: Select an extension from the dropdown list. Incoming calls will be forwarded to the voicemail of the selected extension.

	<ul style="list-style-type: none"> ● “Ring Group”: Select a ring group from the dropdown list as CFN target. ● “Queues”: Select a queue from the dropdown list as CFN target. ● “Voicemail Group”: Select a voicemail group from the dropdown list as CFN target. ● Custom Prompt: The call will be forwarded to a custom prompt. <p>The default setting is “None”.</p>
CFN Time Condition	<p>Select time condition for Call Forward No Answer. The available time conditions are ‘All’, ‘Office Time’, ‘Out of Office Time’, ‘Holiday’, ‘Out of Holiday’, ‘Out of Office Time or Holiday’, ‘Office Time and Out of Holiday’, ‘Specific Time’, ‘Out of Specific Time’, ‘Out of Specific Time or Holiday’, ‘Specific Time and Out of Holiday’.</p> <p>Notes:</p> <ul style="list-style-type: none"> ● “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. ● Specific time can be configured under the Specific Time section. Scroll down the add Time Condition for a specific time. ● Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.
Call Forward Busy	<p>Configure the Call Forward Busy target number. Available options for target number are:</p> <ul style="list-style-type: none"> ● “None”: Call forward deactivated. ● “Extension”: Select an extension from the dropdown list as CFB target. ● “Custom Number”: Enter a customer number as a target. For example: *97 ● “Voicemail”: Select an extension from the dropdown list. Incoming calls will be forwarded to the voicemail of the selected extension. ● “Ring Group”: Select a ring group from the dropdown list as CFB target. ● “Queues”: Select a queue from the dropdown list as CFB target. ● “Voicemail Group”: Select a voicemail group from dropdown list as CFB target. ● Custom Prompt: <p>The default setting is “None”.</p>
CFB Time Condition	<p>Select time condition for Call Forward Busy. The available time conditions ‘All’, ‘Office Time’, ‘Out of Office Time’, ‘Holiday’, ‘Out of Holiday’, ‘Out of Office Time or Holiday’, ‘Office Time and Out of Holiday’, ‘Specific Time’, ‘Out of Specific Time’, ‘Out of Specific Time or Holiday’, ‘Specific Time and Out of Holiday’.</p> <p>Notes:</p> <ul style="list-style-type: none"> ● “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. ● Specific time can be configured under the Specific Time section. Scroll down the add Time Condition for a specific time. ● Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.
Do Not Disturb	<p>If Do Not Disturb is enabled, all incoming calls will be dropped. All call forward settings will be ignored.</p>
DND Time Condition	<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>Notes:</p> <ul style="list-style-type: none"> ● “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. ● Specific time can be configured under the Specific Time section. Scroll down the add Time Condition for a specific time. <p>Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</p>
DND Whitelist	<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"> ● Z match any digit from 1-9. ● N match any digit from 2-9. ● X match any digit from 0-9.
FWD Whitelist	<p>Calls from users in the forward whitelist will not be forwarded. Pattern matching is supported.</p>

	<ul style="list-style-type: none"> • Z match any digit from 1-9. • N match any digit from 2-9. • X match any digit from 0-9.
CC Settings	
Enable CC	If enabled, UCM630X 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.
CC Mode	<p>Two modes for Call Completion are supported:</p> <ul style="list-style-type: none"> • Normal: This extension is used as an ordinary extension. • For Trunk: This extension is registered from a PBX. <p>The default setting is "Normal".</p>
CC Max Agents	<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.</p> <p>The minimum value is 1.</p>
CC Max Monitors	<p>Configure the maximum number of monitor structures that 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.</p> <p>The minimum value is 1.</p>
Ring Simultaneously	
Ring Simultaneously	Enable this option to have an external number ring simultaneously along with the extension. If a register trunk is used for outbound, the register number will be used to be displayed for the external number as the caller ID number.
External Number	<p>Set the external number to ring simultaneously. '-' is the connection character that will be ignored.</p> <p>This field accepts only letters, numbers, and special characters + = * #.</p>
Time Condition for Ring Simultaneously	Ring the external number simultaneously along with the extension based on this time condition.
Use callee DOD on FWD or RS	Use the DOD number when calls are being diverted/forwarded to external destinations or when ring simultaneous is configured.
Monitor privilege control	
Call Monitoring Whitelist	Add members from "Available Extensions" to "Selected Extensions" so that the selected extensions can spy on the used extension using feature code.
Allow Operator Panel Monitoring	Configure whether this extension can be monitored by the Operator Panel administrator.
Seamless transfer privilege control	
Allowed to seamless transfer	Any extensions on the UCM can perform a seamless transfer. When using the Pickup Incall feature, only extensions available on the "Selected Extensions" list can perform a seamless transfer to the edited extension.
PMS Remote Wakeup Whitelist	
Select the extensions that	Selected extensions can set a PMS wakeup service for this extension via feature code.

can set wakeup service for other extensions	
Other Settings	
Ring Timeout	<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 UCM630X. The valid range is between 5 seconds and 600 seconds.</p> <p>Note: If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
Auto Record	<p>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.</p>
Skip Trunk Auth	<ul style="list-style-type: none"> • 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	<p>If ‘Skip Trunk Auth’ is set to ‘By Time’, select a time condition during which users can skip entering the password when making outbound calls.</p>
Dial Trunk Password	<p>Configure personal password when making outbound calls via the trunk.</p>
Support Hot-Desking Mode	<p>Check to enable Hot-Desking Mode on the extension. Hot-Desking allows using the same endpoint device and logs in using extension/password combination. This feature is used in scenarios where different users need to use the same endpoint device during a different time of the day for instance. If enabled, SIP Password will accept only alphabet characters and digits. Auth ID will be changed to the same as Extension.</p>
Enable LDAP	<p>If enabled, the extension will be added to the LDAP Phonebook PBX list. Default is enabled.</p>
Use MOH as IVR ringback tone	<p>If enabled, when the call to the extension is made through the IVR, the caller will hear MOH as a ringback tone instead of the regular ringback tone.</p>
Music On Hold	<p>Specify which Music On Hold class to suggest to the bridged channel when putting them on hold.</p>
Call Duration Limit	<p>Check to enable and set the call limit the duration.</p>
Maximum Call Duration (s)	<p>The maximum call duration (in seconds). The default value 0 means no limit. Max value is 86400 seconds</p>
The Maximum Number of Call Lines	<p>The maximum number of simultaneous calls that the extension can have. 0 indicates no limit.</p>
Outgoing Call Frequency Limit	<p>If enabled, if the number of outbound calls exceed the configured threshold within the specified period, further outbound calls will be not be allowed.</p>
Send PCPID Header	<p>If enabled, this extension's SIP INVITE messages will contain the P-Called-Party-ID (PCPID) header if the callee is a SIP device.</p>

Period (m)	The period of outgoing call frequency limit. The valid range is from 1 to 120. The default value is 1.
Max Number of Calls	Set the maximum number of outgoing calls in a period. The valid range is from 1 to 20. The default value is 5.
Enable Auto-Answer Support	If enabled, the extension will support auto-answer when indicated by Call-info/Alert-info headers.
Call Waiting	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.
Stop Ringing	If enabled, when the extension has concurrent registrations on multiple devices, upon incoming call or meeting invite ringing, if one end device rejects the call, the rest of the devices will also stop ringing. By default, it's disabled.
Email Missed Call Log	If enabled, the log of missed calls will be sent to the extension's configured email address.
Missed Call Type	<p>If Email Missed Calls 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"> • Default: All missed calls will be sent in email notifications. • Missed Internal Call: Only missed local extension-to-extension calls will be sent in email notifications. • Missed External Call: Only missed calls from trunks will be sent in email notifications.

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

SIP Extension Configuration Parameters → Specific Time

Follow Me	
Enable	Configure to enable or disable Follow Me for this user.
Skip Trunk Auth	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.
Music On Hold Class	Configure the Music On Hold class that the caller would hear while tracking the user.
Confirm When Answering	If enabled, call will need to be confirmed after answering.
Enable Destination	Configure to enable destination
Default Destination	The call will be routed to this destination if no one in the Follow Me answers the call.
Use Callee DOD for Follow Me	Use the callee DOD number as CID if configured Follow Me numbers are external numbers.
Play Follow Me Prompt	If enabled, the Follow Me prompt tone will be played
New Follow Me Number	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.
Dialing Order	This is the order in which the Follow Me destinations will be dialed to reach the user.

SIP Extension Configuration Parameters → Follow Me

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.

General	
Extension	The extension number associated with the user.
CallerID Number	Configure the CallerID Number that would be applied for outbound calls from this user. Note: The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
Privilege	Assign permission level to the user. The available permissions are “Internal”, “Local”, “National” and “International” from the lowest level to the highest level. The default setting is “Internal”. Note: Users need to have the same level as or higher level than an outbound rule’s privilege to make outbound calls using this rule.
SIP/IAX Password	Configure the password for the user. A random secure password will be automatically generated. It is recommended to use this password for security purposes.
Voicemail	Configure Voicemail. There are three valid options, and the default option is “Enable Local Voicemail”. <ul style="list-style-type: none">• Disable Voicemail: Disable Voicemail.• Enable Local Voicemail: Enable voicemail for the user.
Voicemail Password	Configure voicemail password (digits only) for the user to access the voicemail box. A random numeric password is automatically generated. It is recommended to use the randomly generated password for security purposes.
Skip Voicemail Password Verification	When a 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.
Send Voicemail Email Notification	Configures whether to send emails to the extension’s email address to notify of a new voicemail.
Attach Voicemail to Email	Configures whether to attach a voicemail audio file to the voicemail notification emails.
Keep Voicemail after Emailing	Only applies if extension-level or global Send Voicemail to Email is enabled.
Disable This Extension	If selected, this extension will be disabled on the UCM630X. Note: The disabled extension still exists on the PBX but cannot be used on the end device.
User Settings	
First Name	Configure the first name of the user. The first name can contain characters, letters, digits, and _.
Last Name	Configure the last name of the user. The last name can contain characters, letters, digits, and _.
Email Address	Fill in the Email address for the user. Voicemail will be sent to this Email address.
User Password	Configure the password for user portal access. A random numeric password is automatically generated. It is recommended to use the randomly generated password for security purposes.

Language	Select the voice prompt language to be used for this extension. The default setting is "Default" which is the selected voice prompt language under Web GUI→PBX Settings→Voice Prompt→Language Settings . The dropdown list shows all the currently available voice prompt languages on the UCM630X. To add more languages to the list, please download the voice prompt package by selecting "Check Prompt List" under Web GUI→PBX Settings→Voice Prompt→Language Settings .
Mobile Phone Number	Configure the Mobile number of the user.

IAX Settings	
Max Number of Calls	Configure the maximum number of calls allowed for each remote IP address.
Require Call Token	Configure to enable/disable requiring call token. If set to "Auto", it might lock out users who depend on backward compatibility when peer authentication credentials are shared between physical endpoints. The default setting is "Yes".
SRTP	Enable SRTP for the call. The default setting is disabled.
ACL Policy	<p>Access Control List manages the IP addresses that can register to this extension.</p> <ul style="list-style-type: none"> ● Allow All: Any IP address can register to this extension. ● Local Network: Only IP addresses in the configured network segments can register to this extension.
Codec Preference	Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p, RTX and VP8.

Call Transfer	
Call Forward Unconditional	Configure the Call Forward Unconditional target number. If not configured, the Call Forward Unconditional feature is deactivated. The default setting is deactivated.
CFU Time Condition	<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"> ● "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. ● Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for a specific time. ● Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.
Call Forward No Answer	Configure the Call Forward No Answer target number. If not configured, the Call Forward No Answer feature is deactivated. The default setting is deactivated.
CFN Time Condition	<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"> ● "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. ● Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for a specific time. ● Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.

Call Forward Busy	Configure the Call Forward Busy target number. If not configured, the Call Forward Busy feature is deactivated. The default setting is deactivated.
CFB Time Condition	<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"> • “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for a specific time. <p>Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</p>
Do Not Disturb	If Do Not Disturb is enabled, all incoming calls will be dropped. All call forward settings will be ignored.
DND Time Condition	The time condition of DND. The DND will take effect while the time condition is satisfied.
DND Whitelist	<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"> • Z match any digit from 1-9. • N match any digit from 2-9. • X match any digit from 0-9.
FWD Whitelist	<p>Calls from users in the forward whitelist will not be forwarded. Pattern matching is supported.</p> <ul style="list-style-type: none"> • Z match any digit from 1-9. • N match any digit from 2-9. • X match any digit from 0-9.
Ring Simultaneously	
Ring Simultaneously	Enable this option to have an external number ring simultaneously along with the extension. If a register trunk is used for outbound, the register number will be used to be displayed for the external number as the caller ID number.
External Number	Set the external number to ring simultaneously. ‘-’ is the connection character that will be ignored.
Time Condition for Ring Simultaneously	Ring the external number simultaneously along with the extension based on this time condition.
Use callee DOD on FWD or RS	Use the callee’s DOD number as CallerID on Outgoing Forwarding or Ring Simultaneously calls.
Monitor Privilege Control	
Call Monitoring Whitelist	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	<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 UCM630X, 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>Note: If the endpoint also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
Auto Record	<p>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.</p>
Skip Trunk Auth	<ul style="list-style-type: none"> • 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	<p>If “Skip Trunk Auth” is set to “By Time”, select a time condition during which users can skip entering the password when making outbound calls.</p>
Dial Trunk Password	<p>Configure personal password when making outbound calls via the trunk.</p>
Enable LDAP	<p>If enabled, the extension will be added to LDAP Phonebook PBX lists.</p>
Music On Hold	<p>Configure the Music On Hold class to suggest to the bridged channel when putting them on hold.</p>
Use MOH as IVR ringback tone	<p>If enabled, when the call to the extension is made through the IVR, the caller will hear MOH as a ringback tone instead of the regular ringback tone.</p>
Call Duration Limit	<p>Check to enable and set the call limit the duration.</p>
Maximum Call Duration (s)	<p>The maximum call duration (in seconds). The default value 0 means no limit. Max value is 86400 seconds</p>
Email Missed Calls	<p>Send a log of missed calls to the extension’s email address.</p>
Missed Call Type	<p>If Email Missed Calls 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"> • Default: All missed calls will be sent in email notifications. • Missed Internal Call: Only missed local extension-to-extension calls will be sent in email notifications. • Missed External Call: Only missed calls from trunks will be sent in email notifications.

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

Follow Me	
Enable	Configure to enable or disable Follow Me for this user.
Skip Trunk Auth	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.

Music On Hold Class	Configure the Music On Hold class that the caller would hear while tracking the user.
Confirm When Answering	If enabled, call will need to be confirmed after answering.
Enable Destination	Configure to enable destination.
Default Destination	The call will be routed to this destination if no one in the Follow Me answers the call.
Use Callee DOD for Follow Me	Use the callee DOD number as CID if configured Follow Me numbers are external numbers.
Play Follow Me Prompt	If enabled, the Follow Me prompt tone will be played.
New Follow Me Number	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.
Dialing Order	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.

General	
Extension	The extension number associated with the user.
Analog Station	Select the FXS port to be assigned for this extension.
Caller ID Number	Configure the CallerID Number that would be applied for outbound calls from this user. Note: The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
Privilege	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". Note: Users need to have the same level as or higher level than an outbound rule's privilege to make outbound calls using this rule.
Voicemail	Configure Voicemail. There are three valid options, and the default option is "Enable Local Voicemail". <ul style="list-style-type: none">● Disable Voicemail: Disable Voicemail.● Enable Local Voicemail: Enable voicemail for the user.
Voicemail Password	Configure voicemail password (digits only) for the user to access the voicemail box. A random numeric password is

	automatically generated. It is recommended to use the randomly generated password for security purposes.
Skip Voicemail Password Verification	When a 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.
Send Voicemail Email Notification	Configures whether to send emails to the extension's email address to notify of a new voicemail.
Attach Voicemail to Email	Configures whether to attach a voicemail audio file to the voicemail notification emails.
Keep Voicemail after Emailing	Only applies if extension-level or global Send Voicemail to Email is enabled.
Emergency CID Name	CallerID name that will be used for emergency calls and callbacks.
Disable This Extension	If selected, this extension will be disabled on the UCM630X. Note: The disabled extension still exists on the PBX but cannot be used on the end device.
User Settings	
First Name	Configure the first name of the user. The first name can contain characters, letters, digits, and _.
Last Name	Configure the last name of the user. The last name can contain characters, letters, digits, and _.
Email Address	Fill in the Email address for the user. Voicemail will be sent to this Email address.
User Password	Configure the password for user portal access. A random numeric password is automatically generated. It is recommended to use the randomly generated password for security purposes.
Mobile Phone Number	Configure the Mobile number of the user.
Language	Select the voice prompt language to be used for this extension. The default setting is "Default" which is the selected voice prompt language under Web GUI→PBX Settings→Voice Prompt→Language Settings . The dropdown list shows all the currently available voice prompt languages on the UCM630X. To add more languages to the list, please download the voice prompt package by selecting "Check Prompt List" under Web GUI→PBX Settings→Voice Prompt→Language Settings .

Analog Settings	
Call Waiting	Configure to enable/disable call waiting feature. The default setting is "No".

User '#' as SEND	If configured, the # key can be used as SNED key after dialing the number on the analog phone. The default setting is "Yes".
RX Gain	Configure the RX gain for the receiving channel of the analog FXS port. The valid range is -30dB to +6dB. The default setting is 0.
TX Gain	Configure the TX gain for the transmitting channel of the analog FXS port. The valid range is -30dB to +6dB. The default setting is 0.
Min RX Flash	Configure the minimum period of time (in milliseconds) that the hook flash must remain unpressed for the PBX to consider the event as a valid flash event. The valid range is 30ms to 1000ms. The default setting is 200ms.
Max RX Flash	Configure the maximum period of time (in milliseconds) that the hook flash must remain unpressed for the PBX to consider the event as a valid flash event. The minimum period of time is 256ms and it cannot be modified. The default setting is 1250ms.
Enable Polarity Reversal	If enabled, a polarity reversal will be marked as received when an outgoing call is answered by the remote party. For some countries, a polarity reversal is used for signaling the disconnection of a phone line and the call will be considered as Hangup on a polarity reversal. The default setting is "Yes".
Echo Cancellation	Specify "ON", "OFF" or a value (the power of 2) from 32 to 1024 as the number of taps of cancellation. Note: When configuring the number of taps, the number 256 is not translated into 256ms of echo cancellation. Instead, 256 taps mean $256/8 = 32$ ms. The default setting is "ON", which is 128 taps.
3-Way Calling	Configure to enable/disable the 3-way calling feature on the user. The default setting is enabled.
Send CallerID After	Configure the number of rings before sending CID. The default setting is 1.
Fax Mode	For the FXS extension, there are three options available in Fax Mode. The default setting is "None". <ul style="list-style-type: none"> • None: Disable Fax. • Fax Gateway: If selected, the UCM630X can support the conversation and processing of Fax data from T.30 to T.38 or T.38 to T.30. only for FXS ports. • Fax Detection: 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 Default Email Address configured in Fax/T.38->Fax Settings.

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

CFN Time Condition	<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"> • “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for a specific time. • Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.
Call Forward Busy	<p>Configure the Call Forward Busy target number. If not configured, the Call Forward Busy feature is deactivated. The default setting is deactivated.</p>
CFB Time Condition	<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"> • “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period. • Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for a specific time. • Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.
Do Not Disturb	<p>If Do Not Disturb is enabled, all incoming calls will be dropped. All call forward settings will be ignored.</p>
DND Time Condition	<p>The time condition of DND. The DND will take effect while the time condition is satisfied.</p>
DND Whitelist	<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"> • Z match any digit from 1-9. • N match any digit from 2-9. • X match any digit from 0-9.
FWD Whitelist	<p>Calls from users in the forward whitelist will not be forwarded. Pattern matching is supported.</p> <ul style="list-style-type: none"> • Z match any digit from 1-9. • N match any digit from 2-9. • X match any digit from 0-9.
CC Settings	
Enable CC	<p>If enabled, UCM630X will automatically alert this extension when a called party is available, given that a previous call to that party failed for some reason.</p>
Ring Simultaneously	
Ring Simultaneously	<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 the caller ID number.</p>
External Number	<p>Set the external number to ring simultaneously. ‘-’ is the connection character that will be ignored.</p>
Time Condition for Ring Simultaneously	<p>Ring the external number simultaneously along with the extension based on this time condition.</p>

Use callee DOD on FWD or RS	Use the callee's DOD number as CallerID on Outgoing Forwarding or Ring Simultaneously calls.
Hotline	
Enable Hotline	If enabled, a hotline dialing plan will be activated, a pre-configured number will be used according to the selected Hotline Type.
Hotline Number	Configure the Hotline Number
Hotline Type	Configure the Hotline Type: <ul style="list-style-type: none"> • Immediate Hotline: When the phone is off-hook, UCM630X will immediately dial the preset number • Delay Hotline: When the phone is off hook, if there is no dialing within 5 seconds, UCM630X will dial the preset number.
Monitor privilege control	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 UCM630X, 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 endpoint 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	<ul style="list-style-type: none"> • 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 a password when making outbound calls.
Dial Trunk Password	Configure personal password when making outbound calls via the trunk.
Enable LDAP	If enabled, this extension will be added to the 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 a ringback tone instead of the regular ringback tone.
Music On Hold	Select which Music On Hold class to suggest to the 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	<p>If Email Missed Calls 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"> • Default: All missed calls will be sent in email notifications. • Missed Internal Call: Only missed local extension-to-extension calls will be sent in email notifications. • Missed External Call: Only missed calls from trunks will be sent in email notifications.

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

Follow Me	
Enable	Configure to enable or disable Follow Me for this user.
Skip Trunk Auth	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.
Music On Hold Class	Configure the Music On Hold class that the caller would hear while tracking the user.
Confirm When Answering	If enabled, call will need to be confirmed after answering.
Enable Destination	Configure to enable destination.
Default Destination	The call will be routed to this destination if no one in the Follow Me answers the call.
Use Callee DOD for Follow Me	Use the callee DOD number as CID if configured Follow Me numbers are external numbers.
Play Follow Me Prompt	If enabled, the Follow Me prompt tone will be played.
New Follow Me Number	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.
Dialing Order	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.

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

CallerID Number	<p>Configure CallerID Number when adding Batch Extensions.</p> <ul style="list-style-type: none"> ○ Use Extension as Number ○ Users can choose to use the extension number as CallerID ○ Use as Number ○ Users can choose to set a specific number instead of using the extension number.
Skip Voicemail Password Verification	When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default, this option is disabled.
Enable Keep-alive	If enabled, the PBX will regularly send SIP OPTIONS to check if host device is online.
Keep-alive Frequency	Configure the keep-alive interval (in seconds) to check if the host is up.
Disable This Extension	Check this box to disable this extension.
Enable SCA	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.
Emergency Calls CID	CallerID number that will be used when calling out and receiving direct callbacks.
Enable Wave	If enabled, this extension number can register, log in and use Wave normally, otherwise it will not be able to use Wave, but the phone function will still be retained.
Sync Contact	If enabled, this extension number will be displayed in the Wave contact, otherwise it will not be displayed, and it cannot be found in the chat, but the user can still dial this number.
Language	Select voice prompt language for this extension. If set to "Default", the global setting for voice prompt language will be used.
Media	
NAT	<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>
Enable Direct Media	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".
DTMF Mode	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.
Alert-info	When present in an INVITE request, the Alert-info header field specifies an alternative ring tone to the UAS.
SRTP	Enable/disable SRTP for RTP stream encryption.
Packet Loss Retransmission	<p>Configure to enable Packet Loss Retransmission.</p> <ul style="list-style-type: none"> ○ NACK ○ NACK+RTX(SSRC-GROUP) ○ OFF
Video FEC	Check to enable Forward Error Correction (FEC) for Video.
FECC	Configure to enable FECC

Audio FEC	Check to enable Forward Error Correction (FEC) for Audio.
ACL Policy	<p>Access Control List manages the IP addresses that can register to this extension.</p> <ul style="list-style-type: none"> ○ Allow All: Any IP address can register to this extension. ○ Local Network: Only IP addresses in the configured network segments can register to this extension. Press "Add Local Network Address" to add more IP segments.
Jitter Buffer	<p>Select jitter buffer method.</p> <ul style="list-style-type: none"> ○ Disable: Jitter buffer will not be used. ○ Fixed: Jitter buffer with a fixed size (equal to the value of "jitter buffer size") ○ Adaptive: Jitter buffer with an adaptive size (no more than the value of "max jitter buffer"). ○ NetEQ: Dynamic jitter buffer via NetEQ.
Codec Preference	Configure the codecs to be used.
Call Transfer	
Presence Status	Select which presence status to set for the extension and configure call forward conditions for each status. Six possible options are possible: "Available" , "Away" , "Chat" , "Custom" , "DND" and "Unavailable" . More details at [PRESENCE].
Call Forward Unconditional	<p>Enable and configure the Call Forward Unconditional target number. Available options for target number are:</p> <ul style="list-style-type: none"> ○ "None": Call forward deactivated. ○ "Extension": Select an extension from dropdown list as CFU target. ○ "Custom Number": Enter a customer number as target. For example: *97. ○ "Voicemail": Select an extension from dropdown list. Incoming calls will be forwarded to voicemail of selected extension. ○ "Ring Group": Select a ring group from dropdown list as CFU target. ○ "Queues": Select a queue from dropdown list as CFU target. ○ "Voicemail Group": Select a voicemail group from dropdown list as CFU target. <p>The default setting is "None".</p>
CFU Time Condition	<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"> ○ "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. ○ Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. ○ Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.

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

Do Not Disturb	<p>If Do Not Disturb is enabled, all incoming calls will be dropped.</p> <p>All call forward settings will be ignored.</p>
DND Time Condition	<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>Notes:</p> <ul style="list-style-type: none"> ○ "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. ○ Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. <p>Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.</p>
DND Whitelist	<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"> ○ Z match any digit from 1-9 ○ N match any digit from 2-9 ○ X match any digit from 0-9.
FWD Whitelist	<p>Calls from users in the forward whitelist will not be forwarded. Pattern matching is supported.</p> <ul style="list-style-type: none"> ○ Z match any digit from 1-9 ○ N match any digit from 2-9 ○ X match any digit from 0-9.
CC Settings	
Enable CC	<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>
CC Mode	<p>Two modes for Call Completion are supported:</p> <ul style="list-style-type: none"> ○ Normal: This extension is used as ordinary extension. ○ For Trunk: This extension is registered from a PBX. <p>The default setting is "Normal".</p>
CC Max Agents	<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>
CC Max Monitors	<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>
Ring Simultaneously	
Ring Simultaneously	<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>

External Number	<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>
Time Condition for Ring Simultaneously	Ring the external number simultaneously along with the extension on the basis of this time condition.
Use callee DOD on FWD or RS	Use the DOD number when calls are being diverted/forwarded to external destinations or when ring simultaneous is configured.
Monitor privilege control	
Allowed to call-barging	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	
Allowed to seamless transfer	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.
Other Settings	
Ring Timeout	<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>Note: If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
Auto Record	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 .
Skip Trunk Auth	<ul style="list-style-type: none"> ○ If set to "yes", users can skip entering the password when making outbound calls. ○ If set to "By Time", users can skip entering the password when making outbound calls during the selected time condition. ○ If set to "No", users will be asked to enter the password when making outbound calls.
Time Condition for Skip Trunk Auth	If 'Skip Trunk Auth' is set to 'By Time', select a time condition during which users can skip entering password when making outbound calls.
Dial Trunk Password	Configure personal password when making outbound calls via trunk.
Enable LDAP	If enabled, the extension will be added to LDAP Phonebook PBX list.
Bind PMS Room	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.
Music On Hold	Specify which Music On Hold class to suggest to the bridged channel when putting them on hold.
Call Duration Limit	The maximum duration of call-blocking.
Maximum Call Duration	The maximum call duration (in seconds). The default value 0 means no limit.

Call Waiting	<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>Note: the option only works when the caller dials the extension directly.</p>
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Batch Add SIP Extension Parameters

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.

General	
Create Number	Specify the number of extensions to be added. The default setting is 5.
Extension Incrementation	Select how much to increment successive extensions. For example, if the value is 2, the extensions will be 1000,1002,1004,.....
Extension	The extension number associated with this particular user/phone.
Permission	<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>Note: Users need to have the same level as or higher level than an outbound rule's privilege in order to make outbound calls from this rule.</p>
CallerID Number	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.
Voicemail	<p>Configure Voicemail. There are three valid options and the default option is "Enable Local Voicemail".</p> <p>Disable Voicemail: Disable Voicemail.</p> <p>Enable Local Voicemail: Enable voicemail for the user.</p>
SIP/IAX Password	<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"> ○ User Random Password. <p>A random secure password will be automatically generated. It is recommended to use this password for security purpose.</p> <ul style="list-style-type: none"> ○ Use Extension as Password. ○ Enter a password to be used on all the extensions in the batch.

Voicemail Password	<p>Configure Voicemail password (digits only) for the users.</p> <ul style="list-style-type: none"> ○ User Random Password. <p>A random password in digits will be automatically generated. It is recommended to use this password for security purpose.</p> <ul style="list-style-type: none"> ○ Use Extension as Password. ○ Enter a password to be used on all the extensions in the batch.
Send Voicemail to Email	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.
Keep Voicemail after Emailing	Only applies if extension-level or global Send Voicemail to Email is enabled.
Auto Record	Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under Web GUI→ CDR → Recording Files .
Skip Voicemail Password Verification	When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default, this option is disabled.
Disable This Extension	Check this box to disable this extension.
Language	Select voice prompt language for this extension. If set to "Default", the global setting for voice prompt language will be used.
IAX Settings	
Max Number of Calls	Configure the maximum number of calls allowed for each remote IP address.
Require Call Token	<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.</p> <p>The default setting is "Yes".</p>
SRTP	Enable/disable SRTP for RTP stream encryption.
ACL Policy	<p>Access Control List manages the IP addresses that can register to this extension.</p> <ul style="list-style-type: none"> ○ Allow All: Any IP address can register to this extension. ○ Local Network: Only IP addresses in the configured network segments can register to this extension.
Codec Preference	Configure the codecs to be used.
Call Transfer	
Call Forward Unconditional	Enable and configure the Call Forward Unconditional target number.

<p>CFU Time Condition</p>	<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"> ○ "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. ○ Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. ○ Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.
<p>Call Forward No Answer</p>	<p>Configure the Call Forward No Answer target number.</p>
<p>CFN Time Condition</p>	<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"> ○ "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. ○ Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. ○ Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.
<p>Call Forward Busy</p>	<p>Configure the Call Forward Busy target number.</p>
<p>CFB Time Condition</p>	<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"> ○ "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. ○ Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. ○ Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.
<p>Do Not Disturb</p>	<p>If Do Not Disturb is enabled, all incoming calls will be dropped.</p> <p>All call forward settings will be ignored.</p>
<p>DND Time Condition</p>	<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>Notes:</p> <ul style="list-style-type: none"> ○ "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period. ○ Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time. ○ Office Time and Holiday could be configured on page System Settings→Time Settings→Office Time/Holiday page.

DND Whitelist	<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"> ○ Z match any digit from 1-9, ○ N match any digit from 2-9, ○ X match any digit from 0-9.
FWD Whitelist	<p>Calls from users in the forward whitelist will not be forwarded. Pattern matching is supported.</p> <ul style="list-style-type: none"> ○ Z match any digit from 1-9, ○ N match any digit from 2-9, ○ X match any digit from 0-9.
Ring Simultaneously	
Ring Simultaneously	<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>
External Number	<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>
Time Condition for Ring Simultaneously	<p>Ring the external number simultaneously along with the extension on the basis of this time condition.</p>
Use callee DOD on FWD or RS	<p>Use the DOD number when calls are being diverted/forwarded to external destinations or when ring simultaneous is configured.</p>
Monitor privilege control	
Allowed to call-barging	<p>Add members from "Available Extensions" to "Selected Extensions" so that the selected extensions can spy on the used extension using feature code.</p>
Seamless transfer privilege control	
Allowed to seamless transfer	<p>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.</p>
Other Settings	
Ring Timeout	<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>Note: If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
Auto Record	<p>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.</p>

Skip Trunk Auth	<ul style="list-style-type: none"> ○ If set to "yes", users can skip entering the password when making outbound calls. ○ If set to "By Time", users can skip entering the password when making outbound calls during the selected time condition. ○ If set to "No", users will be asked to enter the password when making outbound calls.
Time Condition for Skip Trunk Auth	If 'Skip Trunk Auth' is set to 'By Time', select a time condition during which users can skip entering password when making outbound calls.
Dial Trunk Password	Configure personal password when making outbound calls via trunk.
Enable LDAP	If enabled, the extension will be added to LDAP Phonebook PBX list.
Music On Hold	Specify which Music On Hold class to suggest to the bridged channel when putting them 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

Batch Add IAX Extension Parameters

Batch Extension Resetting Functionality




Users can select multiple extensions and reset their settings to default by pressing the reset button

 Reset

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.

Extensions

<input type="checkbox"/>	STATUS	PRESENC...	EXTENSION	NAME	MESSAGE	TYPE	IP AND PORT	EXTENSION I...	OPTIONS
<input type="checkbox"/>	Unavailable	Available	1000		0/0/0	SIP(WebRT...	--		
<input type="checkbox"/>	Idle	Available	1001		0/1/0	SIP(WebRT...	192.168.5.10...		
<input type="checkbox"/>	Unavailable	Available	1002		0/0/0	SIP(WebRT...	--		
<input type="checkbox"/>	Unavailable	Available	1003		0/0/0	SIP(WebRT...	--		
<input type="checkbox"/>	Unavailable	Available	1004		0/0/0	SIP(WebRT...	--		

Total: 5 30 / page Goto 1

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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.

Note

This is the expected behavior when you reset an extension:

- All the data and configuration on the user side will be deleted. That includes user information, call history, call recordings, faxes, voice mails, meeting schedules and recordings, as well as chat history. However, the data related to the user will be kept on the UCM side.

- The extension will be removed from group chats and the messages sent previously by the extension will be kept. However, only other users can search through those messages while the new user of the extension cannot.
- If the extension was in a meeting schedule, the meeting will still be present. The extension will be removed from the meeting and will not be notified about the meeting.

- **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 → Device Management → **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".

Notes

This is the expected behavior when you delete an extension:

- The system will delete all the data of the extension except the CDR and meetings record. All the data on the user side will be erased.
- The extension will be removed from group chats and the messages sent previously by the extension will be kept. However, only other users can search through those messages while the new user of the extension cannot.
- If the extension was in a meeting schedule, the meeting will still be present. The extension will be removed from the meeting and will not be notified about the meeting.

- **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.

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)		

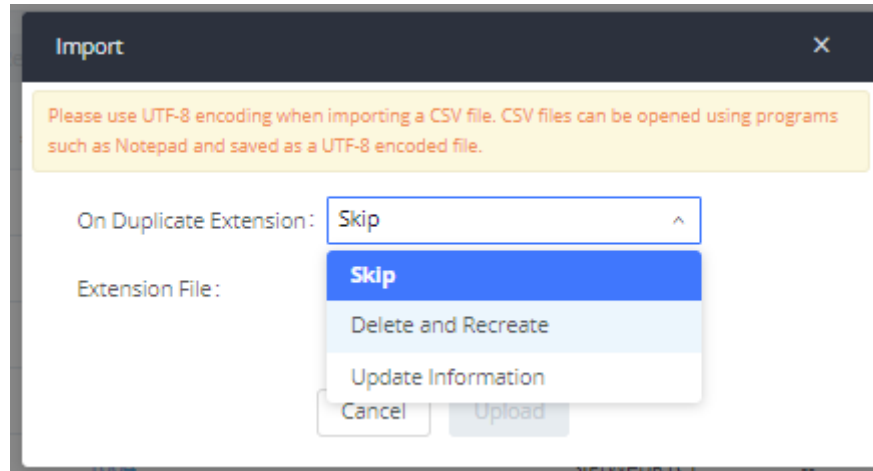
Export Extensions

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.



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.
 - o **Skip:** Duplicate extensions in the csv file will be skipped. The PBX will keep the current extension information as previously configured without change.
 - o **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.
 - o **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	O	P
1	Extension	First Nam	Last Name	Technolog	Enable Vo	CallerID	SIP/IAX P	Voicemail	Skip Voic	Ring Time	Auto Reco	SRTP	Fax Mode	Strategy	Local Sub	Local Sub
2	1000			SIP(WebRT	yes		123456Adm	130270	no		off	no	None	Allow All		
3	1001			SIP(WebRT	no		123456Adm	2426	no		off	no	None	Allow All		
4	1002			SIP(WebRT	yes		123456Adm	771324	no		off	no	None	Allow All		
5	1003			SIP(WebRT	yes	1003	123456Adm	398159	no		off	no	None	Allow All		
6	1004			SIP(WebRT	yes		123456Adm	89737	no		off	no	None	Allow All		

Import File

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

CC Max Monitors	Value from 1-999
Ring simultaneously	Yes/no
External Number	Digits
Time Condition for Ring Simultaneously	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Time Condition for Skip Trunk Auth	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Enable LDAP	Yes/no
Enable T.38 UDPTL	Yes/no
Max Contacts	Values from 1-10
Enable Wave	Yes/no
Alert-Info	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
Do Not Disturb	Yes/no
DND Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Custom Auto answer	Yes/no
Do Not Disturb Whitelist	Empty/digits
User Password	Alphanumeric characters.
First Name	Alphanumeric without special characters.
Last Name	Alphanumeric without special characters.
Email Address	Email address
Language	Default/en/zh
Phone Number	Digits
Call-Barging Monitor	Extensions allowed to call barging
Seamless Transfer Members	Extensions allowed to seamless transfer

SIP extensions Imported File Example

Field	Supported values
Extension	Digits
Technology	IAX
Enable Voicemail	yes/no
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,ILBC,AAL2-G.726-32,ADPCM,G.723,H.263,H.263p,vp8,opus
Permission	Internal/Local/National/International
NAT	yes/no
Call Forward Busy	Digits
Call Forward No Answer	Digits
Call Forward Unconditional	Digits
Require Call Token	Yes/no/auto
Max Number of Calls	Values from 1-512
Dial Trunk Password	Digits
Disable This Extension	Yes/no
CFU Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
CFN Time Condition	
CFB Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Music On Hold	Default/ringbacktone_default
Ring simultaneously	Yes/no
External Number	Digits
Time Condition for Ring Simultaneously	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Time Condition for Skip Trunk Auth	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Enable LDAP	Yes/no
Limit Max time (s)	empty
Do Not Disturb	Yes/no
DND Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Do Not Disturb Whitelist	Empty/digits
User Password	Alphanumeric characters.
First Name	Alphanumeric without special characters.
Last Name	Alphanumeric without special characters.
Email Address	Email address
Language	Default/en/zh
Phone Number	Digits
Call-Barging Monitor	Extensions allowed to call barging
Seamless Transfer Members	Extensions allowed to seamless transfer

IAX extensions Imported File Example

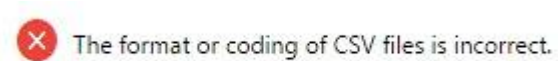
Field	Supported values
Extension	Digits

Technology	FXS
Analog Station	FXS1/FXS2
Enable Voicemail	yes/no
CallerID Number	Digits
Voicemail Password	Digits
Skip Voicemail Password Verification	yes/no
Ring Timeout	Empty/ 3 to 600 (in second)
Auto Record	yes/no
Fax Mode	None/Fax Gateway/Fax Detection
Skip Trunk Auth	Yes/no/bytime
Permission	Internal/Local/National/International
Call Forward Busy	Digits
Call Forward No Answer	Digits
Call Forward Unconditional	Digits
Call Waiting	Yes/no
Use # as SEND	Yes/no
RX Gain	Values from -30→6
TX Gain	Values from -30→6
MIN RX Flash	Values from: 30 – 1000
MAX RX Flash	Values from: 40 – 2000
Enable Polarity Reversal	Yes/no
Echo Cancellation	On/Off/32/64/128/256/512/1024
3-Way Calling	Yes/no
Send CallerID After	1/2
Dial Trunk Password	digits
Disable This Extension	Yes/no
CFU Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
CFN Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
CFB Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Music On Hold	Default/ringbacktone_default
CC Agent Policy	If CC is disabled use: never If CC is set to normal use: generic If CC is set to trunk use: native
CC Monitor Policy	Generic/never
CCBS Available Timer	3600/4800
CCNR Available Timer	3600/7200
CC Offer Timer	60/120
CC Max Agents	Value from 1-999
CC Max Monitors	Value from 1-999
Ring simultaneously	Yes/no

External Number	Digits
Time Condition for Ring Simultaneously	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Time Condition for Skip Trunk Auth	
Enable LDAP	Yes/no
Enable Hotline	Yes/no
Hotline Type	Immediate hotline/delay hotline
Hotline Number	digits
Limit Max time (s)	empty
Do Not Disturb	Yes/no
DND Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Do Not Disturb Whitelist	Empty/digits
User Password	Alphanumeric characters.
First Name	Alphanumeric without special characters.
Last Name	Alphanumeric without special characters.
Email Address	Email address
Language	Default/en/zh
Phone Number	Digits
Call-Barging Monitor	Extensions allowed to call barging
Seamless Transfer Members	Extensions allowed to seamless transfer

FXS Extensions Imported File Example

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.



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.

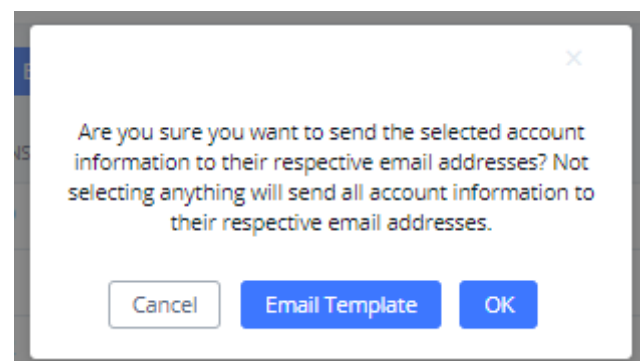
OPTIONS	VALUE
Extension	1000
Status	● Unavailable
Presence Status	Available
Terminal Type	SIP(WebRTC)
CallerID Name	
Message	0/0/0
IP and Port	--
Email Status	To Be Sent
Ring Group	
Call Queue	
Call Queue(Dynamic)	

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.



E-mail Notification – Prompt Information

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

| General Settings

Server Address	192.168.5.147:5060
Account Name	Mia
SIP User ID	1000
Authenticate ID	1000
Authenticate Password	pas1

Account Registration Information

GSWave Settings

Login URL	https://192.168.5.147:8090/#/
Login URL for Public	https://c074ad0a8c94-10671.b.gdms.cloud/#/
Login Name	1000
Login Password	pas1

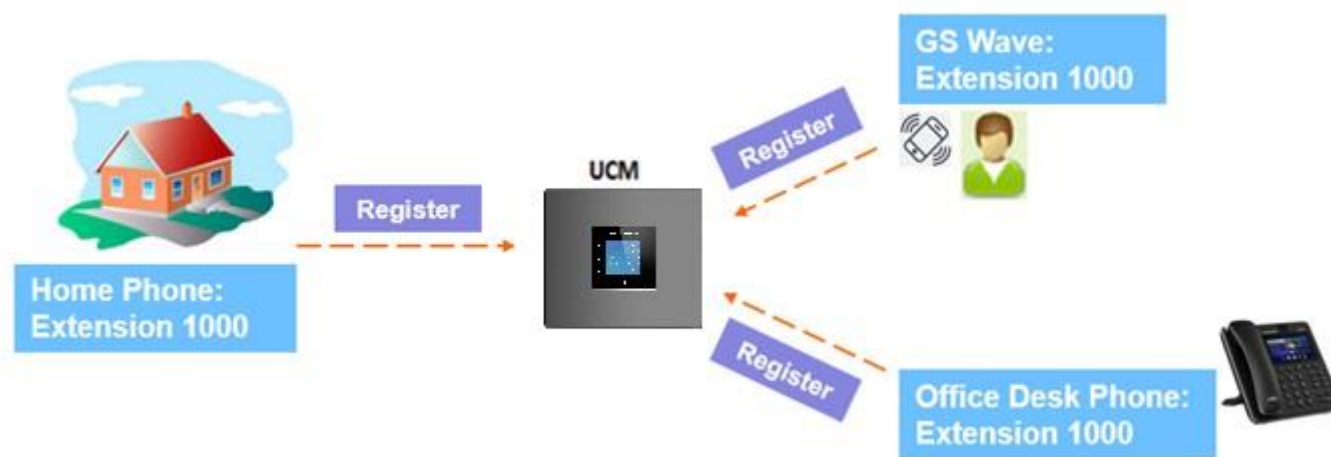


Use Web App to scan qr code and log in

Grandstream Wave 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.



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.

Basic Settings Media Features Specific Time Follow Me Cancel Save

General

* Extension: 1000 CallerID Number: 1000

* Permission: Internal * SIP/IAX Password: *****

AuthID: Voicemail: Local Voicemail

* Voicemail Password: ***** Skip Voicemail Password:

Send Voicemail to Email: Default Verification: Keep Voicemail after Emailing: Default

Enable Keep-alive: * Keep-alive Frequency: 60

Disable This Extension: Enable SCA:

Emergency Calls CID:

User Settings

First Name: Last Name:

Email Address: * User Password: *****

* Language: Default * Concurrent Registrations: 3

Mobile Phone Number:

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.



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**.

- o Click on + Add to create a new extension group.
- o 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.

Create New Extension Group

* Name:

Members:

<input type="checkbox"/> 0 item Available	<input type="checkbox"/> 6 items Selected
Search <input type="text"/>	Search <input type="text"/>
None	<input type="checkbox"/> 1000 <input type="checkbox"/> 4001 <input type="checkbox"/> 4002 <input type="checkbox"/> 4003

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.

General

* Calling Rule Name:

* Pattern:

PIN Groups:

Password:

Disable This Route:

Privilege Level:

PIN Groups with Privilege Level:

Enable Filter on Source Caller ID

Enable Filter on Source Caller

ID:

Custom Dynamic Route:

Outbound Route CID:



Available Extensions/Extension Groups:

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.

FXO Port	<p>Select the channel for the analog trunk.</p> <ul style="list-style-type: none"> ○ UCM6300A: 1 channel ○ UCM6302A: 2 channels ○ UCM6304A: 4 channels ○ UCM6308A: 8 channels
Trunk Name	Specify a unique label to identify the trunk when listed in outbound rules, incoming rules and etc.
Advanced Options	
SLA Mode	Enable this option to satisfy two primary use cases, which include emulating a simple key system and creating shared extensions on a PBX. Enable SLA Mode will disable polarity reversal.
Barge Allowed	<p>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.</p> <p>The default setting is Yes.</p>
Hold Access	<p>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.</p> <p>The default setting is Yes.</p>
Enable Polarity Reversal	If enabled, a polarity reversal will be marked as received when an outgoing call is answered by the remote party. For some countries, a polarity reversal is used for signaling the disconnection of a phone line and the call will be considered as "Hangup" on a polarity reversal. The default setting is "No".
Polarity on Answer Delay	When FXO port answers the call, FXS may send a Polarity Reversal. If this interval is shorter than the value of "Polarity on Answer Delay", the Polarity Reversal will be ignored. Otherwise, the FXO will Onhook to disconnect the call. The default setting is 600ms.
Current Disconnect Threshold (ms)	This is the periodic time (in ms) that the UCM630xA will use to check on a voltage drop in the line. The default setting is 200. The valid range is 50 to 3000.
Ring Timeout	Configure the ring timeout (in ms). Trunk (FXO) devices must have a timeout to determine if there was a Hangup before the line is answered. This value can be used to configure how long it takes before the UCM630xA considers a non-ringing line with Hangup activity. The default setting is 8000.
RX Gain	Configure the RX gain for the receiving channel of analog FXO port. The valid range is from -13.5 (dB) to + 12.0 (dB). The default setting is 0.
TX Gain	Configure the TX gain for the transmitting channel of analog FXO port. The valid range is from -13.5 (dB) to + 12.0 (dB). The default setting is 0.

Use CallerID	<p>Configure to enable CallerID detection.</p> <p>The default setting is "Yes".</p>
Caller ID Scheme	<p>Select the Caller ID scheme for this trunk.</p> <ul style="list-style-type: none"> ○ Bellcore/Telcordia. ○ ETSI-FSK During Ringing ○ ETSI-FSK Prior to Ringing with DTAS ○ ETSI-FSK Prior to Ringing with LR ○ ETSI-FSK Prior to Ringing with RP ○ ETSI-DTMF During Ringing ○ ETSI-DTMF Prior to Ringing with DTAS ○ ETSI-DTMF Prior to Ringing with LR ○ ETSI-DTMF Prior to Ringing with RP ○ SIN 227-BT ○ NTT Japan ○ Auto Detect <p>If you are not sure which scheme to choose, please select "Auto Detect". The default setting is "Bellcore/Telcordia".</p>
Fax Mode	<p>Configures how faxes to this extension will be handled.</p> <ul style="list-style-type: none"> ○ None: Faxes will not be processed. ○ Fax Gateway: Faxes to this extension will be processed and converted from T.30 to T.38 or vice-versa. FXS/FXO ports only. <p>The default setting is None.</p>
FXO Dial Delay (ms)	<p>Configure the time interval between off-hook and first dialed digit for outbound calls.</p>
Auto Record	<p>Enable automatic recording for the calls using this trunk. The default setting is disabled. The recording files can be accessed under Web GUI→CDR→Recording Files.</p>
Disable This Trunk	<p>If selected, the trunk will be disabled and incoming/Outgoing calls via this trunk will not be possible.</p>

<p>DAHDI Out Line Selection</p>	<p>This is to implement analog trunk outbound line selection strategy.</p> <p>Three options are available:</p> <ul style="list-style-type: none"> ○ Ascend <p>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).</p> <ul style="list-style-type: none"> ○ Poll <p>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.</p> <ul style="list-style-type: none"> ○ Descend <p>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).</p> <p>The default setting is "Ascend" mode.</p>
<p>Echo Cancellation Mode</p>	<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"> ○ Default: The NLP limits the signal level to the background noise level when active, and the background noise level adjustment is low. ○ High Noise Level Adjustment: The NLP limits the signal level to the background noise level when active, and the background noise level adjustment is high. ○ Noise Masking: The NLP sends sign noise when active, and the background noise level adjustment is high. ○ White Noise Injection: 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.
<p>Direct Callback</p>	<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>
<p>Tone Settings</p>	
<p>Busy Detection</p>	<p>Busy Detection is used to detect far end Hangup or for detecting busy signal. The default setting is "Yes".</p>
<p>Busy Tone Count</p>	<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>
<p>Congestion Detection</p>	<p>Congestion detection is used to detect far end congestion signal. The default setting is "Yes".</p>
<p>Congestion Count</p>	<p>If "Congestion Detection" is enabled, users can specify the number of congestion tones to wait for. The default setting is 2.</p>

Tone Country	Select the country for tone settings. If "Custom" is selected, users could manually configure the values for Busy Tone and Congestion Tone. The default setting is "United States of America (USA)".
Busy Tone	<p>Syntax:</p> <p>f1=val[@level],[f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]];</p> <p>Frequencies are in Hz and cadence on and off are in ms.</p> <p>Frequencies Range: [0, 4000)</p> <p>Busy Level Range: (-300, 0)</p> <p>Cadence Range: [0, 16383].</p> <p>Select Tone Country "Custom" to manually configure Busy Tone value.</p> <p>Default value:</p> <p>f1=480@-50,f2=620@-50,c=500/500</p>
Congestion Tone	<p>Syntax:</p> <p>f1=val[@level],[f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]];</p> <p>Frequencies are in Hz and cadence on and off are in ms.</p> <p>Frequencies Range: [0, 4000)</p> <p>Busy Level Range: (-300, 0)</p> <p>Cadence Range: [0, 16383].</p> <p>Select Tone Country "Custom" to manually configure Busy Tone value.</p> <p>Default value:</p> <p>f1=480@-50,f2=620@-50,c=250/250</p>
PSTN Detection	Click on "Detect" to detect the busy tone, Polarity Reversal and Current Disconnect by PSTN. Before the detecting, please make sure there are more than one channel configured and working properly. If the detection has busy tone, the "Tone Country" option will be set as "Custom".

Analog Trunk Configuration Parameters

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.

Tone Settings

Busy Detection:

Congestion Detection:

* Tone Country: United States of America (USA)

* 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:

UCM630xA FXO Tone Settings

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

PSTN Detection

Detect model: Auto Detect

Source Channel (to be detected): 1

Destination Channel: 1

* Destination Number:

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.

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

Detect model: Auto Detect

Source Channel (to be detected): 1

Destination Channel: 2

* Destination Number: 123654

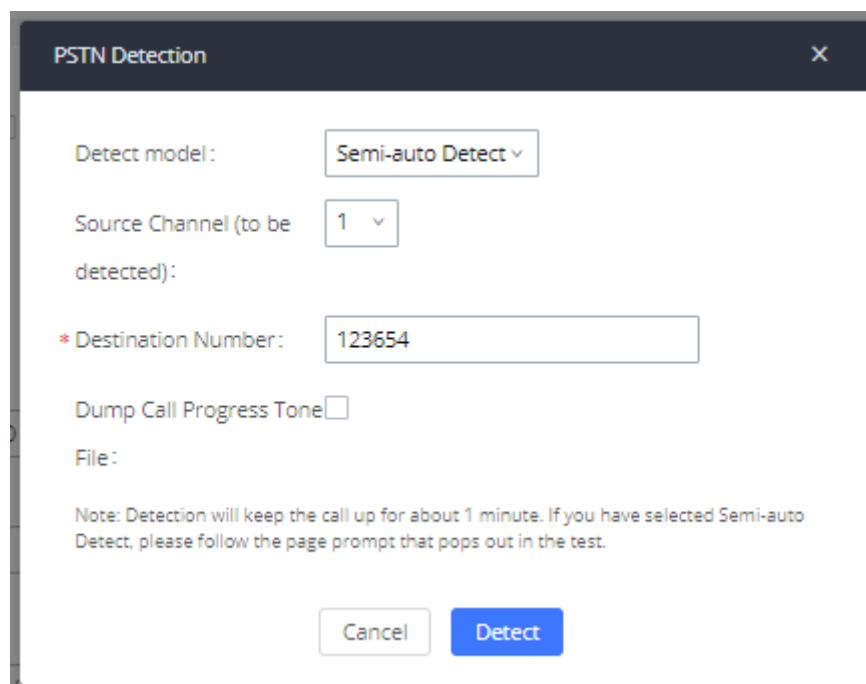
Dump Call Progress Tone

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.

UCM630xA PSTN Detection: Auto Detect

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



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.

Detect Model	<p>Select "Auto Detect" or "Semi-auto Detect" for PSTN detection.</p> <ul style="list-style-type: none"> ○ Auto Detect <p>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.</p> <ul style="list-style-type: none"> ○ Semi-auto Detect <p>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.</p> <p>The default setting is "Auto Detect".</p>
Source Channel	Select the channel to be detected.
Destination Channel	Select the channel to help detect when "Auto Detect" is used.
Destination Number	Configure the number to be called to help the detection.
Dump Call Progress Tone File	Choose whether to save the calling tone file, it is not checked by default.

PSTN Detection for Analog Trunk







- 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 being 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.

<https://documentation.grandstream.com/knowledge-base/sip-trunks-guide/>

The VoIP trunk options are listed in the table below.

Type	Select the VoIP trunk type. <ul style="list-style-type: none"> ● Peer SIP Trunk ● Register SIP Trunk
Provider Name	Configure a unique label (up to 64 characters) to identify this trunk when listed in outbound rules, inbound rules, etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Transport	Configure the SIP Transport method. Using TCP requires local TCP support. Using TLS Requires local TLS support. <ul style="list-style-type: none"> ● UDP ● TCP ● TLS
Auto Record	If enabled, calls handled with this extension/trunk will automatically be recorded.
Keep Original CID	Keep the CID from the inbound call when dialing out. This setting will override the "Keep Trunk CID" option. Please make sure that the peer PBX at the other side supports to match user entry using the "username" field from the authentication line.

Keep Trunk CID	If enabled, the trunk CID will not be overridden by the extension's CID when the extension has CID configured. The default setting is "No".
NAT	Turn on this setting when the PBX is using public IP and communicating with devices behind NAT. If there is a one-way audio issue, usually it is related to NAT configuration or SIP/RTP port support on the firewall.
Disable This Trunk	If checked, the trunk will be disabled. Note: If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
TEL URI	If the trunk has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled.
Caller ID Number	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. Important Note: 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 the user (Register Trunk Only) → CID from inbound call (Keep Original CID Enabled) → Trunk Username/CallerID (Keep Trunk CID Enabled) → DOD → Extension CallerID Number → Trunk Username/CallerID (Keep Trunk CID Disabled) → Global Outbound CID.
CallerID Name	Configure the new name of the caller when the extension has no CallerID Name configured.
Auto Record	If enabled, calls handled with this extension/trunk will automatically be recorded.
Auth ID	Enter the Authentication ID for the "Register SIP Trunk" type.
Direct Callback	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.
RemoteConnect Mode	If enabled, the RemoteConnect related parameters will be set synchronously. Please make sure the trunk host is allocated by GDMS or supports TLS.
Limit Concurrent Calls	If enabled and when the number of concurrent calls exceeds any trunk's configured concurrent call thresholds, an alarm notification will be generated. Note: Please make sure the system alert event "Trunk Concurrent Calls" is enabled.
Concurrent Call Threshold	Threshold of all incoming and outgoing concurrent calls through this trunk.
Outgoing Concurrent Calls Threshold	Threshold of all outgoing concurrent calls passing through this trunk.
Incoming Concurrent Calls Threshold	Threshold of all incoming concurrent calls passing through this trunk.
Total Time Limit For Outbound Calls	
Enable Total Time Limit For Outgoing Calls	When this setting is activated, the user can set a time limit before calls cannot be initiated through this trunk
Period	This setting defines how long until the time allowed for outgoing calls is reset.

	<ul style="list-style-type: none"> • Monthly: The time allowed will reset every month. • Quarterly: The time allowed will reset every 3 months. <p>Example: If the time limit has been set to 4320 minutes, the allowed time will always revert back to 4320 after a month or 3 month based on the period configured.</p>
Total Time	Total time allowed in minutes

Create New SIP Trunk

[SIP Peer Trunk](#)

[SIP Register Trunk](#)

Basic Settings	
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules, etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Auto Record	Enable automatic recording for the calls using this trunk (for SIP trunk only). The default setting is disabled. The recording files can be accessed under Web GUI → CDR → Recording Files .
Keep Original CID	Keep the CID from the inbound call when dialing out, this setting will override the "Keep Trunk CID" option. Please make sure that the peer PBX at the other side supports to match user entry using the "username" field from the authentication line.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by the extension's CID when the extension has CID configured. The default setting is "No".
NAT	Turn on this option when the PBX is using public IP and communicating with devices behind NAT. If there is a one-way audio issue, usually it is related to NAT configuration or SIP/RTP port configuration on the firewall.
Disable This Trunk	If selected, the trunk will be disabled. Note: If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
TEL URI	If the trunk has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled.
Caller ID Number	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. Important Note: When making outgoing calls, the following priority order rule will be used to determine which CallerID will be set before sending out the call: <ul style="list-style-type: none"> • CID from inbound call (Keep Original CID Enabled) → Trunk Username/CallerID (Keep Trunk CID Enabled) → DOD → Extension CallerID Number → Trunk Username/CallerID (Keep Trunk CID Disabled) → Global Outbound CID.
CallerID Name	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
Transport	Configure the SIP transport protocol to be used in this trunk. The default setting is "UDP". <ul style="list-style-type: none"> • UDP

	<ul style="list-style-type: none"> • TCP • TLS
RemoteConnect Mode	If enabled, the RemoteConnect related parameters will be set synchronously. Please make the trunk host is allocated by GDMS or it supports TLS.
Direct Callback	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	
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p and VP8.
Packet Loss Retransmission	Configure to enable Packet Loss Retransmission.
Audio FEC	Configure to enable Forward Error Correction (FEC) for audio.
Video FEC	Configure to enable Forward Error Correction (FEC) for video.
ICE Support	Toggles ICE support. For peer trunks, ICE support will need to be enabled on the other end.
TURN Relay	TURN servers are used for media NAT traversal and will be prioritized if ICE is also enabled.
FECC	Configure to enable Far-end Camera Control
Silence Suppression	If enabled, the UCM will send CN packets for silence suppression after a successful CN negotiation in the SIP SDP. If the client endpoint's OPUS codec supports the reception of DTX packets, the UCM will send DTX packets instead.
SRTP	Enable SRTP for the VoIP trunk. The default setting is "No".
SRTP Crypto Suite	SRTP encryption suite used by UCM for outbound calls. Priority is based on order of configuration.
IPVT Mode	Similar to Enable Direct Media. The PBX will attempt to redirect the RTP media streams to bypass the PBX and to go directly between caller and callee. Primarily for use with trunks to IPVT.
Enable T.38 UDPTL	Enable or disable T.38 UDPTL support.
Special attributes	Carry the ssrc/msid/mid/ct/as/tias/record properties of the SDP. These attributes may cause incompatibility when interconnecting with other devices.
Send PPI Header	If checked, the invite message sent to trunks will contain PPI (P-Preferred-Identity) Header.
Send PAI Header	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.
DOD as From Name	If enabled and "From User" is configured, the INVITE's From header will contain the DOD number.
Passthrough PAI	If enabled and "Send PAI Header" is disabled, PAI headers will be preserved as calls pass

Header	through the UCM.
Send PANI Header	If checked, the INVITE and REGISTER sent to the trunk will contain P-Access-Network-Info header.
Send Anonymous	If checked, the "From" header in outgoing INVITE message will be set to anonymous.
Outbound Proxy Support	Enable to send outbound signal to the proxy instead of the devices directly.
DID Mode	Configure to obtain the destination ID of an incoming SIP call from SIP Request-line or To header.
GIN Registration	If enabled, the UCM will send a GIN REGISTER (generate implicit numbers).
DTMF Mode	<p>Configure the mode for sending the DTMF.</p> <ul style="list-style-type: none"> • RFC4833 (default): DTMF is transmitted as audio in the RTP stream but is encoded separately from the audio stream. Backwards-compatible with RFC2833. • DTMF is transmitted as audio and is included in the audio stream. Requires alaw/ulaw codecs • Info: DTMF is transmitted separately from the media streams. • RFC4733_info: DTMF is transmitted through both RFC4733 and SIP INFO • Auto: DTMF mode will be negotiated with the remote peer, only supports RFC4733 and inband. RFC4733 will be used by default unless the remote peer does not indicate support.
Enable Heartbeat detection	If enabled, the PBX will regularly send SIP OPTIONS to check if the device is online.
The Maximum Number of Call Lines	The number of current outgoing calls over the trunk at the same time. The default value 0 means no limit.
Sync LDAP Enable	<p>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.</p> <ul style="list-style-type: none"> • Sync LDAP Password: Password used for LDAP phonebook encryption and decryption. The password must be the same for both peers to ensure successful syncing. • LDAP Outbound Rule: Specify an outbound rule. The PBX system will automatically modify the remote contacts by adding prefix parsed from this rule. • LDAP Dialed Prefix: System will automatically modify the remote contacts by adding this prefix • LDAP Sync Method: Specifies the sync method of the UCM. When an LDAP sync request is received, the UCM will send the phonebook data via the specified method. • LDAP Last Sync Date: The last successful sync date.
STIR/SHAKEN	<p>Block Spam Calls.</p> <ul style="list-style-type: none"> • Disabled: Disable STIR/SHAKEN. • Outgoing Attest: Enable STIR/SHAKEN attestation for outgoing calls. • Incoming Verify: Enable STIR/SHAKEN verification for incoming calls. • Both: Enable STIR/SHAKEN for both outgoing and incoming calls
Enable CC	Check this box to allow the system to automatically alert this extension when a called party is available, given that a previous call to that party failed for some reason. If the Call Waiting is disabled, the CC service will take effect only for unanswered and timeout calls.

Type	Select the VoIP trunk type. <ul style="list-style-type: none"> • Peer IAX Trunk • Register IAX Trunk
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules, etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by the extension's CID when the extension has CID configured. The default setting is "No".
Username	Enter the username to register to the trunk from the provider when the "Register IAX Trunk" type is selected.
Password	Enter the password to register to the trunk from the provider when the "Register IAX Trunk" type is selected.
Disable This Trunk	If selected, the trunk will be disabled.
Caller ID Number	<p>The number that the trunk will try to use when making outbound calls. CID priority from highest to lowest is as follows:</p> <p>From User (register trunk only) >>> Inbound Call CID (if Keep Original CID is enabled and the call is originally from another trunk) >>> Trunk CID (Keep Trunk CID enabled) >>> DOD CID >>> Extension CID >>> Register Trunk Username (Keep Trunk CID disabled) >>> Global Outbound CID.</p> <p>Note 1: Certain providers may ignore this CID. Note 2: If this CID contains an asterisk (*), call recordings from this trunk might be lost when saving them to NAS storage.</p>
CallerID Name	Configure the new name of the caller when the extension has no CallerID Name configured.

IAX Peer Trunk

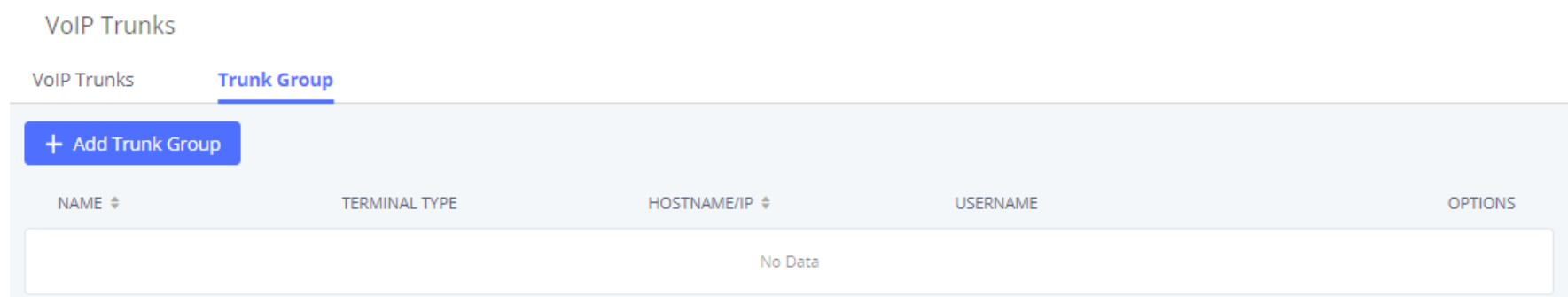
IAX Register Trunk

Basic Settings	
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules, etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by the extension's CID when the extension has CID configured. The default setting is "No".
Disable This Trunk	If selected, the trunk will be disabled.
Caller ID	<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>Important Note: 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 the user (Register Trunk Only) → CID from inbound call (Keep Original CID Enabled) → Trunk Username/CallerID (Keep Trunk CID Enabled) → DOD → Extension CallerID Number → Trunk Username/CallerID (Keep Trunk CID Disabled) → Global Outbound CID.</p>

CallerID Name	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
Username	Enter the username to register to the trunk from the provider.
Password	Enter the password to register to the trunk from the provider.
Advanced Settings	
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p and VP8.
Enable Heartbeat Detection	If enabled, the UCM630X will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
Heartbeat Frequency	When the "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.
Maximum Number of Call Lines	The maximum number of concurrent calls using the trunk. The default setting is 0, which means no limit.

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.



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
Cancel Save

If the host is not a numeric IP address, but the port number is present in the URI, the UCM performs an A or AAAA record lookup of the domain name. If a domain is configured without a port number, the UCM will do an SRV record lookup.

Type: Register SIP Trunk

* Provider Name: Please select a provider

* Host Name:

Transport: UDP

Keep Original CID:

Keep Trunk CID:

NAT:

Disable This Trunk:

TEL URI: Disabled

Need Registration:

Allow outgoing calls if registration fails:

CallerID Name:

* Trunk Registration Number: Trunk Registration Ni / Password / AuthID -

[Add Username](#) +

Line Selection Strategy: Linear

AuthTrunk:

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Trunk Group Configuration

Type	Register Trunk
Provider Name	Configure a unique label to identify the trunk when listed in outbound rules and incoming rules.
Host Name	Enter the IP address or hostname of the VoIP provider's server.
Transport	Configure the SIP Transport method. Using TCP requires local TCP support; using TLS requires local TLS support.
Keep Original CID	Keep CID from the inbound call when dialing out even if option "Keep Trunk CID" is enabled. Please make sure the peer PBX at the other end supports matching user entry using the "username" field from the authentication line.
Keep Trunk CID	Always use trunk CID if specified even if extension has DOD number or CID configured.
NAT	Enable this setting if the IPPBX is using public IP and communicating with devices behind NAT. Note 1: This setting will overwrite the Contact header of received messages, which may affect the ability to establish calls when behind NAT. Consider changing settings in PBX Settings → SIP Settings → NAT instead.
Disable This Trunk	Check this box to disable this trunk
TEL URI	if "Enabled" option is selected, TEL URI and remove OBP from Route cannot be enabled at the same time. If the phone has an assigned PSTN telephone number, this field should be set to "User=Phone". 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.
Need Registration	Whether to register on the external server.

Allow outgoing calls if registration fails	Uncheck to block outgoing calls if registration fails. If "Need Registration" option is unchecked, this setting will be ignored.
CallerID Name	To display the caller ID name of the trunk, you must configure the caller ID number of the trunk.
Trunk Registration Number	The number used to register with the provider server, and the VoIP provider will authenticate the number based on the trunk registration number.
Line Selection Strategy	Linear: Select lines in list order and make Outbound calls. Round Robin: Rotary line selection with memory and making Outbound calls.
AuthTrunk	If enabled, the IPPBX will send a 401 response to the incoming call to authenticate the trunk.
Auto Record	If enabled, calls handled with this extension/trunk will automatically be recorded.
Direct Callback	Allows external numbers the option to get directed to the extension that last called them.
RemoteConnect Mode	If enabled, RemoteConnect-related options will be automatically configured. Please confirm the trunk has a GDMS-assigned address or supports TLS.
Monitor Concurrent Calls	If enabled, the number of concurrent calls on this trunk will be monitored. If the "Trunk Concurrent Calls" system alert is enabled, alert notifications will be generated if the number of concurrent calls exceeds this trunk's configured concurrent call thresholds.
Concurrent Call Threshold	Threshold of all incoming and outgoing concurrent calls in this trunk.
Outgoing Concurrent Call Threshold	Threshold of all outgoing concurrent calls passing through this trunk.
Incoming Concurrent Call Threshold	Threshold of all incoming concurrent calls passing through this trunk.
Enable Total Time Limit For Outbound Calls	If enabled, a limit will be placed on the cumulative duration of outbound calls within a specific period. Once this limit has been reached, further outbound calls from this trunk will not be allowed.

WebRTC Trunks

WebRTC, Web Real-Time Communication, is a real time audio/video chatting framework that allows real-time audio/video chatting through the web browser. WebRTC usually does not refer to the web application itself but to the set of protocols and practices bundled with a graphical interface. Our UCM63xx supports creating WebRTC trunks to use exclusively with web application, this allows the users to join calls and meetings just by clicking a link to a web page.

Below is a figure that shows the options to configure when setting up this feature:

* Trunk Name:	<input type="text" value="GS_WebRTC_Trunk"/>
Disable This Trunk:	<input type="checkbox"/>
Auto Record:	<input checked="" type="checkbox"/>
Enable Concurrent Call Threshold:	<input checked="" type="checkbox"/>
* Incoming Concurrent Call Threshold:	<input type="text" value="150"/>
WebRTC Inbound Link Address:	Automatically generated after saving

WebRTC Trunk Configuration

Trunk Name	Create a unique label to easily identify the trunk for inbound route configuration.
Disable This Trunk	Check this box to disable this trunk.
Auto Record	If enabled, calls handled with this extension/trunk will automatically be recorded.
Jitter Buffer	Select jitter buffer method for temporary accounts such as meeting participants who joined via link. Disable: Jitter buffer will not be used. Fixed: Jitter buffer with a fixed size (equal to the value of "Jitter Buffer Size") Adaptive: Jitter buffer with a adaptive size that will not exceed the value of "Max Jitter Buffer"). NetEQ: Dynamic jitter buffer via NetEQ.
Monitor Concurrent Calls	If enabled, the number of concurrent calls on this trunk will be monitored. If the "Trunk Concurrent Calls" system alert is enabled, alert notifications will be generated if the number of concurrent calls exceeds this trunk's configured concurrent call thresholds.
Incoming Concurrent Call Threshold	Threshold of all incoming concurrent calls passing through this trunk.
WebRTC Inbound Link Address	This link can be embedded onto a web page. Clicking the link will connect to a pre-configured WebRTC trunk destination. You can also enter this link in the browser address bar to directly access and test WebRTC calls.

Important Note

Please note that in order to use WebRTC Trunk feature, you need to have a paid RemoteConnect plan enabled.

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 DID's 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.

Create DOD

*DOD Number: 001122334455

DOD Name: Grandstream_Supp

Add Extension:

Available (15 items):

- 1000
- 1001 "John Snow"
- 1002 "Jack Doe"
- 1003 "Marc Hugzman"
- 1004 "Pieter Rozfield"

Selected (0 item): None

Cancel Save

DOD extension selection

1. 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

Direct Outward Dialing (DOD) 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.

+ Add DOD Import Export

DOD	DOD NAME	EXTENSIONS	OPTIONS
918273645	Test	1000 1001 1002	

Total: 1 10 / page Goto 1

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**.

STATION NAME	STATION	ASSOCIATED SLA TRUNKS	OPTIONS
FXO1	1000	test	

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.

SLA Station Configuration Parameters

Station Name	Configure a name to identify the SLA Station.
Station	Specify a SIP extension as a station that will be using SLA.
Available SLA Trunks	Existing Analog Trunks with SLA Mode enabled will be listed here.
Associated SLA Trunks	Select a trunk for this SLA from the Available SLA Trunks list. Click on to arrange the order. If there are multiple trunks selected, when there are calls on those trunks at the same time, pressing the LINE key on the phone will pick up the call on the first trunk here.
SLA Station Options	
Ring Timeout	Configure the time (in seconds) to ring the station before the call is considered unanswered. No timeout is set by default. If set to 0, there will be no timeout.
Ring Delay	Configure the time (in seconds) for delay before ringing the station when a call first coming in on the shared line. No delay is set by default. If set to 0, there will be no delay.
Hold Access	This option defines the competence of the hold action for one particular trunk. If set to "open", any station could hold a call on that trunk or resume one held session; if set to "private", only the station that places the trunk call on hold could resume the session. The default setting is "open".

Sample Configuration

1. On the 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.

The screenshot shows the configuration page for an analog trunk. At the top, there are two fields: “FXO Port:” with radio buttons for 1 (selected) and 2, and “Trunk Name:” with a text input field containing “test”. Below these is a section titled “Advanced Options” with a sub-section “SLA Mode:” where a checkbox is checked.

Enable SLA Mode for Analog Trunk

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

The screenshot shows a table titled “Analog Trunks”. The table has columns for TRUNKS, DISABLE, TRUNK MODE, ANALOG PORTS, and OPTIONS. A single row is visible with the value “test” in the TRUNKS column, “no” in the DISABLE column, “sla” in the TRUNK MODE column, and “1” in the ANALOG PORTS column. The table also includes a “+ Add” button and pagination controls at the bottom.

Analog Trunk with SLA Mode Enabled

1. 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”.

The screenshot shows a table titled “SLA Station”. The table has columns for STATION NAME, STATION, ASSOCIATED SLA TRUNKS, and OPTIONS. A single row is visible with the value “FXO1” in the STATION NAME column, “1000” in the STATION column, and “test” in the ASSOCIATED SLA TRUNKS column. The table also includes a “+ Add” button and pagination controls at the bottom.

SLA Example – SLA Station

1. 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.
2. 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.

The screenshot shows a table with columns for Mode, Account, Description, and Value. A single row is visible with the value “MPK 1” in the Mode column, “Busy Lamp Field (BLF)” in the Account column, “Account 2” in the Description column, and “1005_fxo1” in the Value column.

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


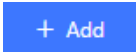

Outbound Routes

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.

Outbound Route Configuration Parameters

Outbound Rule Name	Configure the name of the calling rule (e.g., local, long_distance, and etc.). Letters, digits, _ and – are allowed.
---------------------------	--

Pattern	<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"> ○ [12345-9] ... Any digit in the brackets. In this example, 1,2,3,4,5,6,7,8,9 is allowed. ○ N ... Any digit from 2-9. ○ Wildcard, matching one or more characters. ○ ! ... Wildcard, matching zero or more characters immediately. ○ X ... Any digit from 0-9. ○ Z ... Any digit from 1-9. ○ – ... Hyphen is to connect characters and it will be ignored. ○ [] Contain special characters ([x], [n], [z]) represent letters x, n, z.
Disable This Route	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.
Password	Configure the password for users to use this rule when making outbound calls.
Local Country Code	If your local country code is affected by the outbound blacklist, please enter it here to bypass the blacklist.
Call Duration Limit	Enable to configure the maximum duration for the call using this outbound route.
Maximum Call Duration	Configure the maximum duration of the call (in seconds). The default setting is 0, which means no limit.
Warning Time	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.
Auto Record	If enabled, calls using this route will automatically be recorded.
Warning Repeat Interval	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.
PIN Groups	Select a PIN Group
PIN Groups with Privilege Level	If enabled and PIN Groups are used, Privilege Levels and Filter on Source Caller ID will also be applied.
Privilege Level	<p>Select privilege level for the outbound rule.</p> <ul style="list-style-type: none"> ○ Internal: The lowest level required. All users can use this rule. ○ Local: Users with Local, National, or International level can use this rule. ○ National: Users with National or International level can use this rule. ○ International: The highest level required. Only users with international level can use this rule. ○ Disable: The default setting is “Disable”. If selected, only the matched source caller ID will be allowed to use this outbound route. <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>

<p>Enable Filter on Source Caller ID</p>	<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"> 1. Select available extensions/extension groups from the list. This allows users to specify arbitrary single extensions available in the PBX. 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"> ○ All patterns are prefixed with the "_". ○ Special characters: <p>X: Any Digit from 0-9.</p> <p>Z: Any Digit from 1-9.</p> <p>N: Any Digit from 2-9.</p> <p>.": Wildcard. Match one or more characters.</p> <p>!": Wildcard. Match zero or more characters immediately.</p> <p>Example: [12345-9] – Any digit from 1 to 9.</p> <p>Note: Multiple patterns can be used. Patterns should be separated by comma ",". Example: _X. , _NNXXNXXXXX , _818X.</p>
<p>Outbound Route CID</p>	<p>Attempt to use the configured outbound route CID. This CID will not be used if DOD is configured.</p>
<p>Send This Call Through Trunk</p>	
<p>Trunk</p>	<p>Select the trunk for this outbound rule.</p>
<p>Strip</p>	<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>Example:</p> <p>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>
<p>Prepend</p>	<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>
<p>Use Failover Trunk</p>	
<p>Failover Trunk</p>	<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>Example:</p> <p>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>

Strip	<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>Example:</p> <p>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>
Prepend	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.
Time Condition	
Time Condition Mode	<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>

Failover Trunk Toggles

The screenshot shows the 'Outbound Routes' configuration page. At the top, there is a green informational box. Below it, a row of buttons includes '+ Add', 'Scheduled Sync', 'Outbound Blacklist', 'PIN Groups', 'Failover Trunk Toggles' (highlighted with a red box), 'Import', and 'Export'. Below the buttons is a table with columns: SEQUENCE, NAME, PATTERN, PRIVILEGE LEVEL, and OPTIONS. The table contains two rows of data.

SEQUENCE	NAME	PATTERN	PRIVILEGE LEVEL	OPTIONS
1	out	_2X. _9X.	Internal	[Edit] [Phone] [Trash] [Refresh] [Reset] [Save]
2	Outbound	_X.	Local	[Edit] [Phone] [Trash] [Refresh] [Reset] [Save]

Failover Trunk Toggles

This option controls whether failover trunks will be used if receiving specific responses to outgoing calls.

The screenshot shows the 'Failover Trunk Toggles' configuration dialog. It has a title bar with '< Failover Trunk Toggles', 'Cancel', and 'Save' buttons. The main content area is titled 'No-Failover Response Codes'. It features two list boxes: 'Available' (6 items) and 'Selected' (1 item). The 'Available' list contains codes 403, 404, 408, 480, 503, and 603. The 'Selected' list contains code 486. Navigation arrows are between the lists. A copyright notice is at the bottom.

No-Failover Response Codes

If a call receives the selected response codes, the UCM will not be redirect it to the call route's failover trunk.

Note

Due to the addition of this option, the **Enable 486 to Failover Trunks** option under *PBX Settings* → *General Settings* page has been removed.

Outbound Routes DOD

It is possible to specify DOD number based on Outbound Route, as displayed on the screenshot below. For each outbound route.

SEQUENCE	NAME	PATTERN	PRIVILEGE LEVEL	OPTIONS
1	go_out	_9x	Internal	[Icons]
2	Again_Out	_x	Internal	[Icons]

Outbound Routes Page

DOD	DOD NAME	EXTENSIONS
369369	first dod	1000 1001 1002

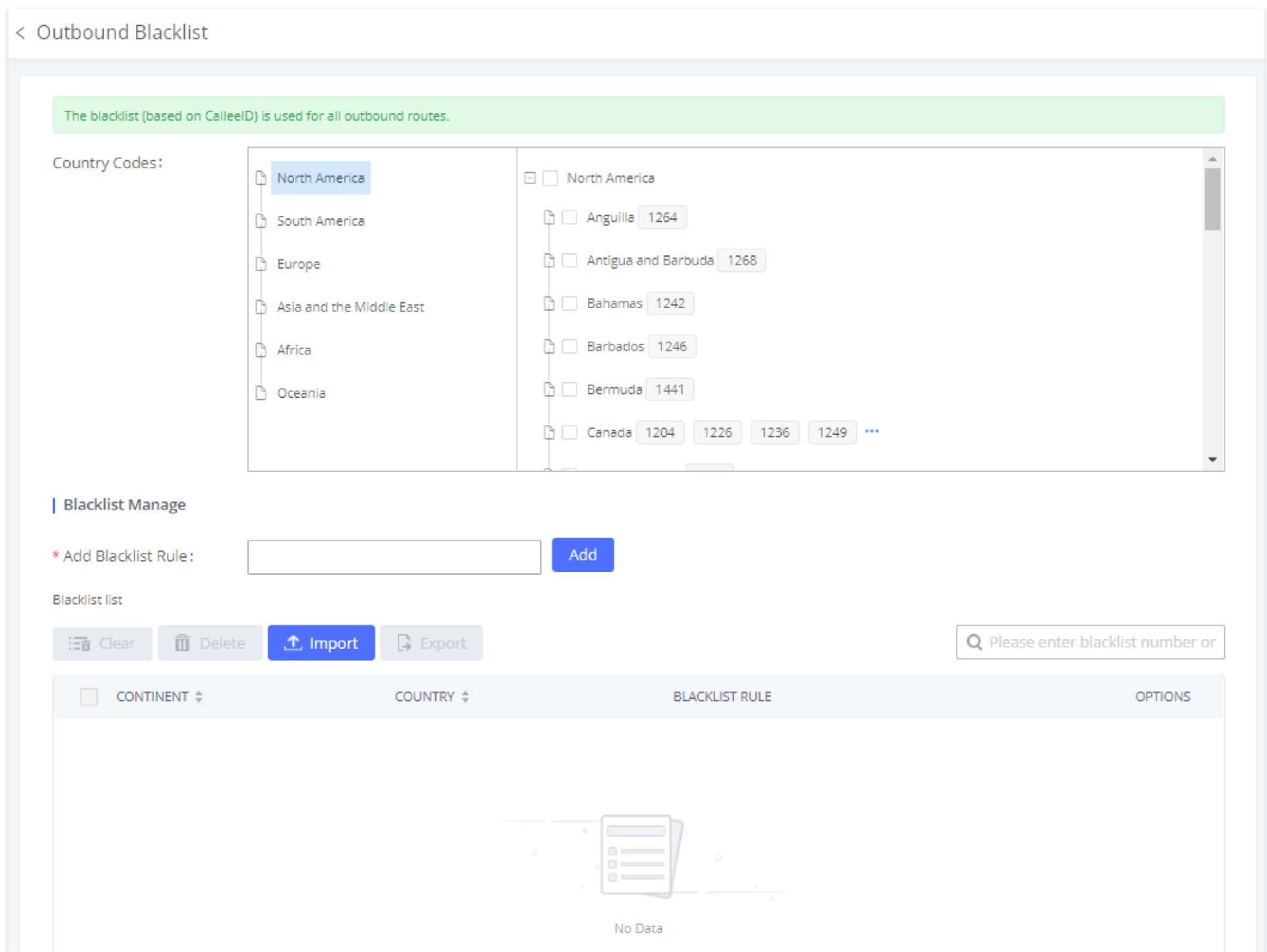
DOD	DOD NAME	EXTENSIONS
852582	second DOD	1000 1001 1002

DOD Configuration by Outbound Route

Outbound Blacklist

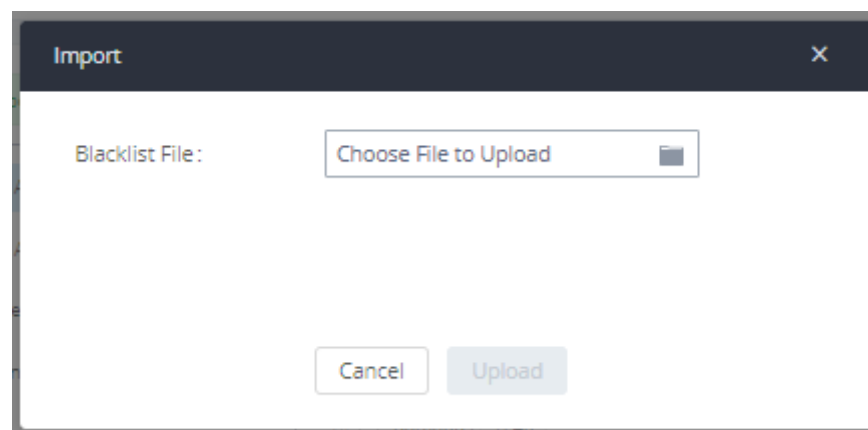
The UCM630xA allows users to configure blacklist for outbound routes. If the dialing number matches the blacklist numbers or patterns, the outbound call will not be allowed. The outbound blacklist can be configured under UCM Web GUI → **Extension/Trunk** → **Outbound Routes**: Outbound Blacklist.

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.



Country Codes

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



Blacklist Import/Export

Scheduled Sync

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

Outbound Routes/Scheduled Sync

Scheduled Sync	Enable the Scheduled Sync feature
Server Address	Enter the TFTP server address. For example, "192.168.1.2:69".
File Name	Specify the file name
Sync Time	Enter the sync time (24hr format). Valid range is 0-23.
Sync Frequency	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**.

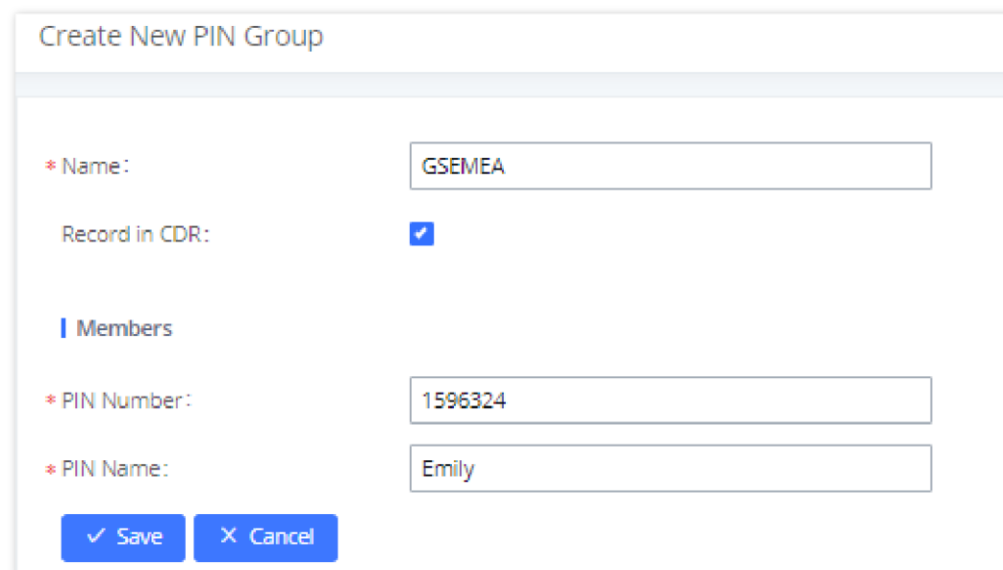
Outbound Routes/PIN Group

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

Once user click on

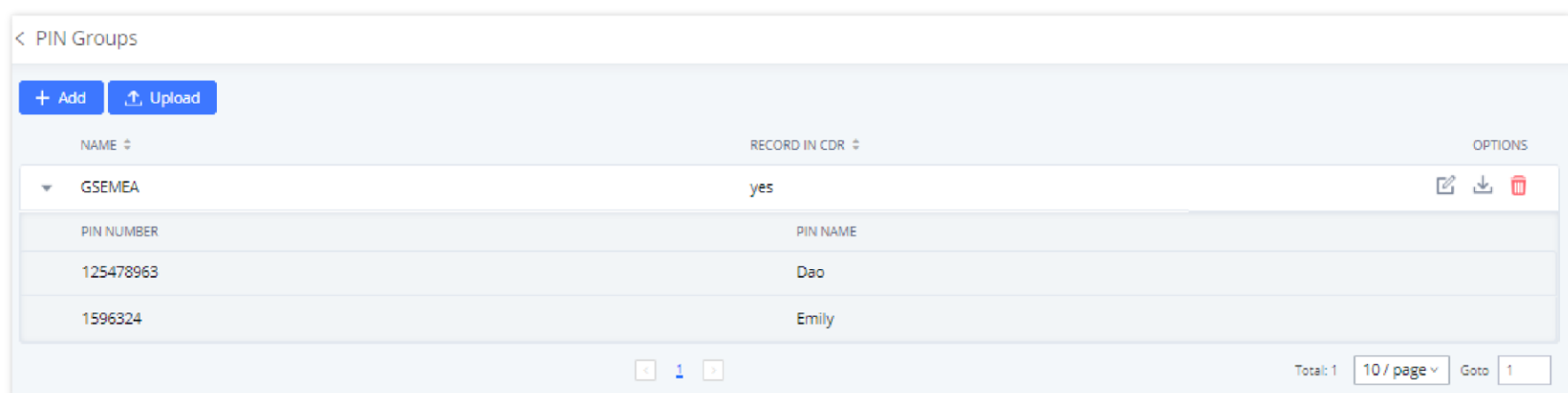
[PIN Groups](#)

the following figure shows to configure the new PIN.



Create New PIN Group

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



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: Privilege Level:

PIN Groups: PIN Groups with Privilege Level:

Password:

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.

CDR Display Filter

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

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/GSEMEA	-
	1000	*36	DIAL	2019-12-03 10:12:37	0:00:19	0:00:19		-

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.

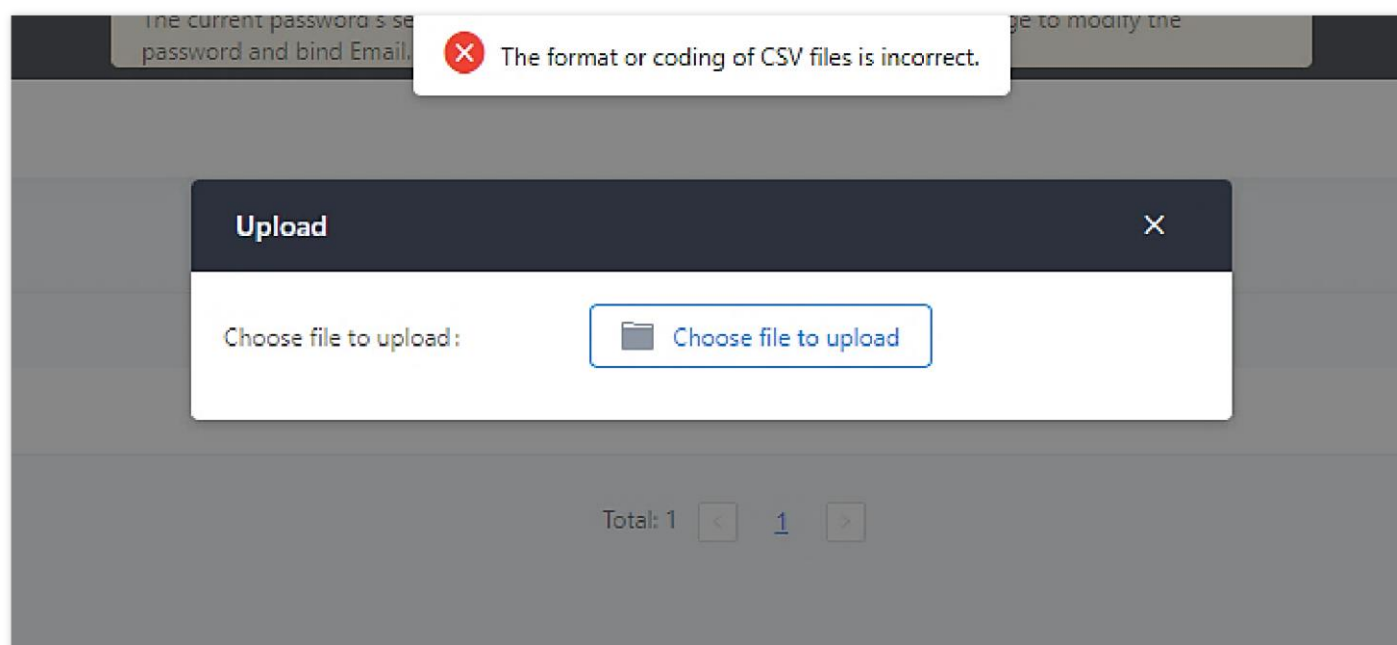
< PIN Groups

NAME	RECORD IN CDR	OPTIONS
GSEMEA	yes	

Total: 1

Importing PIN Groups from CSV files

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



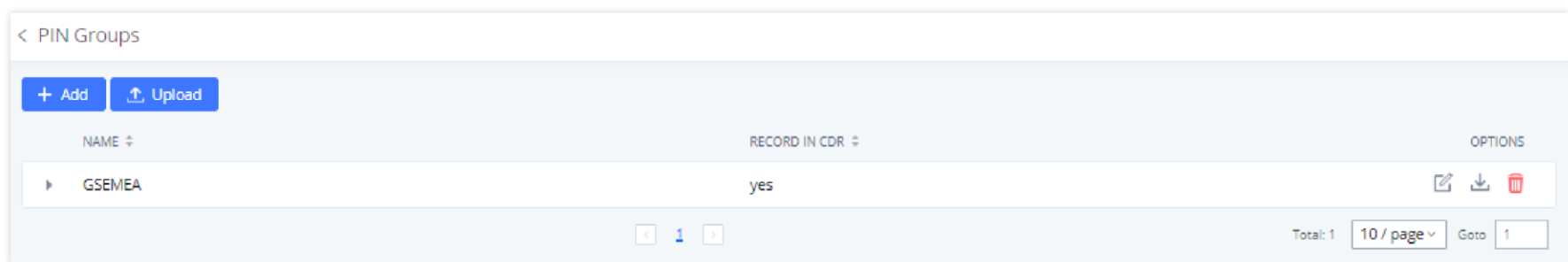
Incorrect CSV File

1. 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				

CSV File Format

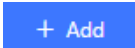


- The top-left value (A1) is the PIN Group name. In this case, it is "ALPHA".
- Row 2 contains the labels for the modifiable fields: pin and pin_name. These values should not be changed and will cause an upload error otherwise.
- Rows 3+ contain the user-defined values with Column A holding the PINs 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.



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

Inbound Rule Configuration Parameters

Trunks	Select the trunk to configure the inbound rule.
Inbound Route Name	Configure the name of the Inbound Route. For example, "Local", "LongDistance" etc.
Pattern	<p>All patterns are prefixed with the "_".</p> <p>Special characters:</p> <p>X: Any Digit from 0-9. Z: Any Digit from 1-9. N: Any Digit from 2-9. “.”: Wildcard. Match one or more characters. “!”: Wildcard. Match zero or more characters immediately. Example: [12345-9] – Any digit from 1 to 9.</p> <p>Notes:</p> <p>Multiple patterns can be used. Each pattern should be entered in a new line.</p> <p>Example:</p> <p>_X. _ NNXXNXXXXX /* 10-digit long distance */</p>
Disable This Route	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.
CID Source	<p>Configures the source of the CID to match with the configured CallerID Pattern.</p> <p>None: CID is not obtained from any source. Only applicable if no CallerID Pattern is configured.</p> <p>DiversionID: CID is obtained from the Diversion header. Only applicable to SIP trunks.</p> <p>CallerID: If the call is from a SIP trunk, the CID is obtained from the From header. Otherwise, the CID will be obtained from other related signaling.</p>
Seamless Transfer Whitelist	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.
Ringback tone	Choose the custom ringback tone to play when the caller reaches the route.
Auto Record	If enabled, calls using this route will automatically be recorded.
Block Collect Call	<p>If enabled, collect calls will be blocked.</p> <p>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".</p>
Alert-Info	Configure the Alert-Info, when UCM receives an INVITE request, the Alert-Info header field specifies an alternative ring tone to the UAS.
Fax Detection	If enabled, fax signals from the trunk during a call will be detected.
Fax Destination	<p>Configures the destination of faxes.</p> <ul style="list-style-type: none"> • Extension: send the fax to the designated FXS/SIP extension (fax machine) or a FAX extension. • Fax to Email: 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 Call Features->Fax/T.38->Fax Settings page. <p>Note: please make sure the sending email address is correctly configured in System Settings->Email Settings.</p>
Auto Answer	If enabled, the UCM will automatically answer calls and receive faxes through the inbound route. If disabled, the UCM will not receive a fax until after the call has been answered. Enabling this option

	will slow down the answering of non-fax calls on the inbound route. The alert tone heard during the detection period can be customized.
Block Collect Calls	<p>If enabled, collect calls will be blocked.</p> <p>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".</p> <p>Note: This is affected by Block Set Calls on the SIP Settings -> General Settings page.</p>
Prepend Trunk Name	If enabled, the trunk name will be added to the caller id name as the displayed caller id name.
Set Caller ID Info	Manipulates Caller ID (CID) name and/or number within the call flow to help identify who is calling. When enabled two fields will show allowing to manipulate the CallerID Number and the Caller ID Name.
CallerID Number	<p>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.</p> <ul style="list-style-type: none"> • `\${CALLERID(num)}`: Default value which indicates the number of an incoming caller (CID). The CID will not be modified. • `\${CALLERID(num):n}`: Skips the first n characters of a CID number, where n is a number. • `\${CALLERID(num):-n}`: Takes the last n characters of a CID number, where n is a number. • `\${CALLERID(num):s:n}`: 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). • n`\${CALLERID(num)}`: Prepends n to a CID number, where n is a number.
CallerID Name	<p>The default string is `\${CALLERID(name)}`, which means the name of an incoming caller, it is a pattern-matching syntax format.</p> <p>A`\${CALLERID(name)}`B means Prepend a character 'A' and suffix a character 'B' to `\${CALLERID(name)}`.</p> <p>Not using pattern-matching syntax means setting a fixed name to the incoming caller.</p>
Enable Route-Level Inbound Mode	<p>Gives uses the ability to configure inbound mode per individual route. When enabled two fields will show allowing to set the Inbound mode and the Inbound mode Suffix.</p> <p>Note: Global inbound mode must be enabled before users can configure route-level inbound mode.</p>
Inbound Mode	<p>Choose the inbound mode for this route.</p> <p>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.</p>
Inbound Mode Suffix	<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>
Inbound Multiple Mode	Multiple mode allows users to switch between destinations of the inbound rule by feature codes. Configure related feature codes as described in [Inbound Route: Multiple Mode]. If this option is enabled, the user can use feature code to switch between different modes/destinations.
CallerID Name Lookup	<p>If enabled, the callerID will be resolved to a name through local LDAP. Note, if a matched name is found, the original callerID name will be replaced. The name lookup is performed before other callerID or callerID name modifiers (e.g., Inbound Route's Set CallerID Info or Prepend Trunk Name).</p> <p>Note: Name lookup may impact system performance.</p>
Dial Trunk	This option shows up only when "By DID" is selected. If enabled, the external users dialing into the trunk via this inbound route can dial outbound calls using the UCM's trunk.
Privilege Level	<p>This option shows up only when "By DID" is selected.</p> <ul style="list-style-type: none"> • Disable: Only the selected Extensions or Extension Groups are allowed to use this rule when enabled Filter on Source Caller ID.

	<ul style="list-style-type: none"> ● Internal: The lowest level required. All users are allowed to use this rule, checking this level might be risky for security purposes. ● Local: Users with Local level, National or International level are allowed to use this rule. ● National: Users with National or International Level are allowed to use this rule. ● International: The highest level required. Only users with an international level are allowed to use this rule.
Allowed DID Destination	<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 is:</p> <ul style="list-style-type: none"> ● Extension ● Conference ● Call Queue ● Ring Group ● Paging/Intercom Groups ● IVR ● Voicemail Groups ● Dial By Name ● All
Default Destination	<p>Select the default destination for the inbound call.</p> <ul style="list-style-type: none"> ● Extension ● Voicemail ● Conference Room ● Call Queue ● Ring Group ● Paging/Intercom ● Voicemail Group ● DISA ● IVR ● External Number ● By DID <p>When “By DID” is used, the UCM will look for the destination based on the number dialed, which could be local extensions, conference, 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"> ● Dial By Name ● Callback
Strip	<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>
Prepend	<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>
Time Condition	
Start Time	<p>Select the start time “hour:minute” for the trunk to use the inbound rule.</p>
End Time	<p>Select the end time “hour:minute” for the trunk to use the inbound rule.</p>
Date	<p>Select “By Week” or “By Day” and specify the date for the trunk to use the inbound rule.</p>
Week	<p>Select the day in the week to use the inbound rule.</p>
Destination	<p>Select the destination for the inbound call under the defined time condition.</p> <ul style="list-style-type: none"> ● Extension ● Voicemail ● Conference Room

- Call Queue
- Ring Group
- Paging/Intercom
- Voicemail Group
- DISA
- IVR
- By DID

When "By DID" is used, the UCM will look for the destination based on the number dialed, which could be local extensions, conference, 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.

Configure the number of digits to be stripped in the "Strip" option.

- Dial By Name
- External Number
- Callback

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 shows the configuration page for an Inbound Route. The 'Trunks' field is set to 'SIPTrunks -- test'. The 'Pattern' field contains '_18005251163'. The 'Default Destination' is set to 'By DID'. The 'Strip' field is set to '8' and the 'Prepend' field is set to '2'. The 'Allowed DID Destination' is set to 'Extension'. Other options like 'Disable This Route', 'Alert-info', 'Fax Detection', 'Block Collect Calls', 'Set CallerID Info', 'Dial Trunk', 'Inbound Multiple Mode', 'Prepend Trunk Name', and 'Enable Route-Level Inbound Mode' are all unchecked. The 'Allowed to seamless transfer' field is empty.

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:

Alert-info: None

Fax Detection:

Block Collect Calls:

Set CallerID Info:

Inbound Multiple Mode:

Allowed to seamless transfer:

Prepend Trunk Name:

Enable Route-Level Inbound Mode:

Default Mode | Mode 1

* Default Destination: Extension 1000

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.

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: Default Mode

* Inbound Default Mode: *61

* Inbound Mode 1: *62

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.

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.

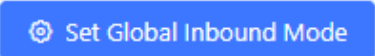
For example, Inbound Default Mode feature code is set to *61 and the Inbound Mode suffix for route 1 is set to 1010. To switch the mode of route 1 to Default Mode, users can dial *611010.

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.

Global Inbound Mode

- 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".



Inbound Mode – Default Mode

- Pressing the key will toggle the inbound mode to "Mode 1", and the button's color will change to red.



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.

Inbound Routes								
<input type="button" value="+ Add"/> <input type="button" value="Blacklist"/> <input type="button" value="Set Global Inbound Mode"/> <input type="button" value="Import"/> <input type="button" value="Export"/> <input type="button" value="Filter"/>								
INBOUND ROUTE NAME	PATTERN	CALLERID PATTERN	INBOUND MODE	INBOUND MODE SUFFIX	TIME CONDITION	TIME	TYPE	DESTINATION
No Data								

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]


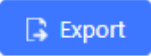
- Inbound Multiple Mode: Yes/No.
- Default Destination: By DID, Extension, Voicemail... User should enter a Default Destination string following the values we have in the Inbound route Default Destination list.
- Destination: An Extension number, Ring Group Extension...
- Default Time Condition.
- Mode 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.

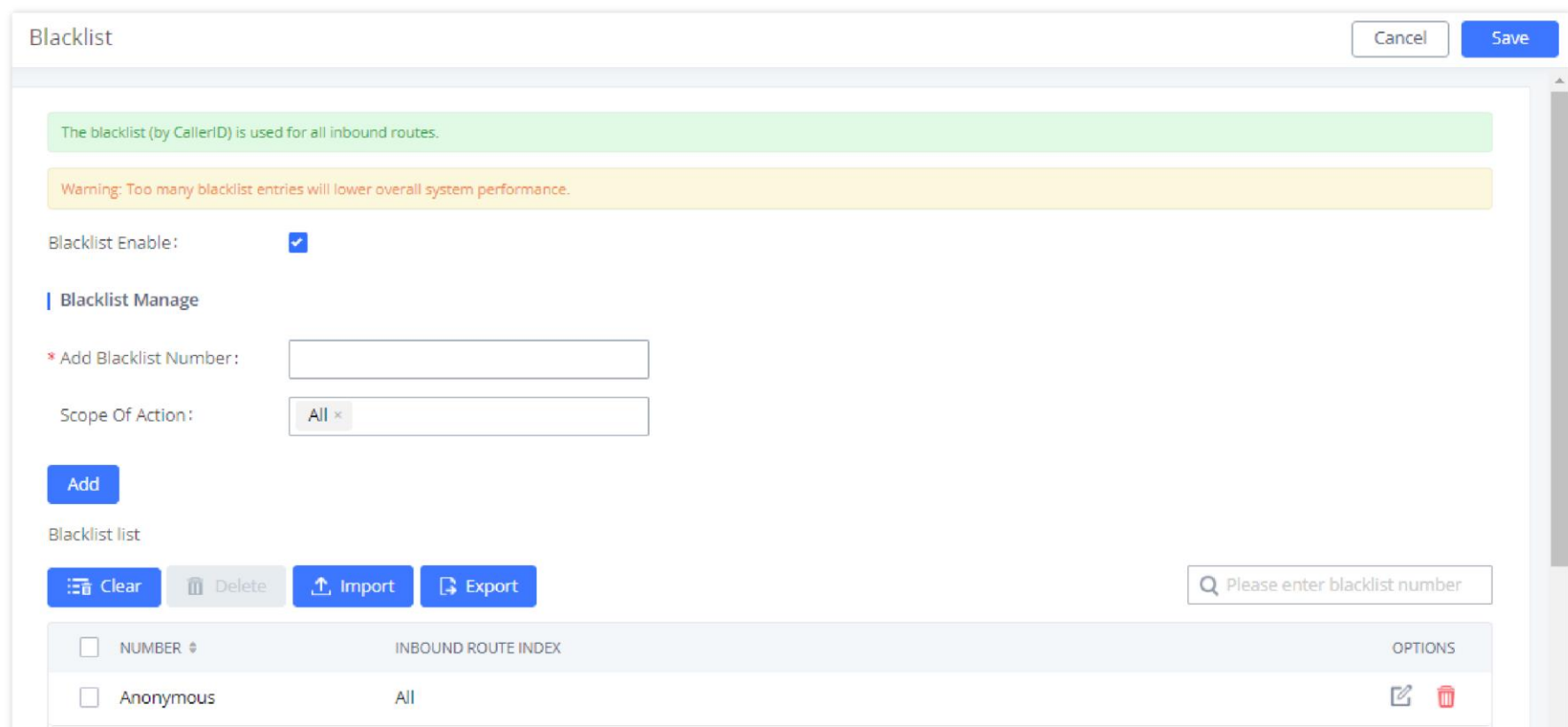
FAX with Two Media

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.



The screenshot shows the 'Blacklist' configuration window. At the top, there are 'Cancel' and 'Save' buttons. Below that, a green message states 'The blacklist (by CallerID) is used for all inbound routes.' followed by a yellow warning: 'Warning: Too many blacklist entries will lower overall system performance.' The 'Blacklist Enable' checkbox is checked. Under 'Blacklist Manage', there is an 'Add Blacklist Number' field and a 'Scope Of Action' dropdown set to 'All'. An 'Add' button is below these fields. The 'Blacklist list' section contains a table with one entry: 'Anonymous' under the 'All' index. Below the table are buttons for 'Clear', 'Delete', 'Import', and 'Export'. A search bar with the placeholder 'Please enter blacklist number' is also present.

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.

	A	B	C	D	E
1	13238680006	12135958547	12136268547	6262357999	
2					
3					
4					
5					

Blacklist csv File



Important 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:

Fax Settings

Enable Error Correction Mode	<p>Configure to enable Error Correction Mode (ECM) for the Fax.</p> <p>The default setting is "Yes".</p>
Maximum Transfer Rate	<p>Configure the maximum transfer rate during the Fax rate negotiation.</p> <p>The possible values are 2400, 4800, 7200, 9600, 12000 and 14400.</p> <p>The default setting is 14400.</p>
Minimum Transfer Rate	<p>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.</p>
Max Concurrent Sending Fax	<p>Configure the concurrent fax that can be sent by UCM6300A. Two modes "Only" and "More" are supported.</p> <ul style="list-style-type: none"> ○ Only <p>Under this mode, the UCM6300A allows only single user to send fax at a time.</p> <ul style="list-style-type: none"> ○ More <p>Under this mode, the UCM6300A supports multiple concurrent fax sending by the users.</p> <p>By default, this option is set to "only".</p>
Fax Queue Length	<p>Configure the maximum length of Fax Queue from 6 to 10.</p> <p>The default setting is 6.</p>
User Information in Fax Header	<p>If enabled this this will give users the option to send a special header in SIP fax messages.</p>
Fax Header Information	<p>Adds fax header into the fax file.</p>
Default Email Address	<p>Configure the Email address to send the received Fax to if user's Email address cannot be found.</p> <p>Note:</p> <p>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.</p>
Template Variables	<p>Fill in the "Subject:" and "Message:" content, to be used in the Email when sending the Fax to the users.</p> <p>The template variables are:</p> <ul style="list-style-type: none"> ○ <code>#{CALLERIDNUM}</code> : Caller ID Number ○ <code>#{CALLERIDNAME}</code> : Caller ID Name ○ <code>#{RECEIVEEXTEN}</code> : The extension to receive the Fax ○ <code>#{FAXPAGES}</code> : Number of pages in the Fax ○ <code>#{VM_DATE}</code> : The date and time when the Fax is received
Send PDF Files Only	<p>If enabled, fax emails will no longer attach TIFF files. Only PDF files will be attached.</p>

Enable Fax Resend	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>OtherFeatures</i> → <i>Fax Sending</i> to allow manual resending.
Max Resend Attempts	Configures the maximum attempts number to resend the fax. Default value is set to 5.
Fax Resend Frequency	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

Advanced Options

SLA Mode:

Enable Polarity Reversal:

Current Disconnect Threshold (ms): 200

* RX Gain:

Use CallerID:

Caller ID Scheme:

Auto Record:

DAHDI Out Line Selection:

Echo Cancellation Mode:

* Trunk Name:

* Ring Timeout:

* TX Gain:

Fax Mode:

* FXO Dial Delay (ms):

Disable This Trunk:

* The Maximum Number of Call Lines:

Direct Callback:

Configure Analog Trunk

1. Go to UCM6300A Web GUI → **Extension/Trunk** → **Extensions** page.
2. Create or edit the extension for FXS port.
 - o **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.

Configure Extension for Fax Machine: FXS Extension

Configure Extension for Fax Machine: Analog Settings

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

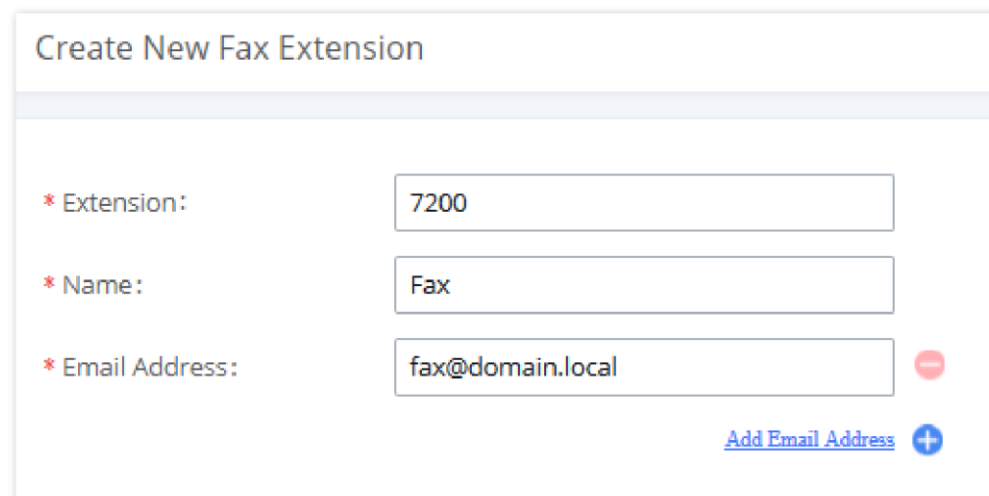
Configure Inbound Rule for Fax

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

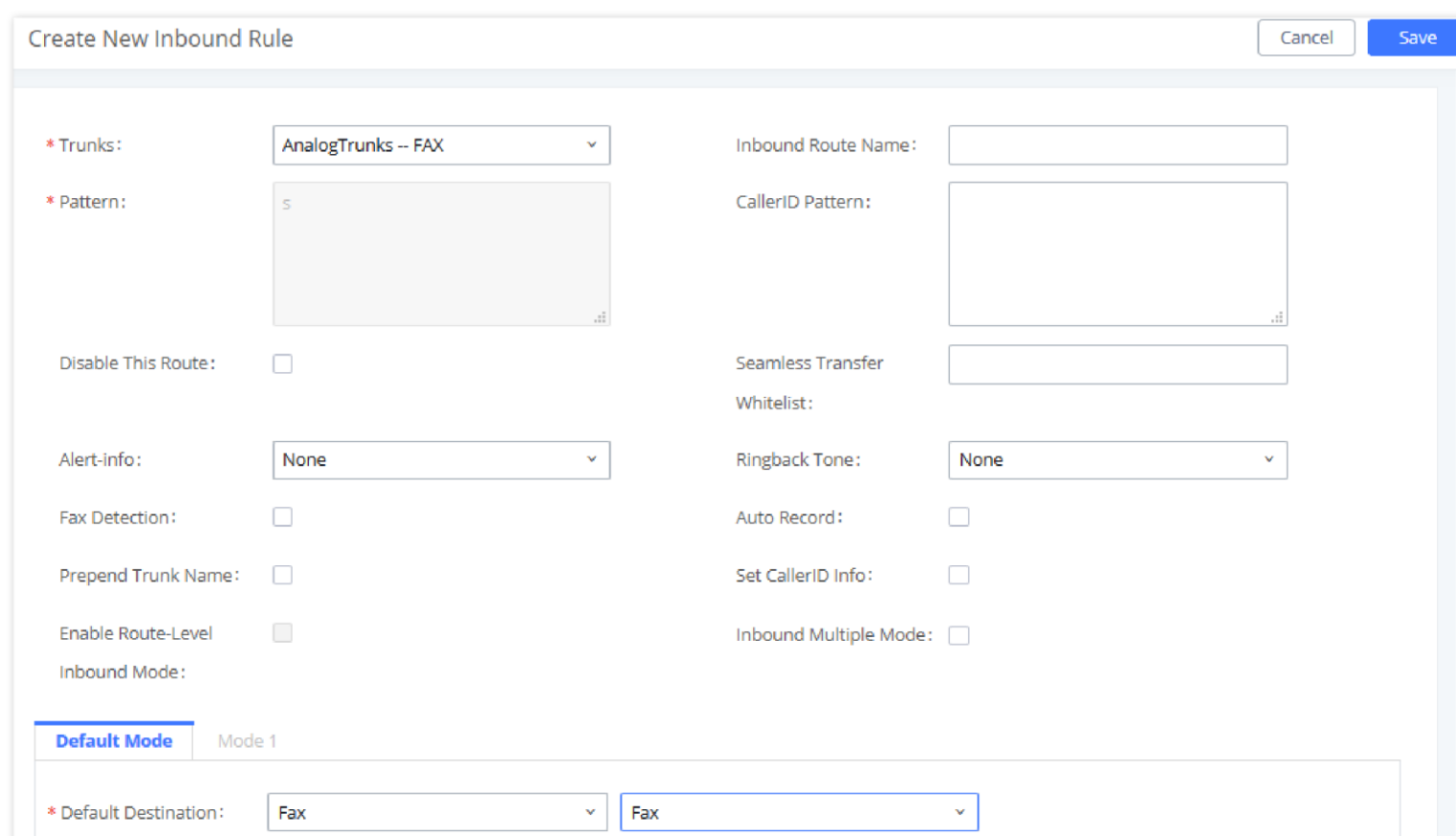
The following instructions describe a sample configuration on how to use Fax-to-Email feature on the UCM6300A.

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 a 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.



Inbound Route to a Fax Extension

5. 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

<input type="checkbox"/>	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	
<input type="checkbox"/>	VFAX-7200-20210125-112246-1611570166.49.tiff	2021-01-25 11:22:46 UTC+01:00	5.69 KB	

List of Fax Files

FAX Sending

Besides the support of Fax machines, The supports also sending Fax via Web GUI access. This feature can be found on Web GUI→**Other 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.

Fax Sending

* External Fax Number:

Fax File:

File Send Progress

Fax Sending in Web GUI

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

Fax Sending

* External Fax Number:

Fax File:

File Send Progress

<input type="checkbox"/>	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%	

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.

UCM630xA series	Number of meeting room	Participant limit
UCM6300A	3	50
UCM6302A	5	75
UCM6304A	7	120
UCM6308A	9	150

Below are the UCM meeting specifications supported for each model:

Room

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

Meeting Room Configuration Parameters

Extension	The number to dial to reach the meeting room.
Meeting Name	The name of the meeting room.
Privilege	Select the permission level for outgoing calls.
Allow User Invite	If enabled, participants can invite other users to the meeting.
Allowed to Override Host Mute	If enabled, participants will be able to unmute themselves if they have been muted by the host.
Auto Record	If enabled, the meeting audio will be recorded and saved as a .WAV file with default filename meeting-\$(Meeting Number)-\${UNIQUEID}. Recordings can be downloaded from the Meeting Recordings page. Note: When this option has been enabled the meeting host cannot stop the recording of the meeting.
Room Password	If meeting room password is configured, meeting participants will need to enter a password to enter the room. Scheduling meetings will not be supported for this room.

Note

Please note that you can't schedule meetings for the rooms which are protected by a password.

Meeting Settings contains the following options:

Meeting Settings

Enable Talk detection	If enabled, the AMI will send the corresponding event when a user starts or ends talking.
DSP Talking Threshold	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.
DSP Silence Threshold	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.
Audio Codec Preference	Configures the preferred codecs for temporary accounts such as meeting participants who joined via link.
Allow New Participants To View Chat History	Configure whether new attendees joining in the middle of a Wave meeting can view the chat content already in the meeting.
Jitter Buffer	<p>Select jitter buffer method for temporary accounts such as meeting participants who joined via link.</p> <ul style="list-style-type: none"> • Disable: Jitter buffer will not be used. • Fixed: Jitter buffer with a fixed size (equal to the value of "Jitter Buffer Size") • Adaptive: Jitter buffer with an adaptive size that will not exceed the value of "Max Jitter Buffer"). • NetEQ: 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.

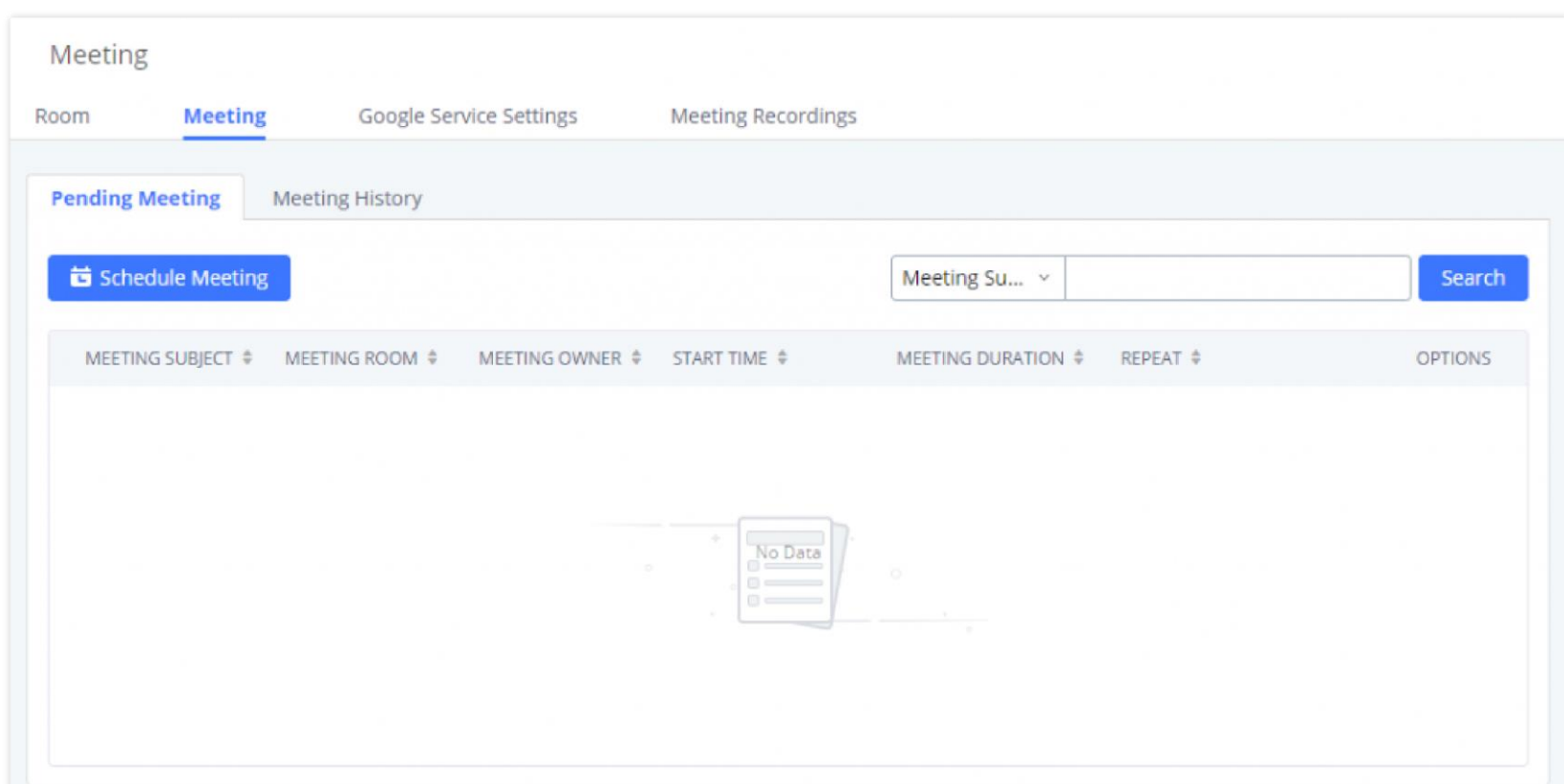
- Click on "Add" to add a new Meeting Schedule.
- Click on the scheduled meeting to edit or delete the event.

Meeting Schedule Parameters

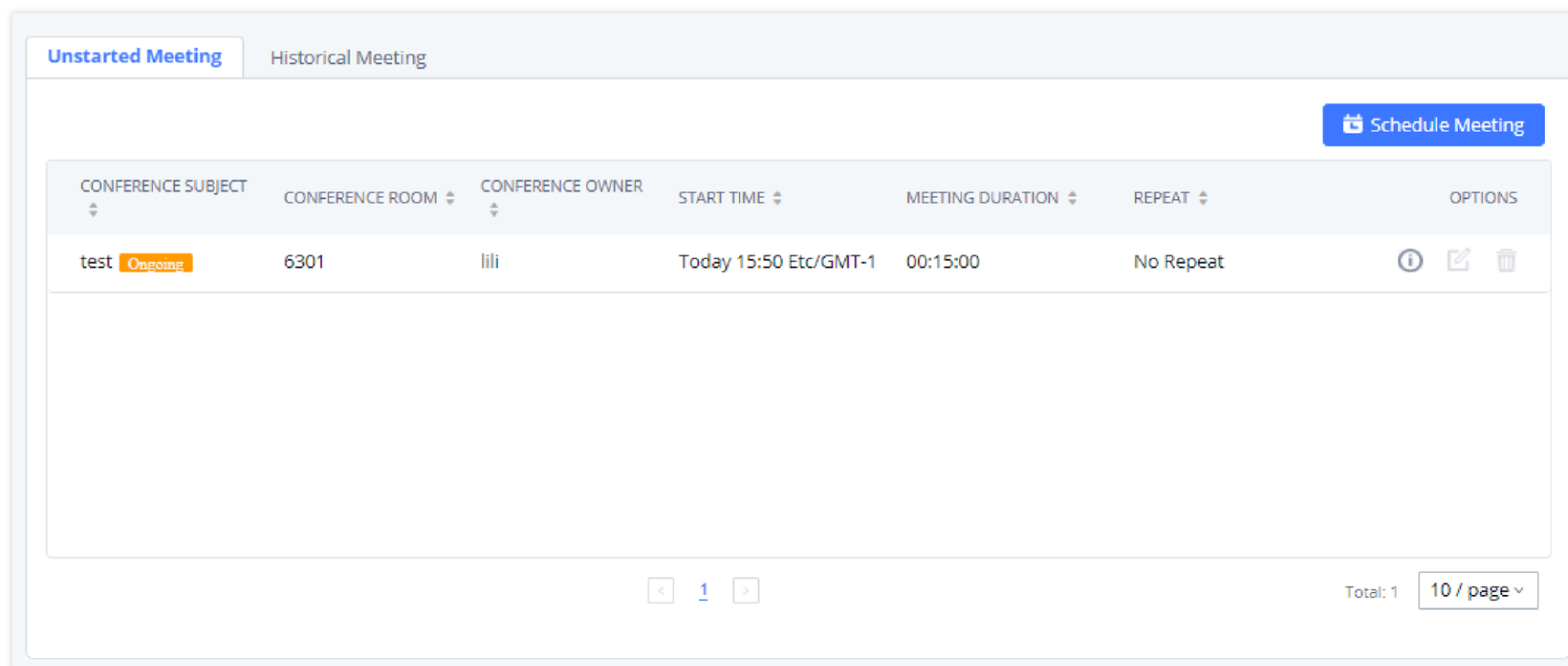
Schedule Options	
Meeting Subject	Configure the name of the scheduled meeting. Letters, digits, Other special characters are also supported. such as #%&@*=-
Meeting Room	Choose which room to have this scheduled meeting. If this option has been enabled, please select an existing room for this meeting. If this option has not been enabled, a new meeting room will be created.
Time	Configure the meeting date and time.
Time Zone	Select the meeting time zone.
Password	Configure the meeting's login password.
Host Password	Configure the Host Password. Note: It is randomly generated when first creating a new meeting Schedule.

Host	Configure Host.
Repeat	Choose when to repeat a scheduled meeting.
Allow User Invite	<p>If this option is enabled, the user can:</p> <ul style="list-style-type: none"> • Press '0' to invite others to join the meeting with invited party's permission • Press '1' to invite without invited party's permission • Press '2' to create a multi-meeting room to another meeting room • Press '3' to drop all current multi-meeting rooms. <p>Note: Meeting host is always allowed to access this menu.</p>
Call Participant	If enabled, the invited participants will be called upon meeting start time.
Allowed to Override Host Mute	If enabled, participants will be able to unmute themselves if they have been muted by the host.
Email Reminder (m)	<p>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.</p> <p>Note: After editing the time of a single recurrence of a scheduled meeting, a cancelation email will now be sent out followed by a meeting update email.</p>
Auto Record	<p>If selected, the meeting will be recorded and saved as either a .WAV or .MKV file. The default filename is meeting-$\{Meeting\ Number\}$-$\{UNIQUEID\}$. Recordings can be downloaded from either the Meeting Recordings or the Meeting Video Recordings page. Video recordings require external storage to be available. When recording a screen share, only the screen share and meeting audio will be recorded.</p> <p>Note: Please note that UCM63XX Audio Series doesn't support Screen Sharing, Whiteboard, or PDF file sharing.</p>
Enable Google Calendar	<p>Select this option to sync scheduled meeting with Google Calendar.</p> <p>Note: Google Service Setting OAuth2.0 must be configured on the UCM630X. Please refer to section [Google Service Settings Support].</p>
Meeting Agenda	Enter information about the meeting, e.g., the purpose of the meeting or the subjects that will be discussed in the meeting.
Invitees	Local extensions, remote extensions, and special extensions are supported.

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



Once the meeting starts, it will be displayed under **Unstarted Meeting** with an "Ongoing" status, as displayed below.

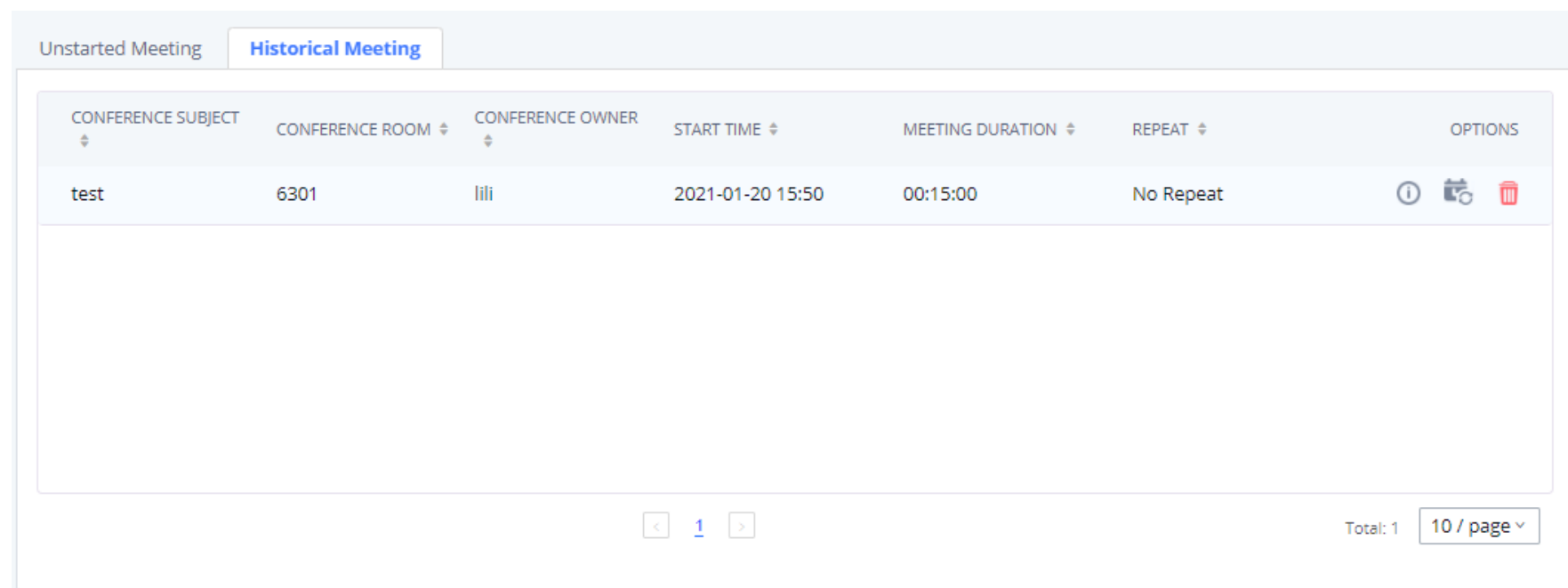


The screenshot shows a web interface with two tabs: "Unstarted Meeting" (active) and "Historical Meeting". A blue "Schedule Meeting" button is in the top right. Below is a table with columns: CONFERENCE SUBJECT, CONFERENCE ROOM, CONFERENCE OWNER, START TIME, MEETING DURATION, REPEAT, and OPTIONS. One row is visible with the following data: test (with an "Ongoing" status tag), 6301, lili, Today 15:50 Etc/GMT-1, 00:15:00, No Repeat, and three icons (info, edit, delete). At the bottom, there are navigation arrows, a page number "1", and a "Total: 1" label with a "10 / page" dropdown.

CONFERENCE SUBJECT	CONFERENCE ROOM	CONFERENCE OWNER	START TIME	MEETING DURATION	REPEAT	OPTIONS
test Ongoing	6301	lili	Today 15:50 Etc/GMT-1	00:15:00	No Repeat	

Meeting Scheduled-Ongoing

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



The screenshot shows the same web interface but with the "Historical Meeting" tab active. The table now shows a meeting with the following data: test, 6301, lili, 2021-01-20 15:50, 00:15:00, No Repeat, and three icons (info, calendar, delete). The "Ongoing" status tag is absent. The bottom navigation and pagination are identical to the previous screenshot.

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	

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.

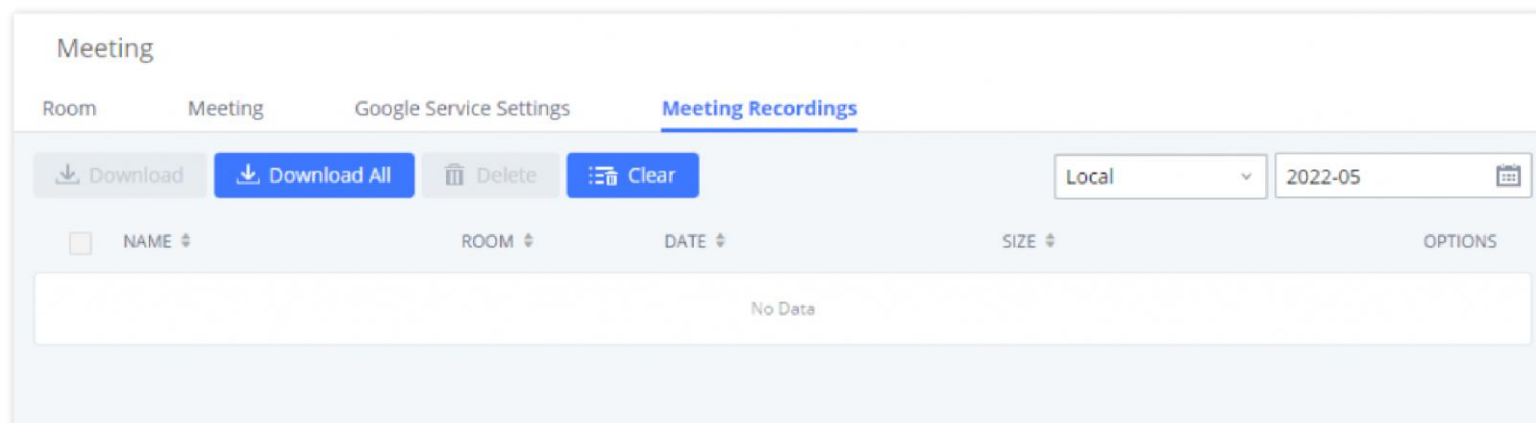
The recording files will be listed as below once available. Users could click on



to download the recording or click on



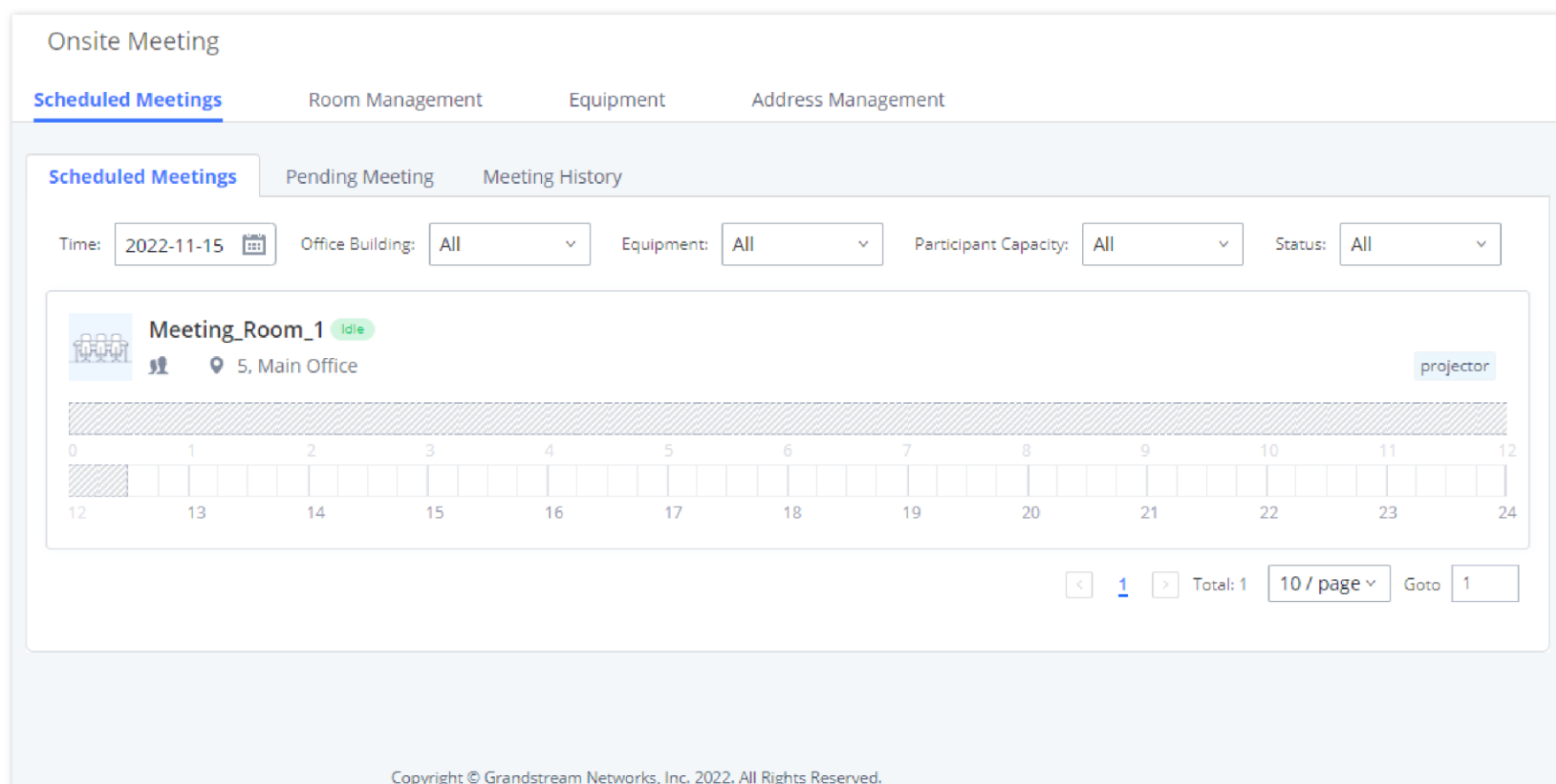
to delete the recording. Users could also delete all recording files by clicking on "Delete All Recording Files" or delete multiple recording files at once by clicking on "Delete" after selecting the recording files.



Meeting Recordings

ONSITE MEETING

For workplaces that require employees to return to physical offices for work, Grandstream UCM offers the Onsite Meeting feature, a new way to stay organized and keep up-to-date with in-person meetings. This feature allows administrators to create and manage onsite meeting rooms, specify meeting room locations, schedule meetings, and add conferencing equipment. The new feature can be found under the **Device Management > Onsite Meeting** page. The first page that appears is the **Scheduled Meetings** page and tab page, which provide an overview of all created meeting rooms. It provides information about the rooms' meeting schedules for the day, their locations, their member capacity, and their equipment.



Scheduled Onsite Meeting

The **Pending Meeting** tab and **Meeting History** tab show detailed information about upcoming meetings and previous meetings respectively. From the **Pending Meeting** tab, users can delete upcoming meetings and extend the duration of ongoing meetings. **The Meeting History** tab will display the last 6 months of onsite meeting.

Onsite Meeting

Scheduled Meetings Room Management Facility Address Management

Scheduled Meetings Pending Meeting Meeting History

Schedule Meeting Meeting Subject Search

MEETING SUBJECT #	MEETING ROOM #	MEETING OWNER #	START TIME #	MEETING DURATION #	REPEAT #	OPTIONS
Quick Meeting Starting soon	BackupRoom	[Avatar]	Today 20:00 Etc/GMT+8	00:30:00	No Repeat	[Info] [Edit] [Delete] [Refresh]
Recent Events	BackupRoom	[Avatar]	2022-11-24 07:00 Etc/GMT+8	01:00:00	No Repeat	[Info] [Edit] [Delete] [Refresh]
Yearly Reports	Primary-Meeting-Room	[Avatar]	2022-12-29 13:00 Etc/GMT+8	01:00:00	No Repeat	[Info] [Edit] [Delete] [Refresh]

Pending Onsite Meeting



When a meeting is scheduled the invitees will receive an invitation via email which contains all the information regarding the onsite meeting. The email will be attached with an .ics file which can be imported to a calendar to mark your calendar of the onsite meeting, please see the screenshot below.

Onsite Meeting Email Invitation

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:

Create New IVR

IVR Configuration Parameters

Basic Settings	
Name	Configure the name of the IVR. Letters, digits, _ and – are allowed.
Extension	Enter the extension number for users to access the IVR.
Dial Trunk	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”.
Auto Record	If enabled, calls to this IVR will automatically be recorded.
Permission	<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 compared 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>

Dial Other Extensions	<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"> ○ Extension ○ Meeting ○ Call Queue ○ Ring Group ○ Paging/Intercom Groups ○ Voicemail Groups ○ Dial by Name ○ All
IVR Black/Whitelist	If enabled only numbers inside of the Whitelist or outside of the Blacklist can be called from IVR.
Internal Black/Whitelist	Contain numbers, either of Blacklist or Whitelist.
External Black/Whitelist	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...
Replace Display Name	If enabled, the UCM will replace the caller display name with IVR name.
Return to IVR Menu	If enabled and if a call to an extension fails, the caller will be redirected to the IVR menu.
Alert Info	When present in an INVITE request, the alert-Info header field specifies an alternative ring tone to the UAS.
Prompt	Select an audio file to play as the welcome prompt for the IVR. Click on "Prompt" to add additional audio file under Web GUI→ PBX Settings → Voice Prompt → Custom Prompt .
Digit Timeout	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. Default timeout is 3s.
Response Timeout	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.
Response Timeout Prompt	Select the prompt message to be played when timeout occurs.
Invalid Input Prompt	Select the prompt message to be played when an invalid extension is pressed.
Response Timeout Prompt Repeats	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.
Invalid Input Prompt Repeats	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.
Language	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→ PBX Settings → Voice Prompt → Language Settings . 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→ PBX Settings → Voice Prompt → Language Settings .
Key Pressing Events	

<p>Key Press Event:</p> <p>Press 0</p> <p>Press 1</p> <p>Press 2</p> <p>Press 3</p> <p>Press 4</p> <p>Press 5</p> <p>Press 6</p> <p>Press 7</p> <p>Press 8</p> <p>Press 9</p> <p>Press *</p>	<p>Select the event for each key pressing for 0-9, *, Timeout and Invalid. The event options are:</p> <ul style="list-style-type: none"> ○ Extension ○ Voicemail ○ Meeting Rooms ○ Voicemail Group ○ IVR ○ Ring Group ○ Queues ○ Page Group ○ Custom Prompt ○ Hangup ○ DISA ○ Dial by Name ○ External Number ○ Callback <p>For each key event, time condition can be configured. At the configured time condition, this IVR key event can be triggered. Office time, holiday time or specific time can be configured for time condition. Up to 5 time conditions can be added for each key.</p> <p>The available time conditions are 'All', 'Office Time', 'Out of Office Time', 'Holiday', 'Out of Holiday', 'Out of Office Time or Holiday', 'Office Time and Out of Holiday' and 'Specific Time'. If 'Specific Time' is selected, a new window will prompt for admin to configure start time, end time and frequency.</p>
<p>Timeout</p>	<p>When exceeding the number of defined answer timeout, IVR will enter the configured event when timeout. If not configured, then it will Hangup.</p>
<p>Invalid</p>	<p>Configure the destination when the Invalid Repeat Loop is done.</p>
<p>Time Condition</p>	<p>Configure the time condition for each key press event, so that it goes to the corresponding destination within a specified time.</p>

Edit IVR: test

Basic Settings **Key Pressing Events** Cancel Save

Press 0

Destination: Extension 3001 Time Condition: Specific Time

TIME	WEEK	MONTH	DAY	OPTIONS
08:00-11:00	Sun Mon Tue Wed Thu Fri Sat	Default	Default	Add +

Press 1

Destination: Select an Option Time Condition: All Time

[Add](#) +

Press 2

Destination: Select an Option Time Condition: All Time

[Add](#) +

Press 3

Destination: Select an Option Time Condition: All Time

[Add](#) +

Press 4

Destination: Select an Option Time Condition: All Time

[Add](#) +

Key Pressing Events

Blacklist/Whitelist 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 black/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:

<input type="checkbox"/> 28 items	Available	<input type="checkbox"/> 2 items	Selected
<input type="checkbox"/>	1000	<input type="checkbox"/>	1001
<input type="checkbox"/>	1003	<input type="checkbox"/>	1002
<input type="checkbox"/>	1004		
<input type="checkbox"/>	1005		
<input type="checkbox"/>	1006		

External Blacklist/Whitelist:

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:

[Add Prompt](#)

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.

Key Pressing Events

Standard Key Event

UCM supports adding time conditions for different key events, so that each key event of the IVR goes to the corresponding destination within a specified time.

Each key event support up to five time conditions, the options available are: All time, Office Time, Out of Office Time, Holiday, Out of Holiday, Out of Office Time or Holiday, Office Time and Out Of Holiday, Specific time.

The screenshot shows the 'Edit IVR: IVR-DISA' window with the 'Key Pressing Events' tab selected. There are two main sections: 'Press 0' and 'Press 1'. Each section has a 'Destination' dropdown menu set to 'Select an Option' and a 'Time Condition' dropdown menu. The 'Time Condition' dropdown for 'Press 0' is open, showing a list of options: 'Office Time', 'Holiday', 'All Time', 'Office Time', 'Out of Office Time', 'Holiday' (highlighted), 'Out of Holiday', 'Out of Office Time or Hold.', and 'Office Time and Out of Holi...'. There are also 'Cancel' and 'Save' buttons at the top right.

Key Pressing Events

Note:

If you select "Specific time", you need to select the start time and the end time.

The frequency supports two options: By week and By Month, by default the specific time does not include the holidays.

The 'Specific Time' dialog box contains the following elements:

- Time:** Two time pickers for 'Start Time' and 'End Time'.
- Frequency:** Radio buttons for 'By Week' and 'By Month'.
- Month Selection:** A 3x4 grid of month abbreviations: Jan, Feb, Mar, Apr; May, Jun, Jul, Aug; Sept, Oct, Nov, Dec.
- Day Selection:** Radio buttons for 'Week' and 'Day'.
- Day Grid:** A 6x7 grid with columns for Mon, Tue, Wed, Thu, Fri, Sat, Sun and rows numbered 1 to 6.
- Excluding Holidays:** A checkbox labeled 'Excluding Holidays'.
- Buttons:** 'Cancel' and 'OK' buttons at the bottom.

Specific Time

Custom Key Event

Users can create custom IVR key press events, vastly increasing the options a business can provide to its customers and improving customer relations and accessibility.

Create New IVR
Cancel Save

Basic Settings
Key Pressing Events

Key Event Type: Standard Custom

Key Events

+ Add Delete Clear

Key: 🗑️

Destination:

Time Condition:

-

+

Other Settings

Timeout

Destination:

Time Condition:

-

+

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Key Pressing Event

This new feature supports the following:

- Up to 100 custom key press events
- Each key combination can contain up to 8 characters (numbers and star (*) only)
- Supports Time Conditions
- Different custom keys can have the same Destination and Time Condition

Note

IVR option Dial Other Extensions will be disabled if using custom IVR keys.

LANGUAGE SETTINGS FOR VOICE PROMPT

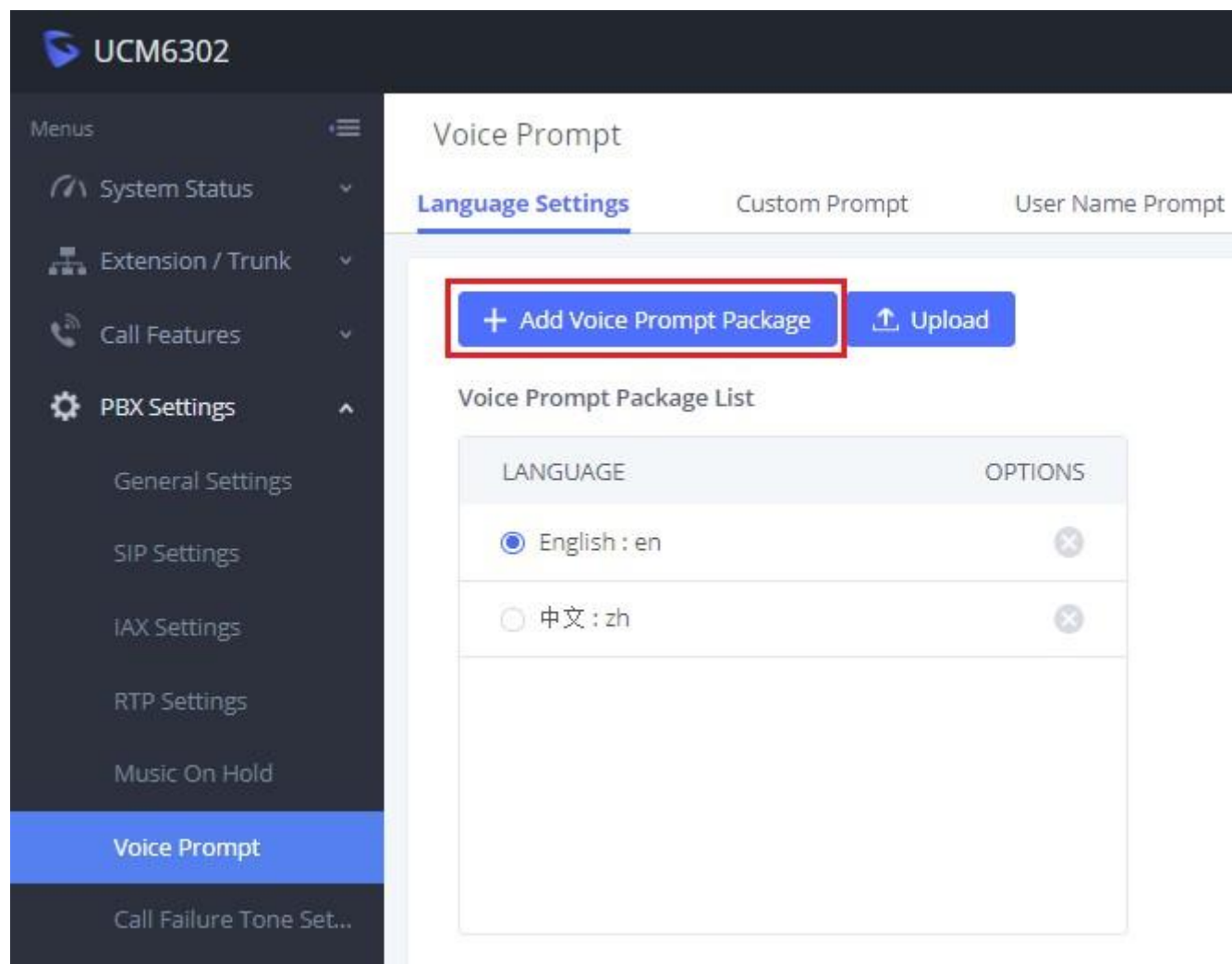
The UCM630xA supports multiple languages in Web GUI as well as system voice prompt. Currently, there are 16 languages supported in system voice prompt: British English, Deutsch, English, Spanish, Spanish (Catalonia), Spanish (Spain), Greek, French, Italian, Dutch, Polish, Portuguese (Brazil), Portuguese (Portugal), Russian, Swedish, Turkish, Czech, Ukrainian, Hebrew, Arabic, and Chinese.

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.



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.

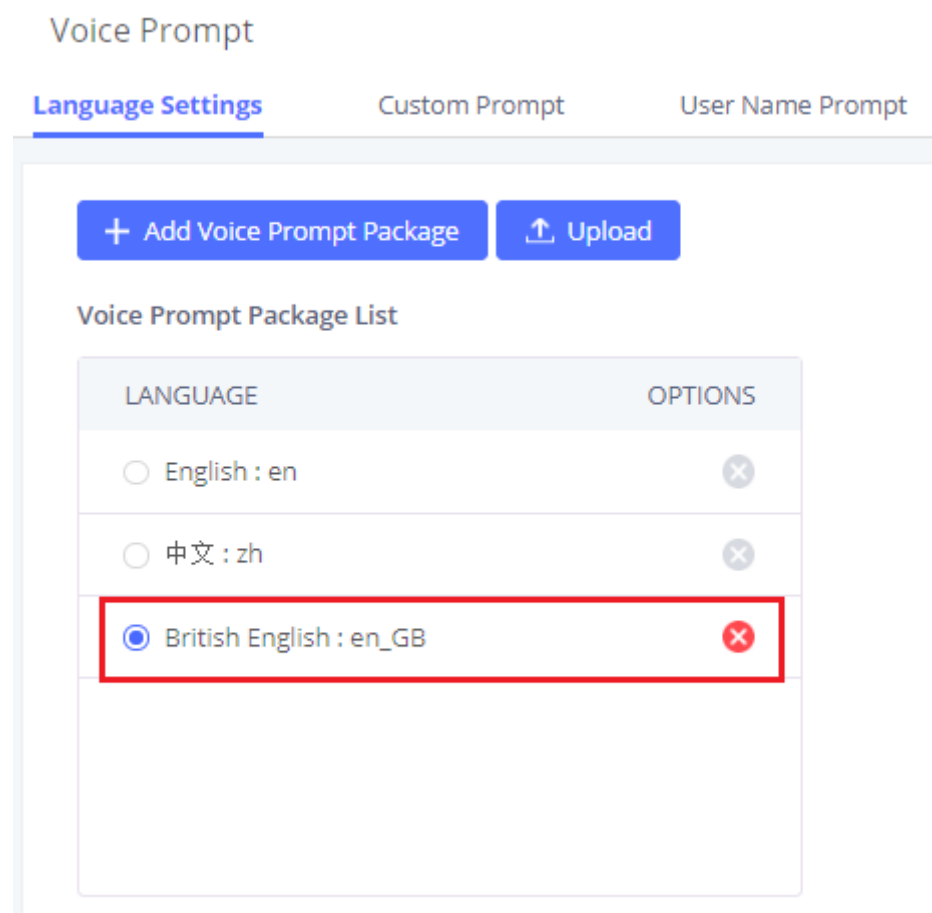
VOICE PROMPT PACKAGE LIST	VERSION (REMOTE / LOCAL)	SIZE	OPTIONS
British English	1.9/-	4.2M	↓
Deutsch	1.8/-	4.2M	↓
English	1.11/1.8	6.0M	⬆
Español	1.10/-	4.4M	↓
Español(Català)	1.8/-	3.1M	↓
Español(Español)	1.8/-	4.2M	↓
Ελληνικά	1.8/-	4.4M	↓
Français	1.8/-	4.1M	↓
Italiano	1.8/-	4.0M	↓

Voice Prompt Package List

Click on



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

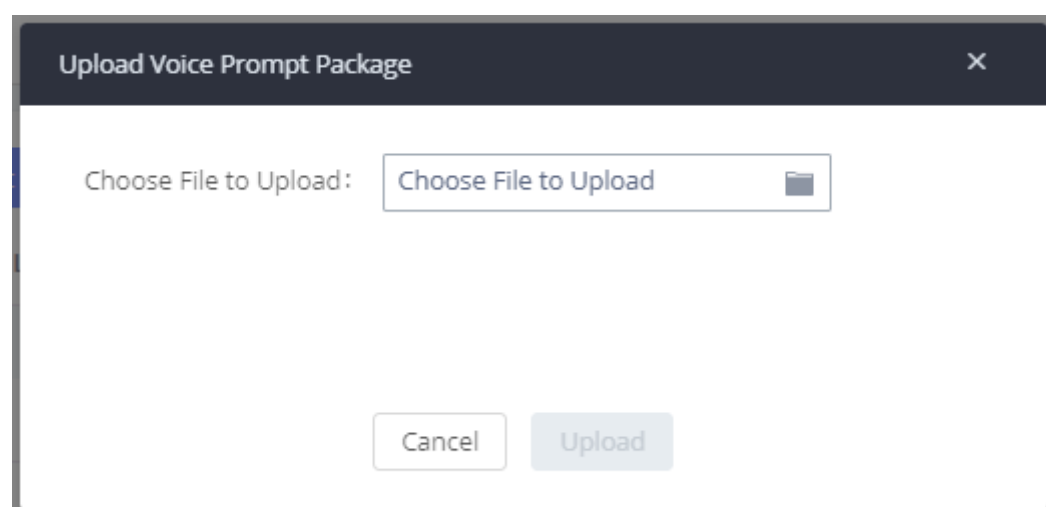


New Voice Prompt Language Added

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.



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.

1. Go under web GUI **PBX Settings** → **Voice Prompt** → **Username Prompt** and click on "**Upload**" button.

2. 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):

Dial "0" for Operator:

Operator Type:

Operator Extension:

* Max Messages Per Folder:

Max Message Time:

Min Effective Message Time:

Announce Message Caller-ID:

Announce Message Duration:

Play Envelope:

Play Most Recent First:

Allow User Review:

Voicemail Remote Access:

Forward Voicemail to Peered UCMs:

Voicemail Password:

Format:

Voicemail Settings

Max Greeting Time (s)	Configure the maximum number of seconds for the voicemail greeting. The default setting is 60 seconds.
Dial '0' For Operator	If enabled, the caller can press 0 to exit the voicemail application and connect to the configured operator's extension.
Operator Type	Configure the operator type; either an extension or a ring group.
Operator Extension	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.
Max Messages Per Folder	Configure the maximum number of messages per folder in users' voicemail. The valid range 10 to 1000. The default setting is 50.
Max Message Time	<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"> ○ 1 minute ○ 2 minutes ○ 5 minutes ○ 15 minutes ○ 30 minutes ○ Unlimited

Min Effective Message Time	<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"> ○ No minimum ○ 1 second ○ 2 seconds ○ 3 seconds ○ 4 seconds ○ 5 seconds <p>Note: Silence and noise duration are not counted in message time.</p>
Announce Message Caller-ID	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".
Announce Message Duration	If enabled, the message duration will be announced at the beginning of the voicemail message. The default setting is "No".
Play Envelope	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".
Play Most Recent First	If enabled, it will play the most recent message first.
Allow User Review	If enabled, users can review the message following the IVR before sending.
Voicemail Remote Access	<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"> ○ <i>Press 1 to leave a message.</i> To leave a message for the extension reached by DID. ○ Press 2 to access voicemail management system. <p>This will allow caller to access any extension VM after entering extension number and its VM password.</p> <p>Note: This option applies to inbound call routed by DID only.</p> <p>The default setting is "Disabled".</p>
Forward Voicemail to Peered UCMs	<p>Enables the forwarding of voicemail to remote extensions on peered SIP trunks.</p> <p>The default setting is "Disabled".</p>
Voicemail Password	Configures the default voicemail password that will be used when an extension is reset.
Format	Warning: WAV files take up significantly more storage space than GSM files.

Voicemail Settings

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.

Main Menu	Sub Menu 1	Sub Menu 2
1 – New messages		1 - Send a reply
		2 - Call the person who sent this message
	3 - Advanced options	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		
2 – Change folders	0 - New messages	
	1 - Old messages	
	2 - Work messages	
	3 - Family messages	
	4 - Friend messages	
	# - Cancel	
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	
0 – Mailbox options	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	

✔ Tips

- While listening to the voicemail, press * or # to rewind and forward the voice message, respectively. Each press will forward or rewind 3 seconds.
- Rewind can go back to the beginning of the message while forward will not work when there are 3 seconds or less left in the voice message.
- Voice guidance will be automatically played when the voicemail is done playing.

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.

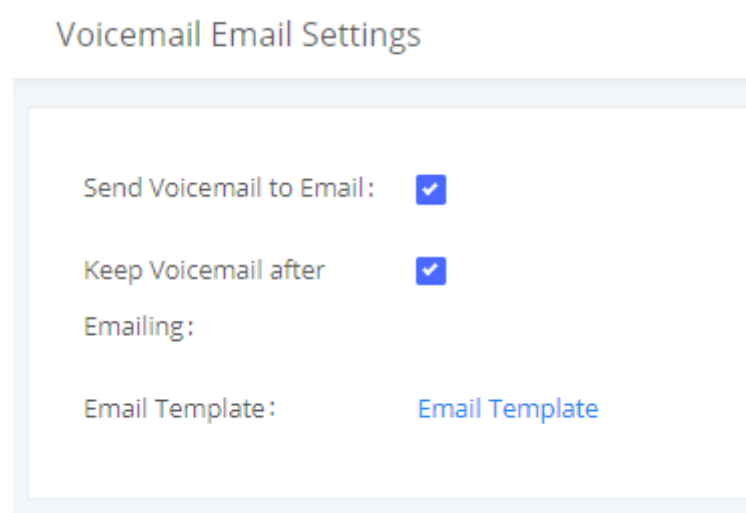
In case if the caller is calling from an internal extension, they will be directly forwarded to the extension’s voicemail box. But if the caller is calling from outside the system and the incoming call is routed by DID to the destination extension, then the caller will be prompted with the choice to 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.

Send Voicemail to Email	If enabled, voicemail will be sent to the user’s email address. Note: SMTP server must be configured to use this option.
Keep Voicemail after Emailing	Enable this option if you want to keep recording files after the Email is sent. The default setting is Enable.
Email Template	<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"> ○ \t: TAB ○ \${VM_NAME}: Recipient’s first name and last name ○ \${VM_DUR}: The duration of the voicemail message ○ \${VM_MAILBOX}: The recipient’s extension ○ \${VM_CALLERID}: The caller ID of the person who has left the message ○ \${VM_MSGNUM}: The number of messages in the mailbox ○ \${VM_DATE}: The date and time when the message is left

Voicemail Email Settings



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.

Voicemail Group

Extension	Enter the Voicemail Group Extension. The voicemail messages left to this extension will be forwarded to all the voicemail group members.
Name	Configure the Name to identify the voicemail group. Letters, digits, _ and - are allowed.

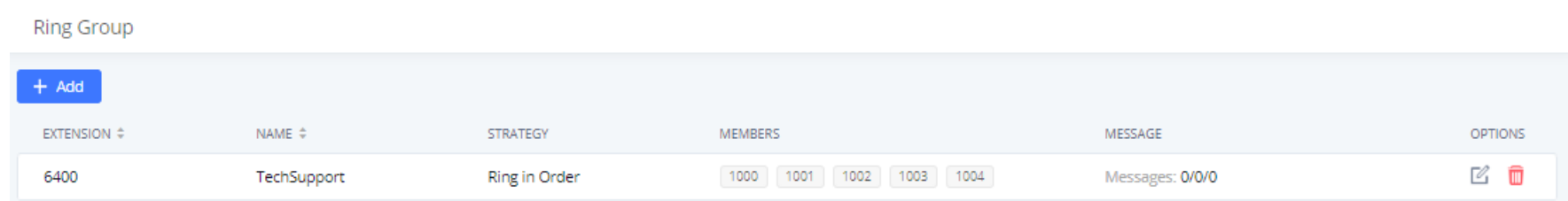
Method	<p>Select the preference for receiving and managing group voicemail.</p> <ul style="list-style-type: none"> • Forwarded: Voicemail will be stored in the group voicemail box, and each voicemail group member will be forwarded a copy of it. • Shared: Voicemail will be stored in the group voicemail box, and voicemail status will be shared among all voicemail group members. If a member deletes a voicemail, it will also be deleted for all members. Likewise, if one member reads a voicemail, it will be considered read for the entire group.
Voicemail Password	Configure the voicemail password for the users to check voicemail messages.
Email Address	Configure the Email address for the voicemail group extension.
Shared Voicemail Status	If enabled, voicemail group status can be monitored via BLF. Green indicates no unread voicemail, and red indicates existing unread voicemail.
Members	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.
Greet Prompt	<p>This voicemail prompt will be played when the callee does not answer within their ring timeout period. Priority: Temporary Prompt > Busy Prompt/Unavailable Prompt > Greet Prompt</p> <p>Sound file must be PCM encoded, 16 bits at 8000Hz mono with mp3/wav format, or raw ulaw/alaw/gsm file with .mp3/.wav/.ulaw/.alaw/.gsm suffix. The file size must be less than 5MB.</p>
Temporary Prompt	<p>This voicemail prompt will be played in all scenarios when it is configured (unregistered, unanswered/ring timeout, busy, DND). Priority: Temporary Prompt > Busy Prompt/Unavailable Prompt > Greet Prompt</p> <p>Sound file must be PCM encoded, 16 bits at 8000Hz mono with mp3/wav format, or raw ulaw/alaw/gsm file with .mp3/.wav/.ulaw/.alaw/.gsm suffix. The file size must be less than 5MB.</p>
Unavailable Prompt	<p>This voicemail prompt will be played when user enters voicemail. Priority: Temporary Prompt > Busy Prompt/Unavailable Prompt > Greet Prompt</p> <p>Sound file must be PCM encoded, 16 bits at 8000Hz mono with mp3/wav format, or raw ulaw/alaw/gsm file with .mp3/.wav/.ulaw/.alaw/.gsm suffix. The file size must be less than 5MB.</p>

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.

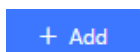
Configure Ring Group

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



Ring Group

- o 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.

Ring Group Name	Configure ring group name to identify the ring group. Letters, digits, _ and – are allowed.
Extension	Configure the ring group extension.
Members	Select available users from the left side to the ring group member list on the right side. Click on ▲ ▼ to arrange the order.
LDAP Phonebook	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.
Ring Strategy	Select the ring strategy. The default setting is “Ring in order”. <ul style="list-style-type: none"> ● Ring Simultaneously: 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 ringing. ● Ring in Order: 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.
Music On Hold	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.
Custom Prompt	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. Note: Users can also refer to the page PBX Settings → Voice Prompt → Custom Prompt , where they could record new prompt or upload prompt files.
Ring Timeout on Each Member	Configure the number of seconds to ring each member. If set to 0, it will keep ringing. The default setting is 60 seconds. Note: The actual ring timeout might be overridden by users if the phone has ring timeout settings as well.
Auto Record	If enabled, calls on this ring group will be automatically recorded. The default setting is No. The recording files can be accessed from WebGUI→ CDR → Recording Files .
Endpoint Call Forwarding Support	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. However, this feature has the following limitations: <ul style="list-style-type: none"> ● This feature will work only when call forwarding is configured on endpoints, not on the UCM. ● 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. ● 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. ● If the forwarded call hits voicemail, the other ring group members will no longer receive the call.
Replace Display Name	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.
Skip Busy Agent	If enabled, skip busy agents regardless of call waiting settings.

Enable Destination	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.
Default Destination	The call would be routed to this destination if no one in this ring group answers the call. Note: 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.
Voicemail	Whether to enable the voicemail for the ring group or not.
Voicemail Password	Configure the voicemail password (only numbers).
Email Address	Fill in the user's Email address (s), the voice message will be sent to this address (s).
Busy Prompt	This voicemail prompt will be played when the callee is in another call or is in DND mode. Priority: Temporary Prompt > Busy Prompt/Unavailable Prompt > Greet Prompt Sound file must be PCM encoded, 16 bits at 8000Hz mono with mp3/wav format, or raw ulaw/alaw/gsm file with .mp3/.wav/.ulaw/.alaw/.gsm suffix. The file size must be less than 5MB.
Greet Prompt	This voicemail prompt will be played when the callee does not answer within their ring timeout period. Priority: Temporary Prompt > Busy Prompt/Unavailable Prompt > Greet Prompt Sound file must be PCM encoded, 16 bits at 8000Hz mono with mp3/wav format, or raw ulaw/alaw/gsm file with .mp3/.wav/.ulaw/.alaw/.gsm suffix. The file size must be less than 5MB.
Temporary Prompt	This voicemail prompt will be played in all scenarios when it is configured (unregistered, unanswered/ring timeout, busy, DND). Priority: Temporary Prompt > Busy Prompt/Unavailable Prompt > Greet Prompt Sound file must be PCM encoded, 16 bits at 8000Hz mono with mp3/wav format, or raw ulaw/alaw/gsm file with .mp3/.wav/.ulaw/.alaw/.gsm suffix. The file size must be less than 5MB.
Unavailable Prompt	This voicemail prompt will only be played when the callee's extension is unregistered. Priority: Temporary Prompt > Busy Prompt/Unavailable Prompt > Greet Prompt Sound file must be PCM encoded, 16 bits at 8000Hz mono with mp3/wav format, or raw ulaw/alaw/gsm file with .mp3/.wav/.ulaw/.alaw/.gsm suffix. The file size must be less than 5MB.

Create New Ring Groups

* Ring Group Name:

* Extension:

Members:

<input type="checkbox"/> 27 items Available	<input type="checkbox"/> 3 items Selected
<input type="checkbox"/> 1000	<input type="checkbox"/> 1002
<input type="checkbox"/> 1001	<input type="checkbox"/> 1003
<input type="checkbox"/> 1005	<input type="checkbox"/> 1004
<input type="checkbox"/> 1006	
<input type="checkbox"/> 1007	

LDAP Phonebook:

<input type="checkbox"/> 0 item Available	<input type="checkbox"/> 0 item Selected
None	None

Ring Group Options

Ring Strategy:

Music On Hold:

Custom Prompt: [Upload Audio File](#)

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:

Sync LDAP Server option

3. 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.

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

Total: 1 10 / page Goto 1

Manually Sync LDAP Server

4. Under Ring Groups setting page, click "Add". Ring Groups can be found under Web GUI→Call Features→Ring Groups.

5. 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

Search

1000

1001

1002

1003

Selected Extensions

Search

None

LDAP Phonebook: 15 Available LDAP

Search

5000(ou=ucm6510,dc=pbx,dc=com)

5001(ou=ucm6510,dc=pbx,dc=com)

5002(ou=ucm6510,dc=pbx,dc=com)

5003(ou=ucm6510,dc=pbx,dc=com)

Selected LDAP

Search

None

Ring Group Options

Ring Group Remote Extension

RESTRICT CALLS

Restrict calls is a feature that can be used to restrict calls between internal extensions besides those in the Allowed List.

This section describes the configuration of this feature in the Call Features→Restrict Calls page.

Create New Restrict Calls

• Name:

Restrict Calls between Extension:

Members:

Available

Search

1001

Selected

Search

1000

Allowed List:

Available

Search

1000



Selected

Search

1001

Restrict Calls

Configure Restrict Calls

- Click on "Add" to add a rule for restrict calls.
- Click on  to edit the rule of restrict calls.
- Click on  to delete the rule of restrict calls.

Name	Configure Restrict call's name
Restrict Calls between extensions	When enabled, members of the group cannot dial other extension, only the numbers in the Allowed List. Note: It's enabled by default.
Members	Configure the members that will not be able to call any extensions besides those in the Allowed List.
Allowed list	Select the extensions that the Members list can be able to call.

PAGING/INTERCOM

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**.

Paging/Intercom Groups

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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2-way Intercom

Name	Configure paging/intercom group name.
Type	Select "2-way Intercom".
Extension	Configure the paging/intercom group extension.
Auto Record	Enable this option to record in WAV format.
Replace Display Name	If enabled, the UCM will replace the caller display name with Paging/Intercom name.
Maximum Call Duration	Specify the maximum call duration in seconds. The default value 0 means no limit.
Custom Prompt	<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>Note: Users can also refer to the page PBX Settings→Voice Prompt→Custom Prompt, where they could record new prompt or upload prompt files.</p>
Members	Select available users from the left side to the paging/intercom group member list on the right.
Paging/Intercom Whitelist	Select which extensions are allowed to use the paging/intercom feature for this paging group.

2-way Intercom Configuration Parameters

1-way Paging

Create New Paging/Intercom Groups

• Name:

• Type:

• Extension:

Auto Record:

Delayed Paging:

Replace Display Name:

• Maximum Call Duration (s):

Custom Prompt: [Upload Audio File](#)

Members:

7 items Available

Search

- 1000
- 1001
- 1002
- 1003
- 1004

0 item Selected

Search

None

Paging/Intercom Whitelist:

7 items Available

Search

- 1000
- 1001
- 1002
- 1003
- 1004

0 item Selected

Search

None

1-way Paging

Name	Configure paging/intercom group name.
Type	Select "1-way Paging".
Extension	Configure the paging/intercom group extension.
Video Broadcast	If checked, video paging will be supported. If the caller sends a video page, the paging group members will be able to receive and view the video.
Auto Record	Enable this option to record in WAV format.
Delayed Paging	Allows the announcement to be played after the configured delay paging. If there are many messages, they will be played in sequence.
LaggedTime(seconds)	Set the delay paging duration, the default is 5 seconds.
Replace Display Name	If enabled, the UCM will replace the caller display name with Paging/Intercom name.
Maximum Call Duration	Specify the maximum call duration in seconds. The default value 0 means no limit.
Custom Prompt	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. Note: Users can also refer to the page PBX Settings→Voice Prompt→Custom Prompt , where they could record new prompt or upload prompt files.
Members	Select available users from the left side to the paging/intercom group member list on the right.
Paging/Intercom Whitelist	Select which extensions are allowed to use the paging/intercom feature for this paging group.

In case the user wants to broadcast a video, these requirements should be respected.

- H.264 video encoding
- .mkv or .tar/.tgz/tar.gz format
- MKV files must be 30 MB file or less
- Compressed files (.tar/.tgz/tar.gz) must be 50 MB or less.
- File name can only contain alphanumeric characters, hyphens (-) and period (.)

If Auto Record is enabled, recorded video pages will be saved in MKV file format. Saved recordings can be found in the *CDR* → *Recordings* → *Video Recordings* page.

Multicast Paging

* Name:

* Type:

* Extension:

Delayed Paging:

Delay (s):

* Maximum Call Duration (s):

Custom Prompt: [Upload Audio File](#)

* Multicast IP Address:

* Port:

Multicast Paging

Paging/Intercom	Select existing paging/intercom groups and multicast communities.
Name	Enter the name of the scheduled Intercom/Paging.
Caller	Once a caller is selected, and the specified start time is reached, the system will contact the caller. If this call is rejected, the page/intercom will be cancelled. If caller is set to None, the system will call all group members and play the configured prompt.
Start Date	Select the date of the start of the paging/intercom.
Start Time	Select the start time of the paging/intercom.
Repeat	Select the repeat interval of the paging/intercom. <ul style="list-style-type: none"> • No Repeat: The intercom/paging will play once on the scheduled date and time. • Everyday: The intercom/paging will play daily starting from the scheduled day and on the time scheduled every day. • Weekly: The intercom/paging will play weekly on the selected day(s) of the week. • Monthly: The intercom/paging will play monthly on the selected date of the month.
Include Holidays	When Every day, Weekly, or Monthly is selected. This option will appear. If enabled, scheduled pages/intercoms will run during holidays. Otherwise, scheduled pages/intercoms displayed on the calendar will not actually be run.
Sync to Google Calendar	This feature cannot be used if Google Services have not been authorized. Please resolve this in the Integrations > Google Services page.

Multicast Community

Multicast community allows creating an extension, which when dialed, can send a preconfigured prompt as a multicast paging to a group of extensions. The user should create first a **Paging/Intercom Group** with **Multicast Paging** as the **Strategy** selected. Please see previous section for more information.

Paging/Intercom > Create New Multicast Community

* Name

* Extension

Delayed Paging

* Maximum Call Duration (s)

Custom Prompt [Upload Audio File](#)

Play Prompt to Caller

* Prompt Playback Count

* Multicast Paging Group

0 Available

Search

None

0 Selected

Search

None

Selected All

Multicast Community Parameters

Paging/Intercom	Select existing paging/intercom groups and multicast communities.
Name	Enter the name of the scheduled Intercom/Paging.
Caller	Once a caller is selected, and the specified start time is reached, the system will contact the caller. If this call is rejected, the page/intercom will be cancelled. If caller is set to None, the system will call all group members and play the configured prompt.
Start Date	Select the date of the start of the paging/intercom.
Start Time	Select the start time of the paging/intercom.
Repeat	<p>Select the repeat interval of the paging/intercom.</p> <ul style="list-style-type: none"> • No Repeat: The intercom/paging will play once on the scheduled date and time. • Everyday: The intercom/paging will play daily starting from the scheduled day and on the time scheduled every day. • Weekly: The intercom/paging will play weekly on the selected day(s) of the week. • Monthly: The intercom/paging will play monthly on the selected date of the month.
Include Holidays	<p>When Every day, Weekly, or Monthly is selected. This option will appear.</p> <p>If enabled, scheduled pages/intercoms will run during holidays. Otherwise, scheduled pages/intercoms displayed on the calendar will not actually be run.</p>

Sync to Google Calendar	This feature cannot be used if Google Services have not been authorized. Please resolve this in the Integrations > Google Services page.
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Scheduled Paging/Intercom

Pending Paging/Intercom

In this page, the user can create scheduled intercom/paging to be played automatically when the time scheduled arrives.

Paging/Intercom > Create New Scheduled Paging/Intercom

* Paging/Intercom

* Name

* Caller

* Start Date

* Start Time -

[Add Start Time](#) +

Repeat

Sync to Google Calendar [Google Services](#)

Paging/Intercom	Select existing paging/intercom groups and multicast communities.
Name	Enter the name of the scheduled Intercom/Paging.
Caller	Once a caller is selected, and the specified start time is reached, the system will contact the caller. If this call is rejected, the page/intercom will be cancelled. If caller is set to None, the system will call all group members and play the configured prompt.
Start Date	Select the date of the start of the paging/intercom.
Start Time	Select the start time of the paging/intercom.
Repeat	<p>Select the repeat interval of the paging/intercom.</p> <ul style="list-style-type: none"> ● No Repeat: The intercom/paging will play once on the scheduled date and time. ● Everyday: The intercom/paging will play daily starting from the scheduled day and on the time scheduled every day. ● Weekly: The intercom/paging will play weekly on the selected day(s) of the week. ● Monthly: The intercom/paging will play monthly on the selected date of the month.
Include Holidays	<p>When Every day, Weekly, or Monthly is selected. This option will appear.</p> <p>If enabled, scheduled pages/intercoms will run during holidays. Otherwise, scheduled pages/intercoms displayed on the calendar will not actually be run.</p>

Sync to Google Calendar

This feature cannot be used if Google Services have not been authorized. Please resolve this in the [Integrations > Google Services](#) page.

Once the paging and intercom has been created, it can be viewed on the same page.

Name	Caller	Paging/Intercom Extension	Paging/Intercom Name	Strategy	Start Time	Repeat	Options
Multicast_Group	1000	1077	Multicast_Group	Multicast Paging	2024-03-07 15:00	No Repeat 15:00	
Paging	1003	1055	Paging	1-way Paging	2024-03-07 18:00	No Repeat 18:00	
Multicast_Group	1006	1077	Multicast_Group	Multicast Paging	2024-03-20 11:00	Wed, Tue, Thu, Fri, Mon 11:00	

Paging/Intercom Schedule

This section displays the schedule of the paging/intercom which have been scheduled. The user can choose to display per day, week, or per month.

Paging/Intercom Groups

Paging/Intercom Groups Multicast Community [Scheduled Paging/Intercom](#)

Pending Paging/Intercom **Paging/Intercom Schedule**

2024 Mar Today < > Mar 1 - 31, 2024 Day Week **Month**

Su	Mo	Tu	We	Th	Fr	Sa
25	26	27	28	29	01	02
03	04	05	06	07	08	09
10	11	12	13	14	15	16
17	18	19	20	21	22	23
24	25	26	27	28	29	30
31	01	02	03	04	05	06

Calendar events:

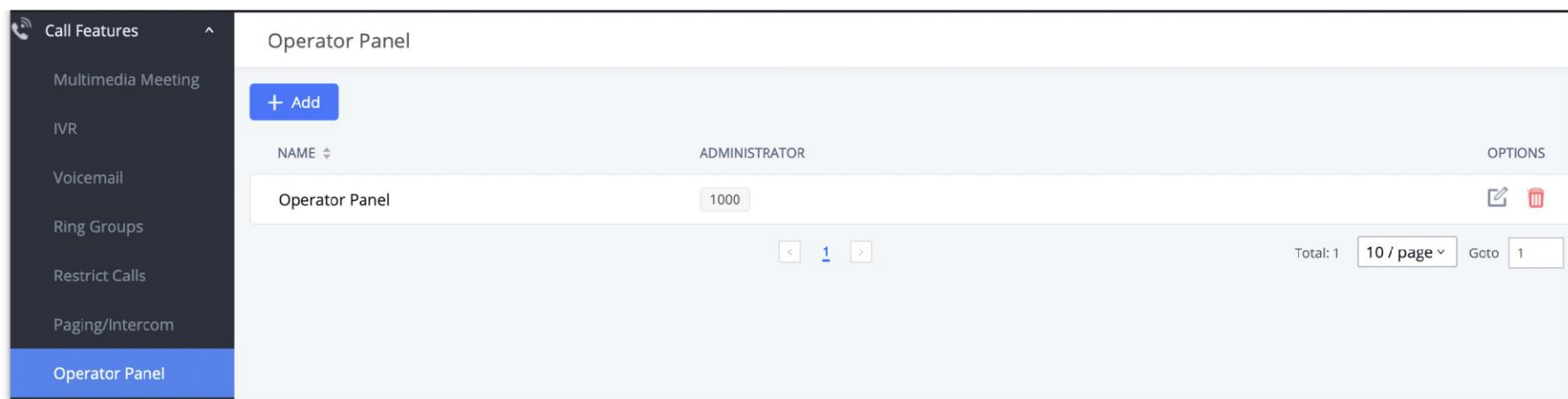
- 03:00 Multicast_C
- 05:00 Paging
- 11:00 Multicast_C
- 11:00 Multicast_C
- 11:00 Multicast_C

OPERATOR PANEL



Configure Operator Panel

Operator Panel settings can be accessed via Web GUI → **Call Features** → **Operator Panel**.

The UCM630xA supports operator panel so that UCM extension can be used as admin to manage calls and activities such as extension status, call queue status, transfer, barge-in, hangup and etc. On Grandstream Wave client, it can display the extensions, ring group, voicemail, call queue, call park status under the management of the extension. This section describes how to configure operator panel.



Operator Panel Configuration Page

- Click on “Add” to create operator panel.
- Click on  to edit the operator panel.
- Click on  to delete the operator panel.

Name	Configure the name for the operator panel created for identification purpose.
Administrator	Assign the administrator for the operator panel. It can be an extension, a extension group or a department. For the selected extension groups and departments, subsequent extensions will automatically become administrators.
Management Module	
Extension	The selected extensions will be supervised by the administrator, and you can choose extensions, extension groups, and departments. For the selected extension groups and departments, subsequent extensions will be automatically supervised by the administrator.
Ring Groups	The selected Ring Groups will be supervised by the administrator. If selecting “All”, all Ring Groups and subsequent updates will be automatically supervised by the administrator.
Voicemail Groups	The selected Voicemail Groups will be supervised by the administrator. If selecting “All”, all Voicemail Groups and subsequent updates will be automatically supervised by the administrator.
Call Queue	The selected Call Queue will be supervised by the administrator. If selecting “All”, all Call Queue and subsequent updates will be automatically supervised by the administrator.
Parking Lot	The selected Parking Lot will be supervised by the administrator. If selecting “All”, all Parking Lot and subsequent updates will be automatically supervised by the administrator.

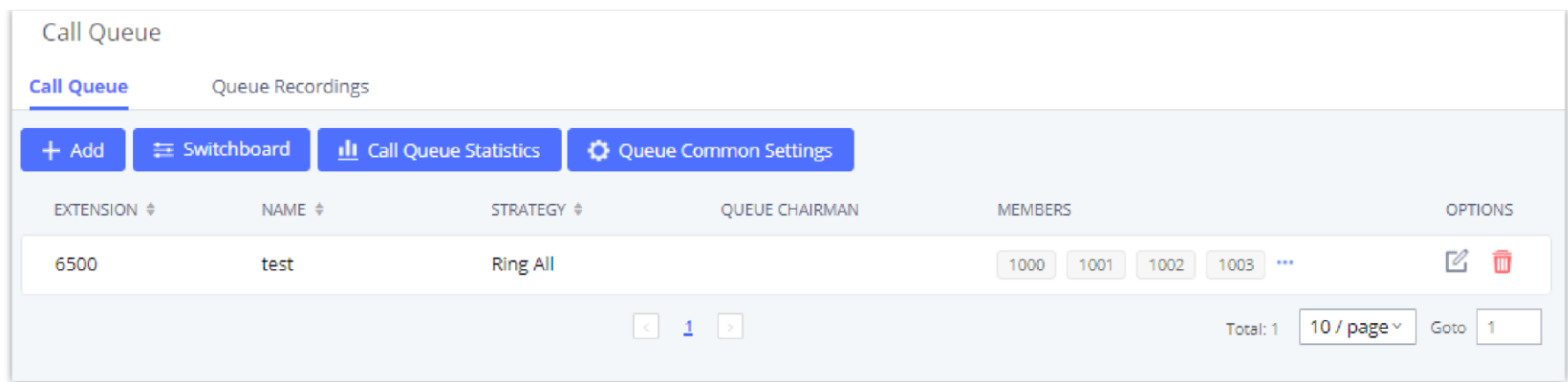
Operator Panel Settings

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**.



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.

Basic Settings	
General	
Extension	Configure the call queue extension number.
Name	Configure the call queue name to identify the call queue.
Strategy	<p>Select the strategy for the call queue.</p> <ul style="list-style-type: none"> ● Ring All: Ring all available Agents simultaneously until one answers. ● Linear: Ring agents in the specified order. ● Least Recent: Ring the agent who has been called the least recently. ● Fewest Calls: Ring the agent with the fewest completed calls. ● Random: Ring a random agent. ● Round Robin: Ring the agents in Round Robin scheduling with memory. <p>The default setting is "Ring All".</p>
Music On Hold	<p>Select the Music On Hold class for the call queue.</p> <p>Note: Music On Hold classes can be managed from Web GUI→PBX Settings→Music On Hold.</p>
Max Queue Length	Configure the maximum number of calls to be queued at once. This number does not include calls that have been connected with agents, only calls that are still in queue. When this maximum value is exceeded, the caller will hear a busy tone and be forwarded to the configured failover destination. Default value is 0 (unlimited).
Agent Rest Time (s)	Configure the amount of time in seconds after ending a call where the agent will not receive additional calls. Once this time has passed, the agent will be able to receive calls again. If set to 0, agents can receive additional calls immediately after ending a call.
Retry Time (s)	Configure the number of seconds to wait before ringing the next agent. The minimum is 1.
Agent Ring Time	Configure the number of seconds to ring an agent. The minimum is 5.

Auto Record	If enabled, the calls on the call queue will be automatically recorded. The recording files can be accessed in Queue Recordings under Web GUI→Call Features→Call Queue .
Welcome Prompt	
Enable	Enable the welcome prompt.
Custom Prompt	Initial tone that plays when the user dials the queue number. Note: The user can upload a custom prompt directly from this parameter.
Play Full Welcome Prompt	If enabled, queue agents will not be rung until after the welcome prompt is done playing. Otherwise, queue agents will be rung while the playing the welcome prompt.
Satisfaction Survey Prompt	
Custom Prompt	After a queue agent hangs up a call, a prompt will play asking the caller to rate their satisfaction on a scale of 1 to 5, with 5 being the highest. Note: The user can upload a custom prompt directly from this parameter.
Max Wait Time	
Max Wait Time	Configures the amount of time a caller will be kept in queue before the the call is automatically routed to the configured Max Wait Time Destination. If set to 0, callers will be kept in queue indefinitely.
Destination	The call will be routed to this destination if no one in this queue answers the call.
Destination Prompt Cycle	
Enable	Enable Destination Prompt Cycle
Destination Prompt Cycle	Configure the voice prompt cycle (in seconds) of this call queue. When playing the voice prompt, you can press 1 to transfer to failover destination.
Custom Prompt	When playing a custom prompt, press 1 to enter the failover destination or continue waiting in queue. Note: The user can upload a custom prompt directly from this parameter.
Destination	After the specified amount of time, the caller will be prompted to press 1 to immediately get redirected to the configured failover destination.
Advanced Settings	
Virtual Queue	
Enable Virtual Queue	If enabled, system will enable a virtual queue for users waiting in queue.
Virtual Queue Mode	When in DTMF mode, pressing 2 will manually trigger virtual queue. When in Timeout mode, virtual queue will automatically be triggered when the configured Virtual Queue Period has passed. DTMF mode and Timeout mode require the caller to manually set a callback number. When in Auto mode, virtual queue will automatically be triggered when the configured Virtual Queue Period has passed. The callback number will automatically be set to the caller's detected CID number.
Virtual Queue Period (s)	The amount of time in seconds that must pass before virtual queue is offered to callers when using Timeout mode or Auto mode.

Virtual Queue Outbound Prefix	System will add this prefix to dialed numbers when calling back users.
Enable Virtual Queue Position Announcement	If enabled, the system will inform callers waiting in the queue of their positions in line.
Enable Virtual Queue Wait Time Announcement	If enabled, the estimated wait time for the call to get answered will periodically be announced to the caller.
Enable Virtual Queue Callback Timeout	If enabled, agents will have a set amount of time to answer a virtual queue callback.
Virtual Queue Welcome Prompt	Upload the file of your welcome prompt of the virtual queue.
Announcement Settings	
Enable Position Announcement	If enabled, the system will inform callers waiting in the queue of their positions in line.
Enable Wait Time Announcement	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.
Announcement Interval	The interval at which caller positions and estimated wait times will be announced.
Agent ID Announcement	If enabled, a system prompt containing the agent ID will be played to the caller when answered by an agent.
Empty Queue	
Leave When Empty	<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"> ● Yes: Callers will be disconnected from the queue if all agents are paused or invalid. ● No: Never disconnect the callers from the queue when the queue is empty. ● Strict: Callers will be disconnected from the queue if all agents are paused, invalid or unavailable.
Dial in Empty Queue	<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"> ● Yes: Callers can always dial into a call queue. ● No: Callers cannot dial into a queue if all agents are paused or invalid. ● Strict: Callers cannot dial into a queue if the agents are paused, invalid or unavailable.
Failover Destination	<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"> ● Play Sound. ● Extension. ● Voicemail. ● Queues. ● Ring Group. ● Voicemail Group. ● IVR ● External Number.
CTI	

Enable Agent Login	Enabling agent login will cause the dynamic agents to be unavailable.
Queue Chairman	The queue chairman can log into his web portal to operate the queue.
Service Level Agreement (SLA)	
Enable SLA	Toggles Service Level Agreement (SLA), which is percentage measurement of the queue group's ability to answer incoming calls within a defined amount of time. If a queue group's calculated SLA percentage is below the configured threshold value, alerts will be generated and sent out via email to the specified recipients. Example: The SLA goal is 80% of calls (Threshold) within 20 seconds (SLA Time). If less than 80% of queue calls are answered within 20 seconds, the specified users will be notified of it.
SLA Time (s)	Configures the amount of time in seconds that agents must answer incoming queue calls within to satisfy service quality requirements. Answering calls past this time will negatively affect the SLA measurement, and an alert will be generated once it hits below the specified SLA alert threshold. Supported values are 1 to 180. Default value is 20.
SLA Alert Email Notification	Enable SLA alert email notification.
Alert Threshold (%)	Configures the SLA alert threshold. If the percentage of queue calls answered within the configured SLA Time go below this value, an alert email will be generated and sent to the configured recipients. Supported values are 1 to 100. Default value is 80.
SLA Alert Interval (m)	Configures the minimum amount of time (in minutes) between alert sending. If a new alert is generated within this period, it will not be sent to recipients until the next alert interval. The valid range is from 1 to 120. The default value is 120.
SLA Alert Email Template	The template of the SLA alert email notifications.
Alert Email Recipients	Send SLA alert notifications to the configured alert email recipients. If a recipient does not have an email address configured, they will not receive the alert notifications.
Other Settings	
Report Hold Time	If enabled, the UCM630X 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".
Replace Display Name	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.
Enable Feature Codes	Enable feature codes option for call queue. For example, *83 is used for "Agent Pause"
Autofill	Configure to enable autofill.
Dynamic Login Password	If enabled, the configured PIN number is required for dynamic agent to log in. The default setting is disabled.
Alert-Info	When present in an INVITE request, the Alert-info header field specifies an alternative ring tone to the UAS.
Agents	
Static Agents	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 ◀ or ▶ to choose.

And use UP and Down arrow to select the order of the agent within the call queue.

Agents limitation:

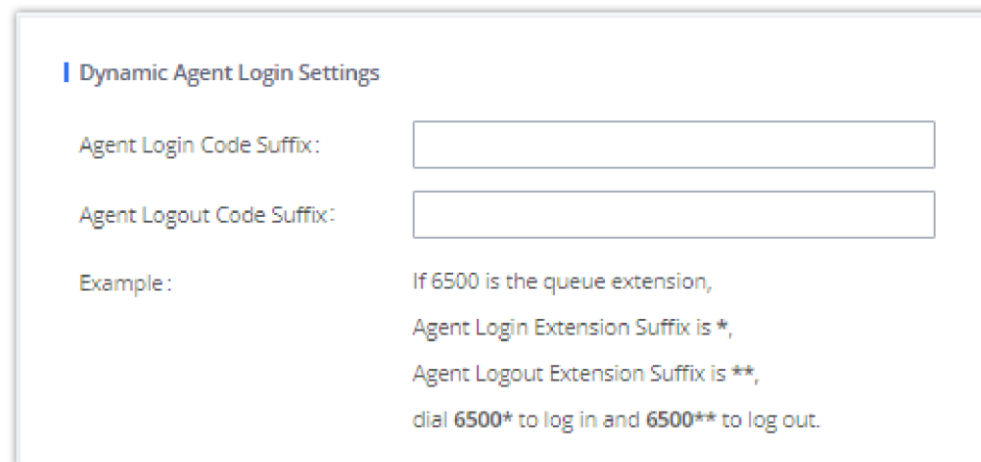
The maximum number of agents which can be assigned varies depending on the model of the UCM6300A used. This limit includes the static and the dynamic agents.

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

UCM Model	Maximum Number of Agents in Call Queue
UCM6300A	25
UCM6302A	50
UCM6304A	80
UCM6308A	160

Table 88: Static Agent Limitation

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.





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.

The agent can also dial the feature with the number of the reason of the pause. E.g., if the agent want to perform a pause for lunch, he/she can dial *831 directly instead of waiting for the IVR response.

- 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 one 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:

Enable Virtual Queue	Enable virtual queue to activate call center features.
Virtual Queue Period	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.
Virtual Queue Mode	<p>Offered to caller after timeout: 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"> ○ Press * to set current number as callback number. ○ Press 0 to set a callback number different than current caller number. ○ Press # to keep waiting on the call queue. <p>Triggered on user request: 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"> ○ Press * to set current number as callback number. ○ Press 0 to set a callback number different than current caller number. ○ Press # to keep waiting on the call queue.
Virtual Queue Outbound Prefix	System will add this prefix to dialed numbers when calling back users.
Enable Virtual Queue Timeout	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.
Write Timeout	Configure the virtual queue callback timeout period in seconds.
Enable Virtual Queue Position Announcement	<p>Enable the announcement of the caller's position periodically.</p> <p>Note: Queue position will now be announced to the caller upon entering the queue.</p>
Position Announcement Interval	Configure the period of time in (s) during which the UCM will announce the caller's position in the call queue.
Enable Virtual Queue Wait Time Announcement	When enabled the UCM will announce the estimated queue wait time to callers if the estimated wait time is longer than 1 minute.
Queue Chairman	Select the extension to act as chairman of the queue (monitoring).
Virtual Queue Welcome Prompt	Click on "Upload Audio File" to upload the VQ welcome prompt.
Enable Agent Login	<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>Notes:</p> <ul style="list-style-type: none"> ○ This feature is currently available only for GXP21xx phones on firmware 1.0.9.18 or greater. ○ After enabling the feature, users need to set the option on GXP21XX phone under "Account→SIP Settings→Advanced Features→Special Feature" to "UCM Call Center". A softkey labeled "UCM-CC" will appear on the bottom of the phone's screen. ○ When this option is enabled, dynamic agent login will be no longer supported. ○ In case of concurrent registrations, changing agent status on one phone (login/logout) will be reflected on all phones.

Queue Auto fill enhancement:

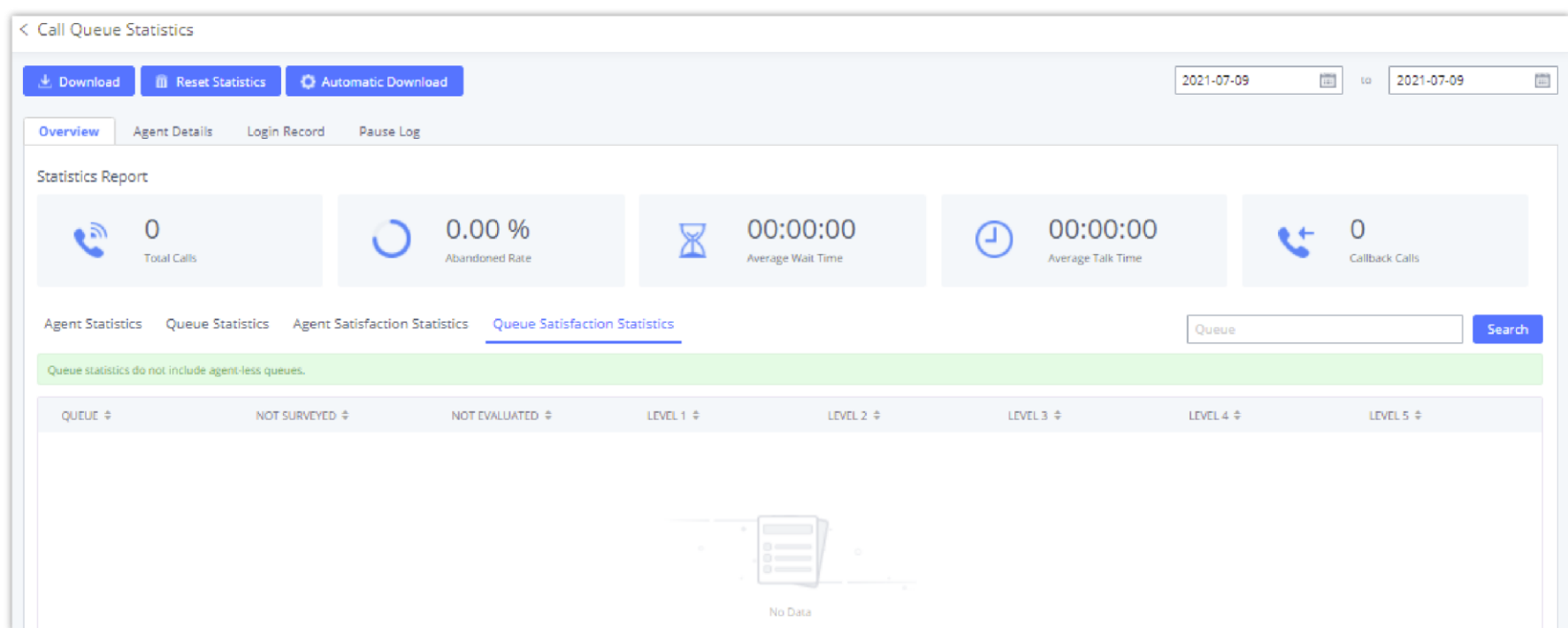
The waiting callers are connecting with available members in a parallel fashion until there are no more available members or no more waiting callers.

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:



Call Queue Statistics

- **Agent statistics:** shows the number of calls and call-related information of agents;
- **Queue Statistics:** counts the number of calls in the queue and information such as calls, waiting, and callback;
- Agent satisfaction statistics used for user’s rating of agents;
- Queue satisfaction statistics counts the score survey statistics.

The overview page performs seat statistics, queue statistics, seat satisfaction statistics, and queue satisfaction statistics according to the business. Agent statistics record the number of calls and call-related information of agents; queue counts the number of calls in the queue and information such as calls, waiting, and callback; agent satisfaction statistics are survey statistics based on user ratings of agents; queue satisfaction statistics are user-queue the score survey 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:

Details				
Queue:	6500	Total Calls:	3	
Answered Calls:	1	Abandoned Calls:	2	
Answered Rate:	33.33 %	Average Wait Time:	00:00:03	
Total Calls Answered Calls Abandoned Calls				
DATE	CALLER ID	ABANDONED	WAIT TIME	TALK TIME
2020-12-02 09:38:09	1005	Yes	00:00:00	00:00:00
2020-12-02 09:38:48	1005	Yes	00:00:00	00:00:00
2020-12-02 09:39:01	1005	No	00:00:03	00:00:04

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](#)

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

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

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.

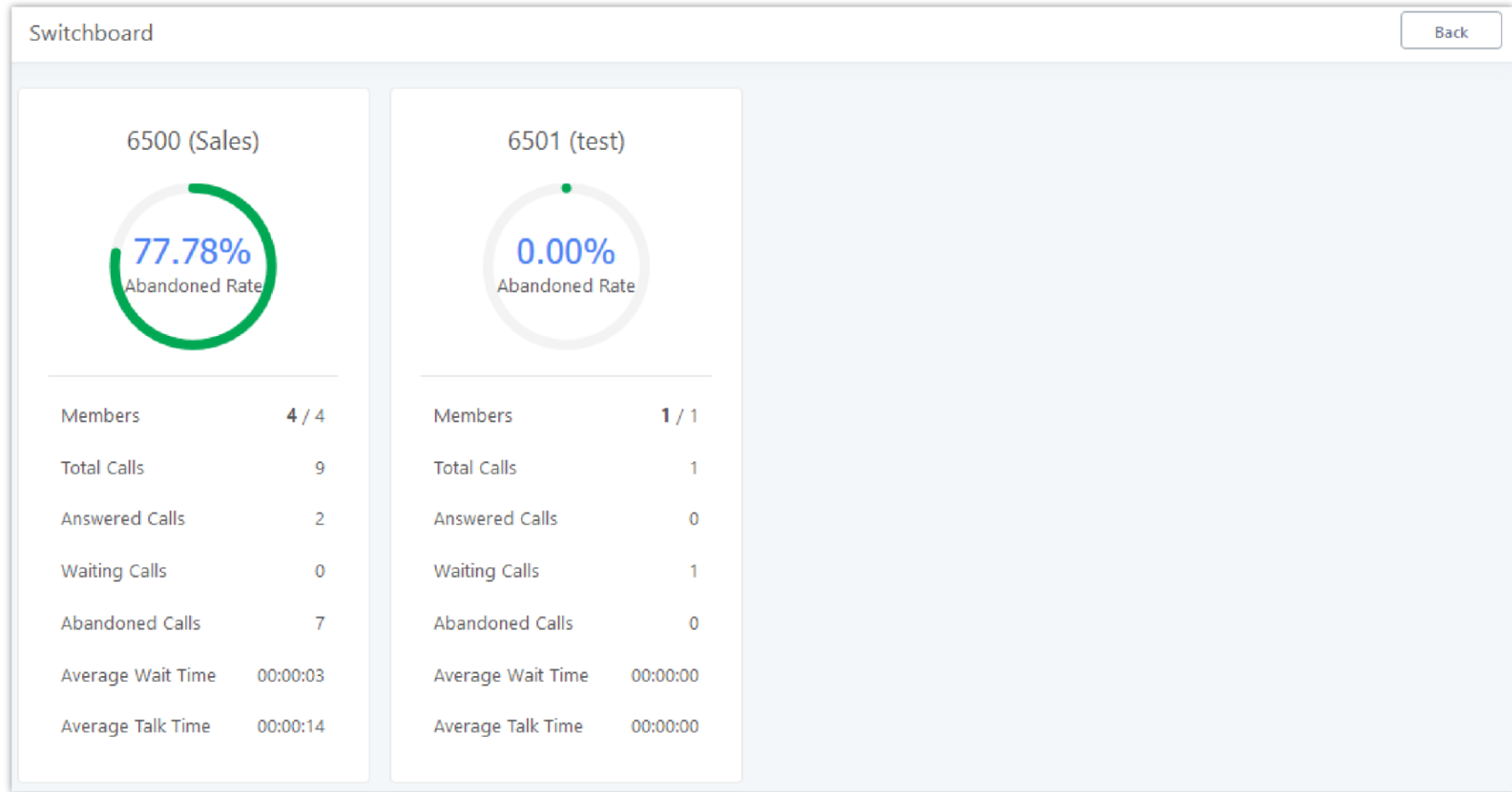
AGENT	QUEUE	PAUSE TIME	RESUME TIME	PAUSE DURATION
2000	6500	2019-11-08 11:32:00	2019-11-08 11:33:33	00:01:33
2000	6500	2019-11-08 11:32:00	2019-11-08 11:33:33	00:01:33

Pause Log

Switchboard

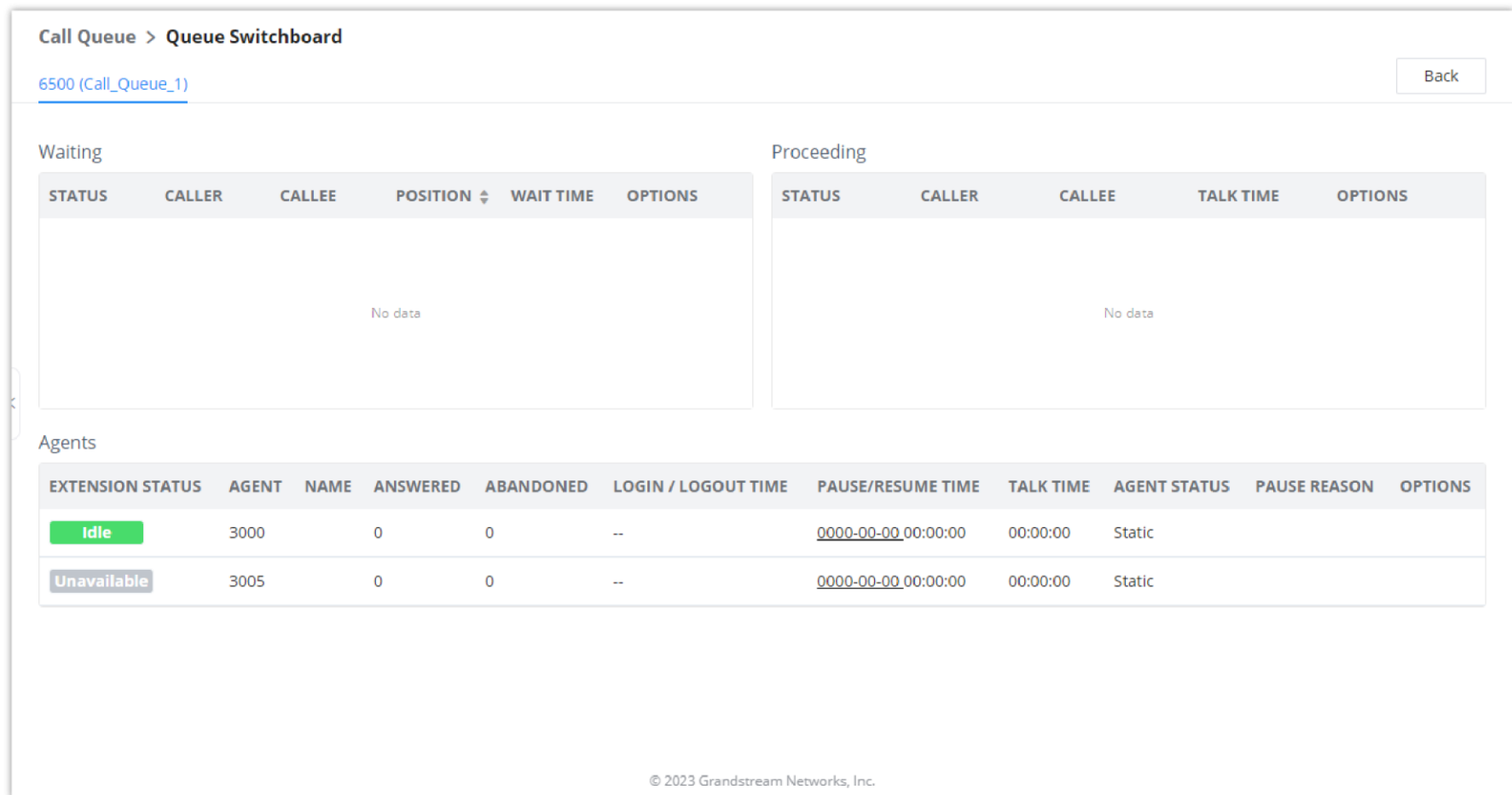
Switchboard is a Web GUI tool for call queue monitoring and management, admin can access to it from the menu **Call Features**→**Call Queue** then press "Switchboard".

Following page will be displayed:




Switchboard Summary

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



Call Queue Switchboard

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

Waiting	This menu shows the current waiting calls along with the caller id and the option to hang-up call by pressing on the  button.
Proceeding	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.

Agents	<p>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).</p> <p>Note: 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.</p>
---------------	--

Switchboard Parameters

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 "Other 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 Chariman is also able to log out dynamic agents from call queues.

Call Queue > Queue Switchboard

6500 (Call_Queue_1) Back

Waiting

STATUS	CALLER	CALLEE	POSITION	WAIT TIME	OPTIONS
No data					

Proceeding

STATUS	CALLER	CALLEE	TALK TIME	OPTIONS
No data				

Agents

EXTENSION STATUS	AGENT	NAME	ANSWERED	ABANDONED	LOGIN / LOGOUT TIME	PAUSE/RESUME TIME	TALK TIME	AGENT STATUS	PAUSE REASON	OPTIONS
Idle	3000		0	0	--	0000-00-00_00:00:00	00:00:00	Static		
Unavailable	3005		0	0	--	0000-00-00_00:00:00	00:00:00	Static		

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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 "Other 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.

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
Ringling	1001	1	1	--	--	00:00:04	Static

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:

Global Queue Settings

Dynamic Agent Login Settings	
Agent Login Code Suffix	<p>Configure the code to dial after the queue extension to log into the queue (i.e. queue extension + suffix).</p> <p>If no suffix is configured, dynamic agents will not be able to log in</p>
Agent Logout Code Suffix	<p>Configure the code to dial after the queue extension to log out of the queue (i.e. queue extension + suffix).</p> <p>If no suffix is configured, dynamic agents will not be able to log out.</p>

Virtual Queue Callback Key Settings	
Enable	Select whether to enable or disable virtual queue callback feature. By default it's disabled.
Call Back Current Number	Press the feature key configured to set your current number as callback number.
Custom Callback Number	Press these feature key configured to set a custom callback number.
Continue Waiting	Press the feature key configured to continue waiting.

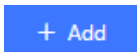


Global Queue Settings

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).

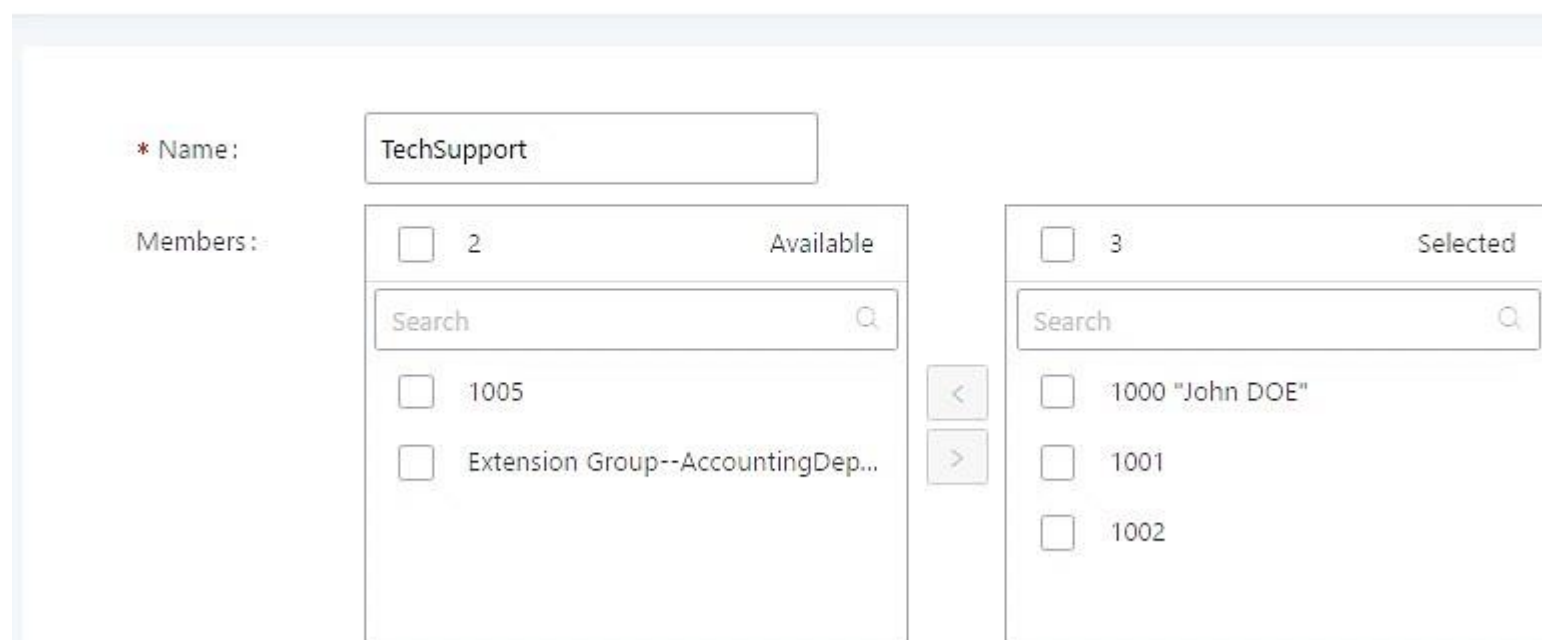
Configure Pickup Groups

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.

Pickup Groups



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"/>			

[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.

Manage Music On Hold

+ Add Download All Music On Hold

Music on Hold Playlists:

Record Delete Upload

<input type="checkbox"/> DISABLED/ENABLED	SOUND FILE	OPTIONS
<input checked="" type="checkbox"/> ON	macroform-cold_day.wav	
<input checked="" type="checkbox"/> ON	macroform-robot_dity.wav	
<input checked="" type="checkbox"/> ON	macroform-the_simplicity.wav	
<input checked="" type="checkbox"/> ON	manolo_camp-morning_coffee.wav	
<input checked="" type="checkbox"/> ON	reno_project-system.wav	

Total: 5 10 / page Goto: 1

[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



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 Web GUI 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:

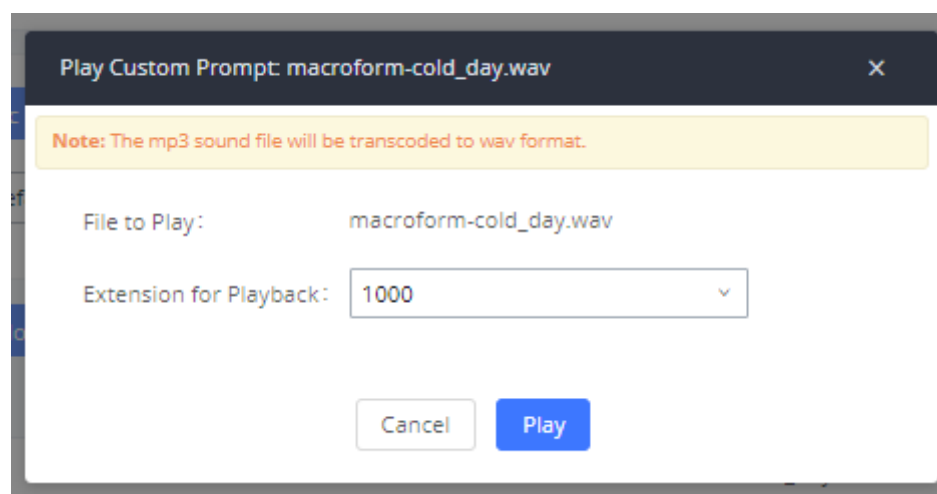
1. Click on the



button for the Music On Hold file.

2. In the prompted window, select the extension to playback and click





Play Custom Prompt

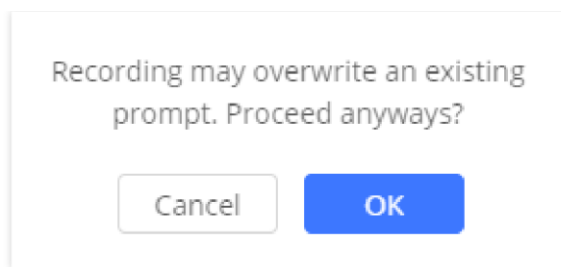
3. The selected extension will ring.
4. Answer the call to listen to the music playback.

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.



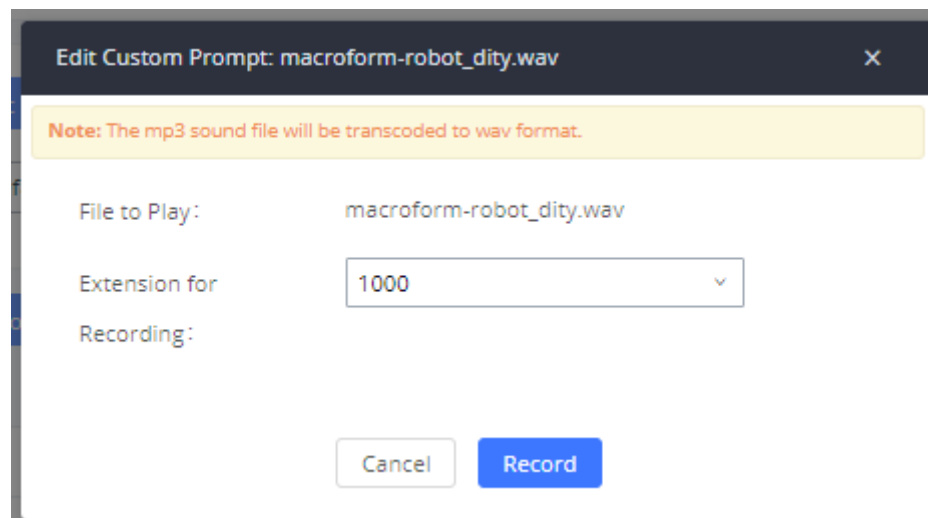
Information Prompt

3. Click

OK

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

Record



Record Custom Prompt

5. Answer the call and start to record your new music on hold.

6. Hang up the call and refresh Music On Hold page then you can listen to the new recorded file.

Note

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://documentation.grandstream.com/knowledge-base/busy-camp-on-2/>

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.

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:

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

SIP Presence Status

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"/>

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**.

STATUS	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	TALK TIME	ACCOUNT CODE	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		-

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




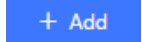
EXTENSION	OPTIONS
1002 for 30 (seconds)	<input type="button" value="X"/> <input type="button" value="↺"/> <input type="button" value="↻"/> <input type="button" value="↷"/>

Edit Follow Me

1. Click on to add local extensions or external numbers to be called after ringing the extension selected in the first step.
2. 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:

Enable	Configure to enable or disable Follow Me for this user.
Skip Trunk Auth	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.
Music On Hold Class	Configure the Music On Hold class that the caller would hear while tracking users
Confirm When Answering	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.
Enable Destination	When enabled, the call will be routed to the default destination if no one in the Follow Me extensions answers the call.
Default Destination	Configure the destination if no one in the Follow Me extensions answers the call. The available options are: <ul style="list-style-type: none"> ○ Extension ○ Voicemail ○ Queues ○ Ring Group ○ Voicemail Group ○ IVR ○ External Number

Follow Me Numbers	The added numbers are listed here. Click on  to arrange the order. Click on  to delete the number. Click on  to delete the number. Click on  to add new numbers.
New Follow Me Number	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.
Dialing Order	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.

Follow Me Settings

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

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

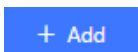
Follow Me Options

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.

Create New Speed Dial

Enable

Destination:

* Speed Dial

Extension:

Default

Destination:

Speed Dial Destinations

The following can be configured as the default destination of a speed dial:

- o **Extension:** Choose the extension to call when dialing the speed dial number.

- **Multimedia Meeting:** Choose the meeting room extension to call when dialing the speed dial number.
- **Voicemail:** Choose the extension which voicemail will be accessed to when dialing the speed dial number.
- **Voicemail Group:** Choose the voicemail group which will be accessed when dialing the speed dial number.
- **IVR:** Choose the IVR which will be accessed when dialing the speed dial number.
- **Ring Group:** Choose the ring group which will be called when dialing the speed dial number
- **Queues:** Choose the queue which will be called when dialing the speed dial number.
- **Paging/Intercom:** Choose the paging/intercom group which will be initiated when the speed dial number is dialed.
- **Fax:** Select the fax extension which will be used as a destination when the speed dial number is dialed.
- **DISA:** Choose DISA as the destination when the speed dial number is dialed.
- **Dial By Name:** Choose Dial By Name as the dialing destination.
- **Announcement:** Choose an announcement as the destination when the speed dial number is dialed.
- **External Number:** Enter an external number to call when the speed dial number is dialed.
- **Custom Number:** Choose a custom number to call when the speed dial number is dialed. This can also be a feature code.

Speed Dial

+ Add

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	

Total: 5 < 1 > 10 / page Goto 1

List of Speed Dial

Import Speed Dial

The user can import speed dial entries from a csv file, this reduces the amount of configuring the same speed dial entries on different UCMs. To do this, please click on "**Import**" as the figure below shows.

Speed Dial

+ Add **Import** Export

SPEED DIAL EXTENSION	SPEED DIAL	DEFAULT DESTINATION	DEFAULT DESTINATION	OPTIONS
123	Enable	Extension	1000	
999	Enable	Voicemail		

< 1 > Total: 2 10 / page Goto 1

Import Speed Dial List

Then select the csv file of the speed dial entries and click **Upload**

Important

Please use UTF-8 encoding when importing a CSV file. CSV files can be opened using programs such as Notepad and saved as a UTF-8 encoded file.

Alert

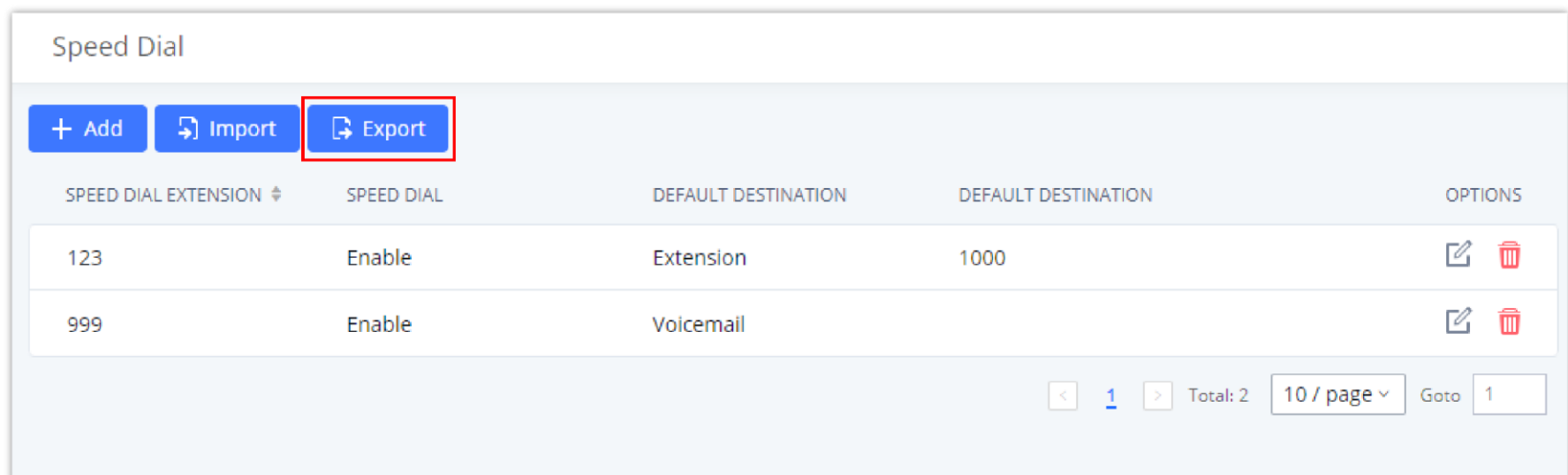
Importing speed dial entries will overwrite the existing speed dials, if you wish to import new speed dial entries to the already existing ones, you will have to export them then combine them together in one file before you import it.





Note

The number of speed dial entries is limited to 1000.

Export Speed Dial

To export speed dial entries, please click on export as the screenshot below shows, then choose the location where to save the csv file.



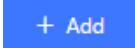


SPEED DIAL EXTENSION	SPEED DIAL	DEFAULT DESTINATION	DEFAULT DESTINATION	OPTIONS
123	Enable	Extension	1000	 
999	Enable	Voicemail		 

Export Speed Dial List

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...:

Create New DISA

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

Name	Configure DISA name to identify the DISA.
Password	<p>Configure the password (digit only) required for the user to enter before using DISA to dial out.</p> <p>Note:</p> <p>The password must be at least 4 digits.</p>
Permission	<p>Configure the permission level for DISA.</p> <p>The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level.</p> <p>The default setting is "Internal".</p> <p>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.</p>
Response Timeout	<p>Configure the maximum amount of time the UCM630xA will wait before hanging up if the user dials an incomplete or invalid number.</p> <p>The default setting is 10 seconds.</p>
Digit Timeout	<p>Configure the maximum amount of time permitted between digits when the user is typing the extension.</p> <p>The default setting is 5 seconds.</p>
Allow Hangup	<p>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".</p>
Replace Display Name	<p>If enabled, the UCM will replace the caller display name with the DISA name.</p>

DISA Settings

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://documentation.grandstream.com/knowledge-base/emergency-calls/>

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**"

Emergency Calls

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:

11 items Available

Search

- 1001 "John Doe"
- 1002
- 1003
- 1004

1 item Selected

Search

- 1000 "James tuan"

Strip:

Prepend:

Auto Record:

Send Recording File:

Email Address: [+](#)

Emergency Number Configuration

The table below gives more description of the configuration Parameters when creating emergency numbers.

Name	Configure the name of the emergency call. For example, "emergency911","emergency211" and etc.
Emergency Number	Config the emergency service number. For example,"911","211" and etc.
Emergency Level	Select the emergency level of the number. Level "3" means the most urgent.
Disable Hunt on Busy	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.
Custom Prompt	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.
Use Trunks	Select the trunks for the emergency call. Select one trunk at least and select five trunks at most.
Members Notified	Select the members who will be notified when an emergency call occurs.
Strip	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.
Prepend	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.

Auto Record	When enabled, emergency call will be automatically recorded.
Send Recording File	When enabled recording files will be sent to the configured email address.
Email Address	The email address to where the recording files will be sent.

Emergency Numbers Parameters

The screenshot shows the 'Emergency Calls' configuration page. It has three tabs: 'Emergency Calls', 'Emergency Recordings', and 'Emergency Location Mapping'. The 'Emergency Calls' tab is active. There is a '+ Add' button. Below it is a table with the following columns: NAME, EMERGENCY NUMBER, EMERGENCY LEVEL, DISABLE HUNT ON BUSY, and OPTIONS. The table contains one row with the value '911' in the NAME and EMERGENCY NUMBER columns, '1' in the EMERGENCY LEVEL column, and 'No' in the DISABLE HUNT ON BUSY column. The OPTIONS column contains edit and delete icons. At the bottom right, it shows 'Total: 1', '10 / page', and 'Goto 1'.

911 Emergency Sample

Emergency Recordings

UCM6300 Series allows recording emergency calls and they can be found under WebUI → **Call Feature** → **Emergency Calls** → **Emergency Recordings**

The screenshot shows the 'Emergency Recordings' configuration page. It has three tabs: 'Emergency Calls', 'Emergency Recordings', and 'Emergency Location Mapping'. The 'Emergency Recordings' tab is active. There are buttons for 'Download', 'Download All', 'Delete', and 'Clear'. There is a dropdown menu set to 'Local' and a date field set to '2022-06'. Below these is a table with columns: NAME, EMERGENCY CALLS, CALLER NUMBER, DATE, SIZE, and OPTIONS. The table is empty and displays 'No Data'. At the bottom, there is a copyright notice: 'Copyright © Grandstream Networks, Inc. 2022. All Rights Reserved.'

Emergency Recordings

Emergency Location Mapping

In compliance with Kari’s Law and the Ray Baum’s Act, UCM’s Emergency Calls feature supports emergency location mapping. This will allow users to associate subnets with emergency location identification numbers (ELINs), which can then be used by E911 service providers for example to determine the location of callers. The new options can be found under **Call Features**→**Emergency Calls**→**Emergency Location Mapping**.

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Emergency Location Mapping

- **ELIN:** The emergency location identification number registered with the E911 provider. This number will be sent out as the emergency call's CID number.
- **Subnet:** The network subnet that the ELIN will be associated with. The ELIN that is sent to E911 providers is based on the subnet that a calling endpoint is registered from. Example: "xxx.xxx.xxx.xxx/24" or "xxxx:xxxx:xxxx:xxxx:xxxx:xxxx:xxxx:xxxx/64".
- **Location:** Location associated with the configured subnet. This is used for the UCM administrator's reference.
- **Geolocation Routing:** Toggles whether to include the *Geolocation* header in the emergency call SIP INVITE message. The *Location* field value will be used as the *Geolocation* header value.

Important Note

Please note that ELIN Mapping is supported only on peer trunks. It would not apply on register trunks.

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.

Callback Configuration Parameters

Name	Configure a name to identify the Callback. (Enter at least two characters)
CallerID Pattern	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. Note: If leaving as blank, all numbers are allowed to use this callback.
Outbound Prepend	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.
Delay Before Callback	Configure the number of seconds to be delayed before calling back the user.
Destination	Configure the destination which the callback will direct the caller to. Two destinations are available: <ul style="list-style-type: none">○ IVR○ DISA 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.

URI	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 -.
Local Extensions	Select the available extensions/Extension Groups listed on the local UCM630xA to be monitored in the event list.
Remote Extensions	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.
Special Extensions	Manually enter the remote extensions in the peer/register trunk to be monitored in the event list. Valid format: 5000,5001,9000

Event List Settings

The screenshot shows the 'Create New Event List' configuration page. It features several sections:

- * URI:** A text input field containing 'test'.
- Event Type:** A dropdown menu set to 'Dialog'.
- Local Extensions:** A list of 9 available extensions (1003 "Betty" through 1007 "Kiki") with checkboxes. A search bar is present above the list. To the right, a 'Selected' list contains 3 items: 1000 "Mia", 1001 "John", and 1002 "Chris".
- Remote Extensions:** Two panels, one labeled 'Available' (0 items) and one labeled 'Selected' (0 items), both containing search bars and the text 'None'.
- Special Extensions:** An empty text input field at the bottom.

Navigation arrows are located between the extension lists.

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.

- 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**.

Create New Dial By Name

* Name:

* Extension:

Custom Prompt: [Upload Audio File](#)

Members:

<input type="checkbox"/> 30 items Available <input type="text" value="Search"/> <input type="button" value="Q"/> <input type="checkbox"/> 1000 <input type="checkbox"/> 1001 <input type="checkbox"/> 1002 <input type="checkbox"/> 1003 <input type="checkbox"/> 1004	<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>
--	--	--

LDAP Phonebook:

<input type="checkbox"/> 1 item Available <input type="text" value="Search"/> <input type="button" value="Q"/> <input type="checkbox"/> ou=GStest,dc=pbx,dc=com	<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>
---	--	--

Options

* Prompt Wait Time:

Query Type: By Last Name + First Name
 By First Name + Last Name

Select Type: By Order By Menu

Create Dial by Name Group

User Settings

First Name:	<input type="text" value="John"/>	Last Name:	<input type="text" value="DOE"/>
Email Address:	<input type="text"/>	* User Password:	<input type="password" value="*****"/>
* Language:	<input type="text" value="Default"/>	* Concurrent Registration...	<input type="text" value="1"/>
Mobile Phone Number:	<input type="text"/>		

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:	<input type="text" value="Dial By Name"/>	<input type="text" value="DialByNameG..."/>
Press 1:	<input type="text" value="Select an Opti..."/>	
Press 2:	<input type="text" value="Select an Opti..."/>	

Dial By Name Group In IVR Key Pressing Events

Edit Inbound Rule Save

<p>* Pattern: <input style="width: 150px;" type="text" value="-"/></p> <p>Disable This Route: <input type="checkbox"/></p> <p>Prepend User Defined Nam... <input type="checkbox"/> <input style="width: 150px;" type="text"/></p> <p>Alert-info: None v</p> <p>Privilege Level: Internal v</p> <p>Allowed to seamless transfe... <input style="width: 150px;" type="text"/></p> <p>Default Mode</p> <p>* Default Destination: Dial By Name v</p>	<p>CallerID Pattern: Separate patterns by commas, such as " _ :</p> <p>Prepend Trunk Name: <input type="checkbox"/></p> <p>Inbound Multiple Mode: <input type="checkbox"/></p> <p>Dial Trunk: <input type="checkbox"/></p> <p>DID Destination: <input style="width: 150px;" type="text"/></p>
---	--

Dial By Name Group In Inbound Rule

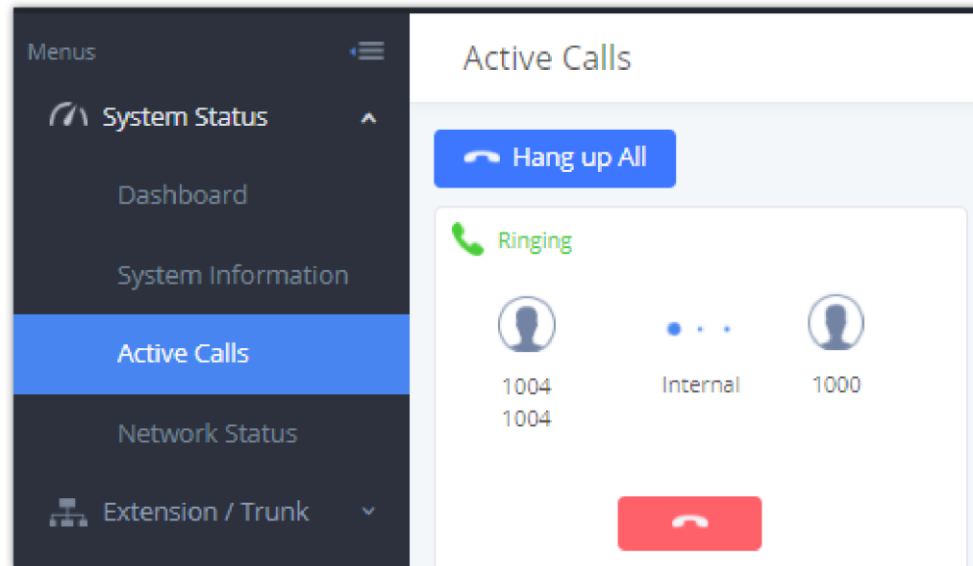
Please refer to [Username Prompt Customization] for User Name Prompt Customization.

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.

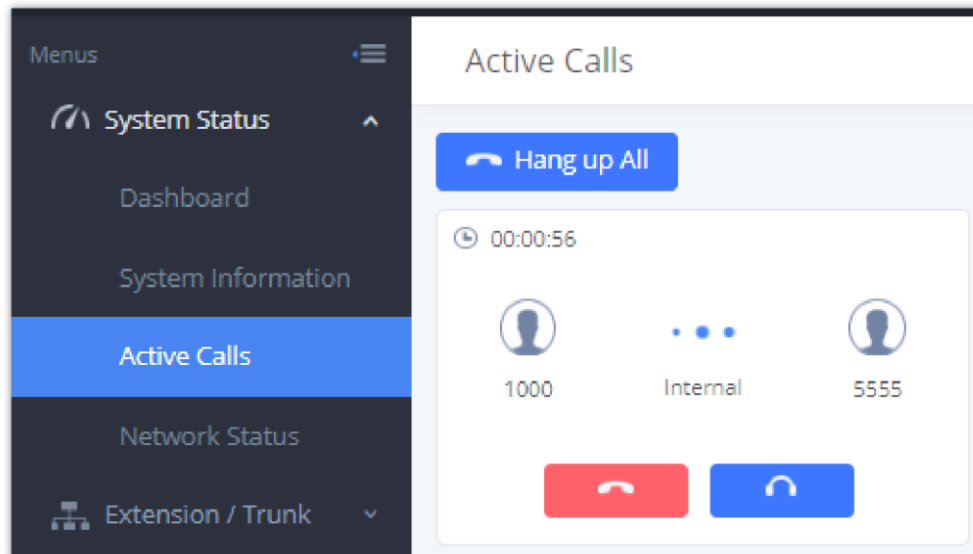
Active Calls Status

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.



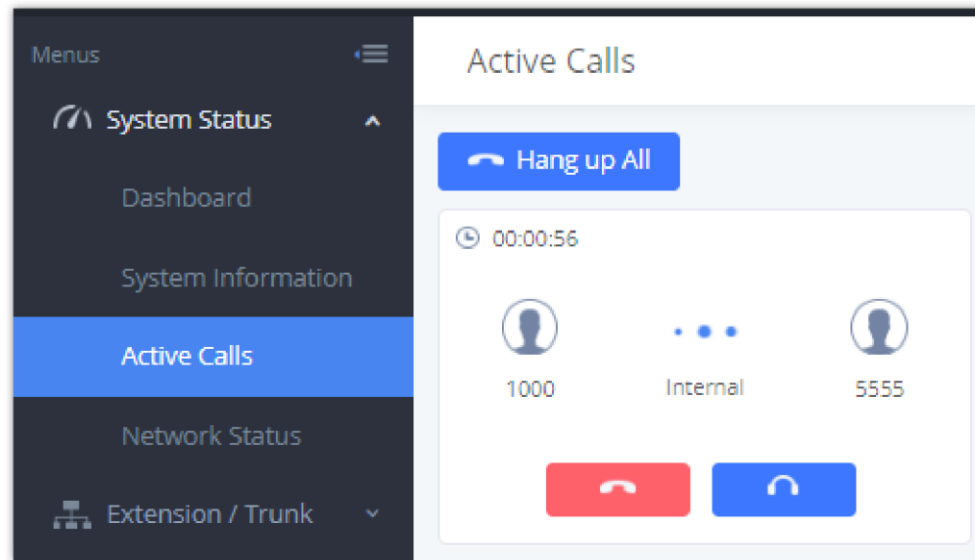
Active Calls – Call Ringing

The following figure shows the call between 1000 and 5555 is established.



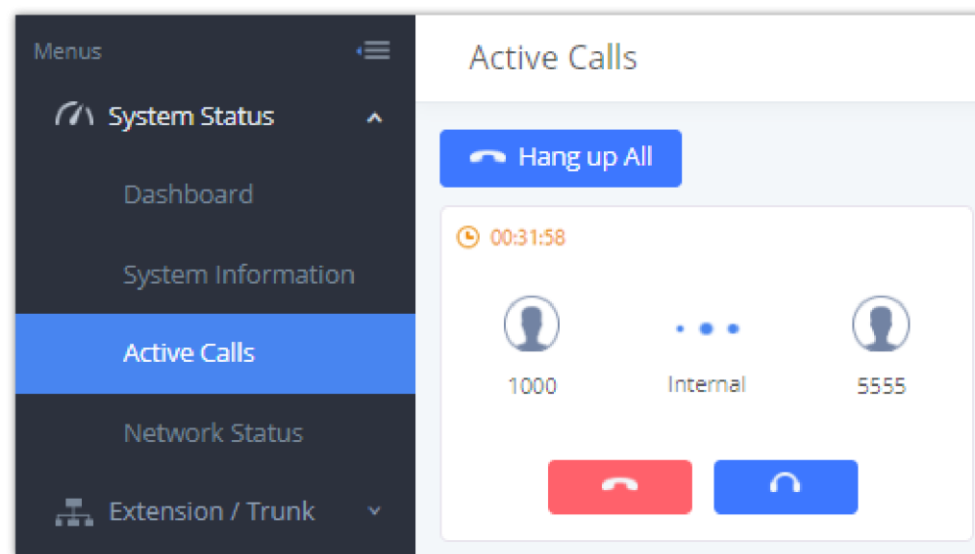
Active Calls – Call Established

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.



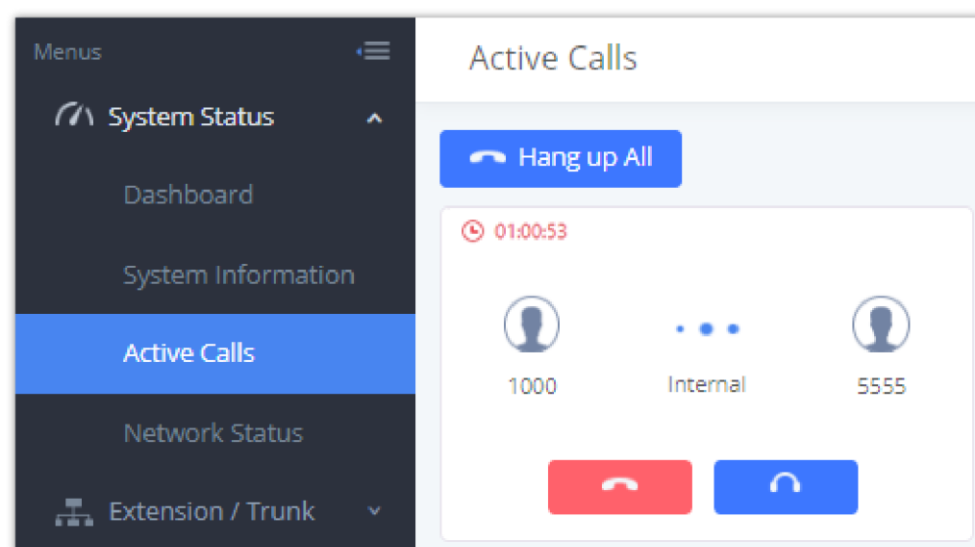
Call Duration: Less than 30 Minutes

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.



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.



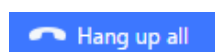
Call Connection more than one hour

Hang Up Active Calls

To hang up an active call, click on



icon in the active call dialog. Users can also click on



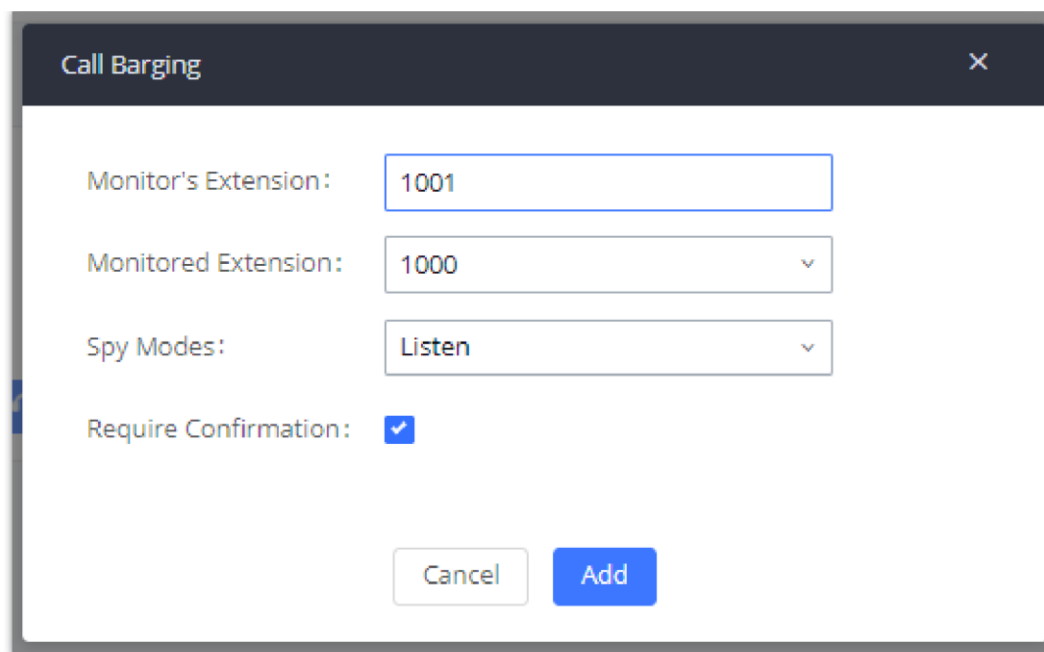
to hang up all active calls.

Call Monitor

During an active call, click on icon



and the monitor dialog will pop up.



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.

1. 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.
2. 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 **[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

UCM630xA Feature Codes

Feature Maps	
Blind Transfer	- Default code: #1

	<p>- 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.</p> <p>- Options:</p> <ul style="list-style-type: none"> • Disable • Allow Caller: Enable the feature code on caller side only. • Allow Callee: Enable the feature code on callee side only. • Allow Both: Enable the feature code on both caller and callee.
Attended Transfer	<p>- Default code: *2</p> <p>- 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.</p> <p>- Options:</p> <ul style="list-style-type: none"> • Disable • Allow Caller: Enable the feature code on caller side only. • Allow Callee: Enable the feature code on callee side only. • Allow Both: Enable the feature code on both caller and callee.
Transfer Dialing Timeout Period (s)	Configures the dial timeout period of blind and attended transfers.
Seamless Transfer	<ul style="list-style-type: none"> • Default code: *44 (Disabled by default). • 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. • 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.
Disconnect	<p>- Default code: *0</p> <p>- Enter the code during active call. It will disconnect the call.</p> <p>- Options:</p> <ul style="list-style-type: none"> • Disable • Allow Caller: Enable the feature code on caller side only. • Allow Callee: Enable the feature code on callee side only. • Allow Both: Enable the feature code on both caller and callee.
Call Park	<p>- Default code: #72</p> <p>- Enter the code during active call to park the call.</p> <p>- Options:</p> <ul style="list-style-type: none"> • Disable • Allow Caller: Enable the feature code on caller side only. • Allow Callee: Enable the feature code on callee side only. • Allow Both: Enable the feature code on both caller and callee.
Feature Code Input Timeout (ms)	Configure the maximum interval (ms) between digits for feature code activation.
Start/Stop Call Recording	<p>-Default code: *3</p> <p>- Enter the code followed by # or SEND to start recording the audio call and the UCM630X will mix the streams natively on the fly as the call is in progress.</p> <p>- Options:</p> <ul style="list-style-type: none"> • Disable • Allow Caller: Enable the feature code on caller side only. • Allow Callee: Enable the feature code on callee side only. • Allow Both: Enable the feature code on both caller and callee.

Enable Recording Whitelist	Enable the Recording Whitelist feature
Recording Operation Whitelist	Select extension in the whitelist that can use the *3 recording function.
Feature Code Digits Timeout	Set the maximum interval (ms) between digits for feature code activation
DND/Call Forward	
Do Not Disturb (DND) Activate	Default code: *77
Do Not Disturb (DND) Deactivate	Default code: *78
Call Forward Busy Activate	- Default Code: *90 - Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward Busy Deactivate	Default Code: *91
Call Forward No Answer Activate	- Default Code: *92 - Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward No Answer Deactivate	Default Code: *93
Call Forward Unconditional Activate	- Default Code: *72 - Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward Unconditional Deactivate	Default Code: *73
Remote Call Forward Enable	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.
Remote DND / Call Forward Settings	
Enable	Enable this option and configure the Whitelist below to allow specific extensions to dial feature codes to set DND or call forwarding for any extension.
Remote Call Forward Busy Enable	Configures and enables CFB for any extension.
Remote Call Forward No Answer Enable	Configures and enables CFNA for any extension.
Remote Call Forward Always	Configures and enables CFU for any extension.

Enable	
Remote DND Enable	Enables Do Not Disturb for any extension.
Remote Call Forward Busy Disable	Disables CFB for any extension.
Remote Call Forward No Answer Disable	Disables CFNA for any extension.
Remote Call Forward Always Disable	Disables CFU for any extension.
Remote DND Disable	Disables Do Not Disturb for any extension.
Whitelist	Extensions in this whitelist can configure DND or call forwarding for any extension via feature codes.
Feature Codes	
Voicemail	
Voicemail Access Code	<ul style="list-style-type: none"> - Default Code: *98 - Enter *98 and follow the voice prompt. Or dial *98 followed by the extension and # to access the entered extension's voicemail box.
My Voicemail	<ul style="list-style-type: none"> - Default Code: *97 - Press *97 to access the voicemail box.
Voicemail Group Access Code	Dial this code to access group voicemail. If password is required, enter password followed by the pound (#) key.
Direct Dial Voicemail Prefix	Prefix used to dial directly to voicemail.
Call Queue	
Agent Pause	<ul style="list-style-type: none"> - Default Code: *83 - Pause the agent in all call queues.
Agent Unpause	<ul style="list-style-type: none"> - Default Code: *84 - Unpause the agent in all call queues.
Dynamic Agent Logout	Log the dynamic agent out of all queues.
Call Pickup	
Pickup on Ringing Prefix	<p>Picks up a ringing call for another extension.</p> <p>Example: If the prefix is **, and there is a call ringing ext 1008, dial **1008 from a different extension to pick up the call to 1008.</p>

Pickup In-call Prefix	Picks up an ongoing call for another extension. Example: If the feature code is *45, and ext 1008 is in a call, dialing *45 and then 1008 following the prompt will take that call. Note: The feature code user must be in the extension's Allowed to seamless transfer list to pick up calls for it.
Pickup Extension	This is the feature code to pick up incoming calls for other extensions in the same pickup group. The default setting is *8.
Call Barging	
Enable Spy	Check this box to enable spy feature codes.
Listen Spy	This is the feature code to listen in on a call to monitor performance. Your line will be muted, and neither party will hear you. The default setting is *54.
Barge Spy	This is the feature code to join in on the call to assist both parties. The default setting is *56.
Whisper Spy	This is the feature code to speak to only one party in the call. For example, you could whisper to employees to help them handle a call. Only an employee on your account will be able to hear you. The default setting is *55.
PMS	
PMS Wakeup Service	Dial this feature code to access PMS Wakeup Service. You can add, update, activate or deactivate PMS Wakeup Service.
PMS Remote Wakeup Service	Dial this code to add, update, activate, and deactivate PMS wakeup service for other extensions.
Update PMS Room Status	2 methods are available: 1. Dial the room status feature code + housekeeper code, listen to the prompt and then the dial the appropriate key for the desired room status. Example: The housekeeper with housekeeper code 0001 dials *230001, listens to the room status options prompt, and then dials 1 to change room status to Available. 2. Dial room status feature code*housekeeper code*desired room status option key to quickly change the room status without needing to go through the system voice prompts. Example: Housekeeper with Housekeeper code 0001 dials *23*0001*1 to change room status Available.
Misc	
Paging Prefix	Configure the paging prefix for paging. For example, if the Paging Prefix is set to *81, dial *816000 to initiate a paging call to extension 6000.
Intercom Prefix	Configure the intercom prefix for intercom calls. For example, if the Intercom Prefix is set to *80, dial *806000 to initiate an intercom call to extension 6000.
Blacklist Add	Follow the voice prompt to add a caller ID to blacklist.
Blacklist Last Caller	Add the last inbound caller ID number to blacklist.
Blacklist Remove	Follow the voice prompt to remove a caller ID from blacklist.
Direct Dial Mobile Phone Prefix	If calling mobile phone numbers is permitted, use this prefix plus the extension number to dial the mobile phone number of this extension directly.

Call Completion Request	If the caller wants to use CC to complete a call, he/she can dial this code. After the CC has been registered successfully, the system will start to monitor the status of the callee. The system will call back the caller when the callee's extension is available.
Call Completion Cancel	If the caller has requested CC successfully, and he/she doesn't need to call back anymore, he/she can dial this code to cancel the request.
Presence Status	Dial this feature code to set the presence status of the extension.
Call Flip	- Default code: *46 - Dial this code to move the call of this extension from another device to the current device.
Wakeup Service	Dial this feature code to access UCM Wakeup Service. You can add, update, activate or deactivate UCM Wakeup Service.
Remote Extension Privilege Update	Whitelisted extensions will be able to use the Remote Extension Privilege Update feature code to remotely change any extension's outgoing call privilege. Note: After this function has been enabled, the extension in the whitelist can set the privilege for outgoing calls of any extension by dialing the feature code.
Remote Extension Privilege Update Whitelist	Remote Extension Privilege Update Whitelist Procedure: 1. Dial *26 on the whitelisted extension, hear the prompt "Change extension's outgoing permission level, please enter the phone number, then enter # key." 2. After the process, voice will prompt "Press 1 to set to internal, press 2 to set to local, press 3 to set to national, press 4 to set to international." 3. After selecting, it will prompt "Change extension XXXX outgoing permission to XXX", and hang up.
Privileged Call	Dial the feature code + extension number, for example *001002 to hang up the current call of extension 1002 with privilege, and then call extension 1002.
Privileged Call Whitelist	The extensions in this white list can use the feature code of the privileged call.
Priority Call	Makes a high priority call to a specified extension by dialing the feature + extension number. If the specified extension is an ongoing call, it will be forcibly hung up to allow the caller to ring the extension.
Priority Call Whitelist	Extension in the whitelist will be allowed to use this feature code to make priority calls.

The UCM630xA also allows user to one click enable / disable specific feature code as shown below:

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.

EXTENSION	NAME	SLOTS	OPTIONS
700	DefaultLot	701-720	
7000	Dales	7001-7022	

Parking Lot

User can create a new Parking lot by clicking on button "Add" :

New Parking Lot

Parking Lot

Parking Lot Extension

- Default Extension: **700**
- During an active call, initiate blind transfer and then enter this code to park the call.

Parking Lot Name	<ul style="list-style-type: none"> ○ Set a name to the parking lot
Parked Slots	<ul style="list-style-type: none"> ○ Default Extension: 701-720 ○ These are the extensions where the calls will be parked, i.e., parking lots that the parked calls can be retrieved.
Use Parklot as Extension	<ul style="list-style-type: none"> ○ 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.
Parking Timeout (s)	<ul style="list-style-type: none"> ○ Default setting is 300 seconds, and the maximum limit is 99.999 seconds. ○ 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.
Music On Hold Classes	Select the Music on Hold Class.
Failover Destination	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.
Ring All Callback on Timeout	If enabled, all registered endpoints of the extension will ring when callback occurs. Otherwise, only the original endpoint will be called back.
Forward to destination on timeout	If enabled, the call will be routed to the configured destination upon timeout. Otherwise, the call will be routed back to the original caller.
Timeout Destination	This option appears once Forward to Destination on Timeout is enabled. Upon park timeout, the call will be routed to the configured destination.
Parking Lot Timeout Alert-Info	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)

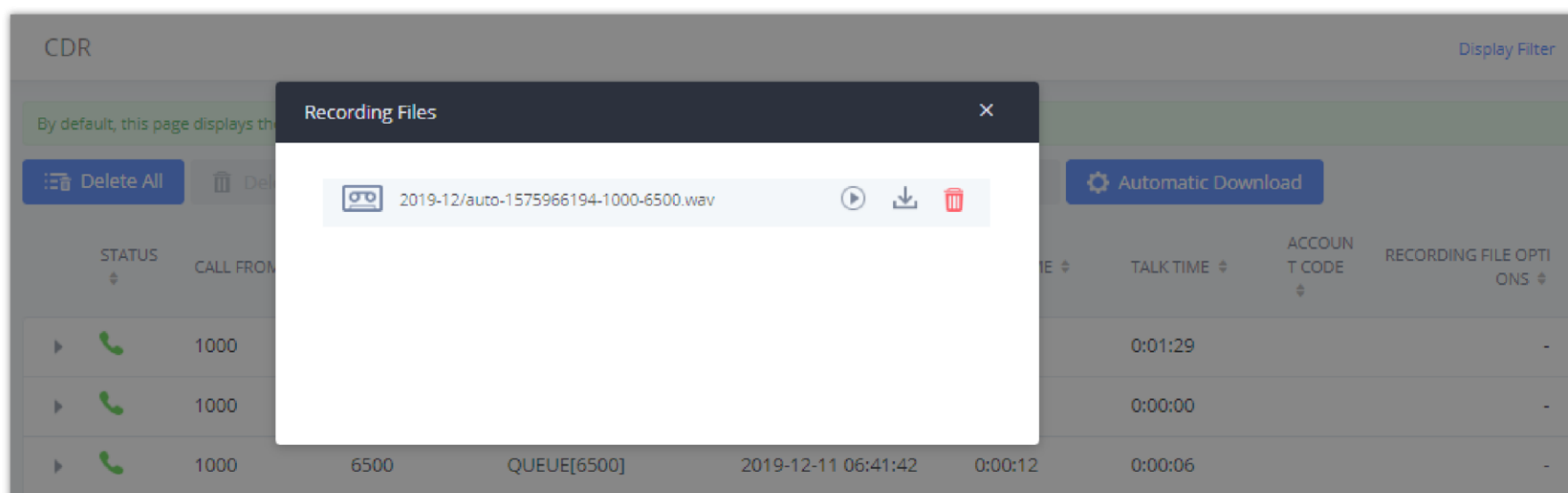


Users can see the CID name information of parked calls. VPK/MPKs must be configured as “Monitored Call Park” with the desired parking lot extension. The display will alternate between displaying the parking lot extension and the call’s CID name. There is no need to configure anything on the UCM.

Call Recording

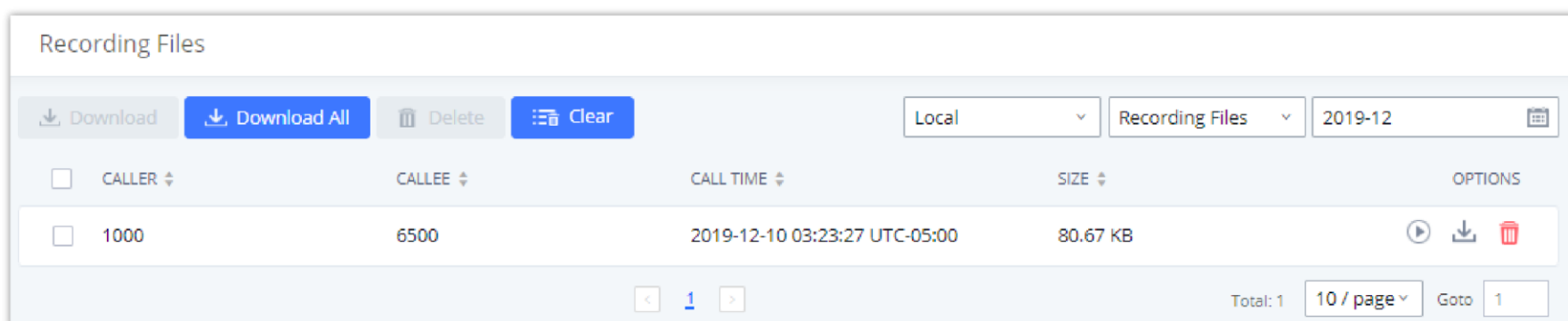
The UCM630xA allows users to record audio during the call. If “Auto Record” is turned on for an extension, ring group, call queue or trunk, the call will be automatically recorded when there is established call with it. Otherwise, please follow the instructions below to manually record the call.

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.



Download Recording File from CDR Page

The above recorded call’s recording files are also listed under the UCM630xA Web GUI→**CDR**→**Recording Files**.



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.

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 active. Under the 'General' section, the 'Enable SCA' checkbox is checked and highlighted with a red box. Other settings include Extension: 1000, Permission: Internal, CallerID Number: 1000, and Voicemail: Local Voicemail.

Enabling SCA option under Extension's Settings

1. 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.

STATUS	SHARED LINE	ROLE	IP AND PORT	SUBSCRIBED	OPTIONS
Unavailable	1000	shared	--	no	+ [edit]

Total: 1 | 10 / page | Goto: 1

SCA Number Configuration

1. Configure the private number as desired.

SCA Private Number Configuration

1. 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.

SCA Options

1. 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:

Add SCA Private Number

Private Number	Configures the private number for the SCA.
Related Shared Line	Display the related shared line.
Enable This Number	Whether enable this private number. If not enabled, this private number is only record in DB, it will not affect other system feature.
Allow Origination from This Number	Enable this option will allow calling from this private number. By default, it is enabled.
Allow Termination to This Number	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:

Editing the SCA Number

Shared Line Number	While SCA is enabled, this number will be the same as the extension number.
Allow Call Retrieve from Another Location	Allows remote call retrieval. Must be enabled in public hold. By default, it is enabled.
Alert All Appearances for Group Paging Calls	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.
Multiple Call Arrangement	Allows simultaneous calls in an SCA group. By default, it is disabled.
Allow Bridging between Locations	Allows location bridging for SCA group. Must be enabled when using the Barge-In feature. By default, it is disabled.
Bridge Warning Tone	Configures the notification in the bridge when another party join. None: No notification sound. Barge-In only: Notification sound will play when another party join. Barge-In and Repeat: Notification sound will play when another party joins and repeat every 30 seconds. By default, it is set to "Barge-In Only".

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 [+ Add](#) 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: [Upload Audio File](#)

Default Destination:

Announcement settings

The table below gives more description of the configuration parameters when creating Announcement.

Name	Configure the name of the Announcement.
Prompt	Choose/upload the audio file to play when the call reaches this announcement.
Default Destination	Select the destination which the call will be forwarded to after the announce has been played.


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

Internal Options/General

General Preferences	
Global Outbound CID	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.
Global Outbound CID Name	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.
Ring Timeout	Configure the number of seconds to ring an extension before the call goes to the user's voicemail box. The default setting is 60 . Note: This is the global value used for each extension if "Ring Timeout" field is left empty on the extension configuration page.
Call Duration Limit	Block calls for the configured duration. If Extensions->Features->Call Duration Limit and Outbound Routes->Call Duration Limit are not configured, General Settings->Call Duration Limit will be used.
Record Prompt	If enabled, users will hear voice prompt before recording is started or stopped. For example, before recording, the IPPBX will play voice prompt "The call will be recorded". The default setting is "No".
Allow External Numbers to Cancel Recording	If enabled, external call parties will be given the option to decline the recording of calls. The IVR will prompt the user to dial *3 in order to cancel the call recording.
Stereo Recording	If enabled, the caller and callee's audio will be split into two channels during call recording. Not applicable to calls with more than 2 parties.
Calling Channel	Configure the audio channels for the calling party and the called party. If the caller is selected as the right channel, the callee will be used for the left channel, and vice-versa. Note: This option will be available when "Stereo Recording" is enabled.
International Call Prefix	When this configuration is empty, International Call Prefix can be empty or +.
Extension Preferences	
Enforce Strong Password	If enabled, a strong password policy will be enforced. This does not affect user login passwords, which must be strong.
Enable Random Password	If enabled, the extension will be created with a randomly generated password.
Send Extension Update Emails	If enabled, an email will be sent to an extension's configured email address after

	creating it or modifying that extension's settings.
Disable Extension Range	If set to "Yes", users could disable the extension range pre-configured/configured on the IPPBX. The default setting is "No". Note: It is recommended to keep the system assignment to avoid inappropriate usage and unnecessary issues.
Extension Ranges	<p>The default extension range assignment is:</p> <ul style="list-style-type: none"> ● User Extensions: 1000-6299 User Extensions is referring to the extensions created under Web GUI→Extension/Trunk→Extensions page. ● Pick Extensions: 4000-4999 This refers to the extensions that can be manually picked from end device when being provisioned by the IPPBX. 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. ● Auto Provision Extensions: 5000-6299 This sets the range for "Zero Config Extension Segment" which is the extensions can be assigned on the IPPBX to provision the end device. ● Meeting Extensions: 6300-6399 This extension range is used for creating meeting rooms. ● Ring Group Extensions: 6400-6499 This extension range is used for ring groups ● Queue Extensions: 6500-6599 This range of extensions is used for queueing ● Voicemail Group Extensions: 6600-6699 This extension range is used for voicemail groups. ● IVR Extensions: 7000-7100 This extension range is used for ● Dial By Name Extensions: 7101-7199 This extension range is used for Dial by Name feature ● FAX Extension: 7200-8200 This extension range is used for T.38 Fax.
	Clicking this button will reset the extension range to their default values.

PBX Settings/RTP Settings

RTP Settings

Internal Options/RTP Settings

RTP Start	Configure the RTP port starting number. The default setting is 10000.
RTP End	Configure the RTP port ending address. The default setting is 20000.
Strict RTP	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".
RTP Checksums	Configure to enable or disable RTP Checksums on RTP traffic. The default setting is "Disable".
ICE Support	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.
STUN Server	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.
BFCP UDP Start	Configure BFCP UDP port starting number. The default setting is 50000.
BFCP UDP End	Configure BFCP UDP port ending number. The default setting is 52999.
BFCP TCP Start	Configure BFCP TCP port starting number. The default setting is 53000.
BFCP TCP End	Configure BFCP TCP port ending number. The default setting is 55999.
TURN Server	Configure TURN server address. TURN is an enhanced version of the STUN protocol and is dedicated to the processing of symmetric NAT problems.
TURN Server Name	Configure turn server account name
TURN Server Password	Configure turn server account password.
Connection Protocol	Protocol used to connect to the TURN server.
Number of ICE Candidates	This configures the number of pre-collected ICE candidates to gather and send to remote peers. The higher the number, the greater the network traffic consumption.

Payload

The UCM630xA payload type for audio codecs and video codes can be configured here.

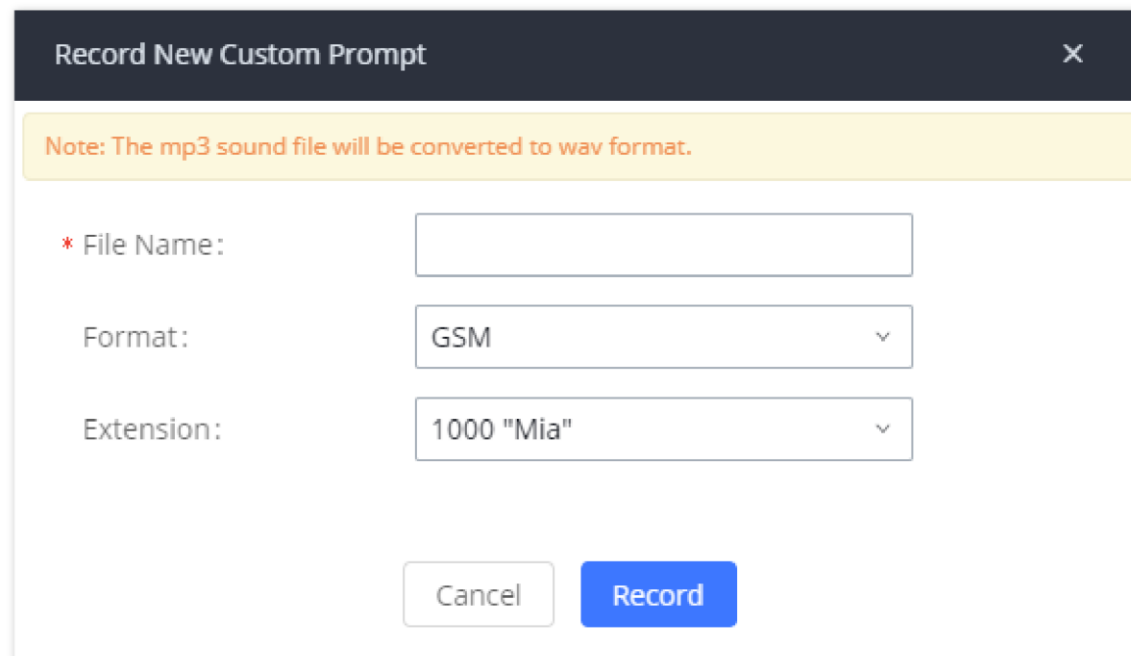
Internal Options/Payload

AAL2-G.726	Configure payload type for ADPCM (G.726, 32kbps, AAL2 codeword packing). The default setting is 112.
DTMF	Configured payload type for DTMF. The default setting is 101.
G.721 Compatible	Configure to enable/disable G.721 compatible. The default setting is Yes.
G.726	Configure the payload type for G.726 if "G.721 Compatible" is disabled. The default setting is 111.
iLBC	Configure the payload type for iLBC. The default setting is 97.
OPUS	Configure the payload type for OPUS. The default setting is 123.
Audio FEC Payload Type	Configure the Audio FEC Payload Type. The default setting is 127
Audio RED Payload Type	Configure the Audio RED Payload Type. Default setting is 122
H.264	Configure the payload type for H.264. The default setting is 99.
H.263P	Configure the payload type for H.263+. The default setting is 100 103.
VP8	Configure the payload type for VP8. The default setting is 108.
Main Video FEC	Configure the Main Video FEC
RTP FECC	Configure the RTP FECC
RTX	Configure the RTX
G.722.1	G.722.1: Low-complexity coder, 24kbps.
G.722.1C	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.



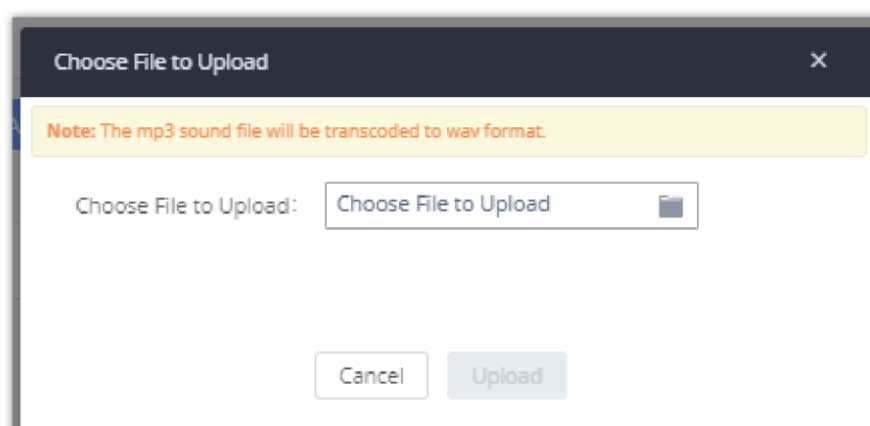
Record New Custom Prompt

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:

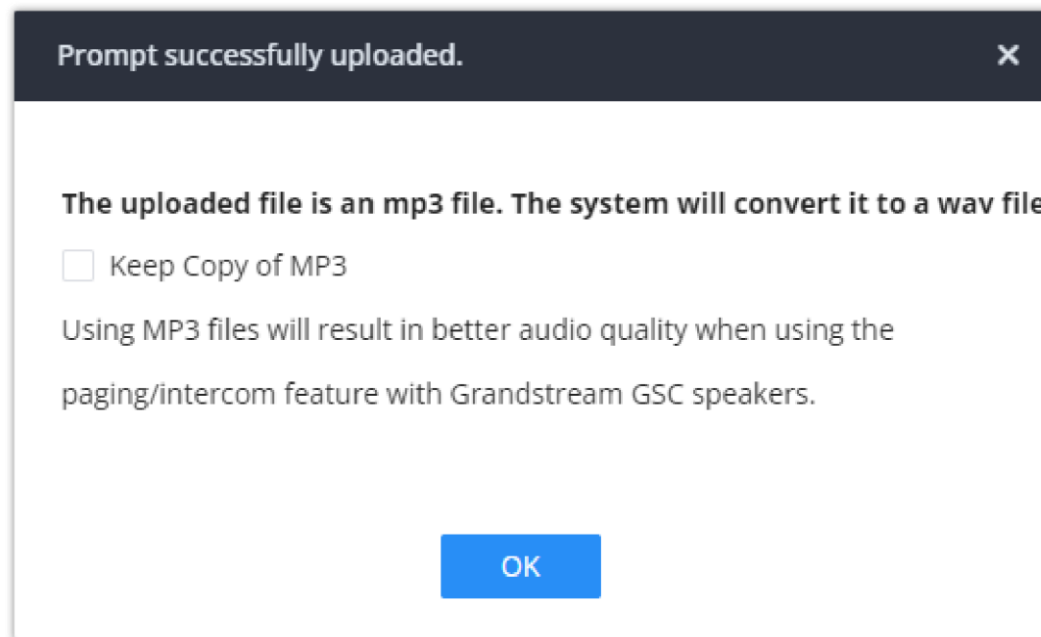
- PCM encoded.
- 16 bits.
- 8000Hz mono.
- In .mp3 or .wav format; or raw/ulaw/alaw/gsm file with .ulaw or .alaw suffix.
- File size under 5M.



Upload Custom Prompt

Click on "choose file to upload" to start uploading. Once uploaded, the file will appear in the Custom Prompt web page.

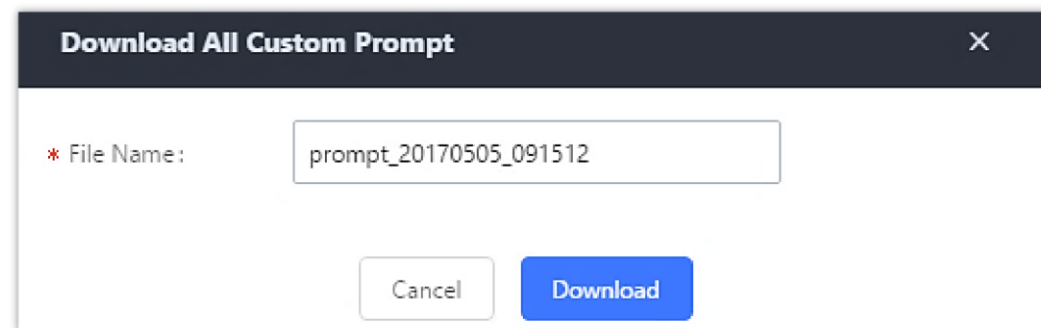
If the file uploaded is mp3, the following prompt will appear



Tick "Keep Copy of MP3" to keep an mp3 copy of the file which can be diffused with certain GSC speakers. Diffusing the MP3 files result in a higher quality sound played by GSC speakers.

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.



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.

Call Failure Tone Settings

SIP Trunk Prompt Tone General Call Failure Tones

Specify the tones to play for various SIP trunk call failure scenarios.

Reset All Default All

400:	<input type="text" value="sip-trunk-out-busy"/>	401:	<input type="text" value="sip-trunk-out-busy"/>
402:	<input type="text" value="sip-trunk-out-busy"/>	403:	<input type="text" value="sip-trunk-out-rejected"/>
404:	<input type="text" value="sip-trunk-out-wrong-nu..."/>	405:	<input type="text" value="sip-trunk-out-busy"/>
406:	<input type="text" value="sip-trunk-out-busy"/>	407:	<input type="text" value="sip-trunk-out-busy"/>
408:	<input type="text" value="sip-trunk-out-busy"/>	410:	<input type="text" value="sip-trunk-out-busy"/>
413:	<input type="text" value="sip-trunk-out-busy"/>	414:	<input type="text" value="sip-trunk-out-busy"/>
415:	<input type="text" value="sip-trunk-out-busy"/>	416:	<input type="text" value="sip-trunk-out-busy"/>
420:	<input type="text" value="sip-trunk-out-busy"/>	421:	<input type="text" value="sip-trunk-out-busy"/>
423:	<input type="text" value="sip-trunk-out-busy"/>	480:	<input type="text" value="sip-trunk-out-busy"/>
481:	<input type="text" value="sip-trunk-out-busy"/>	482:	<input type="text" value="sip-trunk-out-busy"/>
483:	<input type="text" value="sip-trunk-out-busy"/>	484:	<input type="text" value="sip-trunk-out-busy"/>
485:	<input type="text" value="sip-trunk-out-busy"/>	486:	<input type="text" value="sip-trunk-out-busy"/>
487:	<input type="text" value="sip-trunk-out-busy"/>	488:	<input type="text" value="sip-trunk-out-busy"/>
491:	<input type="text" value="sip-trunk-out-busy"/>	493:	<input type="text" value="sip-trunk-out-busy"/>

Reset All Default All

500:	<input type="text" value="sip-trunk-out-server-error"/>	501:	<input type="text" value="sip-trunk-out-server-error"/>
502:	<input type="text" value="sip-trunk-out-server-error"/>	503:	<input type="text" value="sip-trunk-out-server-error"/>
504:	<input type="text" value="sip-trunk-out-server-error"/>	505:	<input type="text" value="sip-trunk-out-server-error"/>
513:	<input type="text" value="sip-trunk-out-server-error"/>		

Reset All Default All

600:	<input type="text" value="sip-trunk-out-busy"/>	603:	<input type="text" value="sip-trunk-out-rejected"/>
604:	<input type="text" value="sip-trunk-out-wrong-nu..."/>	606:	<input type="text" value="sip-trunk-out-busy"/>

SIP Trunk Prompt Tone

General Call Prompt Tones

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 Tones"** page as shown on the following figure:

Call Prompt Tones

SIP Trunk Prompt Tone General Call Prompt Tones Cancel Save

Specify the tones to play for various general call scenarios.

Reset All Default All

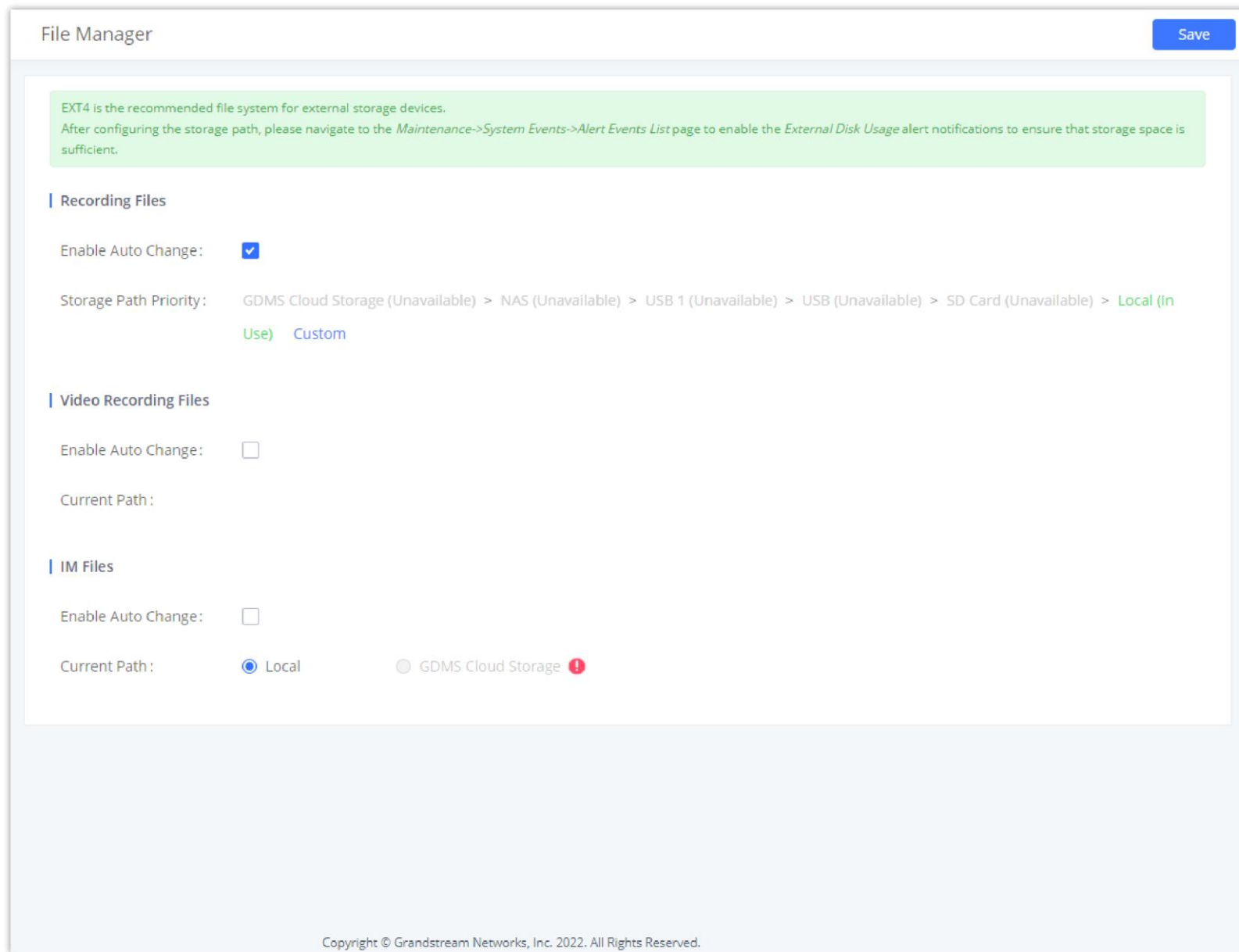
Bad Number:	<input type="text" value="wrong-number"/>	Out Of Service:	<input type="text" value="out-of-service"/>
User Busy:	<input type="text" value="user-busy"/>	Trunk Busy:	<input type="text" value="trunk-busy"/>
No Answer:	<input type="text" value="no-answer"/>	No Permission:	<input type="text" value="no-permission"/>
Do Not Disturb:	<input type="text" value="user-busy"/>	General Failed:	<input type="text" value="general-failed"/>
Call Waiting:	<input type="text" value="Default Ringback Tone"/>		

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General call Failure Prompts

File Manager

UCM supports automatic or manual recording of calls and storage of IM chat files. Files are allowed to be saved in UCM local or external storage devices, and can even be stored in GDMS cloud storage. The chat files are only allowed to be saved in UCM local or external storage devices, users can go to UCM630xA Web GUI→**PBX Settings**→**File Manager** page and select whether to store the recording files in USB Disk, SD card, GDMS or locally on the UCM630xA.



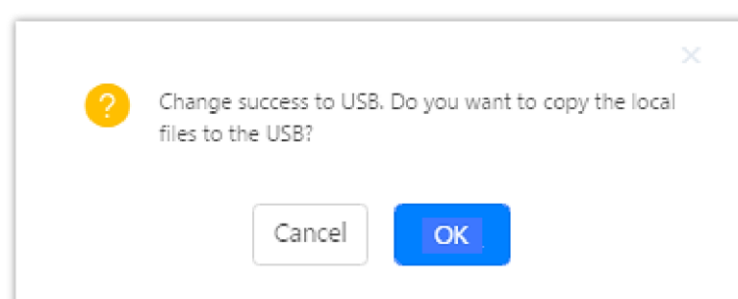
File Manager

Note

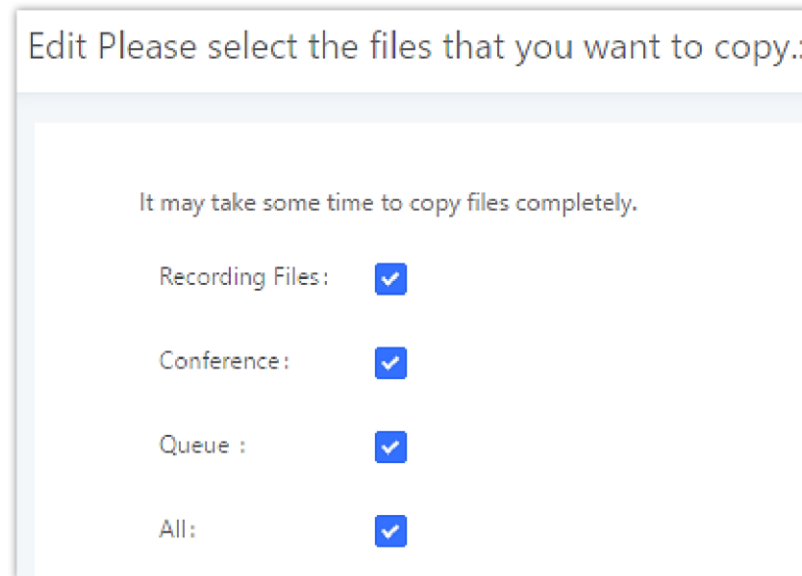
Once a storage device has filled up, the UCM will choose the next available storage device based on the *Storage Path Priority*.

- If “**Enable Auto Change**” is selected, the 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 files will be always saved in the USB Disk.
- When “Enable Auto Change” is enabled, the option “**Storage Path Priority**” will appear. It allows the user to configure the priority of each storage unit in the priority list (The storage on top of the list has the highest priority). The default priority list is *GDMS Cloud Storage > NAS > USB 1 > USB > SD Card > Local*
- If “**Local**” is selected, the files will be stored in UCM630xA internal storage.
- If “**GDMS Cloud Storage**” is selected, data will no longer be stored locally and if you need to listen to the recording, download the file to the computer side and play it offline.

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.



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



Recording Storage Category

On the UCM630xA, 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:

NAS Settings

Enable	Enabled / Disable the NAS recording functionality.
Host	Configure the Domain or IP address of the NAS server. Note: Currently, only IP addresses are supported in the Host/IP field.
Share Name	Specify the name of the shared folder.
Username	Specify the account username to access the NAS server.
Password	Configure the account password to access the NAS server. The password can include letters, number, and special characters.
Security Mode	Select a security mode based on the server settings to ensure proper connection establishment. The default value is ntlmssp.
Status	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.

Important

If Network Storage Device has 1GB of storage space left, it will be considered unavailable the UCM will trigger the external disk usage alert.

SIP SETTINGS

The UCM630xA SIP global settings can be accessed via Web GUI→**PBX Settings**→**SIP Settings**.

SIP Settings/General

SIP Settings/General

Realm For Digest Authentication	Configure the host name or domain name for the UCM630xA. Realms MUST be globally unique according to RFC3261. The default setting is Grandstream.
Bind UDP Port	Configure the UDP port used for SIP. The default setting is 5060.
Bind IPv4 Address	Configure the IPv4 address to bind to. The default setting is 0.0.0.0, which means binding to all addresses.
Bind IPv6 Address	Configure the IPv6 address to bind to. The default is : "[::]" and it means to bind to all IP addresses.
Allow Guest Calls	<p>If enabled, the UCM630xA allows unauthorized INVITE coming into the PBX and the call can be made. The default setting is "No".</p> <p>Warning:</p> <p>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.</p>
Allow Transfer	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".
MWI From	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.
Enable Diversion Header	If disabled, the UCM will not forward the diversion header.
Block Collect Calls	<p>If enabled, collect calls will be blocked.</p> <p>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".</p>

SIP Settings/MISC

SIP Settings/Misc

Outbound SIP Registrations	
Register Timeout	Configure the register retry timeout (in seconds). The default setting is 20.

Register Attempts	Configure the number of registration attempts before the UCM630X gives up. The default setting is 0, which means the UCM630X will keep trying until the server side accepts the registration request.
Trunk Register Period (s)	Configures the time window within which to send initial trunk registration requests. Instead of sending out all initial trunk registration requests at once, requests will be randomly sent out within this period.
Video	
Max Bit Rate (kb/s)	Configure the maximum bit rate (in kb/s) for video calls. The default setting is 384.
Support SIP Video	Select to enable video support in SIP calls. The default setting is "Yes".
Security	
Reject Non-Matching INVITE	If enabled, when rejecting an incoming INVITE or REGISTER request, the UCM630X 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. Default setting is "No".
SDP Attribute Passthrough	
Enable Attribute Passthrough	If enable, and if the service does not know the attribute of FEC/FECC/BFCP, then the attribute will be passthrough.
Early Media	
Enable Use Final SDP	If enabled, call negotiation will use final response SDP.
Ignore 180 Response	If enabled, ringing indication after 183 response will be ignored.
Blind Transfer	
Allow callback when blind transfer fails	If enabled, the UCM will call back to the transferrer when blind transfer fails (reason of failure includes busy and no answer). Note: This feature takes effect only on internal calls.
Blind transfer timeout	Configure the timeout in (s) for the transferrer waiting for the destination to answer. Default is 60s.
Hold	
Forward HOLD Requests	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. Note: Enabling this option may cause hold retrieval issues and MOH to not be heard.

SIP Settings/Session Timer

SIP Settings/Session Timer

Force Timer	If checked, always request, and run session timer.
--------------------	--

Timer	If checked, run session timer only when requested by other UA.
Session Expire	Configure the maximum session refresh interval (in seconds). Default is 1800.
Min SE	Configure the minimum session refresh interval (in seconds). The default setting is 90.

SIP Settings/TCP and TLS

The screenshot shows the 'SIP Settings' window with the 'TCP/TLS' tab active. The settings are as follows:

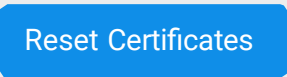
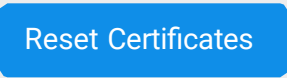
- TCP Enable:**
- TCP Bind IPv4 Address:** 0.0.0.0:5060
- TCP Bind IPv6 Address:** [::]:5060
- TLS Enable:**
- TLS Bind IPv4 Address:** 0.0.0.0:5061
- TLS Bind IPv6 Address:** [::]:5061
- TLS Do Not Verify:**
- TLS Self-signed CA:** TLS.ca (with a 'Reset Certificates' button)
- Private Certificate and Key:**
 - TLS Cert:** TLS.crt
 - TLS Key:** TLS.key (with a 'Reset Certificates' button)
- Cipher Suite:**
 - Cipher Blacklist:**

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TCP/TLS

SIP Settings/TCP and TLS

TCP Enable	Configure to allow incoming TCP connections with the UCM630X. The default setting is "No".
TCP Bind IPv4 Address	Configure the IP address for the 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.
TCP Bind IPv6 Address	Configure the IPv6 address for the TCP server to bind to. "[::]" means bind to all interfaces. The port number is optional with the default being 5060. For example, [2001:0DB8:0000:0000:0000:1428:0000]:5060.
TLS Enable	Configure to allow incoming TLS connections with the UCM630X. The default setting is "Yes".
TLS Bind IPv4 Address	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.

TLS Bind IPv6 Address	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, [2001:0DB8:0000:0000:0000:1428:0000]: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.
TLS Do Not Verify	If enabled, the TLS server's certificate will not be verified when acting as a client. The default setting is "Yes".
TLS Self-Signed CA	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.
	Clicking on this button will reset the certificates.
Private Certificate and Key	
TLS Cert	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. Note: The size of the uploaded certificate file must be under 2MB.
TLS Key	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. Note: The size of the uploaded CA certificate file must be under 2MB.
	Clicking on this button will reset the certificates.
Cipher Suite	
Restrict Cipher List	By default, all SIP TLS encryption suites are in effect on the system, and when turned on, you can configure the encryption suites allowed to be used.
Cipher Suite	Select the encryption suites that are allowed to be used for SIP TLS connections, in the order of priority as configured.

SIP Settings/NAT

SIP Settings/NAT

External Host	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.
Use IP address in SDP	If enabled, the SDP connection will use the IP address resolved from the external host.
External UDP Port	Configure externally mapped UDP port when the PBX is behind a static NAT or PAT.
External TCP Port	Configure the externally mapped TCP port when the UCM630xA is behind a static NAT or PAT.
External TLS Port	Configures the externally mapped TLS port when UCM630xA is behind a static NAT or PAT.

Local Network Address	<p>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.</p> <p>A sample configuration could be as follows:</p> <p>192.168.0.0/16</p>
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SIP Settings/ToS

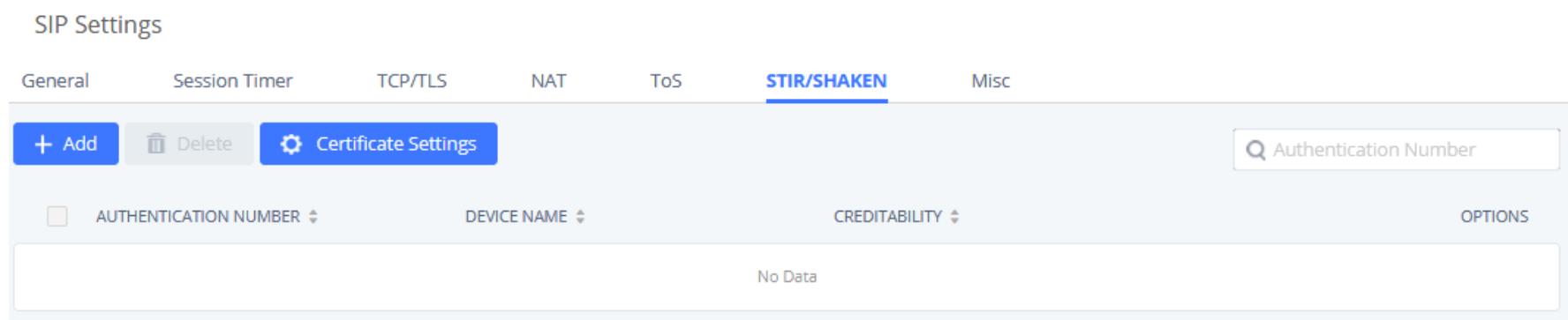
SIP Settings/ToS

ToS for SIP	Configure the Type of Service for SIP packets. The default setting is None.
ToS for RTP Audio	Configure the Type of Service for RTP audio packets. The default setting is None.
ToS for RTP Video	Configure the Type of Service for RTP video packets. The default setting is None.
Default Incoming/Outgoing Registration Time	<p>Configure the default duration (in seconds) of incoming/outgoing registration.</p> <p>The default setting is 120.</p>
Max Registration/Subscription Time	Configure the maximum duration (in seconds) of incoming registration and subscription allowed by the UCM630xA. The default setting is 3600.
Min Registration/Subscription Time	Configure the minimum duration (in seconds) of incoming registration and subscription allowed by the UCM630xA. The default setting is 60.
Enable Relaxed DTMF	Select to enable relaxed DTMF handling. The default setting is "No".
DTMF Mode	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".
RTP Timeout	<p>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.</p> <p>Note: This setting does not apply to calls on hold.</p>
RTP Hold Timeout	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.
RTP Keep-alive	<p>This feature can be used to avoid abnormal call drop when the remote provider requires RTP traffic during proceeding.</p> <p>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.</p> <p>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.</p>
100rel	Configure the 100rel setting on UCM630xA. The default setting is "Yes".
Trust Remote Party ID	Configure whether the Remote-Party-ID should be trusted. The default setting is "No".
Send Remote Party ID	Configure whether the Remote-Party-ID should be sent or not. The default setting is "No".

Generate In-Band Ringing	<p>Configure whether the UCM630xA should generate Inband ringing or not. The default setting is "Never".</p> <ul style="list-style-type: none"> ○ Yes: The UCM630xA will send 180 Ringing followed by 183 Session Progress and in-band audio. ○ No: 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. ○ Never: 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.
Server User Agent	Configure the user agent string for the UCM630xA.
Send Compact SIP Headers	If enabled, compact SIP headers will be sent. The default setting is "No".
Passthrough PAI Header	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.



STIR/SHAKEN

Clicking on the **Add** button will show the following window:

Add Authentication Number
✕

* Authentication Number:

* Device Name:

* Creditability:

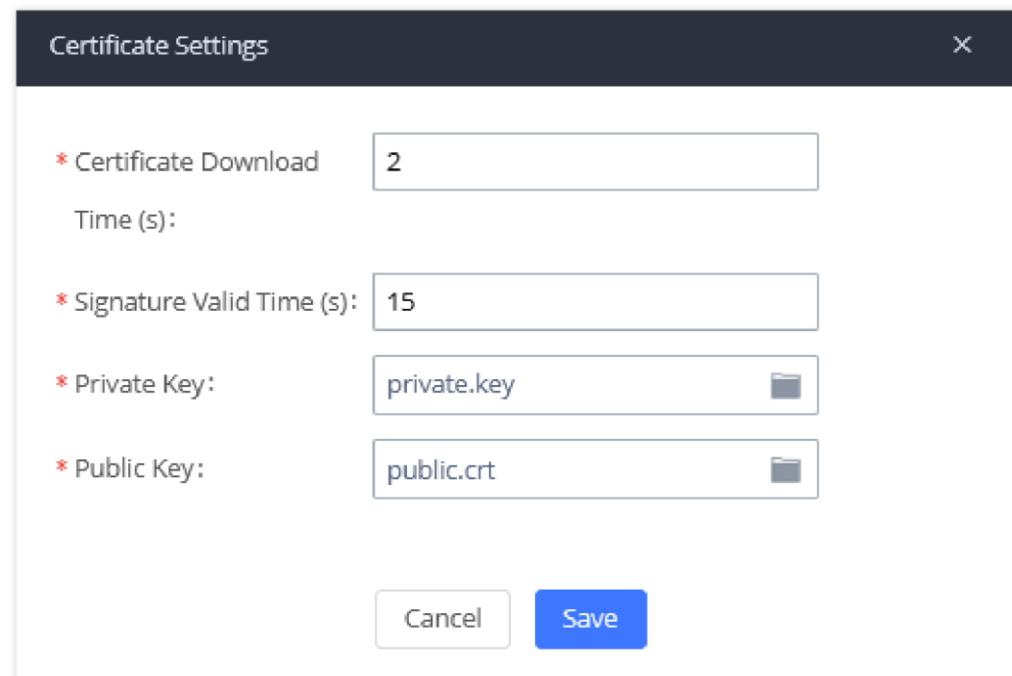
Add Authentication Number

SIP Settings/STIR/SHAKEN – Add Authentication Number Settings

Authentication Number	Configure the Authentication Number.
Device Name	Configure the device name.

Creditability	<p>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:</p> <ul style="list-style-type: none"> ○ 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:



Certificate Settings

SIP Settings/STIR/SHAKEN – Certificate Settings

Certificate Download Time (s)	Configure the public key download timeout period, the default value is 2 seconds.
Signature Valid Time (s)	Configure the validity period of the digital signature, the default value is 15 seconds.
Private Key	<p>Configure the Private key.</p> <p>Note: 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".</p>
Public Key	<p>Configure the Public Key.</p> <p>Note: 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".</p>

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

```

Transparent Call-Info

IAX SETTINGS

The UCM630xA IAX global settings can be accessed via Web GUI→**PBX Settings**→**IAX Settings**.

IAX Settings/General

IAX Settings/General

Bind Port	Configure the port number that the IAX2 will be allowed to listen to. The default setting is 4569.
Bind IPv4 Address	Force IAX2 to bind to a specific address instead of all addresses.
Bind IPv6 address	Configure the IPv6 address to bind to. "[::]" means to bind to all IP addresses.
IAX1 Compatibility	Select to configure IAX1 compatibility. The default setting is "No".
No Checksums	If selected, UDP checksums will be disabled and no checksums will be calculated/checked on systems supporting this feature. The default setting is "No".
Delay Reject	If enabled, the IAX2 will delay the rejection of calls to avoid DOS. The default setting is "No".
ADSI	Select to enable ADSI phone compatibility. The default setting is "No".
Music On Hold Interpret	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.
Music On Hold Suggest	Specify which Music On Hold class to suggest to the bridged channel when putting the call on hold.
Bandwidth	Configure the bandwidth for IAX settings. The default setting is "Low".

IAX Settings/Registration

IAX Settings/Registration

IAX Registration Options	
Min Reg Expire	Configure the minimum period (in seconds) of registration. Default setting is 60.
Max Reg Expire	Configure the maximum period (in seconds) of registration. Default setting is 3600.
IAX Thread Count	Configure the number of IAX helper threads. The default setting is 10.

IAX Max Thread Count	Configure the maximum number of IAX threads allowed. The default is 100.
Auto Kill	If enabled and no ACK is received for new messages after the specified wait time, the connection will be terminated.
Authentication Debugging	If enabled, authentication traffic in debugging will not show. The default is "No".
Codec Priority	<p>Configure codec negotiation priority. The default setting is "Reqonly".</p> <ul style="list-style-type: none"> ○ Caller Consider the callers preferred order ahead of the host's. ○ Host Consider the host's preferred order ahead of the caller's. ○ Disabled Disable the consideration of codec preference all together. ○ Reqonly 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.
Type of Service	Configure ToS bit for preferred IP routing.
IAX Trunk Options	
Trunk Frequency	Configure the frequency of trunk frames (in milliseconds). The default is 20.
Trunk Time Stamps	If enabled, time stamps will be attached to trunk frames. The default is "No".

IAX Settings/Security

IAX Settings/Static Defense

Call Token Optional	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.
Max Call Numbers	Configure the maximum number of calls allowed for a single IP address.
Max Unvalidated Call Numbers	Configure the maximum number of Unvalidated calls for all IP addresses.
Max Call Numbers	Configure to limit the number of calls for a give IP address of IP range.
IP or IP Range	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.

Edit Analog Ports: Signaling Preference
Cancel Update

Port 1: Loop Start

Port 2: Loop Start

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.

ACIM Setting

ACIM Detection: Detect

Detect Option: ERL

Port 1:: 600 Ω

Port 2:: 600 Ω

FXO Ports ACIM Settings

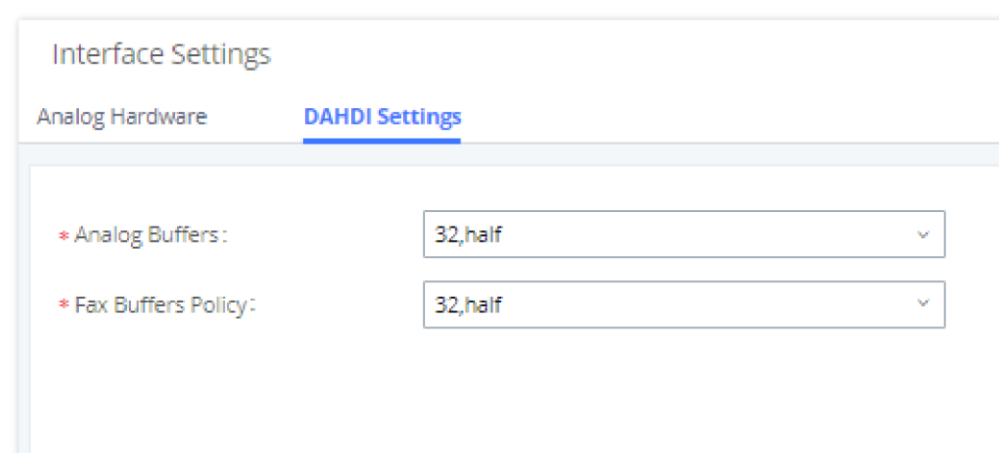
PBX Interface Settings

Tone Region	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)".
Advanced Settings	
FXO Opermode	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)".
FXS Opermode	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)".
FXS TISS Override	Configure to enable or disable override Two-Wire Impedance Synthesis (TISS). The default setting is No. If enabled, users can select the impedance value for Two-Wire Impedance Synthesis (TISS) override. The default setting is 600Ω.
PCMA Override	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. Note: This option requires system reboot to take effect.
Boost Ringer	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".
Fast Ringer	Configure to increase the ringing speed to 25HZ. This option can be used with "Low Power" option. The default setting is "Normal".

Low Power	Configure the peak voltage up to 50V during "Fast Ringer" operation. This option is used with "Fast Ringer". The default setting is "Normal".
Ring Detect	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".
FXS MWI Mode	Configure the type of Message Waiting Indicator on FXS lines. The default setting is "FSK". <ul style="list-style-type: none"> ○ FSK: Frequency Shift Key Indicator ○ NEON: Light Neon Bulb Indicator.
FXO Frequency Tolerance	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.



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.

Contacts

Address book management is under UCM web UI->Maintenance, and it has two sections "Contact Management" and "Department management".

Contact Management

Contact management page displays extension contacts and external contacts information.

- Extension contacts

Extension contacts page shows all the extensions that has "Sync Contact" option enabled in extension settings page. The extension contacts here can be edited or deleted individually or in batch. No new extension contact can be added directly from this page. If an extension contact is deleted from this page, "Sync Contact" option is disabled from this extension. This

will not delete the extension from UCM.

Note

“Delete” extension contact will only remove this extension from extension contact page and it will not sync to contacts on UCM. The extension itself still exists on UCM.

EXTENSION	NAME	DEPARTMENT	EMAIL ADDRESSES	CONTACT PRIVILEGES	OPTIONS
1000		---		All Contacts(Same as Department)	
1001	John Doe	---		All Contacts(Same as Department)	
1002	Jane Doe	---		All Contacts(Same as Department)	
1003	Arthur Morgan	---		All Contacts(Same as Department)	
1004		---		All Contacts(Same as Department)	
1005		---		All Contacts(Same as Department)	
2000		---		All Contacts(Same as Department)	
2001		---		All Contacts(Same as Department)	
2002		---		All Contacts(Same as Department)	
2003		---		All Contacts(Same as Department)	

Extension Contacts

Note

“Delete” extension contact will only remove this extension from extension contact page and it will not sync to contacts on UCM. The extension itself still exists on UCM.

Click Edit icon to configure name, department, email address and etc for each extension contact.

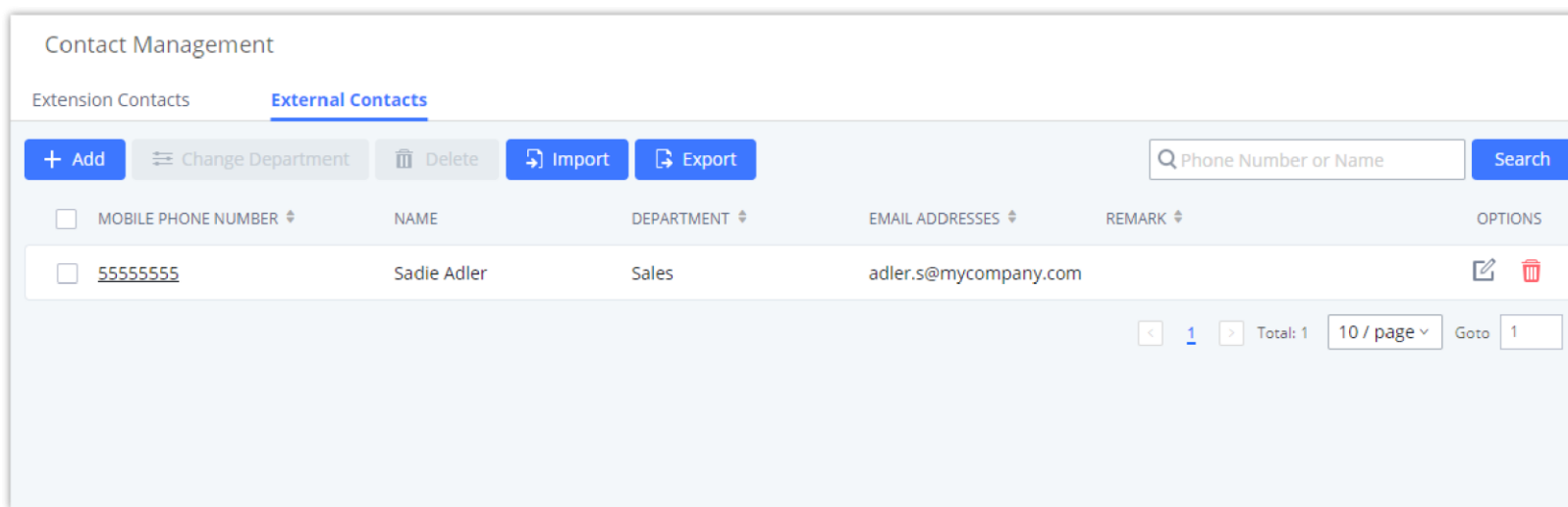
Edit Extension Contact

Extension	Displays extension number.
First Name	Configure first name for the extension contact.
Last Name	Configure last name for the extension contact.

Department	Select department for the extension contact. Department can be created in "Department Management" page.
Department Title	Configure the job title for the extension contact.
Email Address	Configure email address for the extension contact.
Mobile Phone Number	Configure mobile phone number for the extension contact.
Home Number	Configure home number for the extension contact
Fax	Configure Fax for the extension contact.
Same as Department Contact Privileges	When this option is enabled, the contact extension will inherit the same privilege as the department it belongs to.
Contact View Privileges	This option allows configuring privileges for the contact extension. Note: This option will be disabled if " Same as Department Contact Privileges " has been enabled.

- o External contacts

On external contacts page, the admin can create single external contact, import contacts in batch, edit contacts, delete contacts and export contacts.

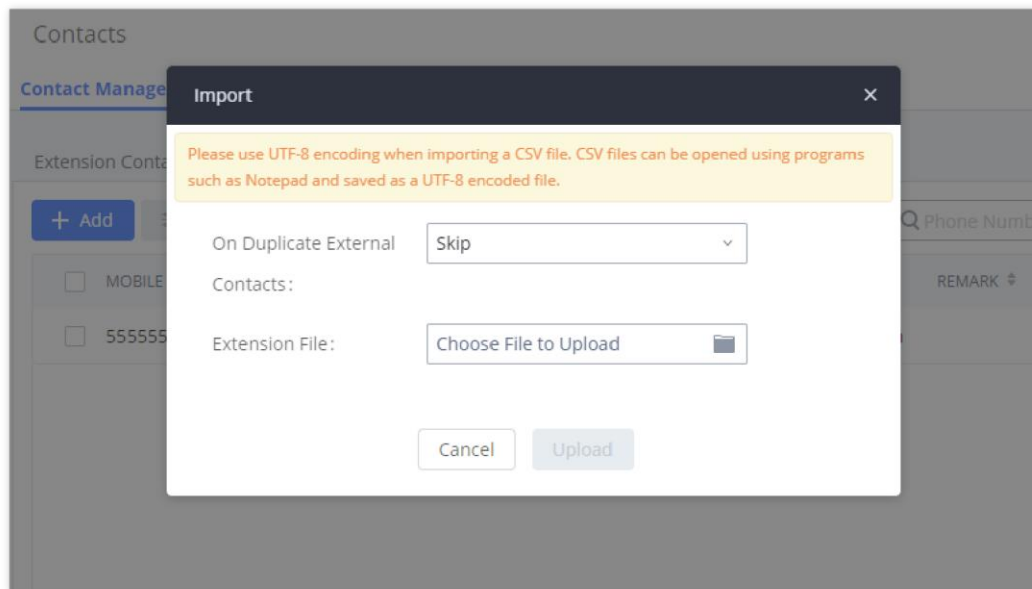


External Contacts

Click on "Export" icon, a CSV format file will be generated with the current external contacts.

Click on "import" icon, then follow the steps below to add external contacts in batch:

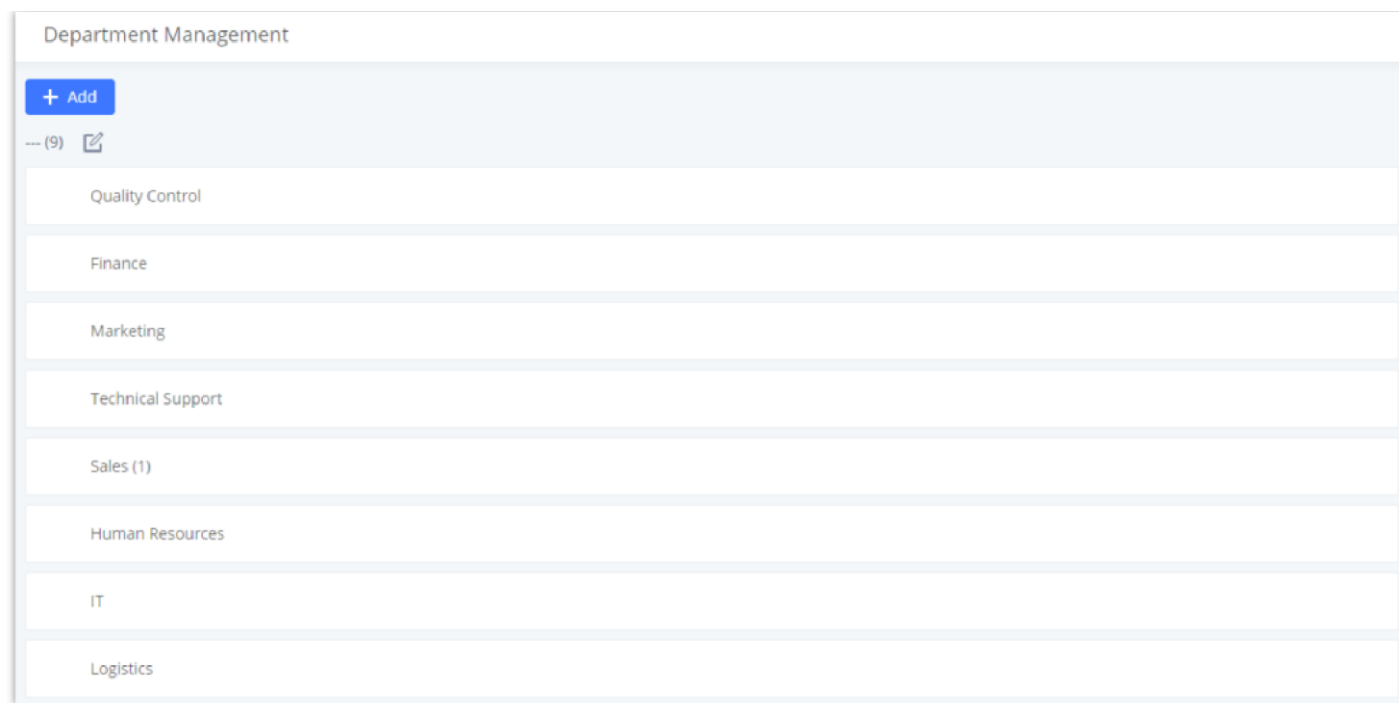
- o **Step 1:** For option "On Duplicate External Contacts", select whether to skip duplicate contact on the imported CSV file or update the duplicate UCM contact with the information in the CSV.
- o **Step 2:** Choose file from local PC to upload.
- o **Step 3:** Click on "Upload".
- o **Step 4:** Click on "Apply" to complete importing external contacts.



Import External Contacts

Department Management




Departments are organizational units that allows organizing extensions within groups that specify the specialty of a the extension owners within a company. This makes finding contacts easier within the UCM contact books.

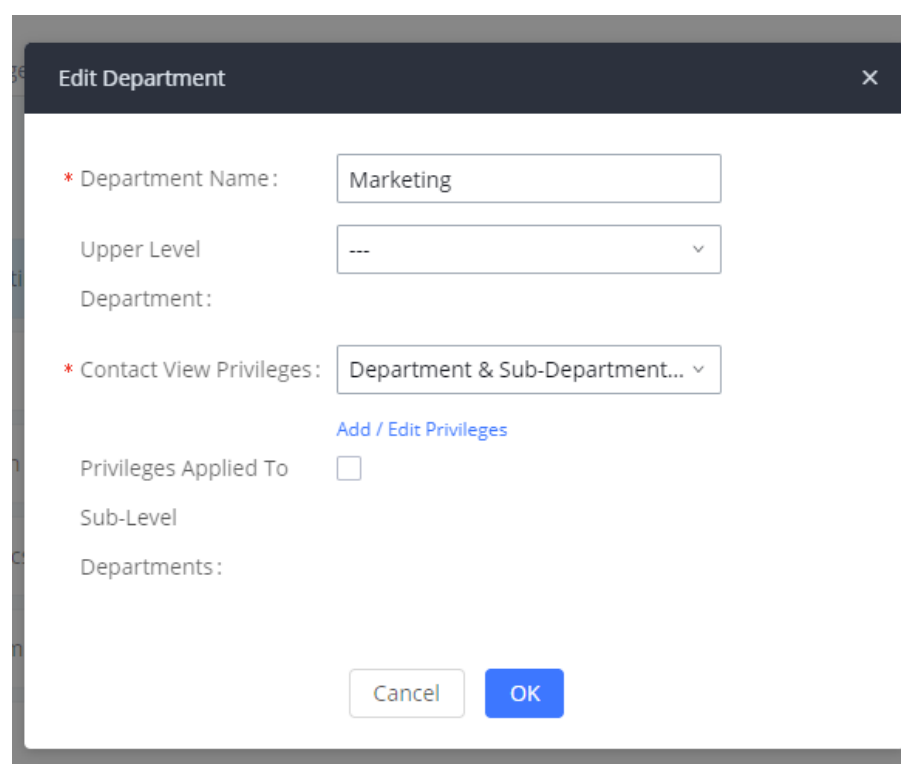


Department Management

Click on "Add" to create a new department. Configure the department name and select the superior department. By default the superior department is the root directory. If the UCM has cloud IM configured, the root directory will be the department in cloud IM.

On the department list:

- Click on  to create sub department.
- Click on  to add member to the department.
- Click on  to edit the department.



Edit Department

Department Name	Enter the name of the department.
Upper Level Department	Select the upper level department if the department being created is a nested department.
Contact View Privileges	<ul style="list-style-type: none"> ● All Contacts: The extensions in this department will be able to see all the contacts. ● Department & Sub-department Contacts: The extensions in this department will only be able to see the contacts which are in the same department or in sub-departments.
Set as Shared Department	Enable this option to share this department across the UCMs which use the same Cloud IM server. To be able to enable this option, make sure that the UCM has a RemoteConnect Plan and is correctly connected to the Cloud IM server.
Share to Following Sites	Pick the sites to which you want to share this specific department.

Note

The user can create up to 100 departments with up to 4 levels of nested departments.

Important

To be able to use shared department, the UCM devices will have to be subscribed to a RemoteConnect plan that offers Cloud IM service. For more information please refer to: <https://ucmrc.gdms.cloud/plans>

Privilege Management

The user can configure custom privileges other than the default ones (All contacts, Departments and sub-departments contacts). These custom privileges allow more flexible ways of allowing contacts to view all or specific contacts from other departments.

UCM admin can add or edit Privilege Management; under UCM web UI → **Contacts Privilege Management**, there are 2 default privileges:

- Visible to all contacts.
- Only the contact person's department and sub-department contacts are visible.

When Cloud IM is enabled on the UCM, a third privilege becomes available to choose:

- Local Contacts: Restricts the contacts shown to the contacts of the local UCM.

NAME	PRIVILEGE STATUS	OPTIONS
All Contacts	● In Use	
Department & Sub-Department Contacts	● Not Applied	

1 / page Total: 2 Goto 1

Privilege Management — Cloud IM Disabled

MESSAGING

Live Chat

Live Chat feature allows to create chat channels that can be embedded on your website to enable your client to reach your customer service more easily. The client can contact your agents by text then

Live Chat > Create New Live Chat

* Name

* Destination

Web Page Language

Visitor Information


Require Visitor Info

Privacy Control

Call Settings

Allow Visitor to Call

Customer Service Agent Information

Avatar 
Please select a file in png, jpg, jpeg format.

* Name

Show Agent's Real Name

Live Chat Configuration

Name	Configure the name of this broadcast.
Sender	Configure the sender of this broadcast.
Message Content	Enter the message to broadcast to recipients. Please keep in mind the display size of recipient endpoints as long messages may be cut off.
Receipients	Select the recipients of the broadcasted message.

Message Broadcast

Message broadcast feature allows the administrator to broadcast a text message to all the endpoints selected by the administrator. The administrator can select departments or individual extensions to boardcast a text message.

Message Broadcast > Send Message Broadcast

* Name

* Sender

* Message Content

* Recipients

Search

Company Contact

- All
- Technical Support
- Quality Control
- Sales
- IT
- Marketing
- Human Resources
- Finance
- 1000 " "

Selected(0)

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Message Broadcast Configuration

Name	Configure the name of this broadcast.
Sender	Configure the sender of this broadcast.
Message Content	Enter the message to broadcast to recipients. Please keep in mind the display size of recipient endpoints as long messages may be cut off.
Receipients	Select the recipients of the broadcasted message.

DEVICE MANAGEMENT

IPC Devices

The UCM admin can add IPC devices and edit accessible extensions so these extensions can view the surveillance streams for the IPC devices.

Click on "Add" to add IPC device.

IP Camera Devices

<input type="checkbox"/> STATUS	DEVICE NUMBER	DEVICE NAME	URL	OPTIONS
<input type="checkbox"/> ● Unmonitored	4001	GSC3620	rtsp://192.168.5.74:554	<input type="button" value="Edit"/> <input type="button" value="Delete"/>

Total: 1
10 / page
Goto

IP Camera Devices

Edit the IPC device settings in the table below.

IPC Devices Settings

Device Number	The number that allowed members can dial to access the IP camera.
Device Name	Enter the name that you want to allocate for the device.
Protocol	The media protocol that will be used.
IP Address	Enter the IP address of the IP camera.
Port	Enter the port of the IP camera. The default is 554
Channel Path	If you want to view the stream of the specified channel, please configure the path of this stream.
Username	If a username and password are set on this device, fill in this field to allow the UCM to access the device.
Password	If a username and password are set on this device, fill in this field to allow the UCM to access the device.
Transmission Protocol	Transport protocol of the IP camera. Default is UDP.
Heartbeat Detection	If enabled, the PBX will regularly send RTSP OPTIONS to check of the device is still online.
Allowed Members	Extensions, Extension Groups, and Departments can be selected to access this IP camera by dialling the configured Device Number.

Create New IP Camera Devices
Cancel Save

General

* Device Number:

* Device Name:

* Protocol:

* IP Address:

* Port:

Channel Path:

Username:

Password:

* Transmission Protocol:

Heartbeat Detection:

User Settings

* Allowed Members:

Company Contact
 All
 1005
 2004
 2003

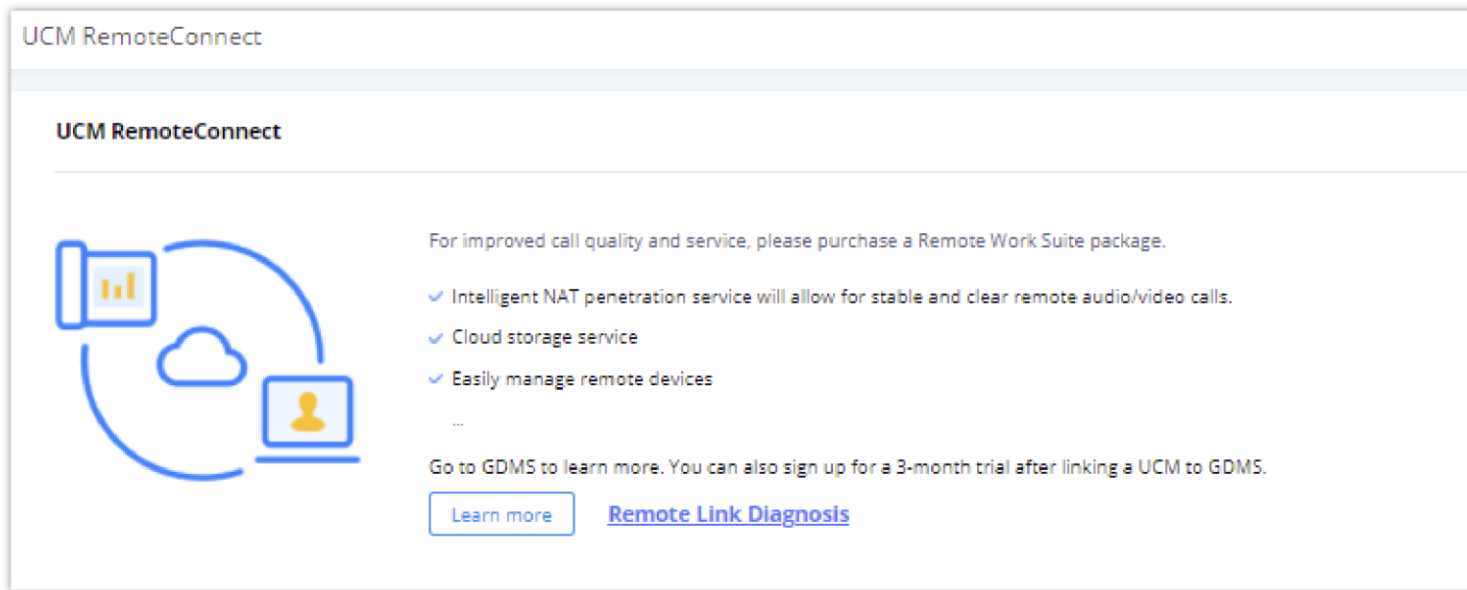
Selected(0)

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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.

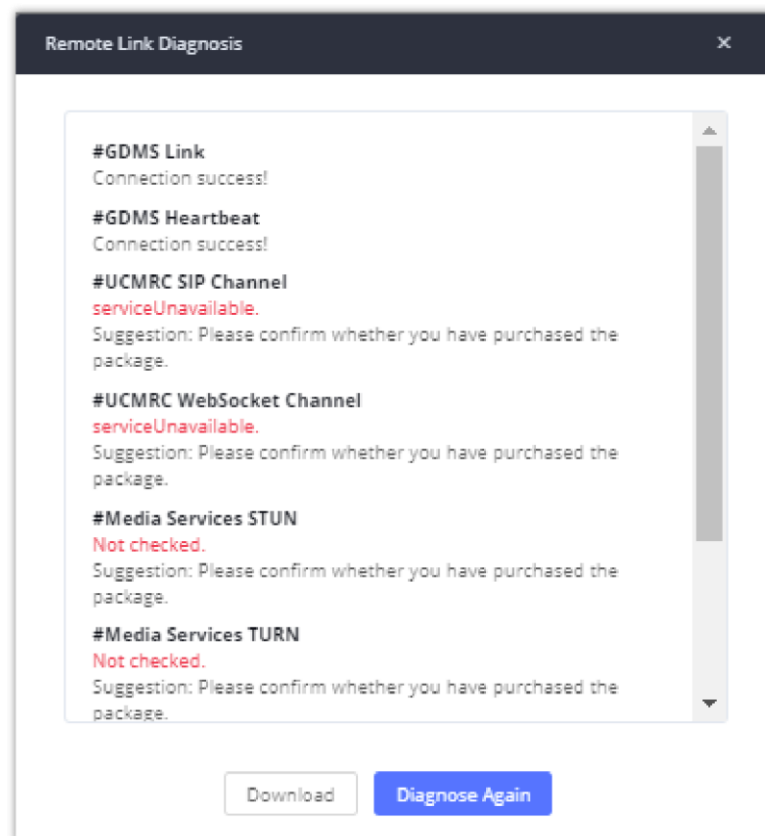
The RemoteConnect can be configured under **WebGUI→RemoteConnect** After purchasing the RemoteConnect package.



RemoteConnect Features

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.

In daily operation, the user can click the "Diagnosis" button to diagnose the remote service system. The specific diagnosis content includes media service (STUN/TURN), GDMS link and heartbeat detection, tunnel service (SIP/Web Socket), Cloud IM, UCM bandwidth speed measurement.



Remote Diagnosis

RemoteConnect

Plan Plan Settings Integrated Customer Service Enterprise UI customization

Subscription Tier:	Basic
Subscription Period:	Permanently Active
Plan Status:	Permanently Active
Max Remote Concurrent Sessions:	2
Max Remote Users:	10
Max Time Per Remote Call/Meeting:	20 minute(s)
Max Cumulative Time for Remote Calls/Meetings Per Day:	120 minute(s)
Max Cumulative Time for Remote Calls/Meetings Per Month:	Unlimited
GDMS Cloud Storage:	0 GB Upgrade
STUN Address:	nat-b.gdms.cloud
Wave RemoteConnect Address:	██████████.a.gdms.cloud
IP Endpoint/Trunk RemoteConnect Address:	██████████.a.gdms.cloud:5061
Wave 3rd Party Plug-ins:	Not supported by the current plan

UCM RemoteConnect – Plan

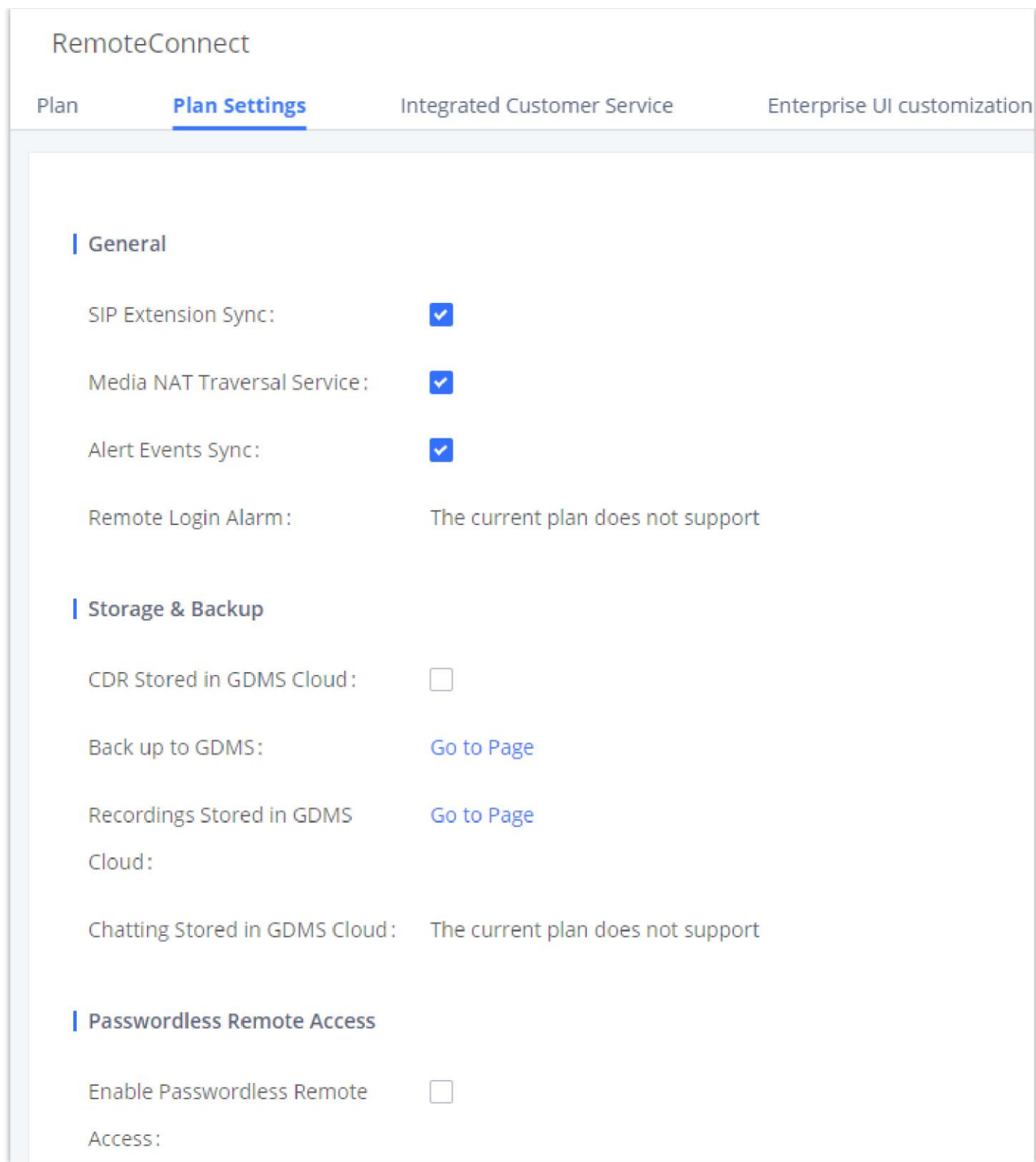
Note

- 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.

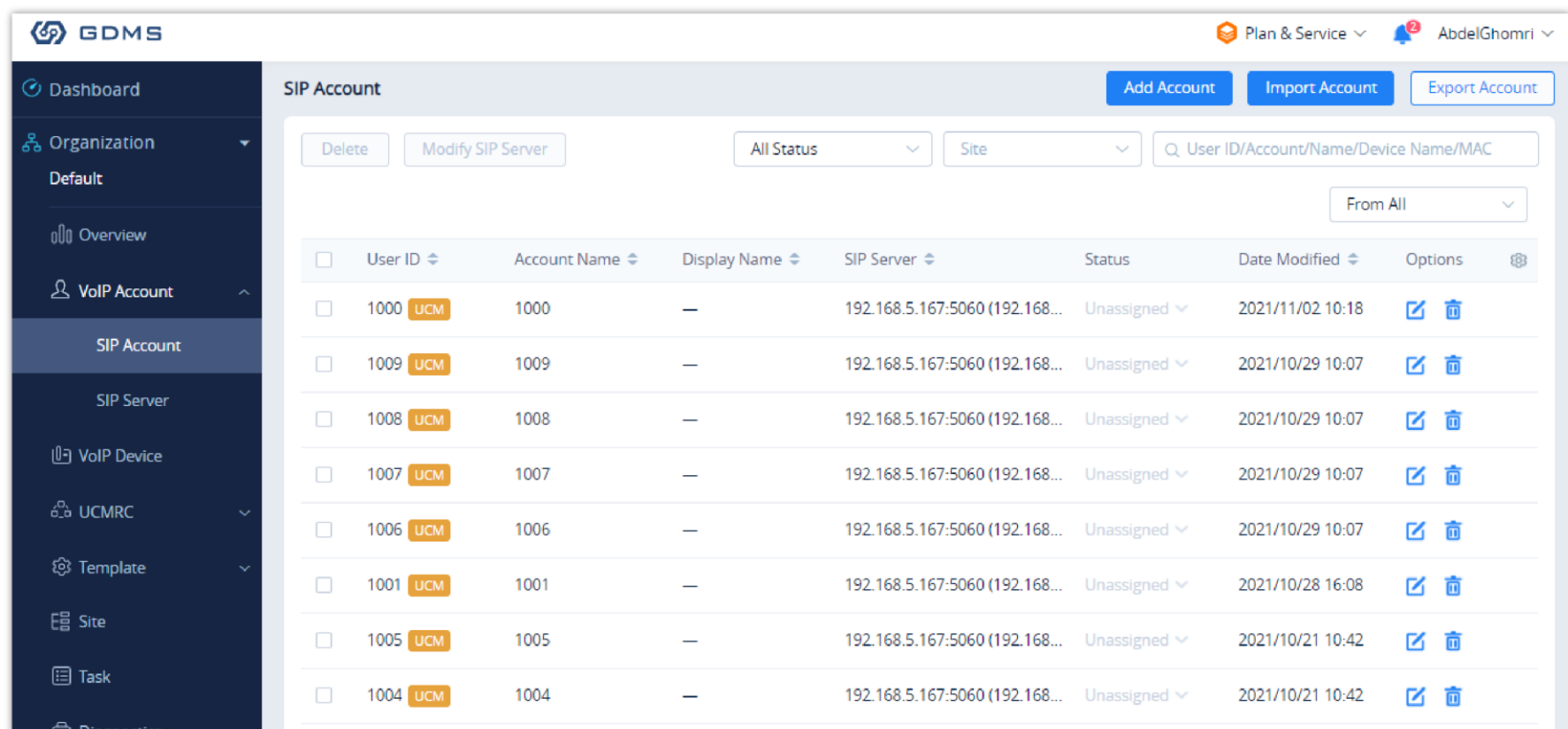
The settings are under UCM **webGUI**→**Value-added Services**→**UCM RemoteConnect**→**Plan Settings**.



UCM RemoteConnect Plan Settings

After adding UCM to the GDMS platform, UCM will synchronize all SIP extensions to the GDMS platform, this allows to use the GDMS platform for account allocation and terminal management.

The accounts synchronized to GDMS platform can be viewed on the GDMS-> VoIP Account->SIP Account page. As shown in the figure below:



UCM SIP Extensions synchronized to GDMS

The Media NAT Traversal provides a fully automatic intelligent external network penetration service to ensure that you can make normal calls/conferences on the external network.

CDR data cloud storage provides a service of dumping to GDMS to prevent CDR from continuously increasing occupying UCM storage space.

Alarm event synchronization is to synchronize the alarm information generated on UCM to the GDMS server.

UCM supports GDMS passwordless remote access. When this button is checked, GDMS remote access UCM does not need to enter the account password, and no login is required.

Allow the administrator/super administrator to open it. After clicking on open with an account, the subsequent passwordless login will use the account. All administrators and super administrators can see whether this UCM is enabled.

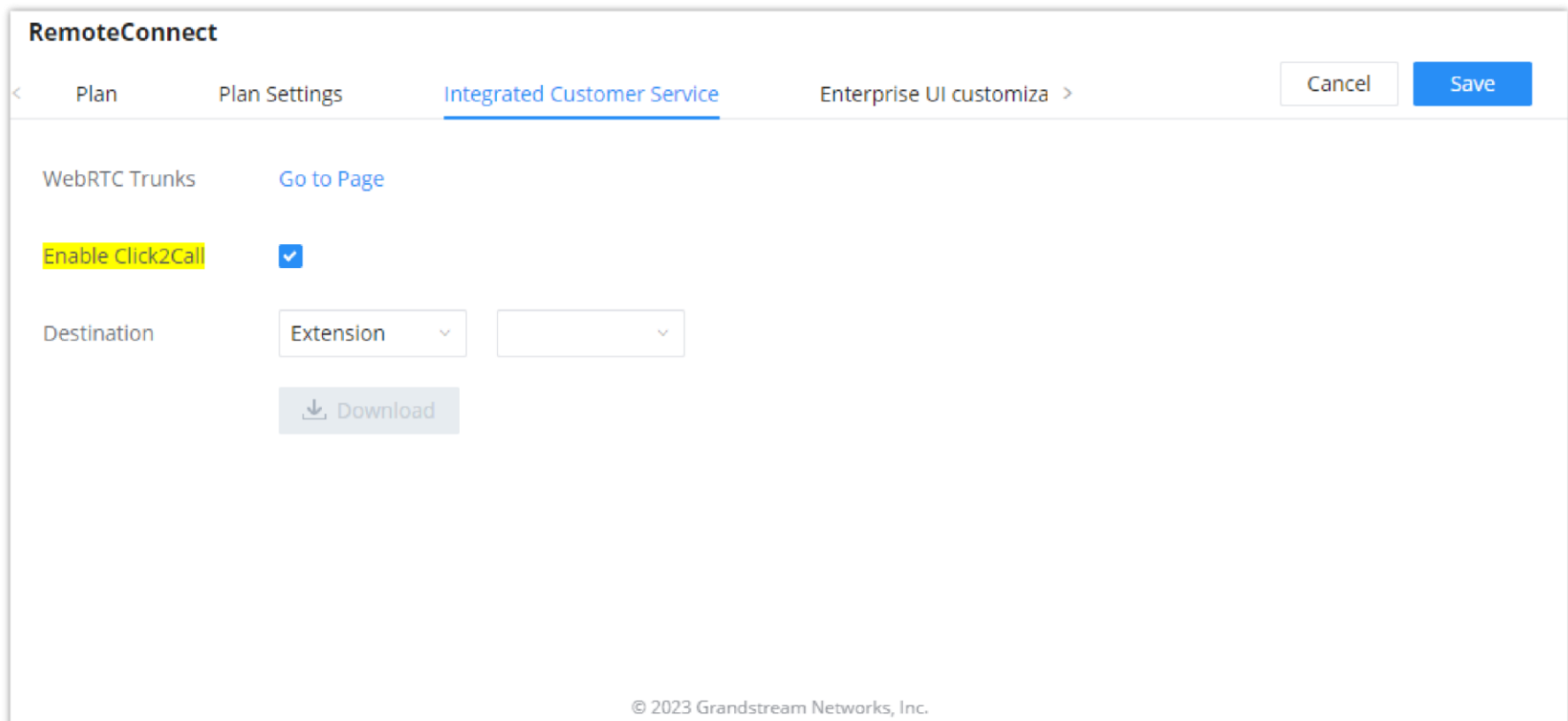
Super administrators can check and uncheck all the exemption lists; administrators can check and uncheck the exemption status of this account, and the corresponding account exemption access function will be closed after cancellation.

Notes

- Deleting an account on GDMS only removes the association between the account and the device, and does not delete the SIP account information on UCM.
- Any creation, deletion or modification of the SIP account on UCM will be automatically synchronized to the GDMS cloud platform.
- After checking the "Media NAT traversal service", the TURN service and other related traversal settings set by the user will not take effect.

Integrated Customer Service

To configure the Integrated Customer Service SDK, go to the **Other Features → UCM RemoteConnect → Integrated Customer Service SDK** page that allows users to download the SDK provided by the customer service system and integrate it on the website, so that the website can contact customer service for call operations. The call queue is used as the customer service number.



The screenshot shows the 'RemoteConnect' configuration page for 'Integrated Customer Service'. The page has a breadcrumb trail: Plan > Plan Settings > Integrated Customer Service > Enterprise UI customiza >. There are 'Cancel' and 'Save' buttons in the top right. The main content area includes:

- 'WebRTC Trunks' with a 'Go to Page' link.
- 'Enable Click2Call' checkbox, which is checked.
- 'Destination' section with a dropdown menu set to 'Extension' and an empty input field.
- A 'Download' button with a download icon.

At the bottom, there is a copyright notice: © 2023 Grandstream Networks, Inc.

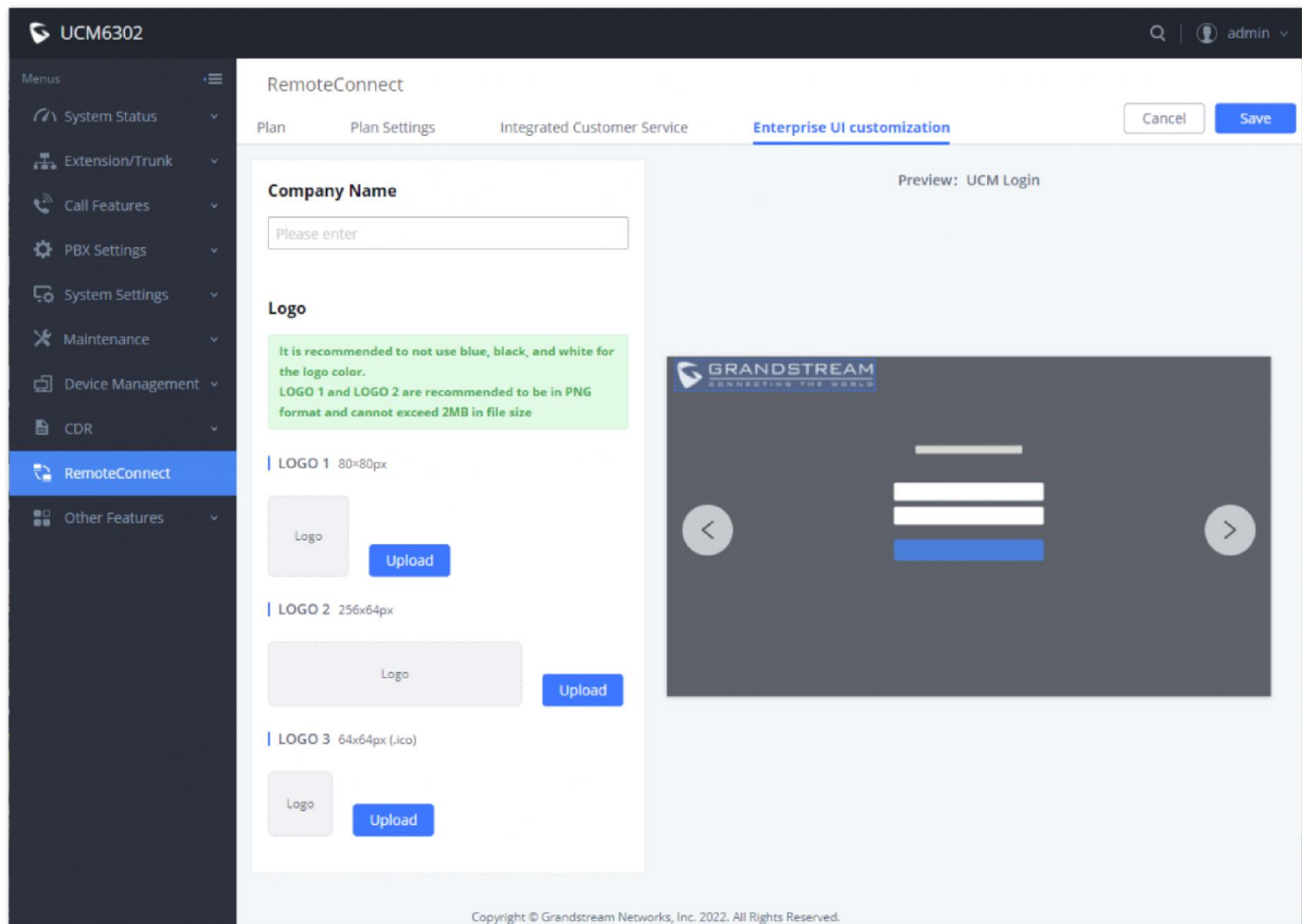
Integrated Customer Service interface

Enabling Click2Call will allow users to initiate a direct call from the web browser by clicking on the call button embedded on the website graphical interface. The calls initiated can be directed to call queues or a specific extension.

Enterprise UI Customization

With a remote connect plan, on the value-added service → UCM RemoteConnect → UI Customization page, users can edit the company name and select a local image file as the new logo. The company name acts on the text part with the trend logo, and the pictures are in different formats and sizes according to the logo position, which are 64*64px (only ico format is supported), 256*256px, 80*80px, which supports users in the "UCM management platform/login", "Reset Password", "Email Template", "Wave_PC", "Wave Login", "Browser Label", "Guide Page" interface preview.

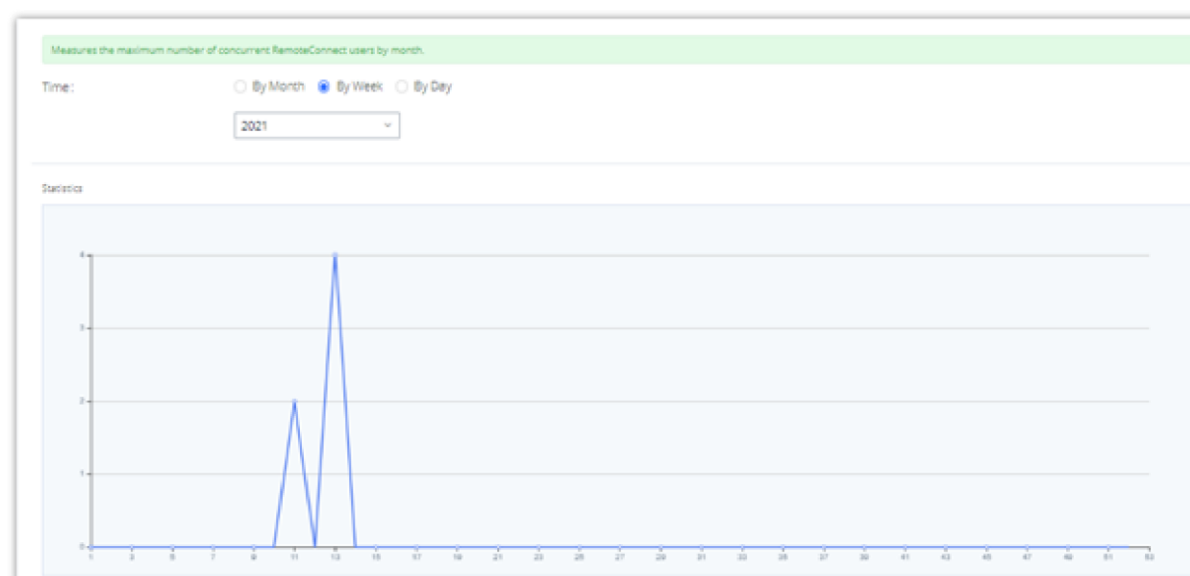
- o LOGO 1: Replaces Browser tab icon
- o LOGO 2: Replaces the Grandstream banner on the top left corner of the management login page and emails.
- o LOGO 3: Replaces the Grandstream logo on the top left corner of the Wave Web interface and UCM management interface.



UI Customization

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 → **RemoteConnect** → **Statistics** page.

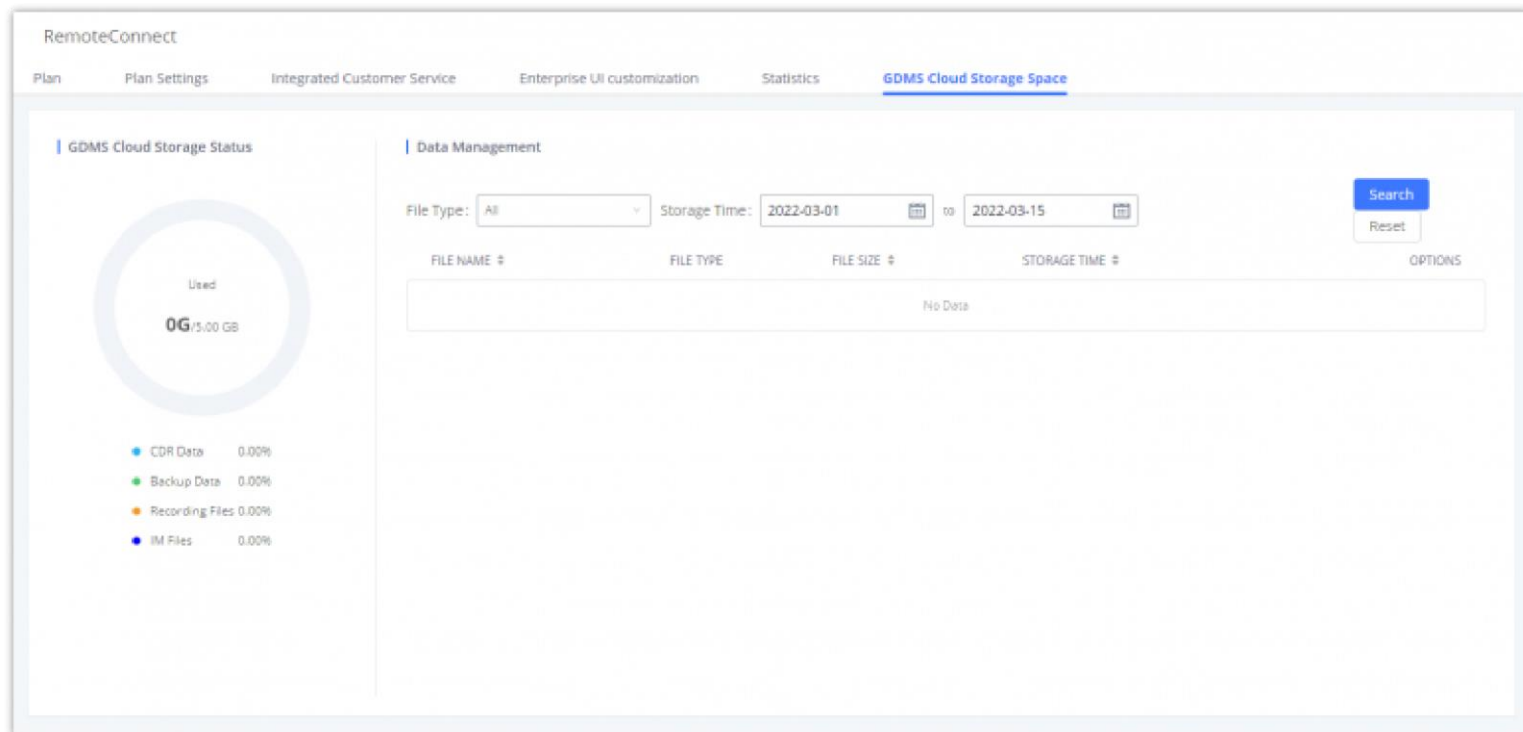


Concurrent Remote Calls

For more information, please visit <http://ucmrc.gdms.cloud/intro.html> and read our [UCM63XXA RemoteConnect guides](#)

GDMS Cloud Storage Space

GDMS Cloud Storage Space feature on the UCM630x offers an overview about how you are using the storage space offered by RemoteConnect. It displays the amount of storage occupied, the amount of free space, also the percentage taken by each type of files. The type of files displayed are the following: CDR Data, Backup Data, Recording Files, and IM Files.



GDMS Cloud Storage Space

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 **Other Features**→**API Configuration**. The API configuration parameters are listed in the table below.

Note: The old version of the API interface only supports cdrapi, recapi and pmsapi functions, and will be removed, please use the new HTTPS API instead.

Configuration Parameters (New)

HTTPS API Settings (New)	
Enable	Enable/Disable API. The default setting is enable.
Username	Configure the username for API Authentication.
Password	Configure the password for API Authentication.
Call Control	If enabled, 3 rd party applications will be able to manage inbound calls via API actions. acceptCall will accept incoming calls while refuseCall will reject them. If no actions are done within 10 seconds, calls will automatically be accepted.
Permitted IP (s)	Sets an IP address Access Control List (ACL) for addresses that are allowed to authenticate as this user. By default this is not set, meaning all IP addresses will be allowed. The format is: "xxx.xxx.xxx.xxx/255.255.255.255".

Configuration Parameters (Old)

HTTPS API Settings (Old)

Basic Settings	
Enable	Enable/Disable API. The default setting is disabled.
TLS Bind Address	Configure the IP 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 8443. 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. The default setting is 0.0.0.0:8443.
Username	Configure the username for TLS authentication.
Password	Configure the password for TLS authentication.
Permitted IP(s)	Specify a list of IP addresses permitted to use the API. This creates an API-specific access control list. Multiple entries are allowed. For example, "192.168.40.3/255.255.255.255" denies access from all IP addresses except 192.168.40.3. By default, this is blank, which indicates that no IP addresses are allowed to use this API.
Other Settings	
TLS Private Key	Upload TLS private key. The size of the key file must be under 2MB. This file will be renamed as 'private.pem' automatically.
TLS Cert	Upload TLS cert. The size of the certificate must be under 2MB. This is the certificate file (*.pem format only) for TLS connection. This file will be renamed as "certificate.pem" automatically. It contains private key for the client and signed certificate for the server.
API Module	
CDR API	Enable/disable CDR API module.
REC API	Enable/disable REC API module.
PMS API	Enable/disable PMS API module.

For more details on CDR API (Access to Call Detail Records), REC API (Access to Call Recording Files) and PMS API, please refer the document in the link here:

- <https://documentation.grandstream.com/knowledge-base/cdr-rec-api/>
- <https://documentation.grandstream.com/knowledge-base/cdr-rec-api/>
- [PMS API](#)

API Queries Supported

The new API supports the queries listed below which will accomplish certain requests and get data about different modules on UCM630xA.

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
callbarge
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

cleanTerminalChatInformation
getSIPAccountQR
getCallQueuesMemberMessage
getQueueCalling

API Configuration Parameters

CDR Real-time Output Settings	
Enable	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.
Server Address	CDR server IP address
Port	CDR server IP port
Upload Prompts User Configuration	
Username	Username used to upload prompts.
Password	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 **Other 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 interface for 'API Configuration'. It has three tabs: 'HTTPS API Settings(New)', 'CDR Real-time Output Settings', and 'Upload Prompts User Configuration' (which is selected). At the top right are 'Cancel' and 'Save' buttons. Below the tabs, the 'Upload Prompts User Configuration' section contains two required fields: '* Username:' with the value 'uploader' and '* Password:' with the value 'uploader123'.

Upload Prompt User Configuration

1. Hash the password of the user configured to an MD5 Encryption format.
2. 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.
3. 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
```

1. 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

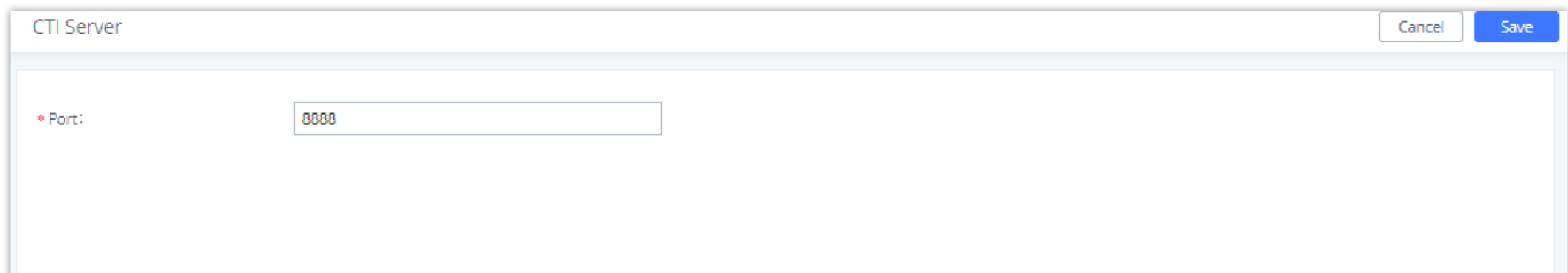
If the File name on the HTTP/HTTPS request exists already on the UCM's Custom voice prompts list the existing file will be overwritten by the new file downloaded from the FTP server.

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→**Other Features**→**CTI Server** as shown on below screenshot:



CTI Server Listening port

More information about GS affinity and CTI Support on Grandstream products series please refer to the following link:

<https://documentation.grandstream.com/knowledge-base/gs-affinity-user-guide/>

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→**Other Features**→**AMI**. For details on how to use AMI on UCM630xA, please refer to the following AMI guide:

<https://documentation.grandstream.com/knowledge-base/ami-asterisk-management-interface/>

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.

Sugar CRM

Configuration page of the SugarCRM can be accessed via admin login, on the UCM WebGUI→**Other Features**→**CRM**.

Sugar CRM Basic Settings

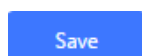
1. Select "Sugar CRM" from the CRM System Dropdown in order to use SugarCRM.

Sugar CRM Settings

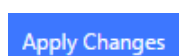
CRM System	Select a CRM system from the dropdown menu, four CRM systems are available: SugarCRM, vTigerCRM, ZohoCRM (v1&v2), Salesforce and ACT! CRM.
CRM Server Address	Enter the IP address of the CRM server.
Add Unknown Number	Add the new number to this module if it cannot be found in the selected module.
Contact Lookups	Select from the " Available " list of lookups and press <input type="radio"/> <input type="radio"/> to select where the UCM can perform the lookups on the CRM tables, Leads, Accounts, and Contacts.

Once settings on admin access are configured:

1. Click on



and



2. Logout from admin access.

3. 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 form titled "CRM User Settings". It contains the following fields:

- Enable CRM:** A checkbox that is checked with a blue checkmark.
- * Username:** A text input field containing "GStest".
- * Password:** A text input field containing "password@123".
- Login Status:** A label with no input field.

CRM User Settings

Vtiger CRM

Configuration page of the VtigerCRM can be accessed via admin login, on the UCM WebGUI→Other Features→CRM.

The screenshot shows a form titled "CRM". It contains the following fields:

- CRM System:** A dropdown menu with "VtigerCRM" selected.
- * CRM Server Address:** A text input field containing "http://vtiger.mydomain.com".
- * Add Unknown Number:** A dropdown menu with "Organizations" selected.
- Contact Lookups:** A list management interface showing "0 item Available" on the left and "3 items Selected" on the right. The selected items are:
 - Look up in Organizatio...
 - Look up in Leads table
 - Look up in Contacts ta...

Vtiger CRM Basic Settings

1. Select "Vtiger CRM" from the CRM System Dropdown in order to use Vtiger CRM.

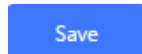
Vtiger CRM Settings

CRM System	Select a CRM system from the dropdown menu, four CRM systems are available: Sugar CRM, Vtiger CRM, Zoho CRM (v2), Salesforce and ACT! CRM.
CRM Server Address	Enter the IP address of the CRM server.
Add Unknown Number	Add the new number to this module if it cannot be found in the selected module.

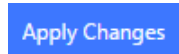
Contact Lookups	<p>Select from the "Available" list of lookups and press</p> <ul style="list-style-type: none"> <input type="radio"/> <input type="radio"/> <p>to select where the UCM can perform the lookups on the CRM tables, Leads, Organizations, and Contacts.</p>
------------------------	--

Once settings on admin access are configured:

1. Click on



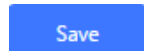
and



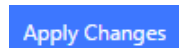
2. Logout from admin access.

3. 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.

CRM User Settings

Enable CRM:

* Username:

* Password:

Login Status:

CRM User Settings

Salesforce CRM

Configuration page of the Salesforce CRM can be accessed via admin login, on the UCM Web GUI→**Other Features**→**CRM**".

CRM

CRM System:

* Add Unknown Number:

Contact Lookups:

1 item Available		2 items Selected
<input type="checkbox"/> Look up in Leads table	<input type="button" value="←"/> <input type="button" value="→"/> <input type="button" value="↑"/> <input type="button" value="↓"/>	<input type="checkbox"/> Look up in Contacts table <input type="checkbox"/> Look up in Accounts table

Salesforce Basic Settings

1. Select "Salesforce" from the CRM System Dropdown in order to use Salesforce CRM.

Once settings are configured:

2. Click on **Save** and **Apply Changes**.
3. Logout from admin access.
4. Login to the UCM as user and navigate under **"User Portal→Other Features→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.

Salesforce User Settings



Salesforce CRM

Configuration page of the Salesforce CRM can be accessed via admin login, on the UCM Web GUI→**Other Features→CRM"**.

Salesforce Basic Settings

1. Select "Salesforce" from the CRM System Dropdown in order to use Salesforce CRM.

Salesforce Settings

CRM System	Select a CRM system from the dropdown menu, four CRM systems are available: SugarCRM, vTigerCRM, ZohoCRM (v1&v2), Salesforce and ACT! CRM.
Add Unknown Number	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 are configured:

1. Click on

Save

and

Apply Changes

2. Logout from admin access.

3. Login to the UCM as user and navigate under "User Portal→Value-added Feature→CRM User Settings".

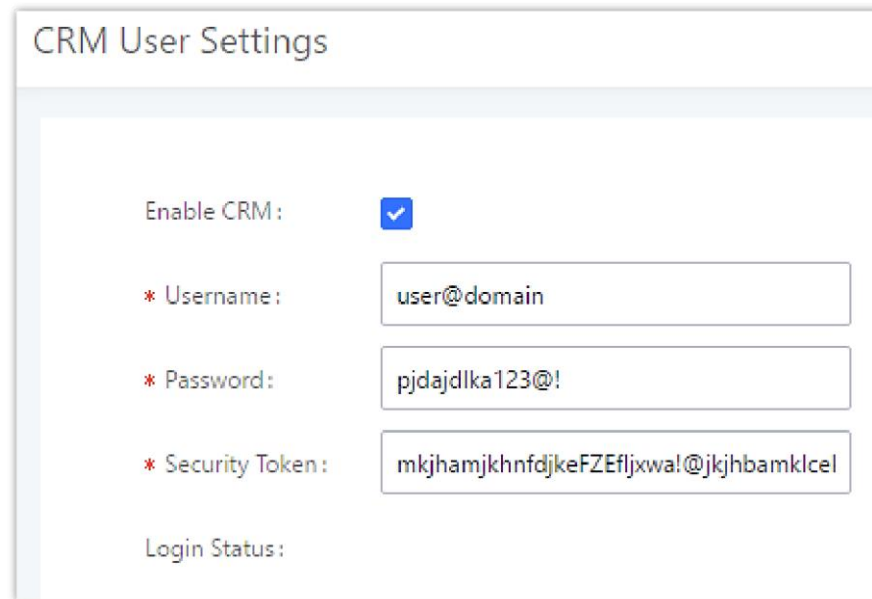
Click on "Enable CRM" and enter the **username**, **password** and **Security Token** associated with the CRM account then click on

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 a web form titled "CRM User Settings". It contains the following fields and values:

- Enable CRM:
- * Username: user@domain
- * Password: pjdajdka123@!
- * Security Token: mkjhamjkhndjkeFZEfljxwa!@jkjhbamklcel
- Login Status:

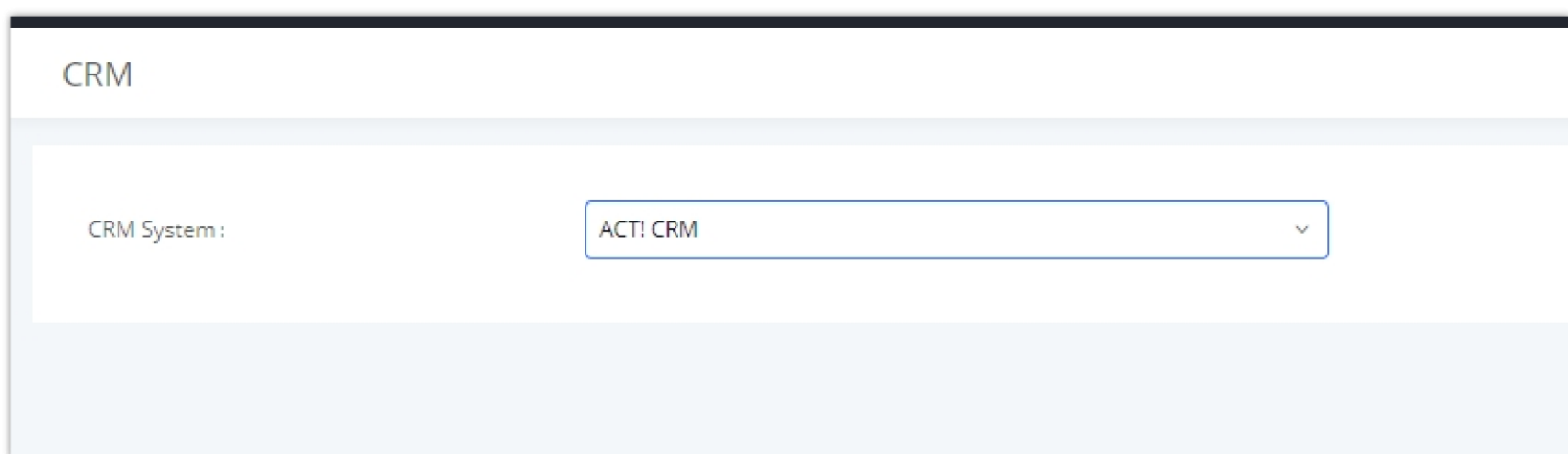
Salesforce User Settings

ACT! CRM

Configuration page of the ACT! CRM can be accessed via admin login, on the UCM Web GUI→Other **Features**→**CRM**".

The configuration steps of the ACT! CRM are as follows:

1. Navigate to Other Features→CRM and select the "ACT! CRM" option.



The screenshot shows a web form titled "CRM". It contains a dropdown menu labeled "CRM System:" with "ACT! CRM" selected.

Enabling ACT! CRM

2. Log into the UCM as a regular user and navigate to **Other 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>

CRM User Settings

Enable CRM:

Username:

Password:

Login Status:

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→**Other 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:

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.

PMS Software

PMSI Middleware

(HMobile Connect)

Grandstream UCM

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

Notes:

1. 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.
2. 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
 - cfw – 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.

1. 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.

PMS

[Basic Settings](#)

PMS Module HSC ▾

Protocol Type HTTP ▾

* UCM Port 8081

* Username

* Password

Sync Guest Name to Phone

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.

PMS

[Basic Settings](#)

PMS Module Mitel ▾

Wakeup Prompt Wake Call ▾ [Upload Audio File](#)

Room Status Update Prompt Default Room Status Update Prom... ▾ [Upload Audio File](#)

* UCM Port 8081

Back Up Voicemail Recordings

Sync Guest Name to Phone

Automatically Clear Wakeup Calls None ▾

Automatically Clear Wave Chat History None ▾

Automatically Reset User/Wave Password

Mitel PMS

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.

PMS

Basic Settings

PMS Module

Wakeup Prompt

Room Status Update Prompt

* UCM Port

Back Up Voicemail Recordings

Sync Guest Name to Phone

Automatically Clear Wakeup Calls

Automatically Clear Wave Chat History

Automatically Reset User/Wave Password

IDS PMS

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://documentation.grandstream.com/knowledge-base/https-api/>, Pmsapi section.

Connecting to PMS

On the UCM WebGUI→**Other Features**→**PMS**→**Basic Settings**” set the connection information for the PMS platform.

PMS Basic Settings

Field	Description
PMS Module	<p>Users can select the desired PMS module from the drop-down list.</p> <ul style="list-style-type: none"> • Hmobile • Mitel • HSC • IDS • PMS API
Wakeup Prompt	<p>A customized prompts that can be played when the wakeup call is answered. To customize it please navigate to PBX Settings → Voice Prompt → Custom Prompt</p>
Username	<p>This username is used to authenticate into the PMS API.</p>

Password	This password is used to authenticate into the PMS API.
Back Up Voicemail Recordings	Back up voicemail recordings to external storage after check-out.
Automatically Clear Wakeup Calls	<p>Scheduled wakeup calls for rooms can be cleared upon checking in or checking out.</p> <ul style="list-style-type: none"> • None: The wakeup calls won't be automatically cleared. • Check out: The wake up calls assigned to the guest will be cleared when they check out. • Check In: The wake up calls assigned to a guest will be cleared when a new client checks in.
Automatically Clear Wave Chat History	If enabled, room Wave chat history will be automatically cleared upon check-in or check-out.
Automatically Reset User/Wave Password	If enabled, the User/Wave password of the room extension will be automatically reset to a random password upon check-out.

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 Management

In Room Management tab, the user can create a room and affect up to two extensions to it. This will appear in Room Status tab, and from there the user can change the Check-in/Check-out

Create New Room
Cancel Save

* Address:

* Room Number:

* Extension 1:

* Extension 2:

* Call Privileges:

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Call Privileges allows the administrator to set the level of call privilege of the room.

Room Status

On this tab the user can check-in and check-out guests.

Check In
✕

Once a customer has checked in, please manage customer information through the PMS system. Please do not modify names, languages, or calling privileges through the UCM system.

Room Number

* Room Status

First Name

Last Name

Guest Account

Guest Category Code

Guest Credit Money

* Arrival Date Select date Select time

* Expected Departure Date Select date Select time

Language

* Call Privileges

Check-in a Client

After clicking "OK" the client entry will be added to the list.

PMS						
Basic Settings	Room Management	Room Status	Call Rate	Wakeup Service	Mini Bar	Housekeeper
Check-in/Check-out History		Custom Room Status Codes				
Room Number	Check-In Status	Check In / Check Out	Room Status	Customer Name	Options	
1000	● Checked in	Check Out	Available		<input type="button" value="🏠"/> <input type="button" value="💰"/>	

Total: 1 < 1 > 10 / page Goto

Room Status

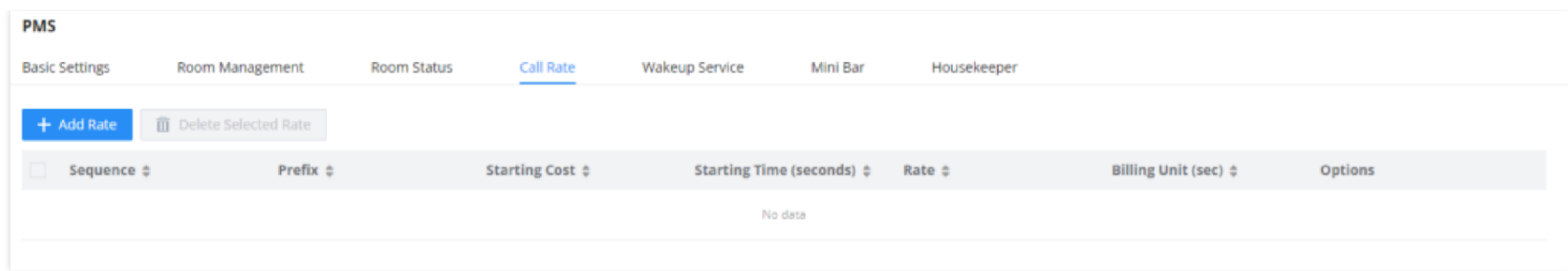
The user can click on **Check-in/Check-out Records** to view the history of the checked-in and checked-out guests.

Note

The Call Privilege configured during a guest's check-in will be reset to the room's default call privilege upon guest check-out.

Call Rate

In Call Rate page, the user can create different call rates for different call types. For example, the user can create a call rate which applies to national calls. The call rates can be differentiated by the prefix set for each call rate, the prefix corresponds to an outbound route pattern which allows national calls. Thus, the call rate applies accordingly.



PMS – Call Rate

Add Call Rate
✕

$Call\ Charge = Starting\ Cost + Rate \times Billing\ Unit$

Prefix

Starting Cost

Starting Time (seconds)

* Rate

* Billing Unit (sec)

Add Call Rate

Prefix	Enter the prefix to be used for outgoing calls that should correspond with an outbound route pattern. If left blank, outgoing calls will not require a prefix, and any number can be dialed.
Starting Cost	Configure the device role. When set as a media server, This UCM's PBX-related features will be disabled.
Starting Time (seconds)	Sets the starting time period for call billing. If the length of a guest's external call does not exceed the starting time, only the starting cost amount will be charged. Example: If the starting cost is set to 0.2, and the starting time is set to 60, the first 60 seconds of a call will be charged a flat amount of 0.20 dollars (or other currency). If the starting time is set to 0 instead, the first 60 seconds will be free.
Rate	Sets the billing rate of a call after the starting time period has ended. This is used with Billing Unit (sec) to calculate the cost of a call (Rate x Billing Unit = Telephone Cost).
Billing Unit (sec)	Sets the billing unit used after the starting time period has ended. This is used with Rate to calculate the cost of a call (Rate x Billing Unit = Telephone Cost). Partial units are rounded up (e.g., If the billing unit is set to 60 seconds, and the call lasted 90 seconds (1.5 units), the guest will be billed for 120 seconds (2 units)).

Note

The user can create up to 500 call rate entries.

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:

Create a New Wakeup Service

PMS Wake up Service

Field	Description
Room Number	Select the room number where to call with a limitation of 63 characters.
Select Time	Set the time of the wakeup call
Action Status	<p>Show the status of the call:</p> <ul style="list-style-type: none"> ○ Programmed: the call is scheduled for the time set ○ Cancelled: the call is canceled ○ Executed: the wakeup call is made <p>Note: Editing an already executed wakeup service will automatically change the service's status to "Programmed".</p>
Type	<ul style="list-style-type: none"> ○ Single: The call will be made once on the specific time. ○ Daily: The call will be repeated every day on the specific time

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	

Wakeup Call executed

Mini Bar

The mini bar feature is used to track the goods which have been consumed by the guests during their stay. This feature allows to add the consumed goods to the bill. Adding the goods to the bill can be done by the housekeeper

Enable Mini Bar
 Increase Mini Bar Usage Code
 Decrease Mini Bar Usage Code
 Global Tax Rate (%)
 Prompt
 Skip Housekeeper and Password Authentication
 Enable Multi-Item Billing

Code ↕	Name ↕	Price (\$) ↕	Tax rate (%) ↕	Options
No data				

Mini Bar

Enable Mini Bar	If enabled, feature codes can be used to increase and decrease usage of Mini Bar items.
Increase Mini Bar Usage Code	Dial this code + the item code to increase usage of the Mini Bar item for billing purposes.
Decrease Mini Bar Usage Code	Dial this code + the item code to reduce usage of the Mini Bar item for billing purposes.
Global Tax Rate (%)	Set the tax rate and configure it for an additional tax charge. If no personal tax is configured for a commodity, the global tax rate of the Mini Bar will prevail.
Prompt	This tone will be played when a housekeeper dials a number to enter the Mini Bar and can be used to indicate the corresponding goods code.
Skip Housekeeper and Password Authentication	If enabled, the default housekeeper code is 0000.
Enable Multi-Item Billing	If enabled, users can enter multiple goods in a single call by separating each good code with star (*).

Local PMS

UCM6300 series offers a local Property Management System to give the user basic management features without having to purchase a PMS for the most basic property management actions. In addition to Room Management, Rooms Status for checking-in and checking-out, Wakeup Service, Mini Bar, and housekeeper functions, the UCM6300 allows several additional functions upon checking-out, like backing up voicemail recordings, clearing wakeup calls and Wave history automatically, in addition to resetting Wave's password. The user can use the Local PMS feature to check-in and check-out clients from the web user interface.

PMS

[Basic Settings](#) Room Management Room Status Call Rate Wakeup Service Mini Bar Housekeeper

PMS Module: Local PMS

Wakeup Prompt: Wake Call [Upload Audio File](#)

Room Status Update Prompt: Default Room Status Update Pro... [Upload Audio File](#)

Back Up Voicemail Recordings:

Sync Guest Name to Phone:

Automatically Clear Phone Call History: None

Automatically Clear Wakeup Calls: None

Automatically Clear Wave Chat History: None

Automatically Reset User/Wave Password:

Review Bill at Check-Out:

Currency Unit: Dollar: \$

[Cancel](#) [Save](#)

Local PMS

Parameter	Description
Wakeup Prompt	A customized prompts that can be played when the wakeup call is answered. To customize it please navigate to PBX Settings → Voice Prompt → Custom Prompt
Room Status Update Prompt	Choose a previously uploaded prompt or upload the prompt which will be played when the room status is changed. If the room status codes have been change, please update the Room Status accordingly.
Back Up Voicemail Recordings	Back up voicemail recordings to external storage after check-out, When Enabled, The user can set the SFTP server for storage purposes by defining the following attributes : <ul style="list-style-type: none"> ● Email Address: Configure the email address to send the backup to. ● Account: Configures the account on the SFTP server. ● Password: Defines the account password ● Server Address: Defines the SFTP server address (e.g., xxx.xxx.xxx.xxx:22). ● Destination Directory: Specify the directory in SFTP server to save the voicemail recordings to. Format: "xxx/xxx/xxx". If this directory does not exist, UCM will create this directory automatically. ● Test the Connection: This option tests the connection to the SFTP server defined.
Sync Guest Name to Phone	Provisions the name of the checked-in guests to endpoints via Zero Config. Requires endpoints to be discoverable and provisionable by Zero Config.
Automatically Clear Phone Call History	Configures whether or not the call history of phones will be automatically cleared upon check-in or check-out. Currently only supported on Grandstream phones. <ul style="list-style-type: none"> ● None: Call history will not be deleted after checking-in or checking-out. ● Check Out: Call history will be delete when the guest checks-out. ● Check In: Call history will be delete when a new guest checks-in.
Automatically Clear Wakeup Calls	Scheduled wakeup calls for rooms can be cleared upon checking in or checking out. <ul style="list-style-type: none"> ● None: The wakeup calls won't be automatically cleared. ● Check out: The wake up calls assigned to the guest will be cleared when they check out. ● Check In: The wake up calls assigned to a guest will be cleared when a new client checks in.

Automatically Clear Wave Chat History	If enabled, room Wave chat history will be automatically cleared upon check-in or check-out.
Automatically Reset User/Wave Password	If enabled, the User/Wave password of the room extension will be automatically reset to a random password upon check-out.
Review Bill at Check-Out	If enabled, a pop-up window with all the charges will appear during guest check-out for reviewing purposes.
Currency Unit	The currency unit for the call rate.

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→**Other 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:

3 items Available

Search

- 1002
- 1003
- 1004

2 items Selected

Search

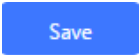
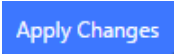
- 1000
- 1001

Create a New Wakeup Service

1. Fill out the required fields and select the members to add to the wakeup group.

Enable Wakeup Service	Enable Wakeup service.
Name	Enter a name (up to 64 characters) to identify the wakeup service.
Prompt	Select the prompt to play for that extension.
Custom Date	If disabled, users can select a specific date and time. If enabled users can select multiple days of the week to perform the wakeup.
Date	Select the date or dates when to performs the wakeup call.
Time	Select the time when to play the wakeup call.
Members	Select the members involved within the wakeup service group.

Wakeup Service

1. Click  and  to apply the changes.

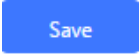
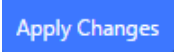
A wakeup service entry is created. The UCM will send a wakeup call to every extension in the member list at the scheduled date and time.

Note: the wakeup service has the following limitation on how many members can be added depending on UCM model.

UCM Model	Max members in a Wakeup Service
UCM6300A	50
UCM6302A	100
UCM6304A	150
UCM6308A	200

Max Wakeup Members

Wake Up Service from User Portal

1. Login to the user portal on the UCM630xA.
2. Wake Up service can be found under "**Other 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  and  to apply the changes.

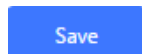
Wake Up Service using Feature Code

1. Login to the UCM as admin.
2. Enable "Wakeup Service" from the WebGUI under "**Call Features**→**Feature Codes**".

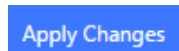
* Listen Spy:	<input type="text" value="*54"/>		* Whisper Spy:	<input type="text" value="*55"/>	
* Barge Spy:	<input type="text" value="*56"/>		* Wakeup Service:	<input type="text" value="*36"/>	<input checked="" type="checkbox"/>
* PMS Wakeup Servi...	<input type="text" value="*35"/>	<input checked="" 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"/>			

Feature Codes

1. Click



and



to apply the changes.

2. 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.

ANNOUNCEMENT CENTER

The UCM630xA supports Announcement 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.

Announcement Center

[+ Add Announcement Center](#)

Code	Name	Options
123	Test_1	

Total: 1 < 1 > 10 / page Goto

[+ Add Group](#)

Number	Name	Members	Options
1	Test	<input type="text" value="5000"/> <input type="text" value="5001"/> <input type="text" value="5002"/>	

Total: 1 < 1 > 10 / page Goto

Announcements Center

Announcements Center Settings

Announcement Center > Create New Announcement Center

* Name

* Code

* Custom Prompt

* Ring Timeout (s)

* Auto Answer

Announce Message Caller-ID

Name	Configure a name for the newly created Announcement Center to identify this announcement center.
Code	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. Note: The combination number must not conflict with any number in the system such as extension number or conference number.
Custom Prompt	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. Note: When uploading a custom, please ensure that the custom prompt file respect the following requirements. <ul style="list-style-type: none"> • The audio file must be less than 5 MB in file size with a file extension of .mp3/. wav/. ulaw/. alaw/. gsm. WAV files must be PCM encoded, 16 bit mono, and 8000Hz. • If uploading a compressed file, the file extension must be .tar/.tgz/.tar.gz, and the file size must not exceed 50MB. File name can only contain alphanumeric characters and special characters -_
Ring Timeout	Configure the ring timeout for the group members. The default value is 30 seconds.
Auto Answer	If set to Yes , the Auto answer will be enabled by the members.
Announce Message Caller-ID	If enabled, the caller's CID number will be announced before playing the uploaded prompt. This CID will also be used as the displayed CID of the call.

Group Settings

Group Settings

Name	Configure a name for the newly created group to identify the group. Note: Name cannot exceed 64 characters.
Number	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. Note: The combination number must not conflict with any number in the system such as extension number or conference number and cannot exceed 64 characters.
Internal Members	Choose the local extensions to add to the group.
LDAP Members	Choose the LDAP contacts to add to the group.
Custom Members	Enter the custom phone numbers to add to the group. Note: The maximum number of custom numbers which can be added are 50 custom number.

Announcements Center feature can be found under Web GUI > **Call Features** > **Announcement Center**. The following example demonstrates the usage of this feature.

1. Click

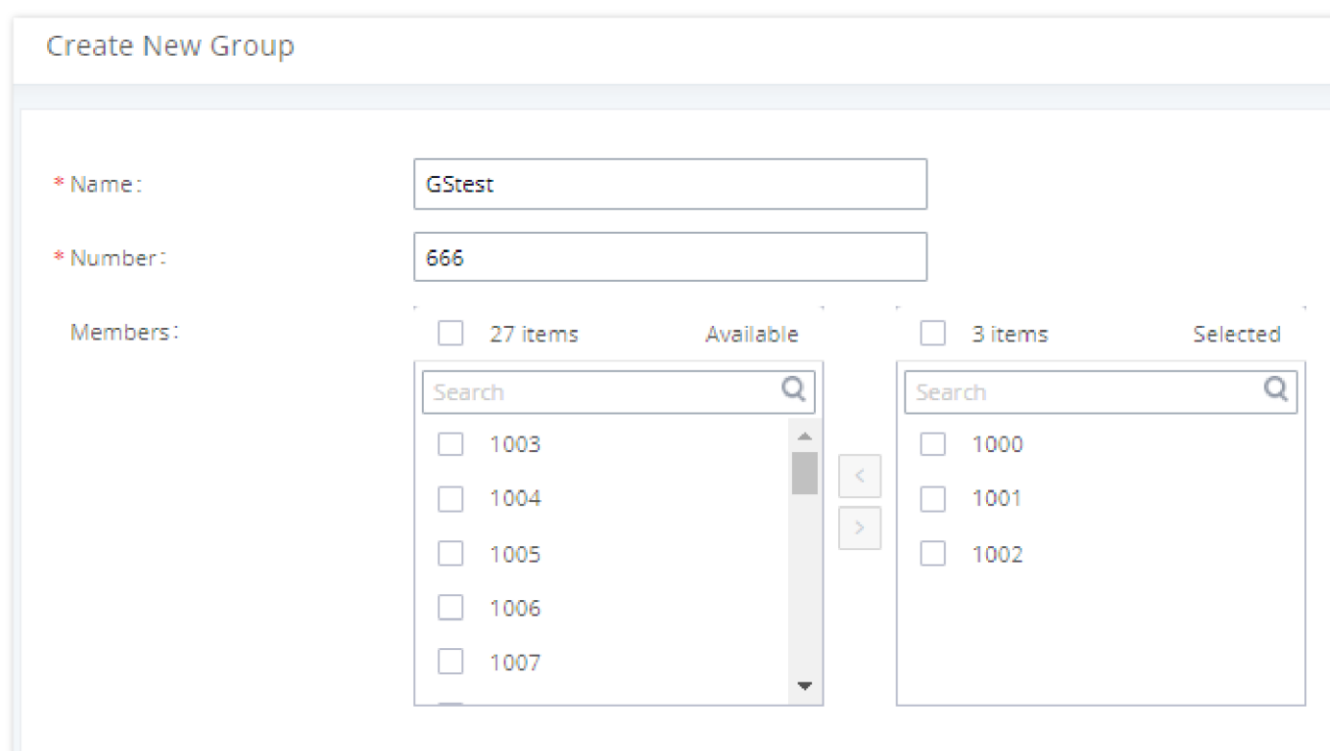


to add new group.

2. Give a name to the newly created group.

3. Create a group number which is used with code to send voice message.

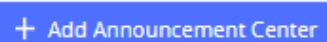
4. Select the extensions to be included in the group, who will receive the voice message.



Announcement Center Group Configuration

In this example, group "Test" has number 666. Extension 1000, 1001 and 1002 are in this group.

1. Click



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

* Name:

* Code:

* Custom Prompt: [Upload Audio File](#)

* Ring Timeout:

* Auto Answer:

Announcements Center Code Configuration

Name	Configure a name for the newly created Announcement Center to identify this announcement center.
Code	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. Note: The combination number must not conflict with any number in the system such as extension number or conference number.
Custom Prompt	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. Note: When uploading a custom, please ensure that the custom prompt file respect the following requirements. <ul style="list-style-type: none"> • The audio file must be less than 5 MB in file size with a file extension of .mp3/. wav/. ulaw/. alaw/. gsm. WAV files must be PCM encoded, 16 bit mono, and 8000Hz. • If uploading a compressed file, the file extension must be .tar/.tgz/.tar.gz, and the file size must not exceed 50MB. File name can only contain alphanumeric characters and special characters -_
Ring Timeout	Configure the ring timeout for the group members. The default value is 30 seconds.
Auto Answer	If set to Yes , the Auto answer will be enabled by the members.
Announce Message Caller-ID	If enabled, the caller's CID number will be announced before playing the uploaded prompt. This CID will also be used as the displayed CID of the call.

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	Edit Delete

◀ 1 ▶
Total: 1 | 10 / page | Goto 1

[+ Add Group](#)

NUMBER	NAME	MEMBERS	OPTIONS
666	GStest	1000 1001 1002	Edit Delete

◀ 1 ▶
Total: 1 | 10 / page | Goto 1

Announcements Center Example

QUEUOMETRICS

QueueMetrics docking tool provides an interface for UCM system and QM docking. Pass the UCM call queue report to QueueMetrics in a richer form. QueueMetrics is a call center control platform that supports login and logout of frequently used agents in the call center, provides call reports, real-time queue monitoring and other functions.

QueueMetrics

Enable QueueMetrics Integration

* QueueMetrics URL

* Username

* Webqloader Password

Partition

Outbound Call Tracking

QueueMetrics

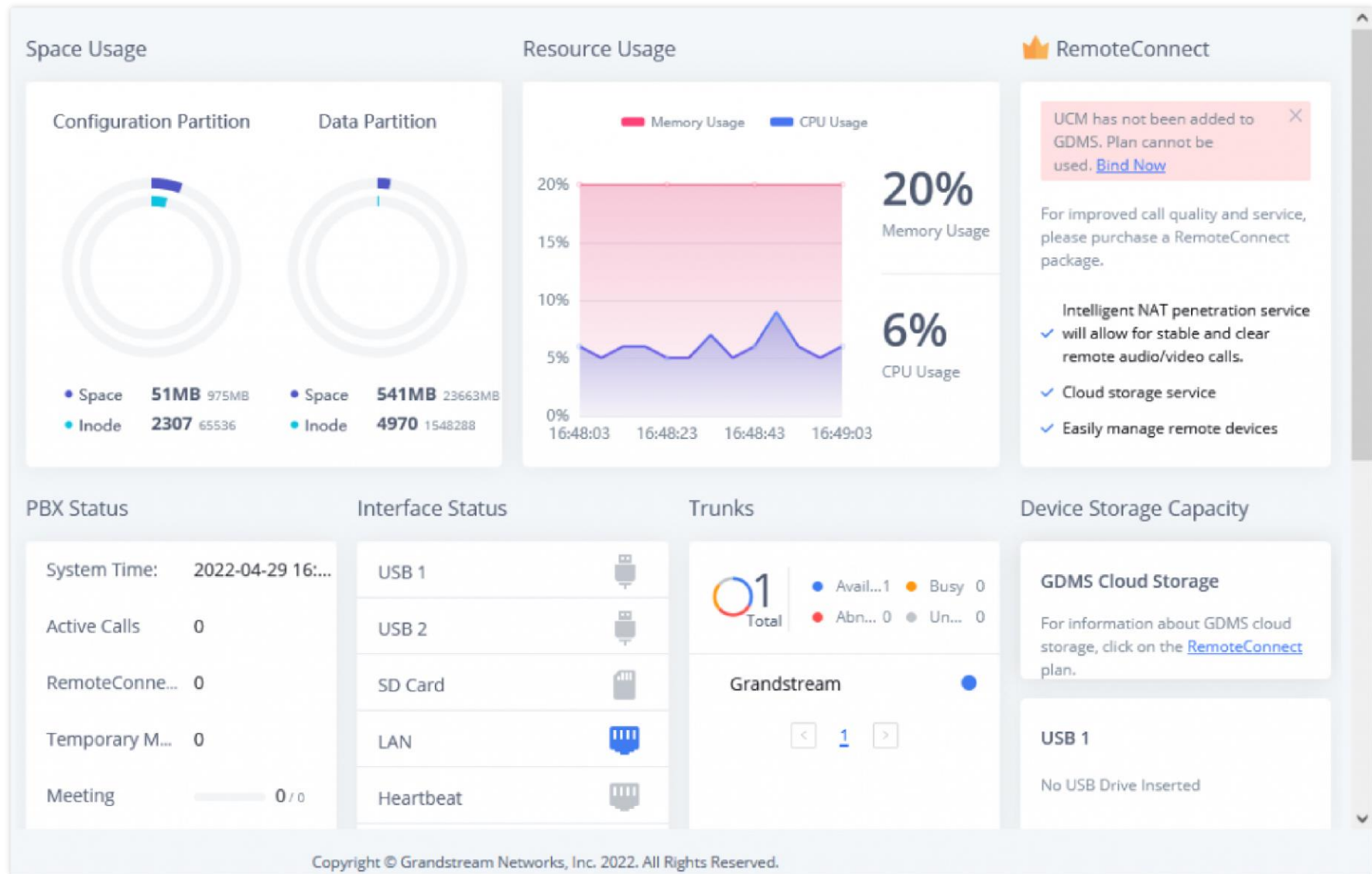
Parameter	Description
Enable QueueMetrics Integration	Tick this box to enable QueueMetrics integration module.
QueueMetrics URL	Enter the URL of the QueueMetrics on-premise server you have installed. (i.e. http://xxx.xxx.xxx.xxx:8080/queuemetrics.).
Username	Please enter the username used to interface with QueueMetrics. This is typically the QueueMetrics webqloader user. Please confirm that the user is enabled to avoid connection failure.
Webqloader Password	Please enter the webqloader password.
Partition	Enter the data storage partition identifier
Outbound Call Tracking	If enabled, QueueMetrics will track the outgoing calls of all extensions. Note: Outbound Call Tracking is available only on the UCM63XX.

Queue Metrics configuration parameters

STATUS AND REPORTING

PBX Status

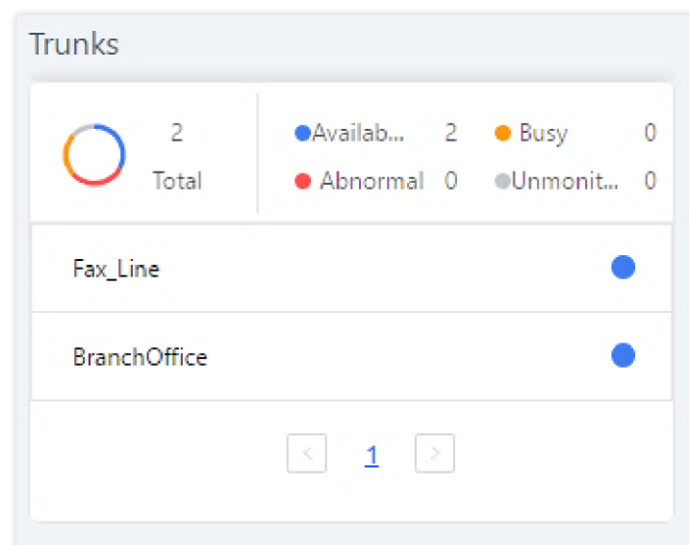
The UCM630xA monitors the status for Trunks, Extensions, Queues, Meeting Rooms, Interfaces and Parking lot. It presents administrators the real-time status in different sections under Web GUI→**System Status**→**Dashboard**.



Status → PBX Status

Trunks

Users could see all the configured trunk status in this section.



Trunk Status

Status	<p>Display trunk status.</p> <ul style="list-style-type: none"> Analog trunk status: <ul style="list-style-type: none"> Available Busy Unavailable Unknown Error SIP Peer trunk status: <ul style="list-style-type: none"> Unreachable: The hostname cannot be reached. Unmonitored: Heartbeat feature is not turned on to be monitored. Reachable: The hostname can be reached. SIP Register trunk status: <ul style="list-style-type: none"> Registered Unrecognized Trunk
Trunks	Display trunk name
Type	<p>Display trunk Type:</p> <ul style="list-style-type: none"> Analog SIP IAX
Username	Display username for this trunk.
Port/Hostname/IP	Display Port for analog trunk, or Hostname/IP for VoIP (SIP/IAX) trunk.



Trunk Status

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"/>	● Ringing	Available	1000		SIP(WebRTC)	192.168.5.199:5070		
<input type="checkbox"/>	● Unavailable	Available	1001		SIP(WebRTC)	--		
<input type="checkbox"/>	● In Use	Available	5555		SIP(WebRTC)	192.168.5.199:63827		










Extension Status

Status	<p>Display extension number (including feature code). The color indicator has the following definitions.</p> <ul style="list-style-type: none"> ○ • Green: Free ○ • Blue: Ringing ○ • Yellow: In Use ○ • Grey: Unavailable
Presence Status	Display the presence status of the extension.
Extension	Display the extension number.
Name	First name and last name of the extension.
IP and Port	Display the IP and port number of the registered device.
Email	<p>Display Email Notification status for the extension.</p> <p>When notification is waiting for be sent, shows  and once sent it will display </p>
Terminal Type	<p>Displays extension type.</p> <ul style="list-style-type: none"> ○ SIP User ○ IAX User ○ Analog User ○ Ring Groups ○ Voicemail Groups













Extension Status

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.

Interface Status	
USB 1	
USB 2	
SD Card	
LAN	
Heartbeat	
FXS	 
FXO	 

UCM6304A Interfaces Status

	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.

Interface Status Indicators

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.

System Status →System Information →General	
Model	Product model.
Part Number	Product part number.
System Time	Current system time. The current system time is also available on the upper right of each web page.
Up Time	System up time since the last reboot.
Boot	Boot version.
Core	Core version.
Base	Base version.
Program	Program version. This is the main software release version.
Recovery	Recovery version.
Lang	Lang version
Wave	Grandstream Wave version

System Status →General

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.

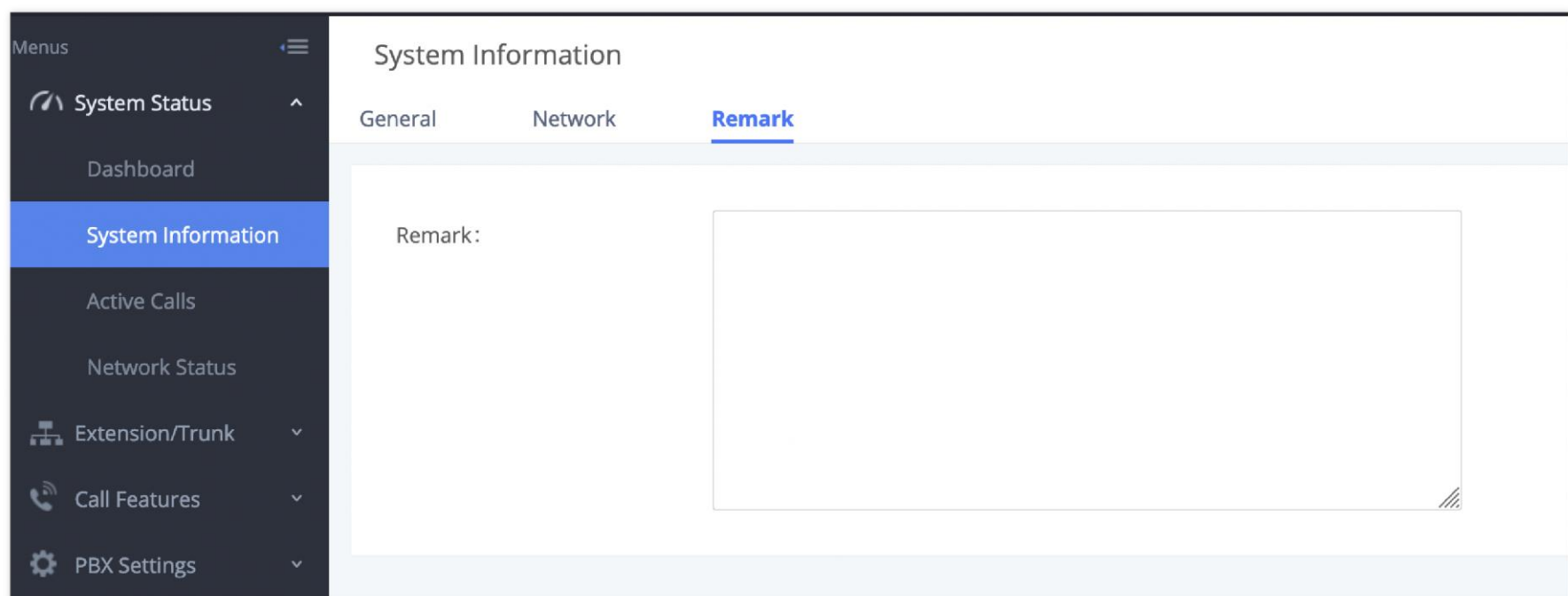
System Status→System Status→Network	
MAC Address	Global unique ID of device, in HEX format. The MAC address can be found on the label coming with original box and on the label located on the bottom of the device.
IPv4 Address	IPv4 address.
IPv6 Address Link	IPv6 address
Gateway	Default gateway address.
Subnet Mask	Subnet mask address.
DNS Server	DNS Server address.
Duplex Mode	Duplex Mode
Speed	Speed

System Status → Network

Remark

The UCM admin could add remark on UCM web UI->System Status->System Information->Remark to log any necessary information for the UCM such as location, technical contacts, important topology information and etc. This could be useful for UCM admin especially when there are multiple UCMs to be managed.

If this UCM has UCMRC service, the remark will also be sync up to GDMS. If this information is edited on GDMS, it will also be updated to the UCM web UI.



System Status → System Information → Remark

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**

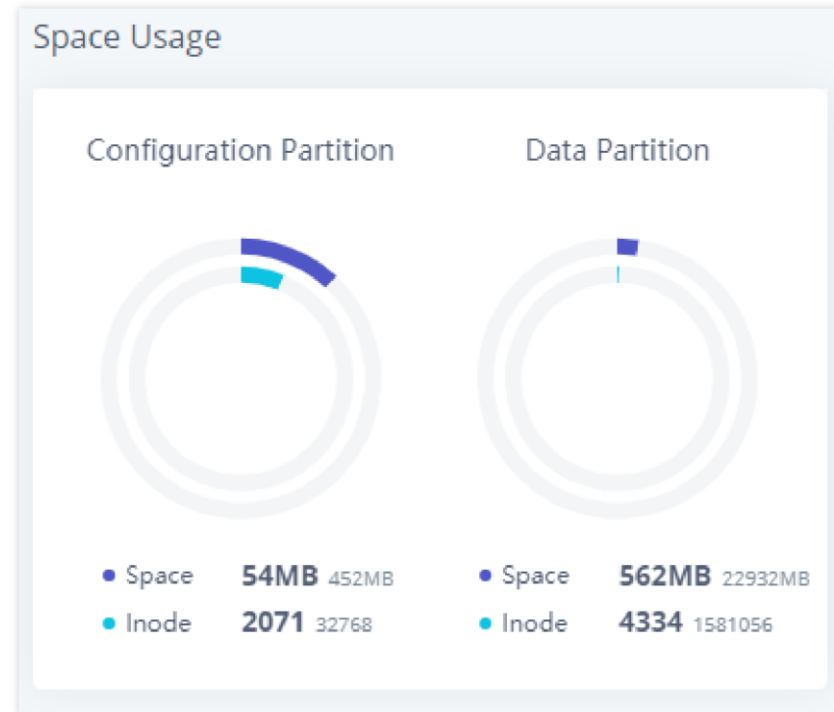
SD Card will display if connected.

Inode Usage includes:

- **Configuration partition**
- **Data partition**

Note:

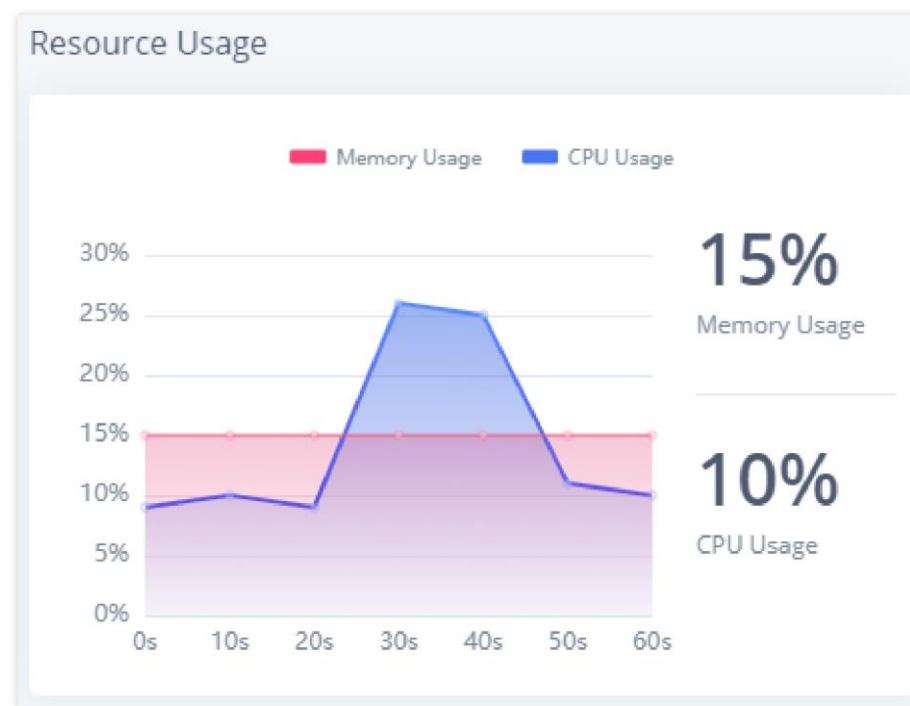
Inode is the pointer used for file reference in the system. The system usually has limited resources of pointers



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.



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:

EVENT NAME	ALERT	EMAIL NOTIFICATION	HTTP NOTIFICATION	PARAMETER SETTINGS
<input type="checkbox"/> Fail2ban Blocking	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	✎
<input type="checkbox"/> Flood Attacks	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	✎
<input type="checkbox"/> Network Traffic Storm	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	✎
<input type="checkbox"/> User Login Banned	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	✎
<input type="checkbox"/> Remote Login	<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	✎
<input type="checkbox"/> User Login Failed	<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"/> Restore Config	<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 Reboot	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	✎
<input type="checkbox"/> CPU Usage Call Control	<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"/> TLS Cert Expiration	<input checked="" type="radio"/> ON	<input type="radio"/> OFF	<input type="radio"/> OFF	✎
<input type="checkbox"/> HA failure warning	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	✎
<input type="checkbox"/> Cloud IM abnormal	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	✎

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Alert Event List

Alert Events
Fail2ban Blocking
Flood Attacks
Network Traffic Storm
User Login Banned
Remote Login
User Login Success
User Login Failed
System Crash
Restore Config
System Update
System Reboot
CPU Usage Call Control
Memory Usage
TLS Cert Expiration
HA Failure
HA Switch

HA Data Sync Failure
Cloud IM Abnormal
Modify Super Admin Password
Data Sync Backup
Local Disk Usage
External Disk Usage
Extended Disk Status
Emergency Calls
SIP Outgoing Call through Trunk Failure
SIP Internal Call Failure
Remote Concurrent Calls
Excessive Outbound Calls
Outbound Call Duration
Trunk Outbound Call Duration Usage
Trunk Concurrent Calls
Register SIP trunk failed
SIP Peer Trunk Status
Register SIP failed
SIP Lost Registration

Alert Events

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.

<https://documentation.grandstream.com/knowledge-base/ucm-remoteconnect-user-guide/>

Click on

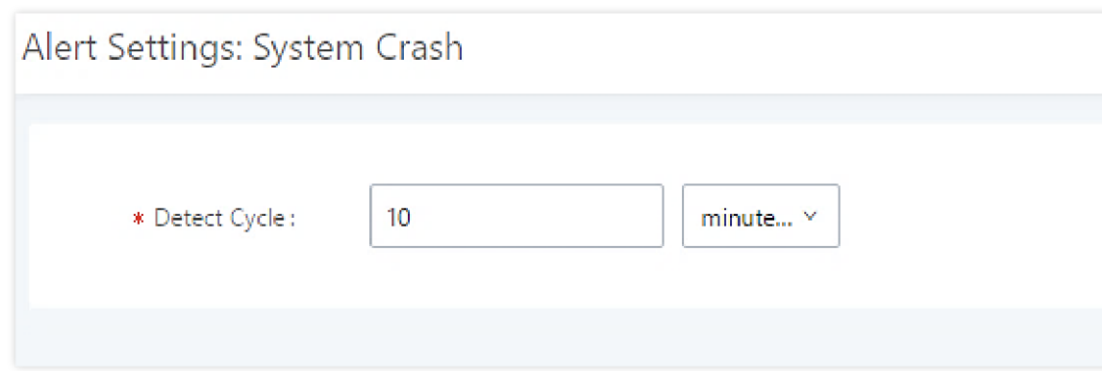


to configure the parameters for each event. See examples below.

1. **Fail2ban blocking:** If the system Fail2ban is blocking, the event will be recorded in the alert log.
2. **Flood Attacks:** An alert will be generated in case a DDoS attack attempt is detected by the UCM. The event will be registered in the alert log and it will be pushed to the GDMS.
3. **Network Traffic Storm:** An alert will be generated in case there is an excessive amount of packets on the LAN. Network Traffic Storms consume the resources of the network components and saturate the bandwidth, which will bring the whole network to a halt. This event will be registered in events log and a notification will be pushed to the GDMS.
4. **User login banned:** If user login is blocked, the event will be recorded in the alert log.

5. **Remote Login:**

6. **System Crash:**



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



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.

7. **Restore Config:** Once the system configuration is restored, the configuration restoration event will be recorded in the alert log.

8. **System Update:** Once the system is upgraded, the system upgrade event will be recorded in the alarm log.

9. **System Reboot:** UCM will detect the system restart and will send an alert for it. There are two kinds of reboots that the UCM detects, normal and abnormal reboots. Normal reboots are the reboots that are done when you press the restart button on the web UI, reboot that occur after updating the firmware, HA backup reboot etc... Abnormal reboots are the reboots that occur due to a system failure. Normal reboots are registered in the alert log and they are not pushed to GDMS, while abnormal reboots are registered in the alert list and are pushed to GDMS.

10. **TLS Cert Expiration:** Starting 7 days before the HTTP Server TLS certificate in the UCM device expires, an expiration countdown notification is sent every day; the certificate has expired, an expiration notification is sent; after the alarm notification is generated, a valid new certificate is uploaded, and a notification to restore the TLS certificate is generated.

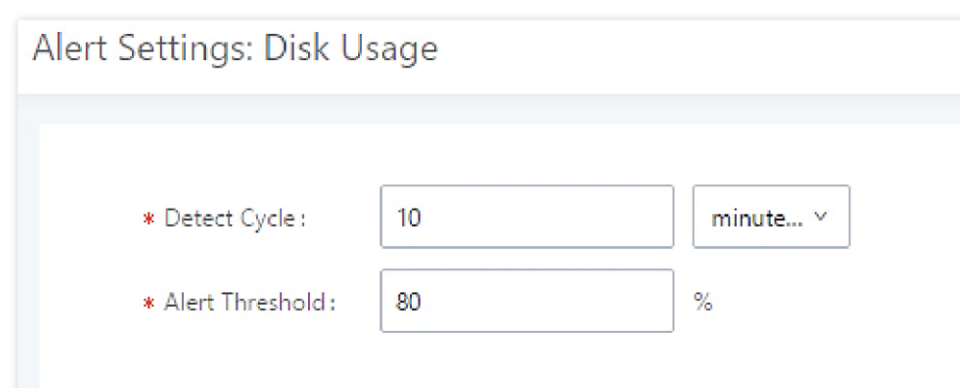
11. **HA failure warning:** After the HA dual-system hot backup disaster recovery function is enabled in the UCM device, the HA fault alarm is automatically turned on. When the device has a software and hardware-related fault, an HA fault alarm is generated.

12. **Cloud IM Abnormal:** An alert message will be generated if the Cloud IM encounter any issue or exhibit any abnormal behavior.

13. **Modify Super Admin Password:** Once the super administrator password is modified, the system will record the password modification event in the alarm log.

14. **NAS:** If the system network disk is abnormal, the event will be recorded in the alarm log.

15. **Disk Usage :**



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.

! If the threshold is exceeded, any behavior of operating the disk will be rejected, including stopping file upload, IM writing, recording and CDR recording.

16. Memory Usage

The screenshot shows a configuration window titled "Alert Settings: Memory Usage". It contains a single field labeled "Alert Threshold:" with a text input box containing the number "80" and a dropdown menu set to "%".

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.

17. External Disk Usage

The screenshot shows a configuration window titled "Alert Settings: External Disk Usage". It contains two fields: "* Detect Cycle:" with a text input box containing "10" and a dropdown menu set to "minute..."; and "* Alert Threshold:" with a text input box containing "80" and a dropdown menu set to "%".

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.

18. **External disk status:** If the external disk of the system is Connected/Disconnected, the event will be recorded in the alarm log.

19. **CPU Usage Call 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.

20. **Emergency Calls:** If the system generates an emergency call, the event will be recorded in the alert log.

21. Register SIP trunk failed

The screenshot shows a configuration window titled "Alert Settings: Register SIP trunk failed". It contains one field: "* Detect Cycle:" with a text input box containing "1" and a dropdown menu set to "minute(s)".

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.

22. **SIP peer trunk status:** If the SIP peer trunks status is abnormal, the event will be recorded in the alert log.

23. **SIP Outgoing Call through Trunk Failure:** If the system SIP trunk outgoing call fails, the event will be recorded in the alert log.

24. **Register SIP failed**

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.

25. **SIP lost registration:** If System SIP extension registration is lost, the event will be recorded in the alert log.

26. **SIP Internal Call Failure:** If the system SIP extension call fails within the office, the event will be recorded in the alert log.

27. **High Frequency Outgoing Call:** When an extension initiates calls frequently, an alert will be logged in the alert log and a notification will be pushed to the GDMS and through email as well.

28. **Remote concurrent calls:** If the remote concurrent call fails, the event will be recorded in the alert log.

29. **Trunk Outbound Call Duration Usage:**

30. **Trunk Concurrent calls:** When the system detects that the number of concurrent calls of a certain relay exceeds the threshold set by the relay within a certain period of time, the event will be recorded in the alarm log. Calls are not restricted if the threshold is exceeded.

31. **User login success:** Successful user login events will be recorded in the alert log.

32. **User login failed:** User login failure events will be recorded in the alert log.

33. **Data Sync Backup:** If the system performs data synchronization and backup abnormalities, the event will be recorded in the alert log.

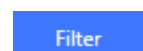
Alert Log

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

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



. 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 for Alert Log' form. It has four rows of input fields:

- 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 with 'All' selected.
- Type:** A dropdown menu with 'All' selected.

 At the bottom right, there are two buttons: 'Reset' and 'Filter'.

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.

Super Admin Email	Configure the email addresses to send alert notifications to. Up to 10 email addresses can be added.
Admin Email	Configure the email addresses to send alert notifications to. Up to 10 email addresses can be added.
Email Template	Please refer to section Email Templates
Protocol	Protocol used to communicate with the server. HTTP or HTTPS. Default one is HTTP .
HTTP Server	The IP address or FQDN of the HTTP/HTTPS server.
HTTP Server Port	HTTP/HTTPS port
Warning Template	Customize the template used for system warnings. By default: <code>{"action":"\${ACTION}","mac":"\${MAC}","content":"\${WARNING_MSG}"}</code>
Notification Template	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> Note: The notification message with " action:0 " will be sent periodically if Notification Interval is set.

Notification Interval	<p>Modifies the frequency at which notifications are sent in seconds.</p> <p>No notifications will be sent if the value is "0". Default value: 20</p>
Template Variables	<p><code>\${MAC}</code> : MAC Address</p> <p><code>\${WARNING_MSG}</code> : Warning message</p> <p><code>\${TIME}</code> : Current System Time</p> <p><code>\${CPU_USED}</code> : CPU Usage</p> <p><code>\${MEM_USED}</code> : Memory Usage</p> <p><code>\${ACTION}</code> : Message Type. Refer to [Alert Events]</p> <p><code>\${DISK_USED}</code> : Disk Usage</p> <p><code>\${EXTERNAL_DISK_USED}</code> : Disk Usage</p>

Alert Contact

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 web interface titled "CDR" with a "Hide Filter" link in the top right. The interface contains several input fields and checkboxes for filtering CDR data:

- Start Time:** Two dropdown menus for "Select date" and "Select time".
- End Time:** Two dropdown menus for "Select date" and "Select time".
- Caller Number:** A text input field.
- Caller Name:** A text input field.
- Callee Number:** A text input field.
- Account Code:** A text input field.
- Source Trunk Name:** A text input field.
- Destination Trunk Name:** A text input field.
- Action Type:** A text input field.
- Export File Data:** A text input field.
- Extension Group:** A dropdown menu.
- Call Type:** Four checkboxes: "Inbound Calls", "Outbound Calls", "Internal Calls", and "External Calls".
- Status:** Four checkboxes: "Answered", "No Answer", "Busy", and "Failed".

At the bottom of the form are two buttons: "Filter" (in blue) and "Reset". A green footer note reads: "By default, this page displays the CDR entries from the current month. Use the 'Filter' button to specify a time range."

CDR Filter

Call Type	<p>Groups the following:</p> <ul style="list-style-type: none"> ○ Inbound calls: Inbound calls are calls originated from a non-internal source (like a VoIP trunk) and sent to an internal extension. ○ Outbound calls: Outbound calls are calls sent to a non-internal source (like a VoIP trunk) from an internal extension. ○ Internal calls: Internal calls are calls from one internal extension to another extension, which are not sent over a trunk. ○ External calls: External calls are calls sent from one trunk to another trunk, which are not sent to any internal extension.
Status	<p>Filter with the call status, the available statuses are the following:</p> <ul style="list-style-type: none"> ○ Answered ○ No Answer ○ Busy ○ Failed
Source Trunk Name	<p>Select source trunk(s) and the CDR of calls going through inbound the trunk(s) will be filtered out.</p>
Destination Trunk Name	<p>Select destination trunk(s) and the CDR of calls going outbound through the trunk(s) will be filtered out.</p>
Action Type	<p>Filter calls using the Action Type, the following actions are available:</p> <ul style="list-style-type: none"> ○ Announce ○ Announcement page ○ Dial ○ Announcements ○ Callback ○ Call Forward ○ Meeting ○ DISA ○ Follow Me ○ IVR ○ Page ○ Parked Call ○ Queue ○ Ring Group ○ Transfer ○ VM ○ VMG ○ VQ_Callback ○ Wakeup ○ Emergency Call ○ Emergency Notify ○ SCA

Extension Group	Specify the Extension Group name to filter with.
Export File Data	<p>Select the fields that will be exported, the following fields are available:</p> <ul style="list-style-type: none"> <input type="radio"/> Account Code <input type="radio"/> Session <input type="radio"/> Premier caller <input type="radio"/> Action type <input type="radio"/> Source trunk name <input type="radio"/> Destination trunk name <input type="radio"/> Caller number <input type="radio"/> Caller ID <input type="radio"/> Caller name <input type="radio"/> Callee number <input type="radio"/> Answer by <input type="radio"/> Context <input type="radio"/> Start time <input type="radio"/> Answer time <input type="radio"/> End time <input type="radio"/> Call time <input type="radio"/> Talk time <input type="radio"/> Source channel <input type="radio"/> Dest channel <input type="radio"/> Call status <input type="radio"/> Dest channel extension <input type="radio"/> Last app <input type="radio"/> Last data <input type="radio"/> AMAFLAGS <input type="radio"/> UIQUEID <input type="radio"/> Call type <input type="radio"/> NAT
Account Code	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.
Start Time	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.
End Time	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.

Caller Number	<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>For example:</p> <p>3XXX: It will filter out CDR that having caller number with leading digit 3 and of 4 digits' length.</p> <p>3.: It will filter out CDR that having caller number with leading digit 3 and of any length.</p>
Caller Name	<p>Enter the caller name to filter the CDR report. CDR with the matching caller name will be filtered out.</p>
Callee Number	<p>Enter the callee number to filter the CDR report. CDR with the matching callee number will be filtered out.</p> <p>Note: The "Callee Number" filter field supports specifying Pattern (example: 3XXX) or using Leading digits (example: 3.) as filtering options.</p>

CDR Filter Criteria

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		-

Call Report

The CDR report has the following data fields:

- o **Start Time**

Format: 2019-12-11 09:53:03

- o **Action Type**

Example:

IVR

DIAL

WAKEUP

- o **Call From**

Example format: 5555

- o **Call To**

Example format: 1000

- o **Call Time**

Format: 0:00:11

- o **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**

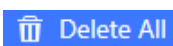
Click on "Download Search Result(s)" to export the records filtered out to a .csv file.

- **Download All Records**

Click on "Download All Records" to export all the records to a .csv file.

- **Delete All**

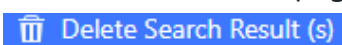
Click on



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



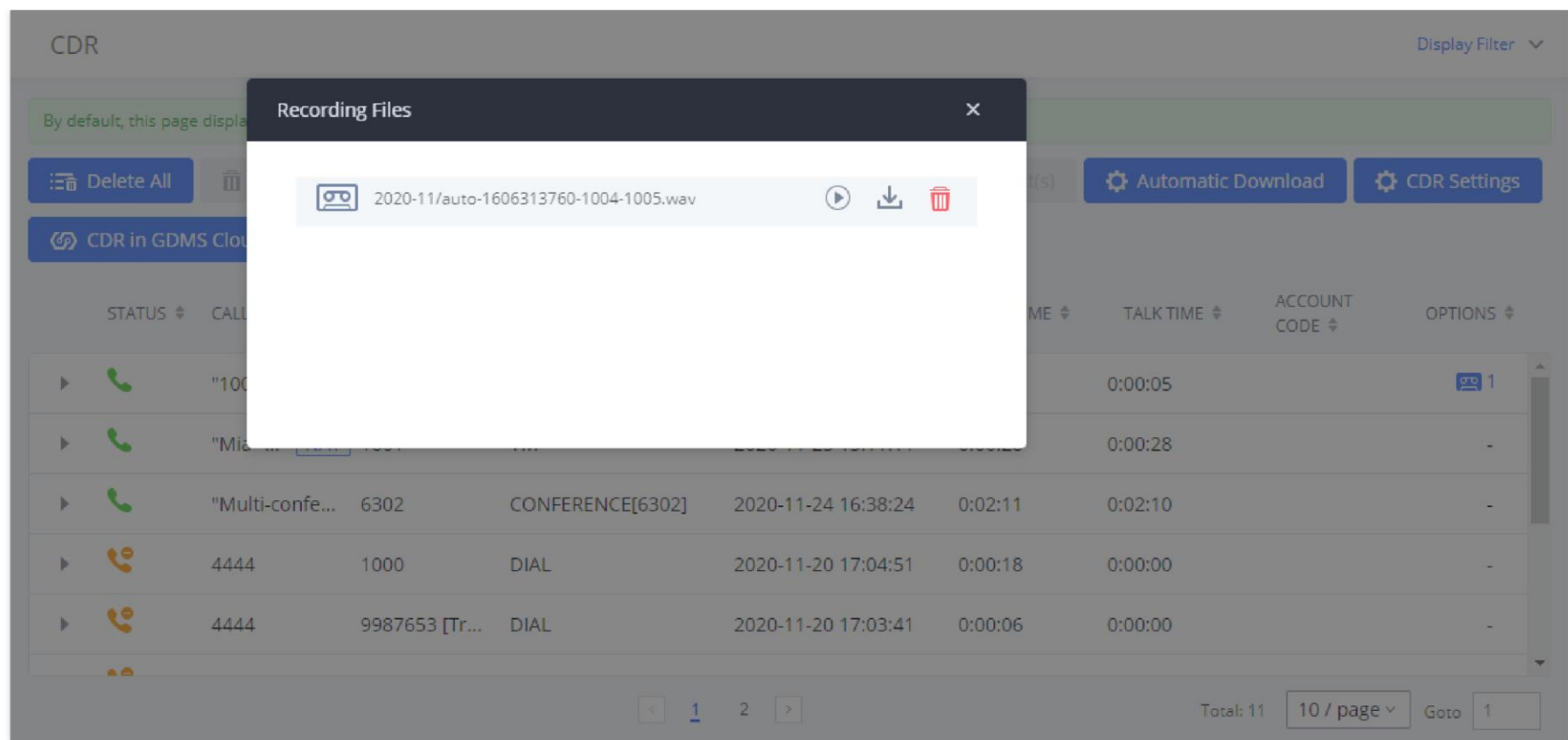
to play the recording file; click on



to download the recording file in .wav format; click on



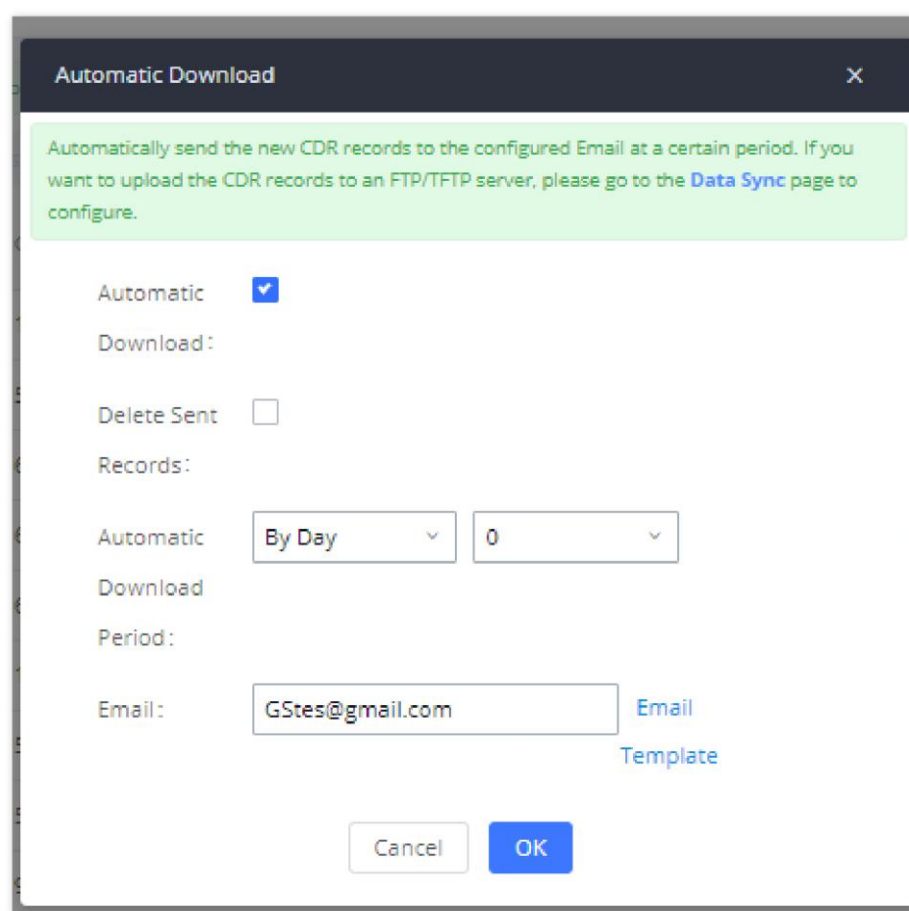
to delete the recording file (the call record entry will not be deleted).



Call Report Entry with Audio Recording File

o **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.



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 CODE	RECORDING FILE OPTIONS
▶	5555	1000	DIAL	2019-12-11 09:53:03	0:00:11	0:00:06		-

CDR Report

STATUS	PREMIER CALLER	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME †	TALK TIME	ACCOUNT CODE	RECORDING FILE OPTIONS
	1000	"abllili lolo" 1000...	9985632	DIAL	2019-12-10 03:23:14	0:00:00	0:00:00		-
	1000	"abllili lolo" 1000...	6500	QUEUE[6500]	2019-12-10 03:23:14	0:00:00	0:00:00		1
	1000	"abllili lolo" 1000...	5555	QUEUE[6500]	2019-12-10 03:23:14	0:00:13	0:00:07		-

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

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

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

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-XXXXXX: 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.

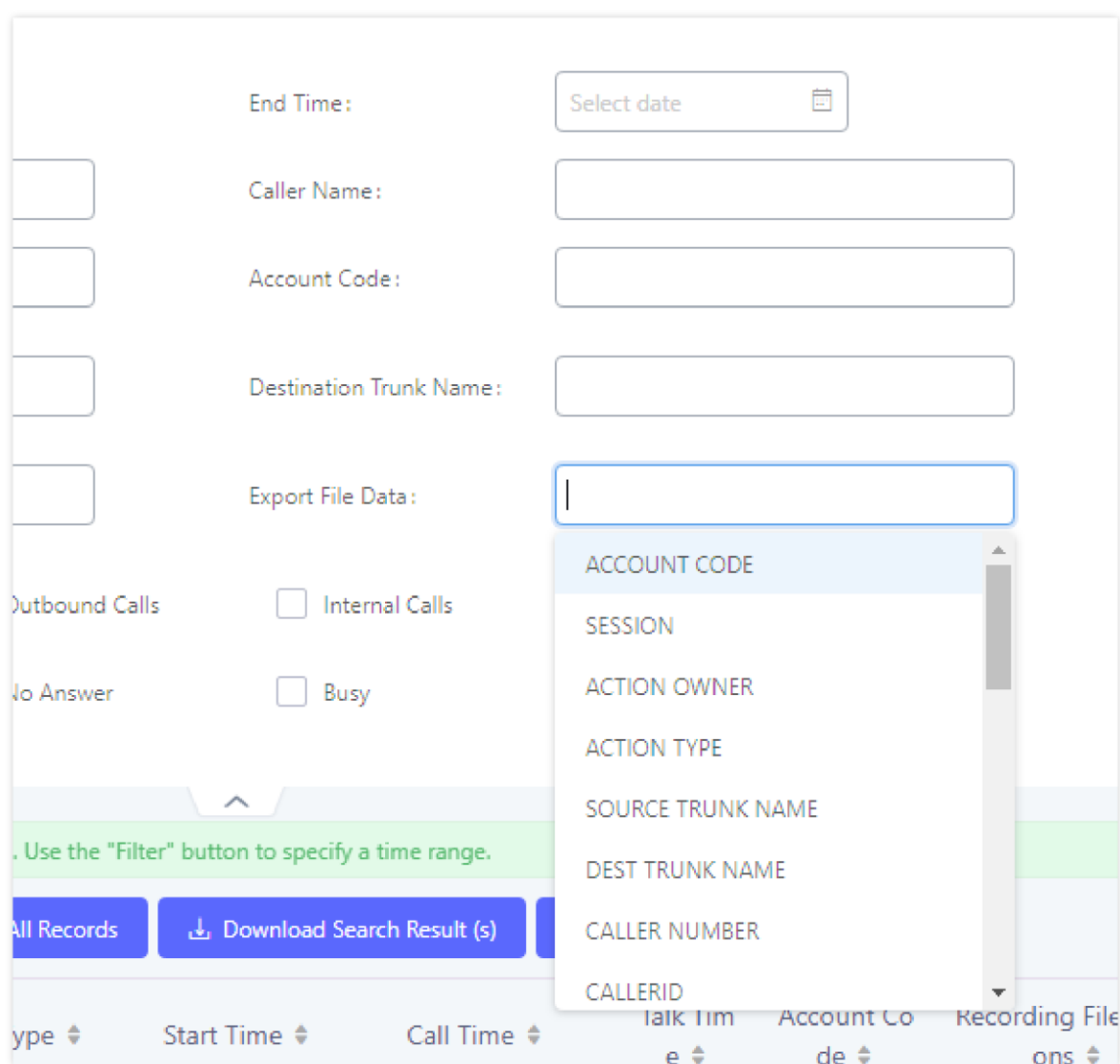
Playback: play some prompts to you, such as 183 response or run into an IVR.

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

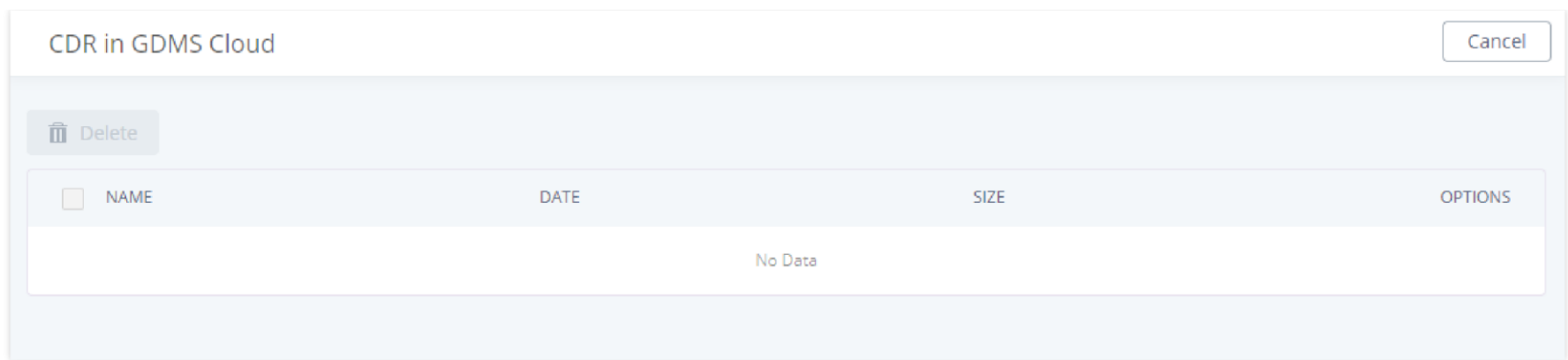
Users can select the data they want to see in exported CDR reports by first clicking on the *Filter* button on the CDR page under **CDR→CDR** and selecting the desired information in the *Export File Data* field.



CDR Export File data

CDR in GDMS Cloud

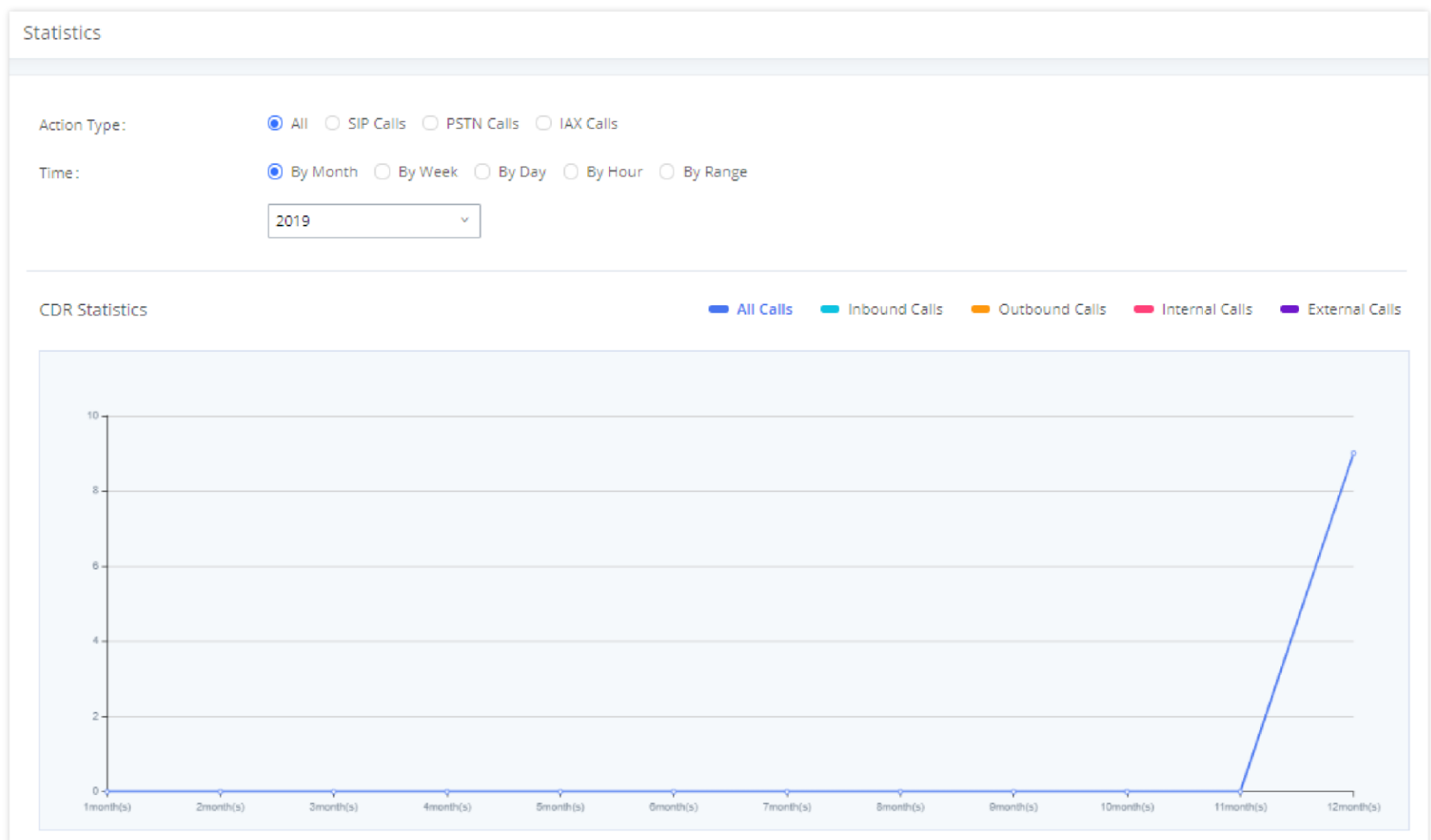
Cloud Storage for CDR Record which can be displayed under **CDR → CDR in GDMS Cloud**.



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.



CDR Statistics

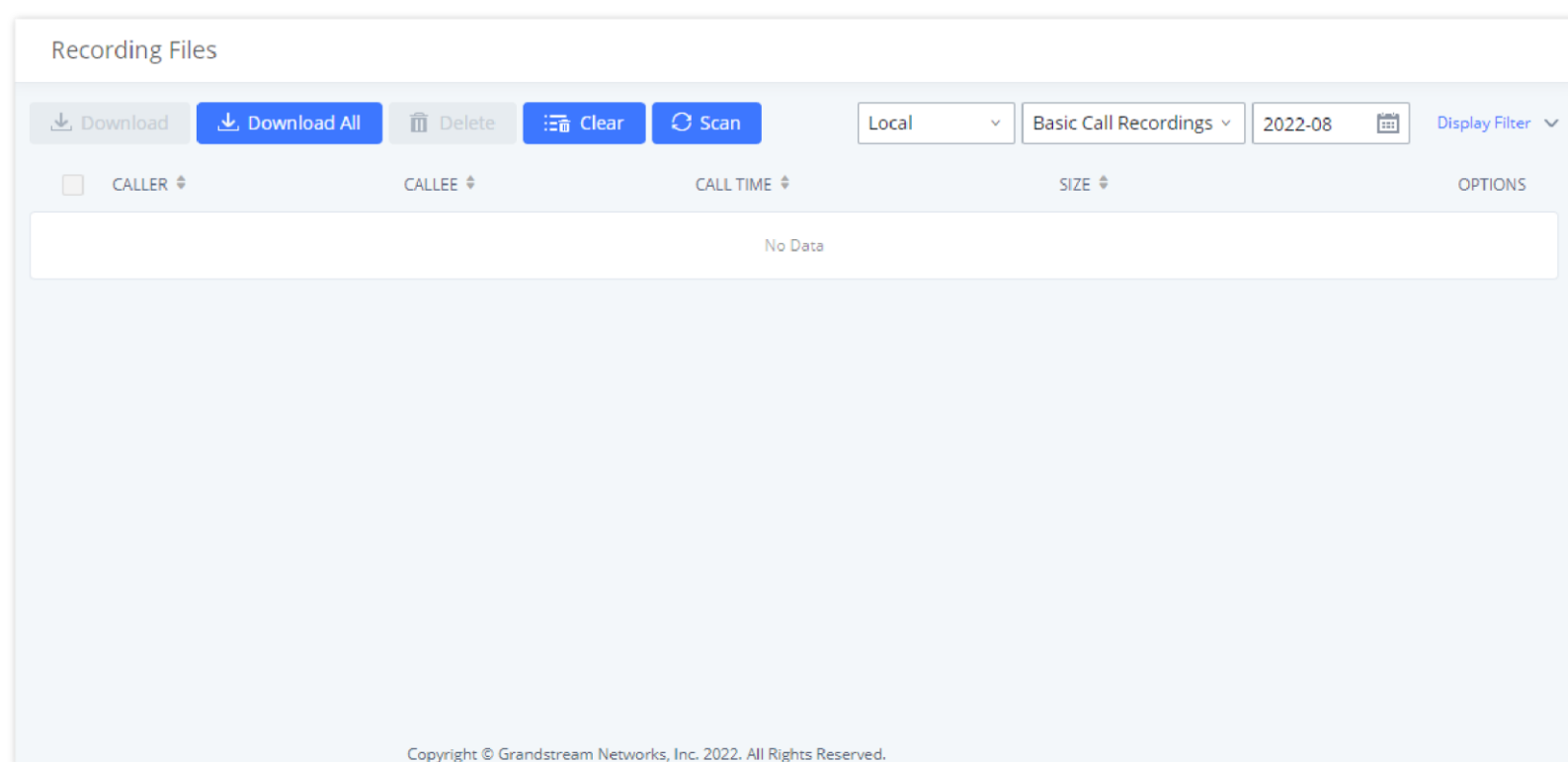
Trunk Type	<p>Select one of the following trunk type.</p> <ul style="list-style-type: none"> <input type="radio"/> All <input type="radio"/> SIP Calls <input type="radio"/> PSTN Calls
Call Type	<p>Select one or more in the following checkboxes.</p> <ul style="list-style-type: none"> <input type="checkbox"/> Inbound calls <input type="checkbox"/> Outbound calls <input type="checkbox"/> Internal calls <input type="checkbox"/> External calls <input type="checkbox"/> All calls

Time Range	<ul style="list-style-type: none"> ○ By month (of the selected year). ○ By week (of the selected year). ○ By day (of the specified month for the year). ○ By hour (of the specified date). ○ By range. For example, 2016-01 To 2016-03.
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CDR Statistics Filter Criteria

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.



CDR → Recording Files

- Click on **"Download"** to batch-download the selected recording files.
- Click on **"Download All"** to download all the recording files.
- Click on **"Delete"** to batch-delete the selected recording files.
- Click on **"Clear"** to delete all the recording files.
- Click on **"Scan"** to retrieve the file information and display all the recording files on external storage. The UCM automatically retrieves the info of the first 5000 files from external storage already. This button can be used when the number of files stored on the external storage exceeds 5000 files and it requires manual file scanning.
- Select either **"USB Disk"** or **"Local"** to show recording files stored on external or internal storage, depending on the selected storage space.
- Select whether to show call recordings, queue recordings or conference 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 other features like Call Queue, Wakeup Service and CRM.

Users also can access their personal data files (call recordings, Voicemail Prompts ...).

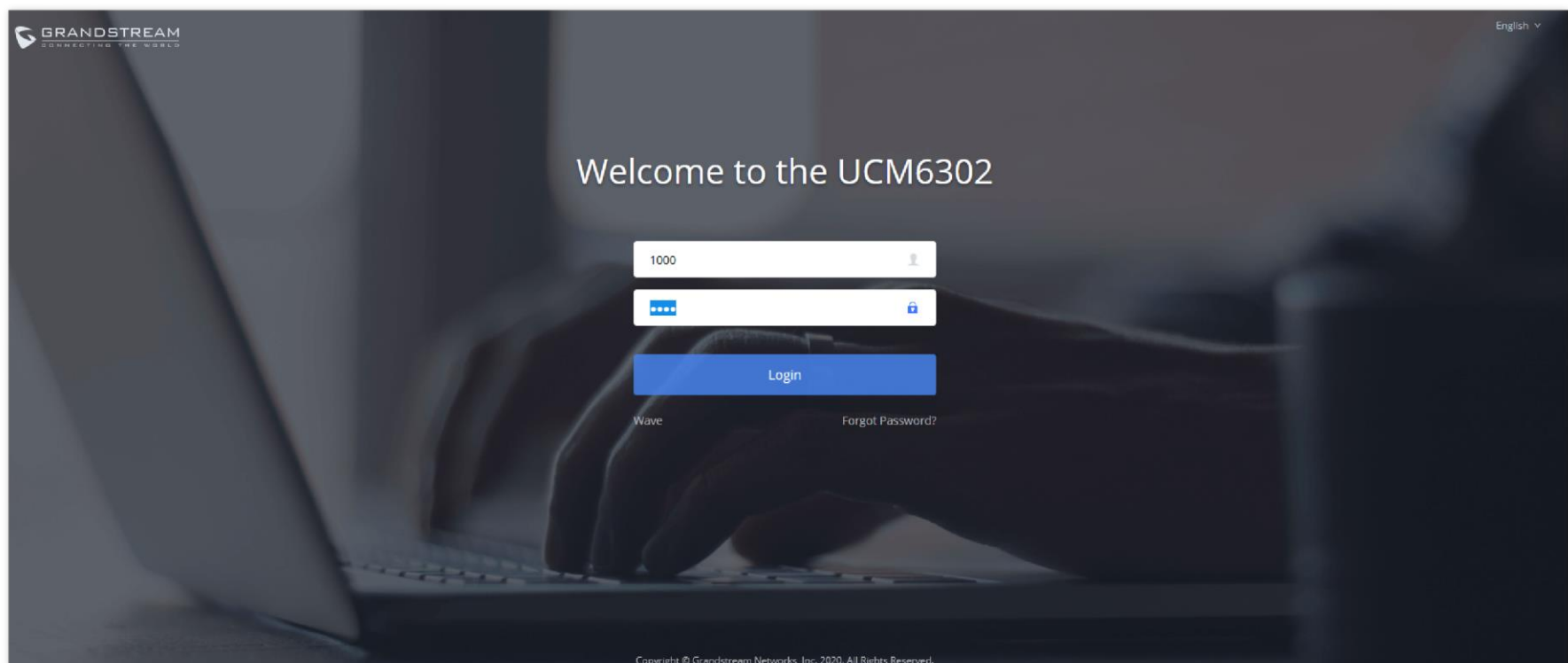
The login credentials are configured by Super Admin. The following figure shows the dialog of editing the account information by Super Admin. The Username must be the extension number and it is not configurable, and the password is set on "User Password" field and it should not be confused with the SIP extension password.

The screenshot shows a web form titled "Edit User Information: 1000". The form contains the following fields:

* User Name:	1000	* User Password:	mYpassWord!
Privilege:	Consumer	Department:	Support
Fax:		Email Address:	user1000@domain.local
First Name:	John	Last Name:	DOE
Home Number:		Mobile Phone Numb...:	

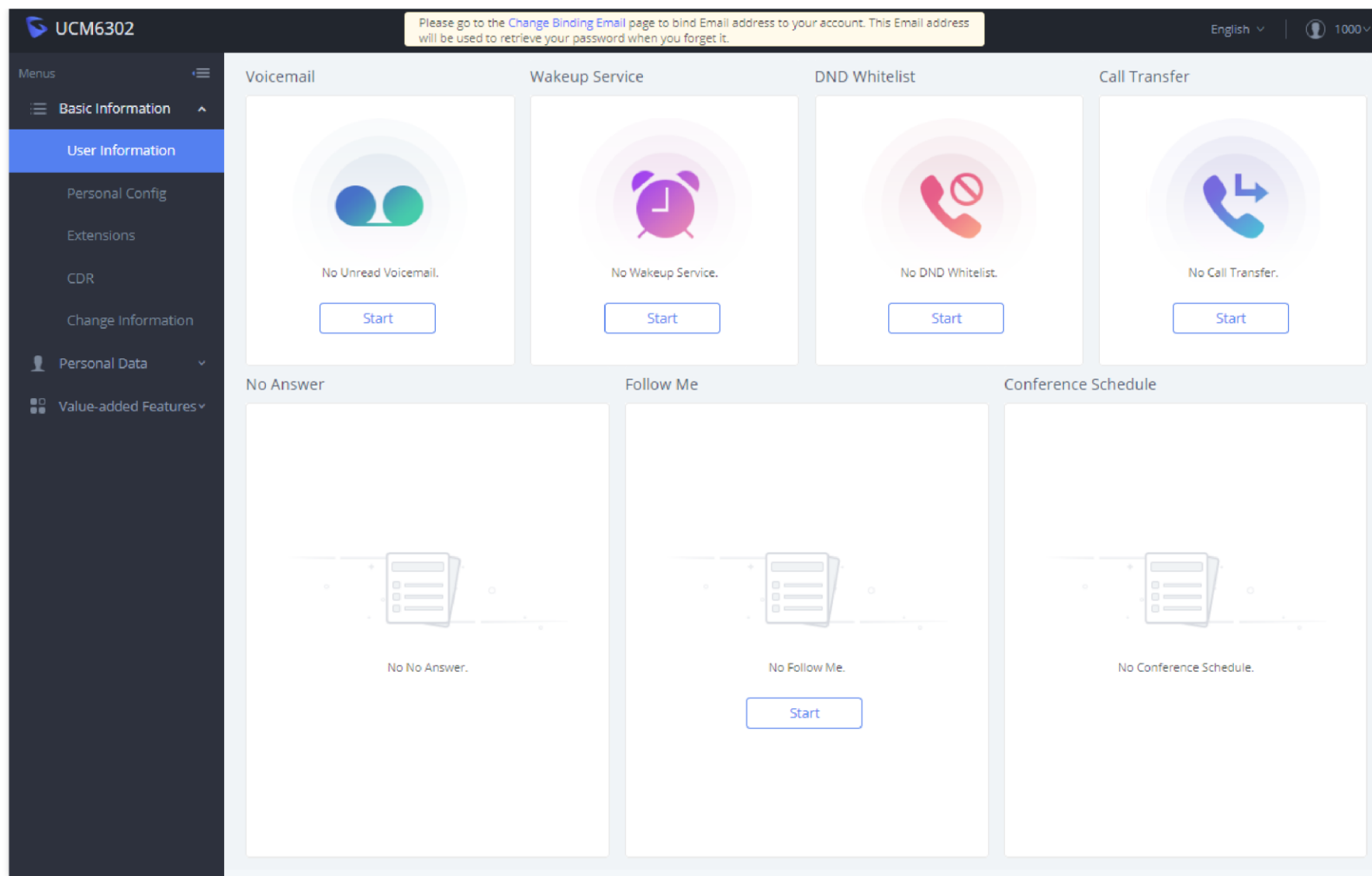
Edit User Information by Super Admin

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



User Portal Login

After login, the Web GUI display is shown as below.



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.

Other Features

On this section, the user has access to manage and use all rich features which includes.

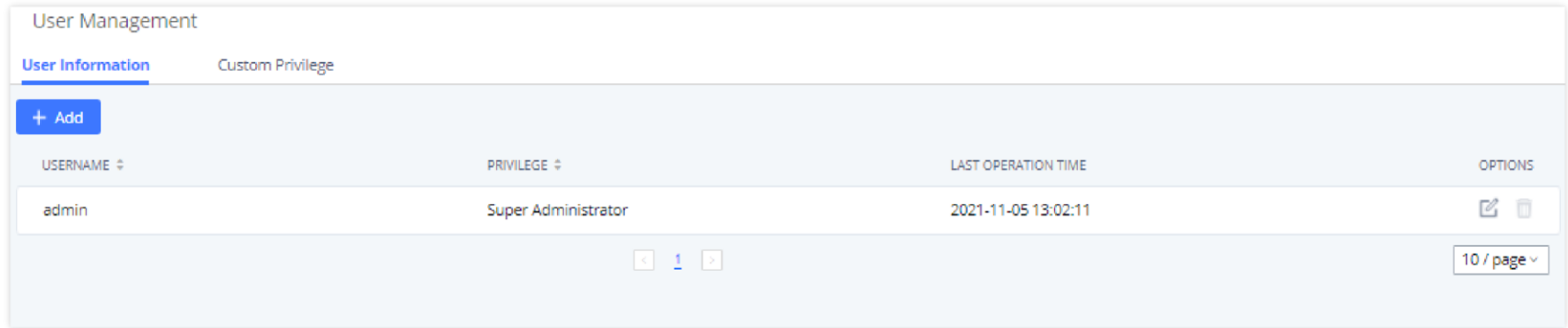
- + If user is a member of call queue, they can check the queue's activity from the "Call Queue" section.
- + Create and enable Wake Up service.
- + Enable and configure CRM connection to either SugarCRM or Salesforce.

For the configuration parameter information in each page, please refer to [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

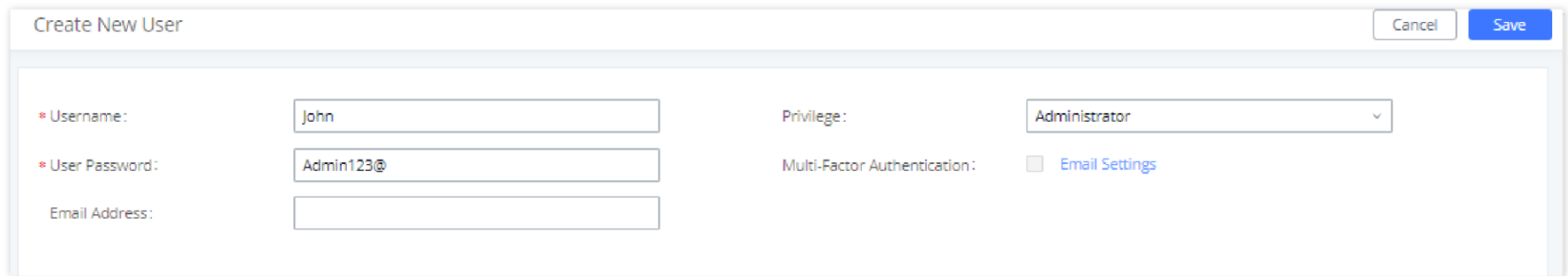
User management is on Web GUI→**Maintenance**→**User Management** page. User could create multiple accounts for different administrators to log in the UCM630xA Web GUI. Additionally, the system will automatically create user accounts along with creating new extensions for extension users to login to 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 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.



Create a New User

Username	Configure a username to identify the user which will be required in Web GUI login. Letters, digits, and underscore are allowed in the username.
User Password	Configure a password for this user which will be required in Web GUI login. English input is allowed without space, ' and " .
Privilege	This is the role of the Web GUI user. When super admin creates new user, "Administrator" or customized privilege can be selected.
Multi-Factor Authentication	If this authentication is enabled, the user account needs to be verified with an MFA code every time it logs in to enhance the security of the product.
Email Address	Configure the email address for the user. This is optional.





User Management → Create New User

Once created, the Super Admin can edit the users by clicking on



or delete the user by clicking on



User Management			
User Information		Custom Privilege	
+ Add			
USERNAME	PRIVILEGE	LAST OPERATION TIME	OPTIONS
admin	Super Administrator	2021-11-05 13:08:50	 
John	Administrator	--	 

10 / page

User Management – New Users

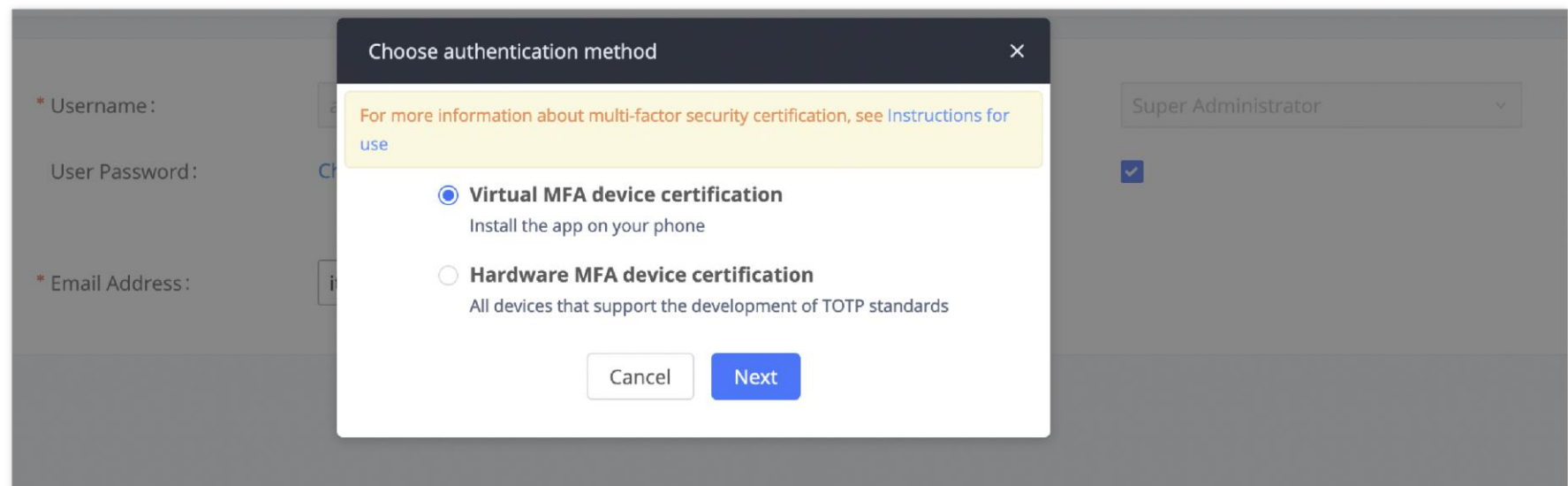
Multi-Factor Authentication

To enhance the security for UCM, super admin and admin can select to use multi-factor authentication method for login to protect the login information. Virtual MFA and hardware MFA are supported and can be selected. Once enabled, the user must use MFA code to verify before login.

Notes:

- The user cannot enable or disable MFA for another different user.
- Super admin can edit user settings for admin but cannot edit Multi-Factor Authentication option. MFA option is only viewable for super admin when super admin edits other users.
- When the user sees MFA enabled, only this user can disable or enable it again.
- Email address and email settings are required before enabling Multi-Factor Authentication. Please ensure email setting has "Client" type configured. Otherwise, MFA cannot be enabled.

Please refer to MFA how to guide [here](#) for more information.



MFA Settings

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

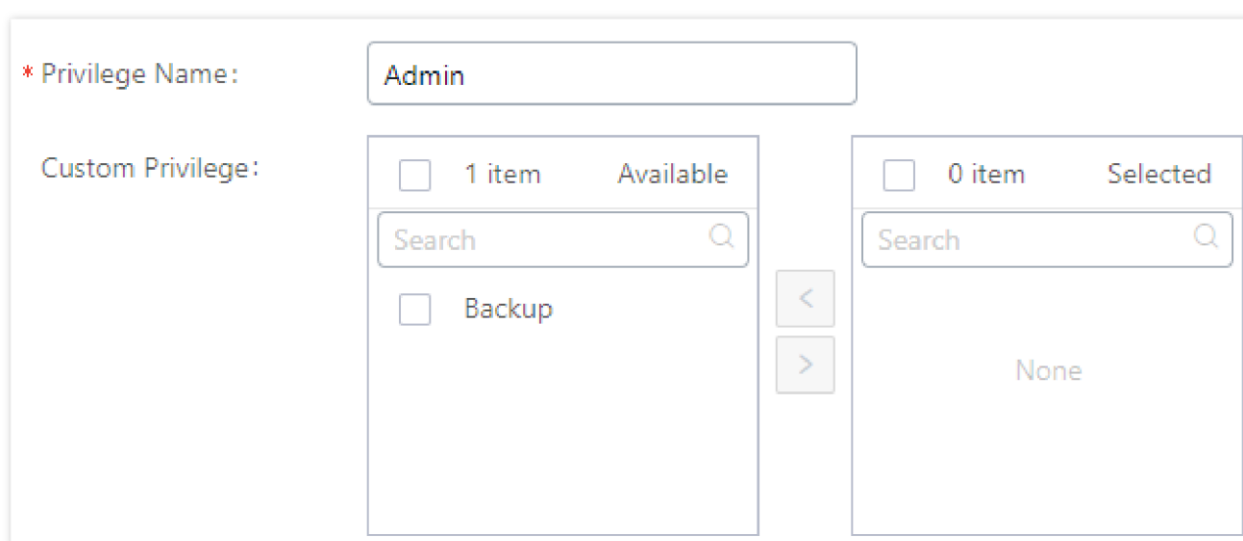
Settings→User Management→Operation Log

- "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



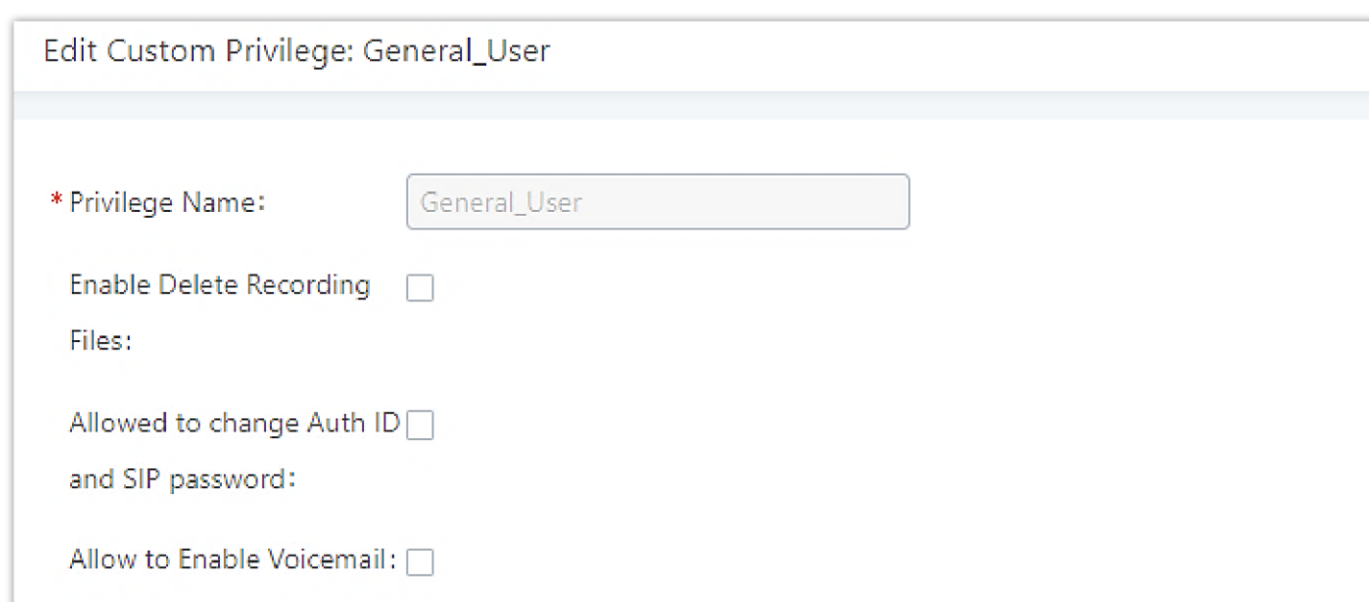
to edit the "Admin" account and include backup operation permission for these types of users.



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 other features. For more details; please refer to <https://documentation.grandstream.com/knowledge-base/user-portal/>.
- The Super Admin 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.



General User

Custom Privilege

The Super Admin user can create users with different privileges. 38 items are available for privilege customization.

1. API Configuration
2. Backup

3. Callback
 4. Call Queue
 5. Queue Statistics
 6. Queue Recordings
 7. CDR Recording Files
 8. CDR Records
 9. CDR Statistics
 10. Dial By Name
 11. DISA
 12. Emergency Calls
 13. Event List
 14. Extensions
 15. Extension Groups
 16. Outbound Routes
 17. Inbound Routes
 18. Fax/T.38
 19. Feature Codes
 20. IVR
 21. Paging/Intercom
 22. Parking Lot
 23. Pickup Groups
 24. PMS – Wakeup Service
 25. Ring Groups
 26. Restrict Calls
 27. SCA
 28. Speed Dial
 29. System Status
 30. System Events
 31. LDAP Server
 32. Time Settings
 33. Meeting
 34. Voicemail
 35. Voice Prompt
 36. Scheduled Call
 37. Contacts
 38. Zero Config
 39. Announcement.
 40. RemoteConnect
-

Create New Custom Privilege

Log in UCM630xA as super admin and go to **Maintenance**→**User Management**→**Custom Privilege**, create privilege with customized available modules.


When you add CDR Records and CDR Recording Files custom privileges, additional privileges will appear (All Deletion of CDR and Allow Deletion of DCR Recordings , respectively). This offers more flexibility on the privileges that the admin assigns to the user.

Allow Deletion of CDR and CDR Recordings

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

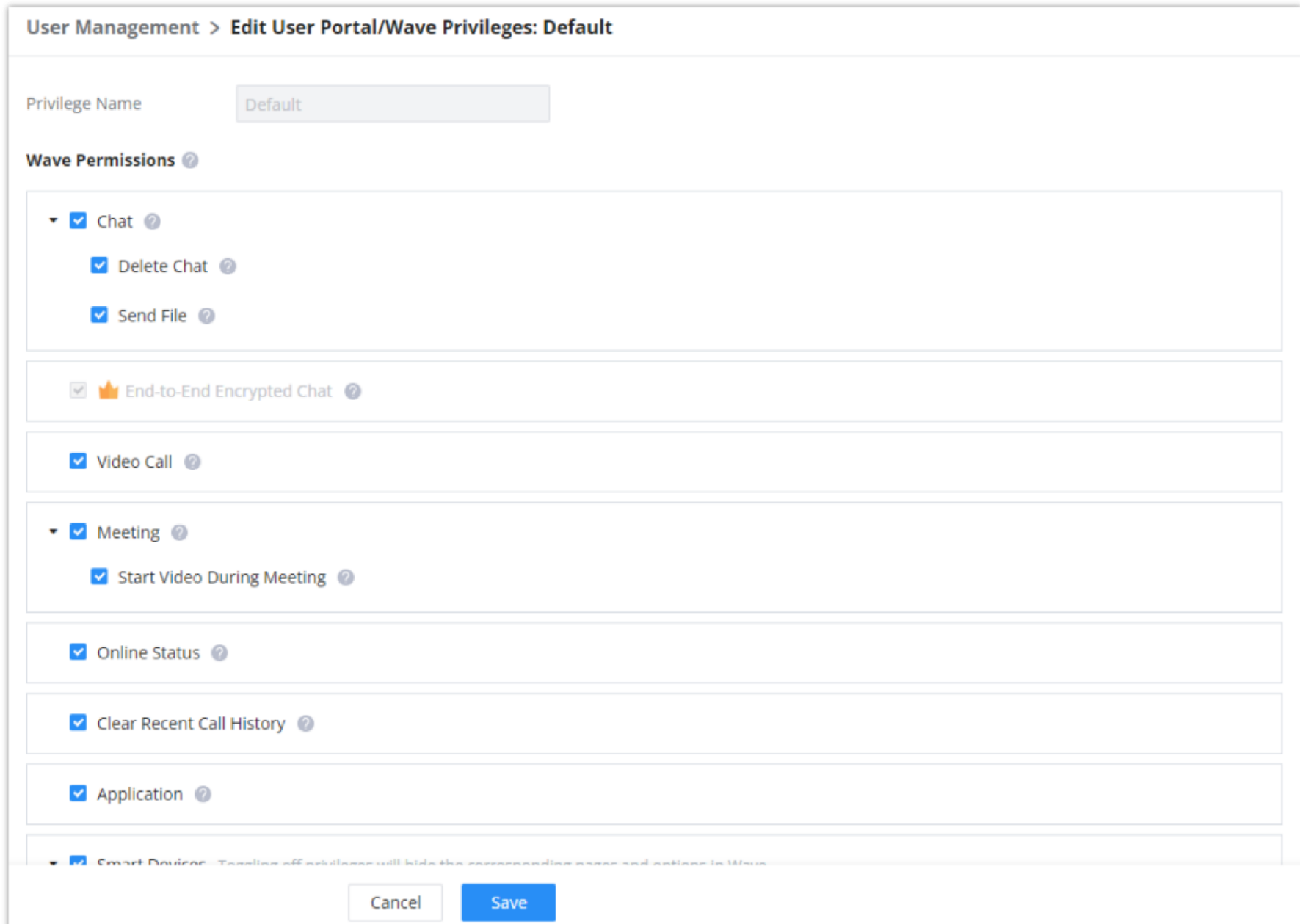
When there are multiple Web GUI users created, concurrent multi-user login is supported on the UCM630xA. Multiple users could edit options and have configurations take effect simultaneously. However, if different users are editing the same option or making the same operation (by clicking on "Apply Changes"), a prompt will pop up as shown in the following figure.

 Operating too frequently or other users are doing the same operation. Please retry after 15 seconds.

Multiple User Operation Error Prompt

User Portal/Wave Privilege

The user can create customize privileges related to an extension's User Portal and Wave. The created privilege can be affected to the extensions to limit or allow them to use certain functionalities related to Wave and the User Portal.



User Management > Edit User Portal/Wave Privileges: Default

Privilege Name

Wave Permissions ?

- Chat ?
 - Delete Chat ?
 - Send File ?
- End-to-End Encrypted Chat ?
- Video Call ?
- Meeting ?
 - Start Video During Meeting ?
- Online Status ?
- Clear Recent Call History ?
- Application ?
- Smart Devices ?

Cancel Save

User Portal/Wave Privileges

Wave Permissions

- **Chat:** Toggles ability to use the Wave Chat feature.
 - **Delete Chat:** Toggles support for Wave to delete chats and chat history. This data will only be deleted on the Wave client side.
 - **Send File:** Toggles file/image sending support in Wave chat. If disabled, users will still be able to download, view and forward chat files.
- **End-to-End Encrypted Chat*:** Toggles ability to use the Wave End-to-End Encrypted Chat feature.
- **Video Call:** Toggles ability to use the Wave Video Call feature.
- **Online Status:** Toggles ability to set Wave online status such as "Busy", "Appear Away", "Do Not Disturb", "Appear Offline", etc. If unchecked, the status will be displayed as only either "Online" or "Idle".
- **Clear Recent Call History:** Toggles ability to delete recent call history entries and entire recent call history on Wave.
- **Application:** Toggles ability to access the "Applications" page under Wave Desktop and Wave Web.
- **Smart Devices:** Toggling off privileges will hide the corresponding pages and options in Wave.
 - **Door System**
 - **Monitor**
 - **Call Device (CTI)**

- **3rd Party Applications**

- **App Store:** Toggles ability to access the Wave App Store. If unchecked, the App Store will be hidden, but installed apps can still be used.
- **Pre-installed Apps*:** Configure Wave pre-installed add-ins and related settings.

User Portal/Wave Privileges

- **Account Settings:** If unchecked, the *User Portal > Basic Information > Account Settings* page and the *Wave > Sidebar > User > Account Settings* option will be hidden.
- **Extension Settings:** If unchecked, the extension's *User Portal->Basic Information->Extensions* page and the *Wave > Sidebar > User > Call Settings* option will be hidden.
 - **Do Not Disturb:** Toggles ability to set DND through the User Portal.
 - **SIP/IAX Password & AuthID:** Toggles ability to access the **SIP/IAX Password** and **AuthID** settings under the *User Portal > Basic Information > Extensions > Basic Settings* page.
 - **Configuration Voicemail**
- **Manage Recordings:** Toggles ability to view recordings through the User Portal and Wave, including the recordings in call logs, meeting details, and Wave Application tab.
 - **Delete Recordings:** Toggles ability to delete recordings through the User Portal and Wave. For Wave, this includes the ability to delete call logs, meeting details, and recordings.
- **Personal Data:** Toggling off privileges will hide the corresponding pages and options in the User Portal and Wave.
 - CDR
 - Follow Me
 - Voicemail
 - Recordings files
 - Fax Files
 - SCA
- **Other Features:** Toggling off privileges will hide the corresponding pages and options in the User Portal and Wave.
 - Fax Sending
 - Call Queue
 - Schedule Call

*: Features which are marked by an asterisk are a part of RemoteConnect plans.

User Endpoint Access History

The User Endpoint Access History tab allows the administrator to view the access history of all extensions, the time on which the access has occurred, the IP addresses from which the extensions were accessed, and whether they were accessed from the User Portal, Wave Web/Desktop, or mobile. Extension access from the SIP endpoints won't be logged in this page.

User Management					
User Information	Custom Privilege	User Portal/Wave Privileges	<u>User Endpoint Access History</u>		
EXTENSIONS ⇅	NAME ⇅	EXTENSION TYPE ⇅	TERMINAL TYPE	LAST OPERATION TIME	IP ADDRESS
1000		SIP(WebRTC)	Android/iOS	2022-11-14 18:15:35	192.168.5.137
1001	Arthur Morgan	SIP(WebRTC)	Wave Web/Desktop Android/iOS User portal	2022-11-11 10:18:22 2022-11-14 16:10:37 2022-08-10 09:26:40	192.168.5.119 192.168.5.76 192.168.5.111
1002	Bonnie MacFarlan	SIP(WebRTC)	Wave Web/Desktop Android/iOS User portal	2022-11-11 10:18:22 2022-11-14 16:53:03 2022-08-23 15:49:30	192.168.5.93 192.168.5.79 192.168.5.111
1003	Catherine Braitwaite	SIP(WebRTC)	Wave Web/Desktop Android/iOS User portal	2022-10-18 14:14:12 2022-11-14 20:34:59 2022-11-17 09:19:44	192.168.5.111 192.168.5.76 192.168.5.168
1004	John Marston	SIP(WebRTC)	Wave Web/Desktop Android/iOS	2022-08-16 15:32:37 2022-11-16 11:26:44	192.168.5.111 192.168.5.91
1005	Abigail Roberts	SIP(WebRTC)	Wave Web/Desktop Android/iOS	2022-11-16 17:42:47 2022-11-16 18:11:34	192.168.5.79
1006	Mary-Beth Gaskill	SIP(WebRTC)	Wave Web/Desktop Android/iOS	2022-11-16 16:21:14 2022-11-16 17:37:50	192.168.5.165 192.168.5.79
1007	Hosea Matthews	SIP(WebRTC)	Wave Web/Desktop	2022-11-16 17:50:35	192.168.5.63

Total: 8

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Login Settings

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](#)

Enter Old Password	Enter the Old Password for UCM630xA
Change Password	Enable Change Password
Enter New Password	Enter the New Password for UCM630xA
Re-enter New Password	Retype the New Password for UCM630xA
Change Username	Enable Change Username
Please enter the username	Enter the Username
Email Address	The Email address is the User Email Address. It is used for receiving password information if the user forgets his password.

Change Password

Change Username

UCM630xA allows users now to change Super Administrator username.

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**

Change Binding Email

Email Address	Email Address is used to retrieve password when password is lost
----------------------	--

Change Binding Email option

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.

The screenshot shows the 'Login Settings' window with the 'Login Security' tab selected. It contains three input fields: 'User Login Timeout' (10), 'Maximum number of login attempts' (5), and 'User ban period' (5). Below these are two tables: 'Login Banned User List' and 'Login Whitelist', both currently empty with 'No Data' displayed. A search bar is present in the 'Login Banned User List' section.

Login Timeout Settings

Remote Login

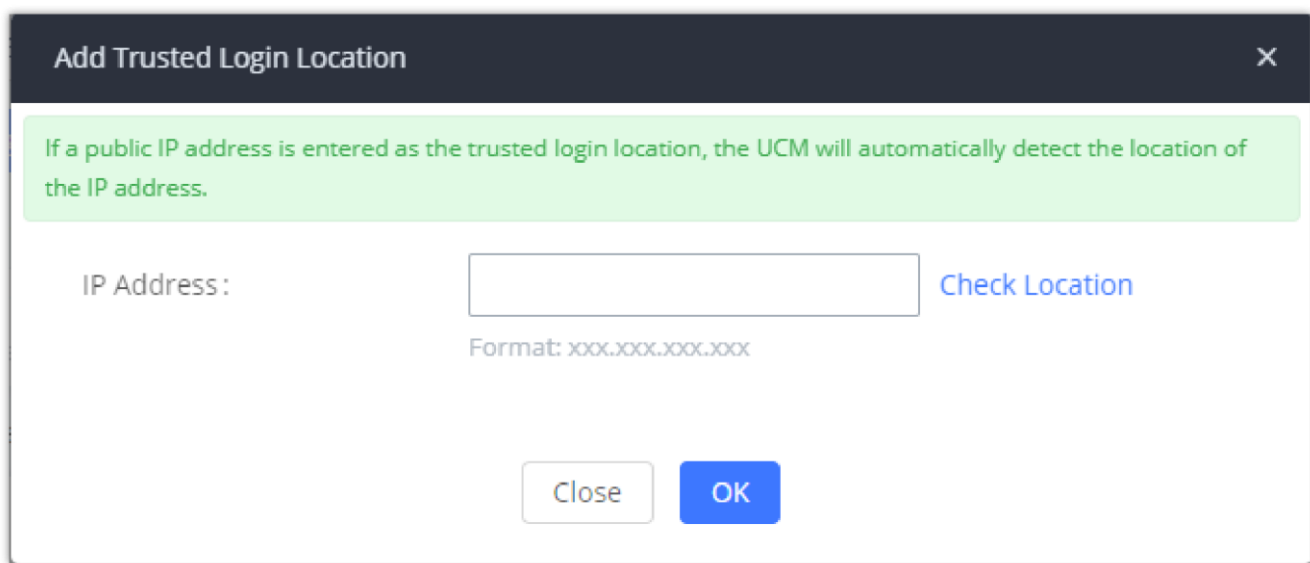
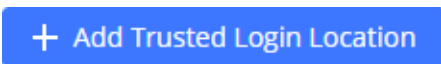
This feature allows the user to manage trusted login locations, also, verifying where login sessions were initiated from, this is very important since, in this type of scenario, the UCM6300 would be directly connected to the Internet, and the public IP address would be used for the remote login. This feature adds a layer of visibility and control, thus enhancing the security of the UCM.

In this tab there are two types of lists of locations:

- Trusted Login Locations: These are the trusted login locations that are added manually by the admin. Any added trusted login location will not generate any remote login alert upon the first time login.
- Other Login Location: This list will show all the remote login locations that are not trusted, logging in for the first time from an untrusted login location will generate an alert, but the subsequent remote logins from the same location will not generate alerts.

The screenshot shows the 'Login Settings' window with the 'Remote Login' tab selected. It features a '+ Add Trusted Login Location' button and a table with two rows: 'Trusted Login Locations (1/5)' and 'Other Login Locations'. The table has columns for 'LOGIN LOCATION' and 'OPTIONS'. A pagination control shows '1' of 1 items.

To add a trusted login location, the user must click on



Then add the public IP address of the location, click on “**Check Location**” to verify if it’s the correct location then click “**OK**”.

Note
The system administrator can add up to 5 Trusted Login Locations, while Other Login Locations can have an unlimited number of entries.

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”.

DATE	USERNAME	IP ADDRESS	RESULTS	PAGE OPERATION	SPECIFIC OPERATION	REMARK
2022-08-04 15:48:02	admin	192.168.5.111 (Marrakesh, Marrakech-Safi, MA)	Operation successful	Extensions: Login	Username: admin.	Click to modify notes
2022-08-04 15:47:53	admin	192.168.5.111	Wrong account or password!	Extensions: Login	Username: admin.	Click to modify notes
2022-08-04 15:47:43	admin	192.168.5.111	Wrong account or password!	Extensions: Login	Username: admin.	Click to modify notes
2022-08-04 15:17:52	admin	192.168.5.111 (Marrakesh, Marrakech-Safi, MA)	Operation successful	Extensions: Login	Username: admin.	Click to modify notes
2022-08-04 14:44:19	admin	192.168.5.111 (Marrakesh, Marrakech-Safi, MA)	Operation successful	Extensions: Login	Username: admin.	Click to modify notes
2022-08-04 14:26:53	admin	192.168.5.111 (Marrakesh, Marrakech-Safi, MA)	Operation successful	Extensions: Login	Username: admin.	Click to modify notes
2022-08-04 14:14:39	admin	192.168.5.111 (Marrakesh, Marrakech-Safi, MA)	Operation successful	Extensions: Login	Username: admin.	Click to modify notes
2022-08-04 12:26:36	admin	192.168.5.111	This trusted login location already exists.	addCommonLoginAddr	Details	Click to modify notes
2022-08-04 12:26:26	admin	192.168.5.111	This trusted login location already exists.	addCommonLoginAddr	Details	Click to modify notes
2022-08-04 12:15:54	admin	192.168.5.111 (Marrakesh, Marrakech-Safi, MA)	Operation successful	Extensions: Login	Username: admin.	Click to modify notes

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.

Date	The date and time when the operation is executed.
Username	The username of the user who performed the operation
IP Address	The IP address and geographical location from which the operation has been made.
Results	The result of the operation.
Page Operation	The page where the operation is made. For example, login, logout, delete user, create trunk and etc.
Specific Operation	Click on the hyperlinked operation detail to reveal more details.
Remark	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".

Operation Logs Filter

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

To delete operation logs, users can perform filtering first and then click on

Delete Search Result (s)

to delete the filtered result of operation logs. Or users can click on

Clear

to delete all operation logs at once.

Upgrading

During its lifetime, the UCM6300 Audio Series models will receive regular updates. These updates include new features and functionalities, bug fixes, security patches, and general improvements to the security and performance of the device. It is highly recommended that you keep the firmware of your UCM up to date.

Before proceeding with the firmware upgrade, please visit <https://www.grandstream.com/support/firmware> and download the latest firmware available for the UCM. Once the file has been download, please decompress the file, you will get a .bin file, that's the file which will uploaded to the UCM.

To upgrade the firmware of the UCM6300 Audio Series, please access the web UI of the UCM, then navigate to **Maintenance** → **Upgrade**

Upgrade Firmware

1. Click on "Choose File to Upload" field in front of "Firmware File Path"
2. Use the window to navigate to where the firmware file is located then select it.
3. Once the firmware file has been selected, the file will be uploaded to the UCM and the upgrade will begin.
4. A prompt will appear to reboot the device to finalize the upgrade.

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.

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, SFTP, 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

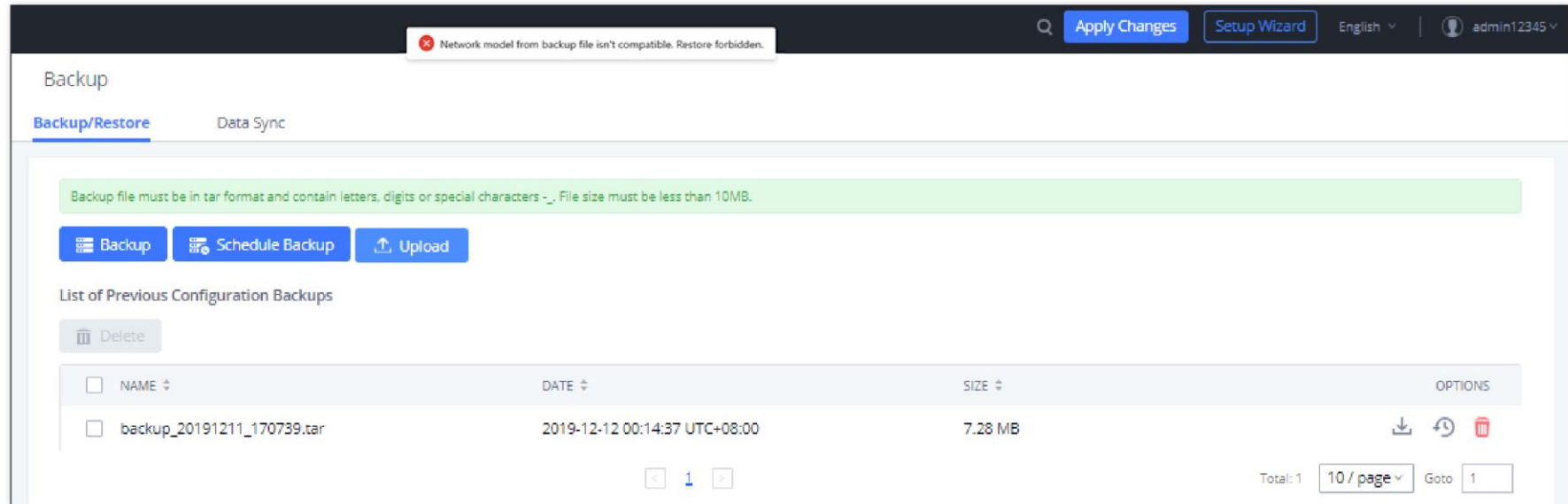


to upload backup file from the local device to UCM630xA. The uploaded backup file will also be displayed in the web page

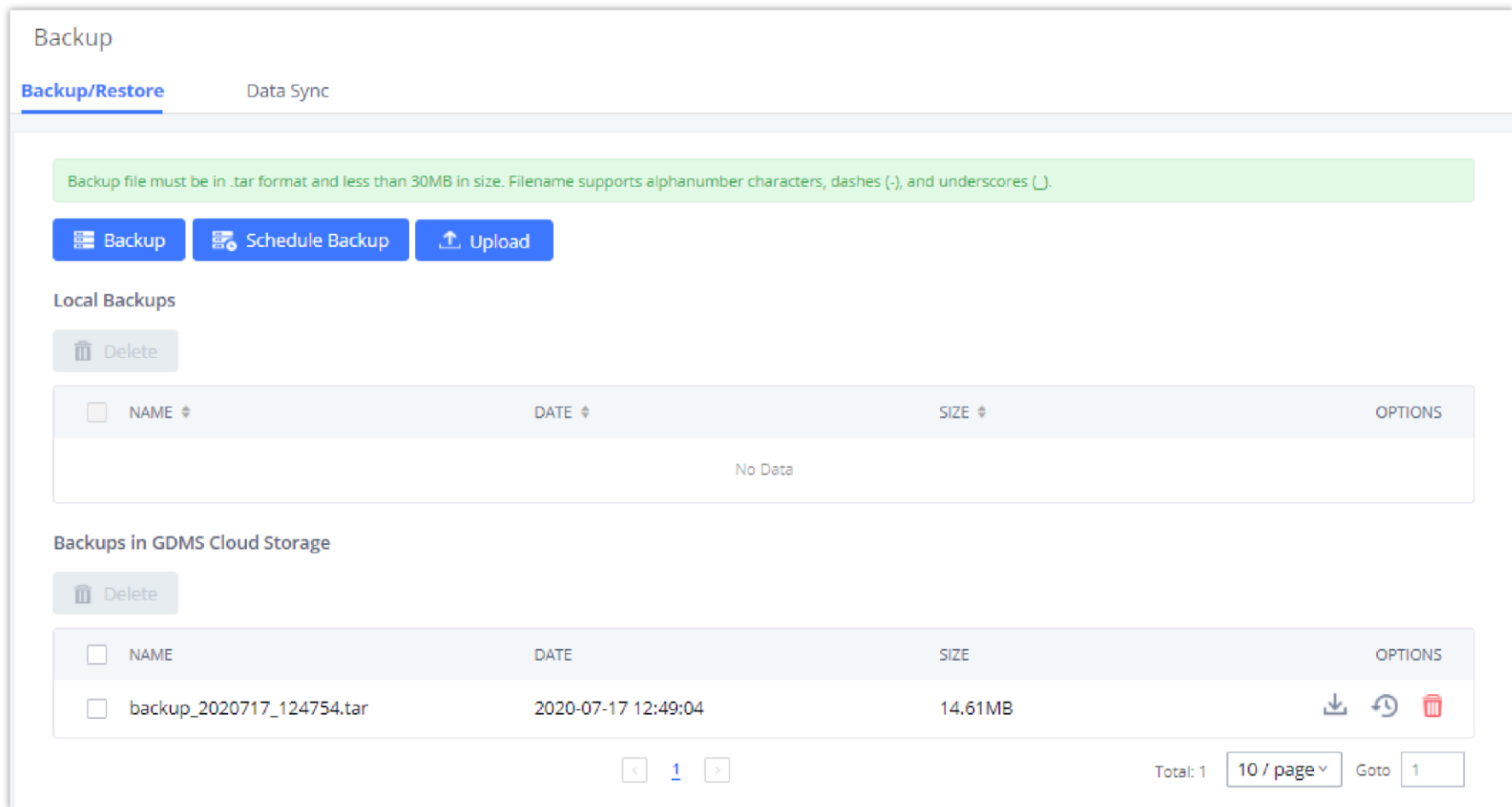
and can be used to restore the UCM630xA.

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:

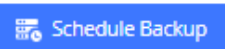


Restore Warning



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

[+ Test Connection](#)

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 Recording Files

Voice Mail Fax

* Account:

Password:

* Server Address:

Destination Directory:

* Sync Time:

[+ Test Connection](#) [+ Synchronize All Data](#)

Data Sync Log

[Clean](#)

No record to view

Data Sync

Enable Data Sync	Enable the auto data sync function. The default setting is "No".
Account	Enter the Account name on the SFTP backup server.
Password	Enter the Password associate with the Account on the SFTP backup server.
Server Address	Enter the SFTP server address.
Destination Directory	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.
Sync Time	Enter 0-23 to specify the backup hour of the day.

Before saving the configuration, users could click on

[+ Test Connection](#)




. 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

[+ Synchronize All Data](#)

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.

Note

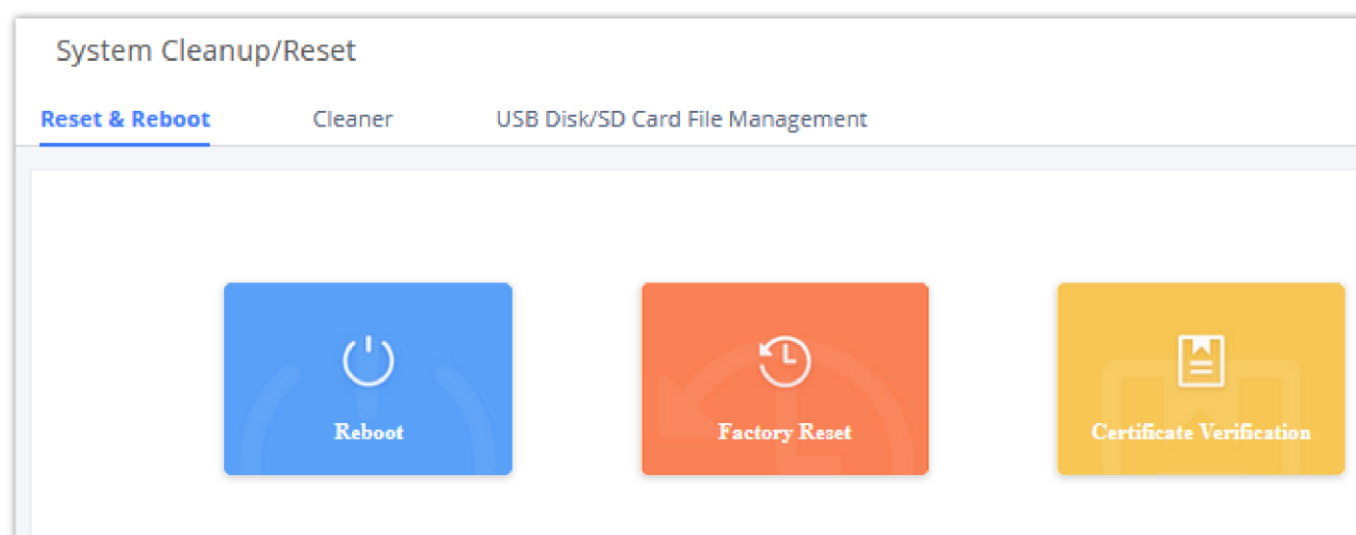
The uploaded backup file must be a tar file with no special characters like *,!,#,@,&,\$,%^^,(,),/,\\,space in the file name. The uploaded back file size must be under 10MB.

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.



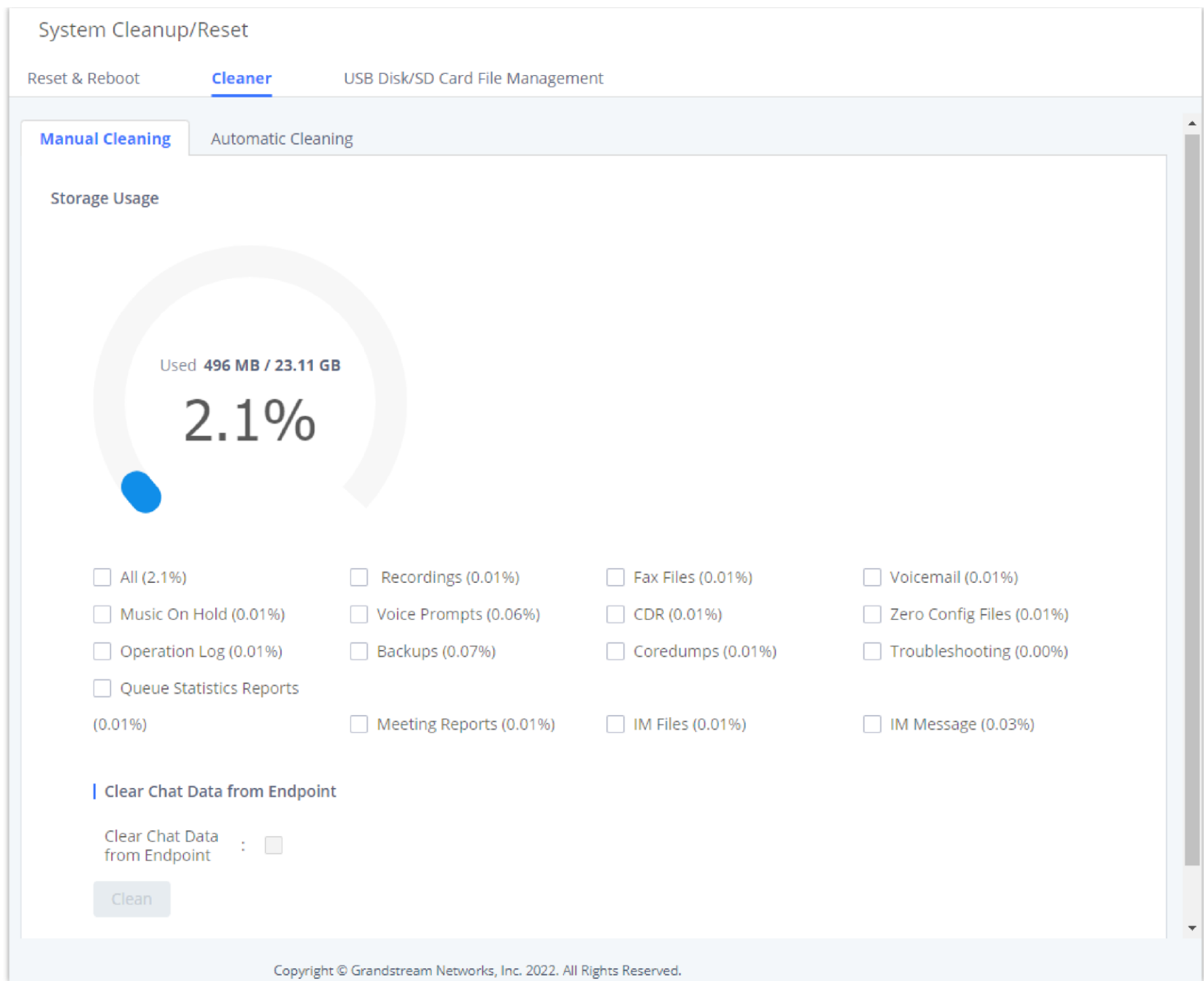
Reset and Reboot

- 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.



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.

Automatic Cleaning

Automatic Cleaning Configuration

Enable CDR Cleaner	Enable the CDR Cleaner function.
CDR Clean Time	Enter 0-23 to specify the hour of the day to clean up CDR.
Cleaning Conditions	<p>By Schedule: 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>Keep Last X Records: 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>Keep Last X Days: Delete all entries older than X days.</p>
Clean Interval	Enter 1-30 to specify the day of the month to clean up CDR when By Schedule is selected as Cleaning Conditions .

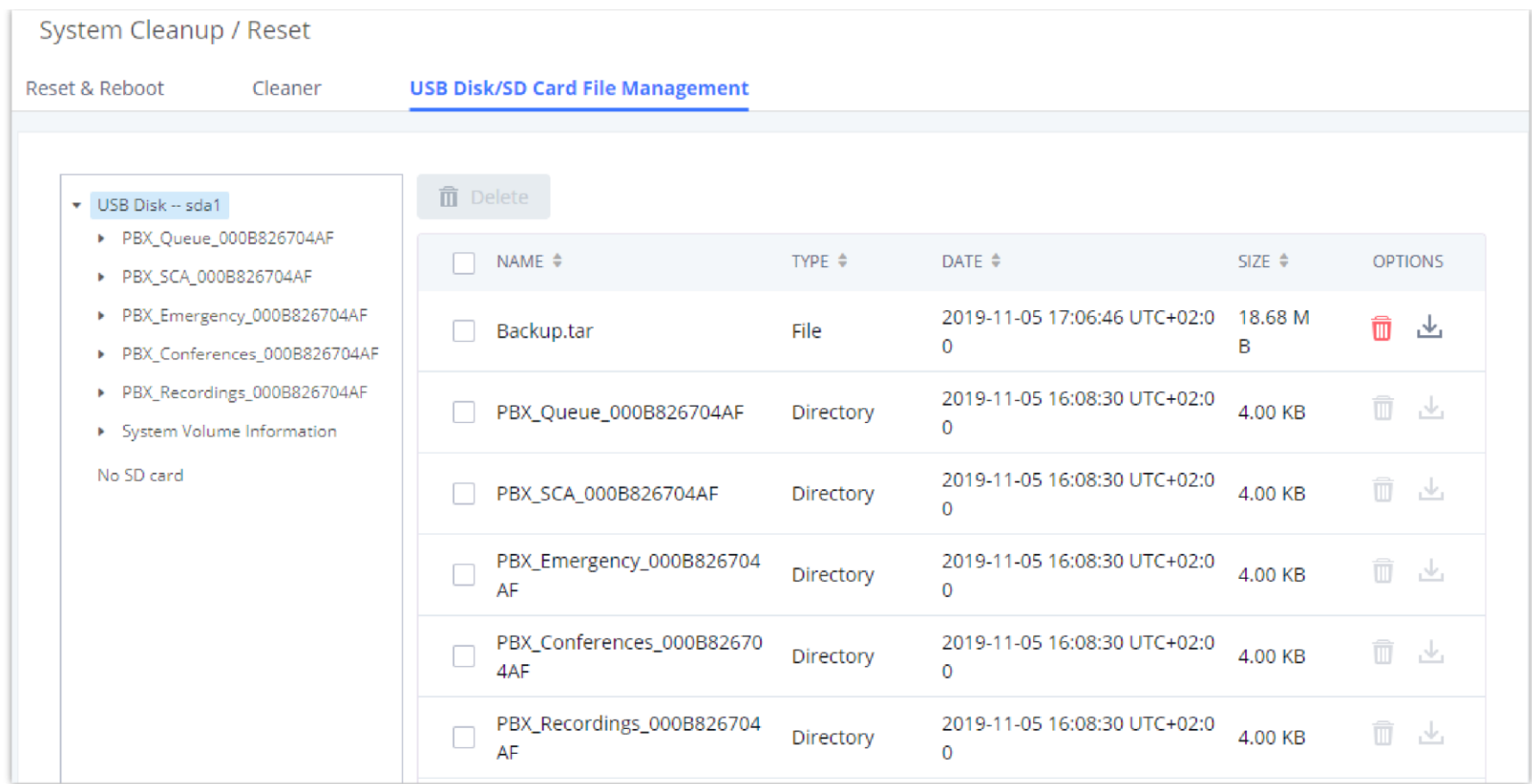
Max Entries	Set the maximum number of CDR entries to keep when Keep Last X Records is selected as Cleaning Conditions .
Keep Last X Day	Enter the number of days of call log entries to keep when Keep Last X days is selected as Cleaning Conditions .
Enable Queue Statistics Report Cleaner	Enable scheduled queue log cleaning. By default, is disabled.
Queue Statistics Report Cleaner Clean Time	Enter the hour of the day to start the cleaning. The valid range is 0-23.
Cleaning Conditions	<p>By Schedule: 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>Keep Last X Records: 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>Keep Last X Days: Delete all entries older than X days.</p>
Clean Interval	Enter how often (in days) to clean queue logs when By Schedule is selected as Cleaning Conditions . The valid range is 1-30.
Max Entries	Set the maximum number of Queue Statistics entries to keep when Keep Last X Records is selected as Cleaning Conditions .
Keep Last X Day	Enter the number of days of call log entries to keep when Keep Last X days is selected as Cleaning Conditions .
Enable Meeting Statistics Report Cleaner	Enable scheduled Meeting log cleaning. By default, is disabled.
Cleaning Conditions	<ul style="list-style-type: none"> ○ By Schedule: If the clean interval is 3, cleaning will be performed every 3 days to remove all records that were generated 3 days ago. ○ Keep Last X Records: 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.) ○ Keep Last X Days: Delete all entries older than X days.

Clean Interval	Enter how often (in days) to clean queue logs when By Schedule is selected as Cleaning Conditions . The valid range is 1-30.
Max Entries	Set the maximum number of Meeting Statistics Report entries to keep when Keep Last X Records is selected as Cleaning Conditions .
Keep Last X Day	Enter the number of days of call log entries to keep when Keep Last X days is selected as Cleaning Conditions .
Enable File Cleaner	Enter the Voice Records Cleaner function.
Clean Files in External Device	If enabled the files in external device (USB/SD card) will be atomically cleaned up as configured.
Choose Cleaner File	Select the files for system automatic clean. <ul style="list-style-type: none"> ○ Basic Call Recording Files. ○ Meeting Recording Files. ○ Call Queue Recording Files. ○ Voicemail Files. ○ Backup Files.
Clean time	Enter the hour of the day to start the cleaning. The valid range is 0-23.
Cleaning Conditions	<ul style="list-style-type: none"> ○ By Schedule: If the clean interval is 3, cleaning will be performed every 3 days to delete all files. ○ By Threshold: 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. ○ Keep Last X Days: Delete all files older than X days.
File Clean Interval	Enter 1-30 to specify the day of the month to clean up the files.
File Clean Threshold	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.
Keep Last X Days	Automatically delete all recordings older than this x days when the threshold is reached. If not set, all data is cleared
Cleaner Log	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.



USB/SD Card Files Cleanup

USB/SD Card Files Cleanup

Current Path	Displays the current path.
Directory	Select the directory user want to clean.
Delete Selected File	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.

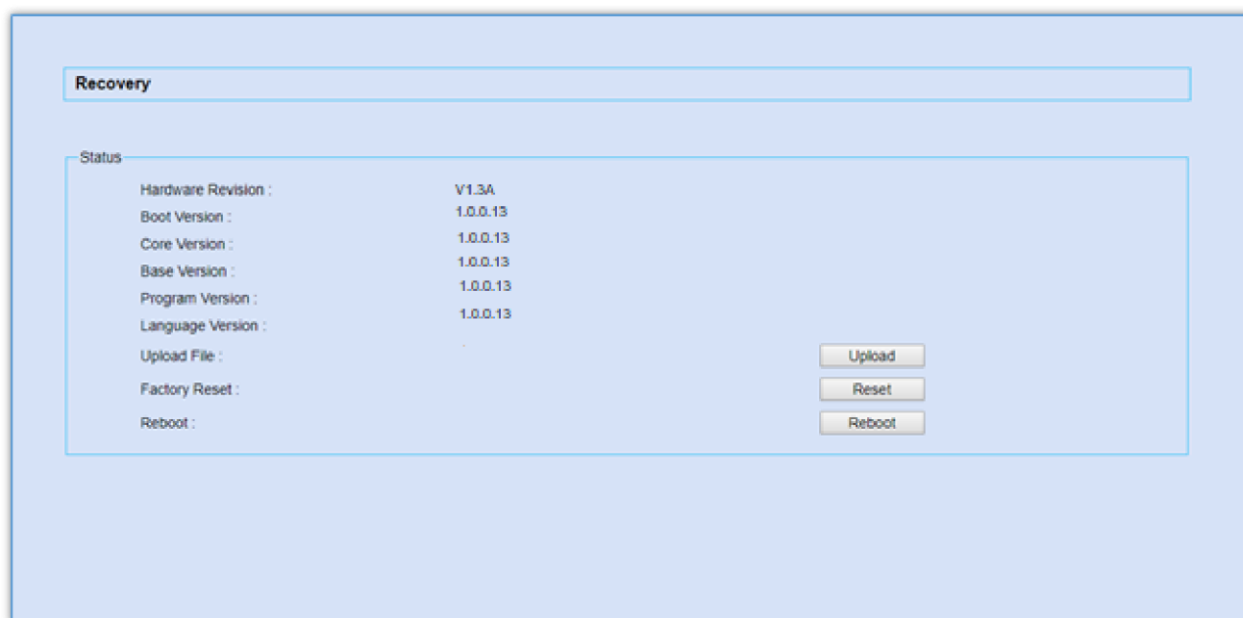
1. To access recovery mode on UCM, please follow below steps:
2. Remove the power from the unit and keep the network cable connected.
3. Press using a PIN the reset button and keep holding.
4. Plug back the power supply while maintaining the reset button pressed.
5. Wait for couple of seconds until you 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:



UCM6302A Recovery Web Page

Make sure to enter the correct admin password, and press login to access the recovery mode page :



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.</p>

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

Ethernet Capture allows capturing the traffic of the UCM for troubleshooting purposes. To access Ethernet Capture feature, please navigate to **Maintenance** → **Network Troubleshooting** → **Ethernet Capture**



Ethernet Capture

The capture packets can be stored locally and downloaded for analysis. However, if the user is diagnosing a randomly-occurring issue, he/she can run a continuous packet capture which can be limited by the size of the packet capture and the number of packet capture instances

! Important

When the maximum packet capture file size is reached, a new packet capture file will be created. When the maximum number of capture files number is reached, then the UCM will delete the oldest file created file and replace it with the new one

Capture Type	Ethernet Capture: Gets a packet capture of all network traffic going through the device. WebSocket Capture: Gets a packet capture of WebSocket protocol. Mainly used for troubleshooting Wave Web calling and conferencing issues.
---------------------	---

Interface Type	Select the network interface to monitor.
Capture Filter	Enter the filter to obtain the specific types of traffic, such as (host, src, dst, net, proto...).
Storage Location	<ul style="list-style-type: none"> • Local: Store the captured packets in the local storage. • SFTP Server: Save the capture trace to a SFTP server. Please make sure that SFTP is correctly configured under PBX Settings -> Online Storage -> SFTP Server • External Storage: Save the capture trace in a usb flash drive or an SD card. This requires that a USB flash drive or SD card to be plugged into the UCM. File formats supported are FAT32 and ExFat.
Save to External Storage	When or more external storage units are connected to the UCM6300 series, the user will be able to pick which one to use. Note: This option is available only when you choose "External Storage" as the storage destination of the capture trace.
Destination Directory	When SFTP is selected, this option will appear. Please enter the directory path in which you would like to store the captured packets.
Packet Capture Size	This option appears only when "External Storage" or "SFTP" options are selected. Define the packet capture size, the option available are: 50MB, 100MB, and 200MB.
Number of Packet Capture	Define the maximum number of the packets captured. The available options are 5, 10, and 20 packets.
	Start capturing network traffic.
	Stop capturing network traffic.
	Download the captured packets. This option can only be used when the captured packets are stored locally.
Enable SRTP Debugging	Check this box to troubleshoot calls encrypted with TLS/SRTP.

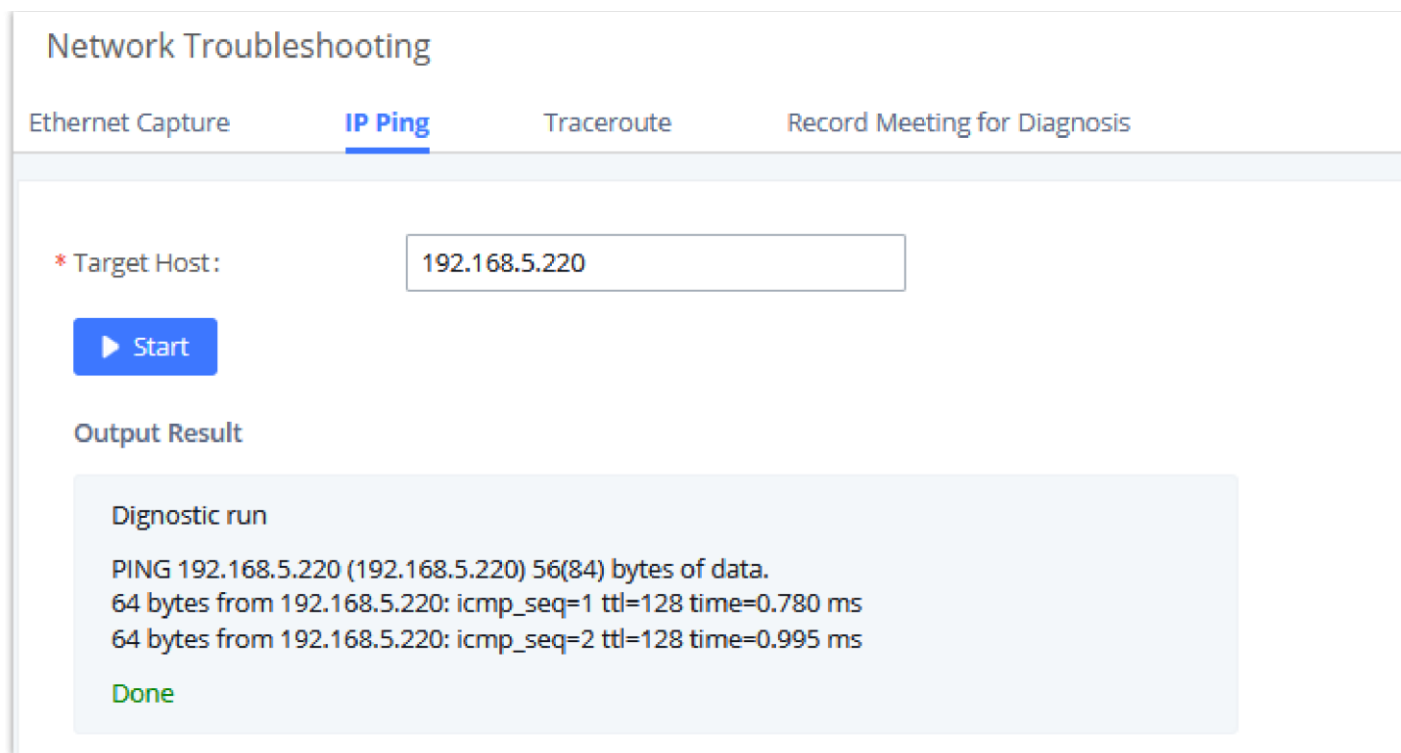
The output result is in .pcap format. Therefore, users could specify the capture filter as used in general network traffic capture tool (host, src, dst, net, protocol, port, port range) before starting to capture the trace.

Note

Capture files saved on external devices will now have "capture" prepended to file names.

IP Ping

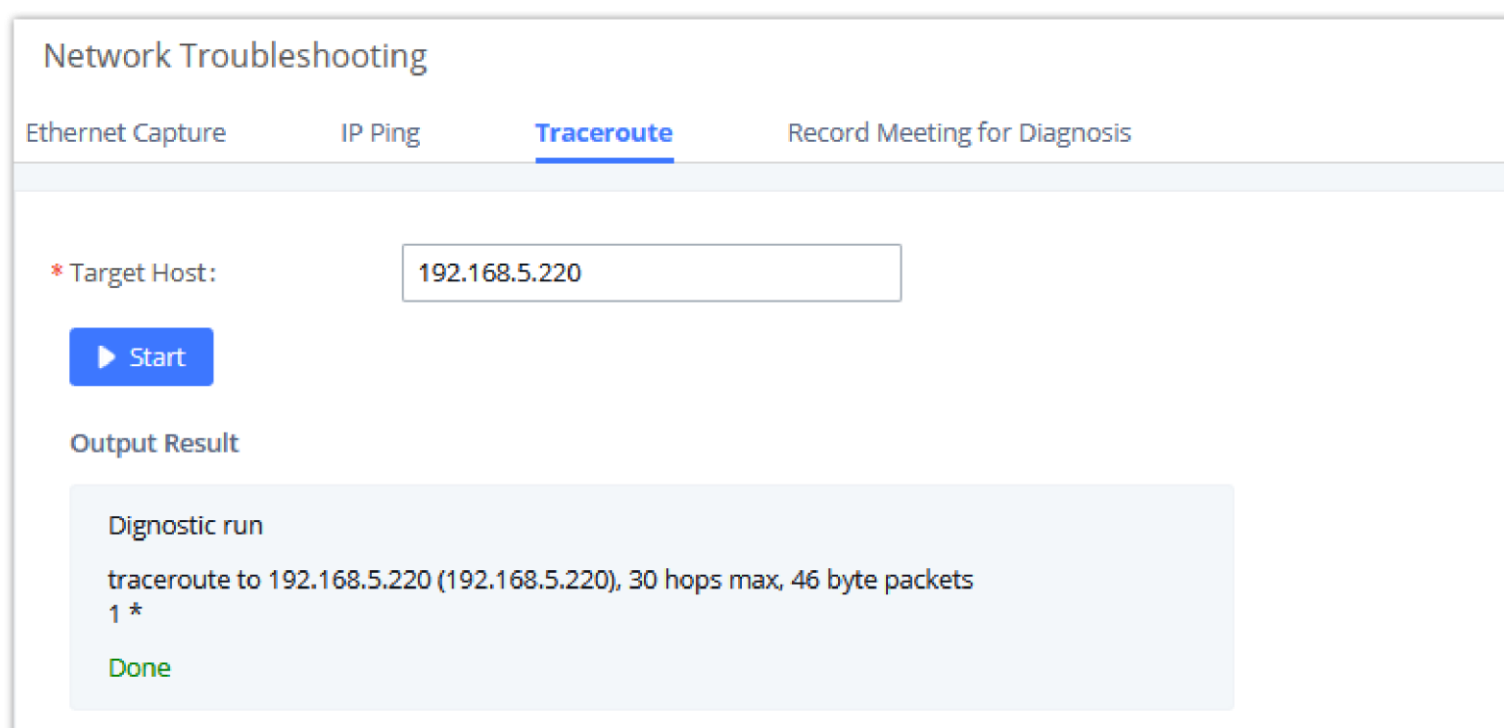
Enter the target host in host name or IP address. Then press "Start" button. The output result will dynamically display in the window below.



Ping

Traceroute

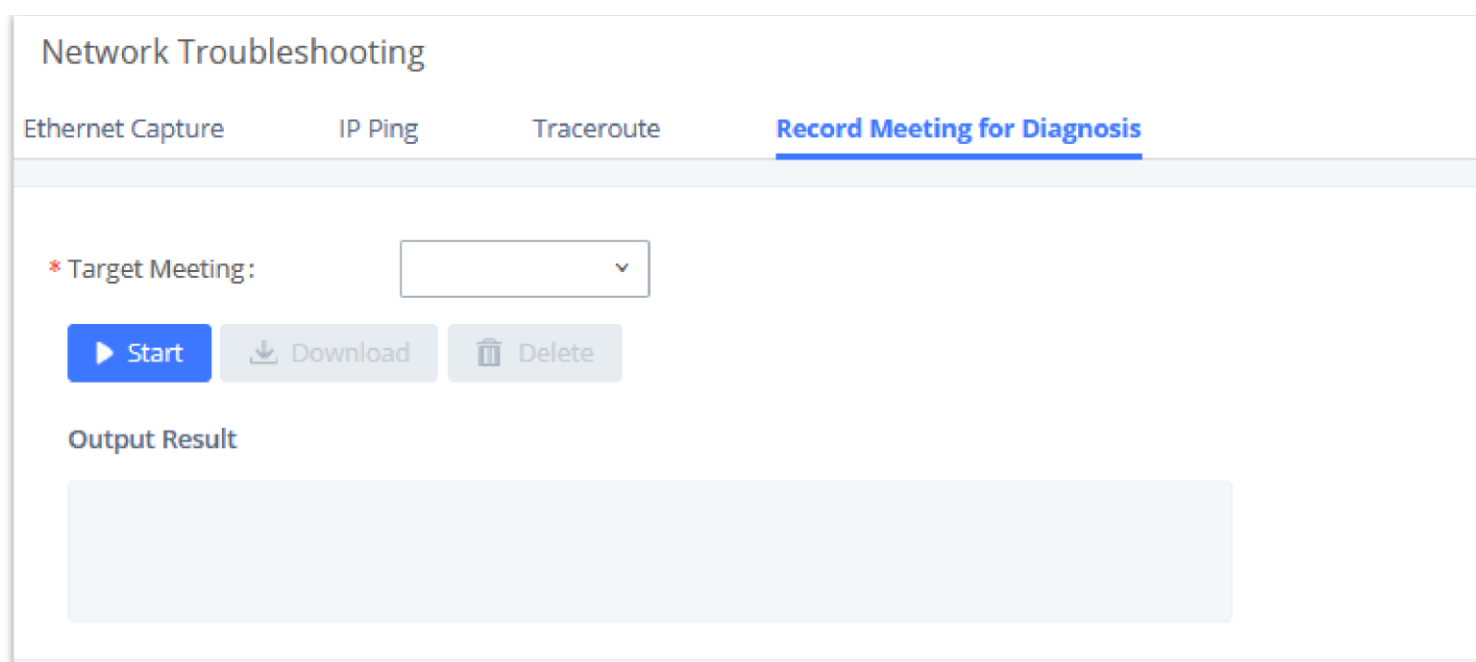
Enter the target host in host name or IP address. Then press "Start" button. The output result will dynamically display in the window below.



Traceroute

Record Meeting for Diagnosis

Enter the target meeting, support the ongoing meeting, and then click the "Start" button to capture the recording diagnosis of the meeting members in progress. The output result will be automatically displayed below, click the "download" button to download to the local. After the download is complete, immediately click the "Delete" button to clear the system content.



Record Meeting for Diagnosis

Signaling Troubleshooting

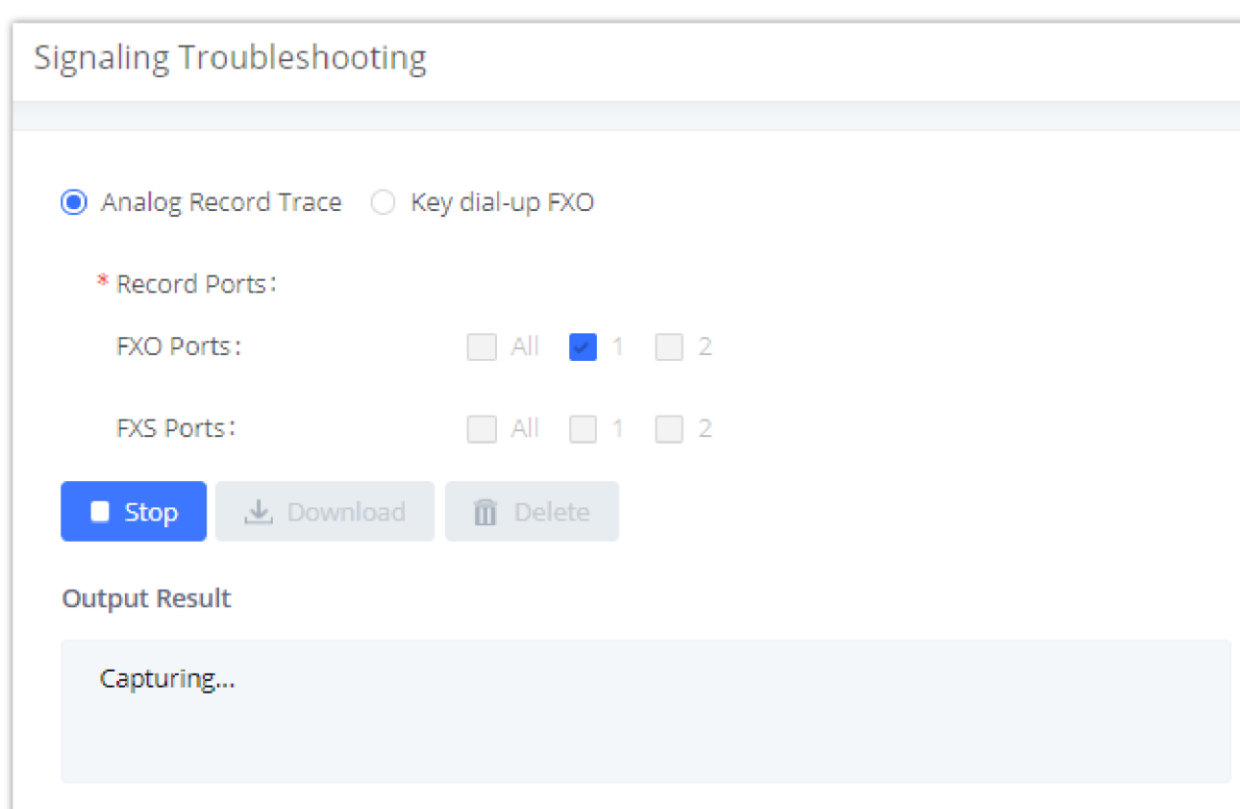
Analog Record Trace

- o **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".
5. Click on "Download" to download the analog record trace.



Troubleshooting Analog Trunks

- o 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".

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.

1. After capturing the trace, users can download it for basic analysis. Or you can contact Grandstream
2. Technical support in the following link for further assistance if the issue is not resolved.

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

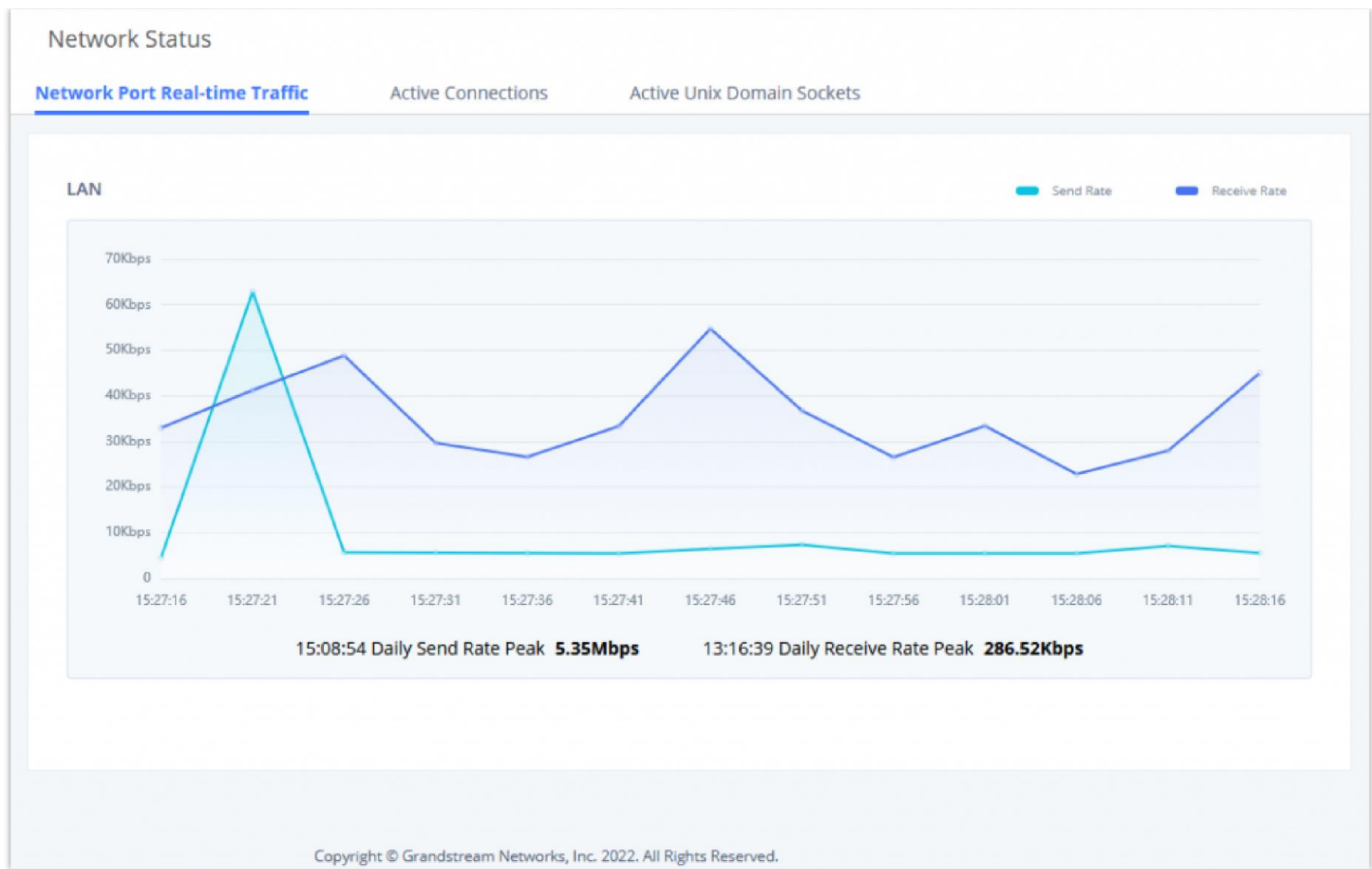
Service Check

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.

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

APPENDIX A: RFC STANDARDS USED IN THE UCM6300 AUDIO SERIES

- **RFC 3261** SIP: Session Initiation Protocol
- **RFC 3262** Reliability of Provisional Responses in SIP
- **RFC 3263** Session Initiation Protocol (SIP): Locating SIP Servers
- **RFC 3264** An Offer/Answer Model with the Session Description Protocol
- **RFC 3515** The Session Initiation Protocol (SIP) Refer Method
- **RFC 3311** The Session Initiation Protocol (SIP) UPDATE Method
- **RFC 4028** Session Timers in the Session Initiation Protocol (SIP)
- **RFC 2976** The SIP INFO Method
- **RFC 3842** A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
- **RFC 3892** The Session Initiation Protocol (SIP) Referred-By Mechanism
- **RFC 3428** Session Initiation Protocol (SIP) Extension for Instant Messaging
- **RFC 4733** RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals
- **RFC 4566** SDP: Session Description Protocol
- **RFC 2617** HTTP Authentication; Basic and Digest Access Authentication
- **RFC 3856** A Presence Event Package for the Session Initiation Protocol (SIP)
- **RFC 3711** The Secure Real-time Transport Protocol (SRTP)
- **RFC 5245** Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translator (NAT) Traversal for Offer/Answer Protocols
- **RFC 5389** Session Traversal Utilities for NAT (STUN)
- **RFC 5766** Traversal Using Relays around NAT (TURN): Relay Extensions to Session Traversal Utilities for NAT (STUN)
- **RFC 6347** Datagram Transport Layer Security Version 1.2
- **RFC 6455** The WebSocket Protocol
- **RFC 8860** Sending Multiple Types of Media in a Single RTP Session
- **RFC 4734** Definition of Events for Modem, Fax, and Text Telephony Signals

- **RFC 3665** Session Initiation Protocol (SIP) Basic Call Flow Examples
- **RFC 3323** A Privacy Mechanism for the Session Initiation Protocol (SIP)
- **RFC 3550** RTP: A Transport Protocol for Real-Time Applications

CHANGELOG

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 1.0.25.9

- Add Portugal Portuguese voice prompt support. [[LANGUAGE SETTINGS FOR VOICE PROMPT](#)]
- Add Wi-Fi related settings to GHP-W series template.

Firmware Version 1.0.25.7

- Multicast Community has been added. [[Multicast Community](#)]
- Scheduled Paging UI has been reworked. [[Scheduled Paging/Intercom](#)]
- Added support for playing MP3 files over GSC35XX devices. [[Upload Custom Prompt](#)]
- Added support for PMS billing system for Local PMS. [[Call Rate](#)]
- Mini bar has been improved. [[Mini Bar](#)]
- Added ability to set a Feature Code as a the Default Destination. [[Speed Dial](#)]
- Added ability to add External Number and LDAP numbers on Announcement Center. [[Announcement](#)]
- Added Fax Sending privilege when creating custom privileges. [[Custom Privilege](#)]
- Added enabling TURN Relay for a specific extension. [[Create New SIP Extension](#)]
- HA Data Sync Failure alert event has been added to the Alert Events List. [[Alert Events List](#)]
- SIP endpoints firmware can now be updated using the Zero Config page. [[Managing Discovered Devices](#)]
- Clearing the call history upon guest check-in or check-out has been added. [[Local PMS](#)]
- Onsite Meeting email invitations will include now an ics file. [[Onsite Meeting](#)]
- "Send File" Wave privilege has been added. [[User Portal/Wave Privilege](#)]

Firmware Version 1.0.23.17

- No major change.

Firmware Version 1.0.23.15

- Increased Fail2Ban list limit to 50 entries. [[Fail2ban](#)]
- NAS password field now supports special characters "&,\$'^`>|", [[PBX Settings/NAS](#)]
- Speed Dial limit has been increased to 1000. [[Speed Dial](#)]

Firmware Version 1.0.23.12

- Added support for limiting chat history viewing for newly added participants. [[Room](#)]
- Added support for including company number when scheduling a meeting. [[Meeting Schedule](#)]
- Added support for tracking outbound calls for QueueMetrics.
- Added support for displaying call queue names on GRP phones.
- Added support for auto answer on inbound route level to detect fax call. [[Inbound Rule Configurations](#)]
- Added support for updating the call forwarding status when enabled via feature code. [[Feature Codes](#)]
- Added support for controlling diversion header. [[SIP Settings](#)]
- Added "Extension Group" as a custom privilege. [[Custom Privilege](#)]

- Added ability to retrieve recording files saved on NAS/GDMS cloud through API.
- Added support for manually backing up the UCM configuration to SFTP. [[Backup/Restore](#)]
- Added disaster recovery for High Availability. [[HA](#)]
- Added feature code to blacklist last called number. [[Feature Codes](#)]
- Added ability to share voicemail to group members. [[VOICEMAIL](#)]
- Added ability to delete voicemails for all voicemail group from one member. [[VOICEMAIL](#)]
- Added ability to monitor voicemail groups with voicemail key on endpoints. [[VOICEMAIL](#)]
- Added "Ignore Safe Operational Flow" to Network Port Traffic Control. [[Network Settings](#)]
- Added support for domains in API ACL.
- Added support for configuring chat filesize limit. [[IM Settings](#)]
- Added support for "announce message caller-ID" in Announcement Center. [[Announcement Center Settings](#)]
- Added support for preventing Wave chat deletion. [[User Portal/Wave Privilege](#)]
- Added "Priority Call" feature code. [[Feature Codes](#)]
- Added support for updating time zones under Time Settings. [[Time Settings](#)]
- Added Live Chat feature. [[Live Chat](#)]
- Added Message Broadcast feature. [[Message Broadcast](#)]

Firmware Version 1.0.21.9

- Added SMS service support. [[SMS Settings](#)]
- Added support for creating shared departments. [[Department Management](#)]
- Added support for RADIUS login. [[RADIUS](#)]
- Added support for private intercom. [[Configure Private Intercom](#)]
- Added support for setting an extension as the default destination in Click2call. [[Integrated Customer Service](#)]
- Added support for SIP TLS cipher suite. [[SIP Settings/TCP and TLS](#)]
- Added support for continuous packet capture when using USB/SD card storage or SFTP. [[Ethernet Capture](#)]
- Added support for resetting TLS certificates to the default ones. [[SIP Settings/TCP and TLS](#)]
- Added support for SSH token. [[SSH Access](#)]
- Added support for subscribing to a voicemail group. [[Configure Voicemail Group](#)]
- Added support for setting separate call forwarding conditions for external and internal calls. [[Create New SIP Extension](#)]
- Added support for forwarding calls to a custom prompt. [[Create New SIP Extension](#)]
- Added support for external numbers to opt out of being recorded when calling into the UCM. [[PBX Settings](#)]
- Added support for caller name look up. [[Inbound Routes](#)]
- The name of the agent will now be displayed in the switchboard. [[Switchboard](#)]
- Agent pause can now be performed quickly by dialing the respective feature code and the corresponding pause reason without having to interact with the IVR. [[Configure Call Queue](#)]
- Agent pause reason will now be displayed in the switchboard. [[Switchboard](#)]
- Added support for enabling the welcome prompt to be played simultaneously with background music while the agent phone is ringing. [[Configure Call Queue](#)]

Firmware Version 1.0.19.10

- Specific Time configuration is now included in the extension exports

Firmware Version 1.0.19.9

- Optimized various system processes
- Added Onsite Meeting feature. [[Onsite Meeting](#)]
- Added ability to customize extension call waiting tone [[General Call Prompt Tones](#)]

- Updated Zoho CRM authentication process [[Zoho Telephony](#)]
- {VM_DATE} date value format has been changed to MM/dd/yyyy hh:mm:ss from DDD yyyy MMM hh:mm:ss. [[Configure Fax/T.38](#)] [[Voicemail Email Settings](#)]
- Added Device Name \${DEVICE_NAME} variable to Alert Events and Emergency Calls email templates
- Added Geolocation header support [[Emergency Location Mapping](#)]
- Added P-Called-Party-ID header option to the *Add/Edit Extension* -> *Features* page [[Create New SIP Extension](#)]
- Added *Allow Operator Panel Monitoring* extension option to toggle whether the Operator Panel can monitor the extension. [[Create New SIP Extension](#)]
- Added *Basic* extension export option [[Export Extensions](#)]
- *Allow Call-barging Extension List* option changed to *Call Monitoring Whitelist* [[Create New SIP Extensions](#)] [[Create New IAX Extension](#)]
- Added *Silence Suppression* option to *Extensions/VoIP Trunks* page [[Create New SIP Extension](#)] [[VoIP Trunk Configuration](#)]
- If a storage device is full, the UCM will mark it as unavailable and automatically change file storage path to the next available location based on the *Storage Path Priority*. Previously, UCM would change the file storage path to its own local storage if external storage was full. [[File Manager](#)]
- Added new commands related to call queue and Wave [[API Configuration Parameters](#)]
- Added support for multiple API (new) users [[API Configuration Parameters](#)]
- Added the *Default Certificate Auto Renewal* option. If enabled, the default browser certificate will be automatically renewed after 398 days (the max certificate validity period of Chrome, Firefox, and Safari browsers). User-defined certificates are not affected. [[HTTP Server](#)]
- Added ability to sync local IM data to Cloud IM [[Cloud IM Service](#)]
- Added ability create custom IVR key presses [[Custom Key Event](#)]
- Added *Chat Data from Endpoint* option to the Maintenance -> System Cleanup/Reset -> Cleaner page. If enabled, this option will clean out chat data from Wave clients at the same time as the UCM's server-side automatic/manual cleaning of chat data. [[Cleaner](#)]
- Added support for meeting room passwords. However, meetings cannot be scheduled for rooms with meeting password enabled. [[Room](#)]
- Meeting kick warning interval has been changed from 30 mins to 20 mins. Note: This kick warning will only play when there is only 1 person in a meeting room, and if they do not opt to stay in the meeting room after the warning, they will be removed from the meeting room after 5 minutes.
- ARP will now be used instead of ping to check NAS connectivity.
- Upgrade logs will now contain firmware version information.
- Queue chairmen can now log out dynamic agents
- Added option to automatically reset user/Wave password upon check-in/check-out. [[Local PMS](#)]
- Added option to clear Wave chat history automatically upon check-in or check-out. [[Local PMS](#)]
- Added Local PMS functionality [[Local PMS](#)]
- Check-out will no longer reset the "Skip Voicemail Password Verification" extension setting
- Added ability to assign two extensions to a room [[Room Management](#)]
- Added option to clear scheduled wakeup calls on both check-in and check-out. [[Local PMS](#)]
- Added the ability to change the default call privilege of a room. A room's privilege will be reset to this value after a guest checks out of it.
- Added support for stereo audio recording [[General Settings](#)]
- Added option to route calls based on a caller's Diversion header value [[Inbound Rule Configuration](#)]
- Added ability to control whether to use failover trunks based on the call response codes [[Failover Trunk Toggles](#)]
- Added support for H.264 with multiple payload types in SDP
- When receiving an INVITE with no SDP, following INVITEs with SDP will offer H.264 1080p resolution by default.
- In the scenario where an inbound external call is forwarded from an extension to an external number, the Contact header will now use the CID of the forwarding extension instead of the caller's CID.

- Removed *External Device Usage Threshold* option. If a connected NAS has only 1GB remaining available storage space, it will be considered unavailable and trigger the external disk usage alert. [[NAS](#)]
- Added *User Endpoint Access History* page [[User Endpoint Access History](#)]
- Added User Portal/Wave privilege control [[User Portal/Wave Privilege](#)]
- *Dial Trunk* option has been renamed to *Dial External Number* and moved to the *Dial Other Extensions* section
- The Wave Welcome email will now use the port number configured in System Settings->HTTP Server->Wave Settings->Port if the Wave Settings->External Host value is not a RemoteConnect address or does not contain a port number.
- Added links to relevant online documentation to various pages of the UCM webUI.
- Added Phonebook VMPK mode to GRP261x template
- Added Firmware tab for improved firmware management [[Firmware](#)]
- Added ability delete downloaded base model templates in the Model Update page
- Added ability to search for templates via the device model name
- Added ability to select either LAN1 or LAN2 to scan for devices on when using dual network method

Firmware Version 1.0.17.11

- Several system process optimizations

Firmware Version 1.0.17.8

- Updated python version to 3.8 and related processes.
- Improved speed of applying changes
- Updated lighttpd version to 1.4.61.
- The Privilege Name field now supports parenthesis ().
- The Contacts page has been moved to its own category in the sidebar.
- Added <https://www.zohoapis.in> option to CRM Server Address list.
- Added Channel Path option for accessing specific IP camera channels via URL. [[Device Management](#)]
- Improved alert email sending process.
- Updated Remote Registration email template.
- Emergency calls will no longer be restricted by the RemoteConnect call limit.
- The default highest priority codec is now G.722.
- Added Remote Extension Privilege Update feature code and Remote Extension Privilege Update Whitelist field to allow specified users to remotely change extensions' privileges. [[Feature Codes](#)]
- Automatic file migration after file storage path failure to the next storage location in the storage priority path.
- Users can now customize the storage path priority for recordings, and IM files. [[File Manager](#)]
- FXO FSK CID detection now uses spandsp.
- If SIP extensions synced from UCMs are deleted on GDMS, they will no longer be synced again.
- If HA is enabled, the HA cluster IP address will now be provisioned as the config server to endpoints instead of the active UCM's IP address.
- Added support for configuring inbound route blacklist via HTTPS API.
- If dialing into a Dial by Name directory, the call will end automatically after failing 3 times.
- Added Server Type option to the LDAP Server→LDAP Phonebook→Phonebook Download Configurations page. Users can select between LDAP and Active Directory. [[LDAP Server](#)]
- Added Department field to LDAP phonebook contacts. [[LDAP Server](#)]
- Added meeting room extensions to LDAP phonebooks. [[LDAP Server](#)]
- External Contacts created from the Contacts page will now be added to the system's internal LDAP phonebook. [[LDAP Server](#)]
- Added Remote Login tab to the Maintenance→Login Settings page. [[Maintenance](#)]

- Created new Meetings Settings page under the Multimedia Meetings page and moved several meeting-related options to it. [[Multimedia Meeting](#)]
- Regular meeting participants can now invite other members to join the meeting by dialing 1 if "Allow User Invite" is enabled. [[Multimedia Meeting](#)]
- Meetings will become "Pending" after rescheduling.
- Pending meetings are now sorted by start time by default.
- Added the Allowed to Override Host Mute option to the Edit Meeting Room and Schedule Meeting pages to allow participants to unmute themselves even after the meeting host mutes them. [[Multimedia Meeting](#)]
- Added support for user authentication. [[OpenVPN](#)]
- Operation Log entries will now contain the IP address and location information from which the operation originated. [[Operation Log](#)]
- Added option Automatically Clear Wakeup Calls for deleting scheduled wakeup calls after either guest check-in and check-out. [[PMS Features](#)]
- Users can now dial the Update PMS Room Status feature code, the maid code, and the room status code all at once to change room status. [[PMS Features](#)]
- Added a Scan button to manually retrieve the list of recordings on external storage. The UCM automatically displays up to 5000 recordings on attached external storage, but pressing this button will allow the UCM to display more. [[Recording Files](#)]
- Added ability to batch delete cloud storage files.
- If IP endpoints cannot connect to the GDMS TURN server via UDP, UCM will use TCP to connect them.
- Added Trunk Registration Period (s) option to SIP Settings->Misc. [[SIP Settings](#)]
- Added option Special Attributes to the Extension/Trunk->VoIP Trunks->Edit SIP Trunk->Advanced Settings page. If enabled, the following attributes will be included in the SIP SDP: ssrc, msid, mid, ct, as, tias, record. Enabling this may cause compatibility issues with non-Grandstream devices. [[VoIP Trunk Configuration](#)]
- Added CEI msid for audio calls.
- Added trickle-ice param to SIP OPTION's 200 OK.
- profile-level-id will be added to 200 OK when receiving INVITEs without SDP.
- Added ability to import/export speed dials. [[Speed Dial](#)]
- Improved processes to avoid duplicate alerts for the following events: Registered SIP Trunk failed, Local Disk Usage and External Disk Usage.
- Separated Allow Deletion of CDR and Recordings option to Allow Deletion of CDR and Allow Deletion of CDR Recordings. [[User Management](#)]
- Added the Call Waiting option to the User Portal.
- Added support for voicemail message seeking. When listening to voicemail, users can press star (*) to rewind 3 seconds or pound (#) to fast forward 3 seconds. [[Voicemail](#)]
- Added Line Selection Strategy option for Trunk Groups. [[VoIP Trunks](#)]
- Changed register trunk Username field name to Trunk Registration Number. [[VoIP Trunks](#)]
- [Web] General web UI improvements
- [Web] Added Help option under the username dropdown menu that will redirect to the UCM6300 Series FAQ.
- [Web] Updated some tooltips.
- [Web] Optimized search functionality
- [Zero Config] Added GMT+2:00 (Israel) option to Time Zone drop down list in all Zero Config pages.
- [ZeroConfig] Added support for WP22 and WP825 model templates.

Firmware Version 1.0.15.13

- Added SNMP monitoring feature. [[SNMP](#)]
- Added support to configure the time of a holiday. [[Holiday](#)]
- Added ability to determine the maximum total call duration per trunk for outbound calls. [[VoIP Trunk Configuration](#)]

- Added contact viewing privilege (independent from Department Contact Privilege). [[Contact Management](#)]
- Added support for agent ID announcement. [[Configure Call Queue](#)]
- Added support for Service Level Agreement for Call Queue. [[Service Level Agreement](#)]
- Added support for changing the Meeting room's name. [[Room](#)]
- Added WebRTC Trunk feature. [[WebRTC Trunks](#)]
- Extension data cleaning has been improved. [[Search and Edit an Extension](#)]
- Added support for SRTP Crypto Suite. [[VoIP Trunk Configuration](#)]
- STIR/SHAKEN has been improved. [[STIR/SHAKEN](#)]
- Added support for displaying the extension that initiated an emergency call in the emergency email notification. [[EMERGENCY](#)]
- Added support for collecting ICE candidates when an RTP connection is requested. [[RTP Settings](#)]
- Flood Attacks and Network Traffic Storm alerts have been added to the Alert Events List. [[Alert Events List](#)]
- Added support for Network Port Traffic Control for the ports of the UCM63xx Audio Series. [[Network Settings](#)]
- Added Support for limiting the frequency of calls that can be made in a period of time. [[Create New SIP Extension](#)]
- Added support for storing the local chat files in the GDMS. [[File Manager](#)]
- UCM RemoteConnect plan expiry screen has been improved.
- GDMS Cloud Storage Space details can now be viewed in the RemoteConnect menu. [[GDMS Cloud Storage Space](#)]

Firmware Version 1.0.13.9

- Added option to enable/disable DND status remotely for an extension. [[CALL FEATURES](#)]
- Added local proxy in IM settings. [[Cloud IM Service](#)]
- Added support for enabling/disabling auto audio recording for meeting. [[Auto Record](#)]
- Added privilege management for contacts. [[Privilege Management](#)]
- Improved fail2ban blacklist display. [[Fail2ban](#)]
- Improved email template for scheduling meeting. [[Email Templates](#)]
- Contacts sync-up between UCM and end points (wave/IP phones). [[LDAP Settings](#)]
- Added ability to specify DOD number based on outbound route. [[Outbound Routes DOD](#)]
- Fixed an issue where updating model templates will result in deleting the existing ones.
- Added support to use TURN Relay as an option to allow hosts behind NAT firewalls to communicate. [[VoIP Trunk Configuration](#)]

Firmware Version 1.0.11.10

- Added Operator Panel. [[OPERATOR PANEL](#)]
- Added time condition support for IVR key events. [[Key Press Event](#)]
- Added cloud IM abnormal alert event. [[Alert Events List](#)]
- Support setting to choose whether to play Follow Me.
- Add Fail2Ban whitelist comment information. [[Fail2ban](#)]
- Support Call Flip feature code. [[Feature Codes](#)]
- Added Multi-Factor Authentication for UCM login. [[Multi-Factor Authentication](#)]
- Added support for IoT device management. [[DEVICE MANAGEMENT](#)]
- Added option to enable and disable password-less remote access. [[UCM RemoteConnect Plan Settings](#)]
- Added option "Stop Ringing". [[Stop Ringing](#)]
- Add option "Email Missed Call Log". [[Email Missed Call Log](#)]
- Added remark for UCM system status. [[Remark](#)]
- Added option to enable and disable virtual queue call back keys settings. [[Virtual Queue Callback Key Setting](#)]

- Added Contacts section. [[Contacts](#)]
- Add option "Security Mode" for NAS settings. [[Security Mode](#)]
- Removed display for consumer users in user management page. [[User Management](#)]
- Support custom ignoring 180 response after 183 response. [[SIP Settings/MISC](#)]
- Added option to enable IPv6 for HA settings. [[Enable IPv6](#)]

Firmware Version 1.0.9.10

- Added Support for import/export Zero Config. [[Global Policy](#)]
- Added support for enable Wave and Sync Contact under the extension. [[Create New SIP Extension](#)]
- Added support for Custom time supplement time conditions. [[Create New SIP Extension](#)]
- Added support for Call Restriction. [[RESTRICT CALLS](#)]
- Added support for Queue Metrics. [[QUEUE METRICS](#)]
- Added support for CDR API add whitelist. [[Permitted IP \(s\)](#)]
- Added support for Call queue satisfaction survey. [[Queue Statistics](#)].
- The old API Configuration is reopened for use. [[HTTPS API Settings \(Old\)](#)]
- Custom permissions support the function of deleting CDR and recording files. [[Custom Privilege](#)]
- Added support to adjust recording file storage path. [[File Manager](#)]
- Added support to High Availability feature on UCM6300A series. [[HA](#)]
- Paging/Intercom supports delayed paging. [[Configure Paging/Intercom](#)]
- UCMRC remote service diagnosis. [[Remote Diagnosis](#)]
- Support LDAP to automatically update the phone book. [[LDAP Automatic Update Cycle](#)]
- Support meeting room automatic gain control. [[Meeting AGC](#)]

Firmware Version 1.0.7.12

- Added support for email reminder when editing the time of a scheduled meeting. [[Email Reminder \(m\)](#)]
- Improved extension status syncing process to the IM server.

Firmware Version 1.0.7.9

- This is the initial version.