

CloudUCM - User Manual

INTRODUCTION

CloudUCM is a cloud PBX solution that provides a scalable and secure business communication and collaboration platform with powerful features and integrations that enable teams to be more productive than ever before. This Cloud PBX unifies all business communication into one centralized solution that provides voice and video calling, meetings, chat, data, analytics, mobility, surveillance, facility access, intercoms and more. CloudUCM supports all SIP endpoints and the Wave app for desktop, mobile, and web, allowing teams to communicate and collaborate from anywhere on nearly any device. This scalable solution can be easily expanded at any time without the need for extra equipment, provides enterprise-level security and reliability, and supports powerful third-party integrations and expansions. By providing a state-of-the-art suite of communication and collaboration features, bank-grade security, advanced customization, and a variety of plan options, CloudUCM is the ideal PBX solution for small-to-medium sized businesses, retail, hospitality, and residential deployments.

TECHNICAL SPECIFICATIONS

The following table resumes all the technical specifications including the protocols/standards supported, voice codecs, telephony features, and languages for the CloudUCM.

Supported UC Endpoints and Client Devices	Support all SIP endpoint Wave app for desktop (Windows 10+, macOS 10+), web (Firefox, Chrome, Safari, Edge, Opera) and mobile (Android & iOS) Google Chrome extensions.
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
Built-in SBC	Free All plans default to supporting built-in SBC services to protect CloudUCM systems from external attacks
Call with WebRTC Trunk	Supports mobile and desktop web browsers: Chrome, Edge , Safari, Firefox, Opera Supports mobile application which built-in WebRTC WebView, such as Whatsapp, Facebook, Weixin and more
Collaboration	Audio and Video Meetings/Conferences, Instant Messaging and Group Chats with End-to-End Encryption, File Sharing, Screen Sharing, In-Meeting Chat, Voice Detection, Meeting Recording, Polls, Surveys, Message status, Advance Whiteboard with Multiplayer Annotation, Meeting Assistant, Onsite Meeting Room Scheduling, and more
Customer Service Support	<ul style="list-style-type: none">• Supports integration with third-party customer service platforms, including Whatsapp and Telegram. And built-in live chat• Includes a built-in live online web chat platform to provide customer service• Provides a web link that can be added to any web page or any browser that supports WebRTC• Compatible with computers, mobile browsers, and mobile apps
Customer Relationship Management (CRM)	Supports integration with ACT!, Bitrix24, Freshdesk, Hubspot, Salesforce, Sugar, Vtiger, Zendesk, Zoho, Dynamics 365, and more.

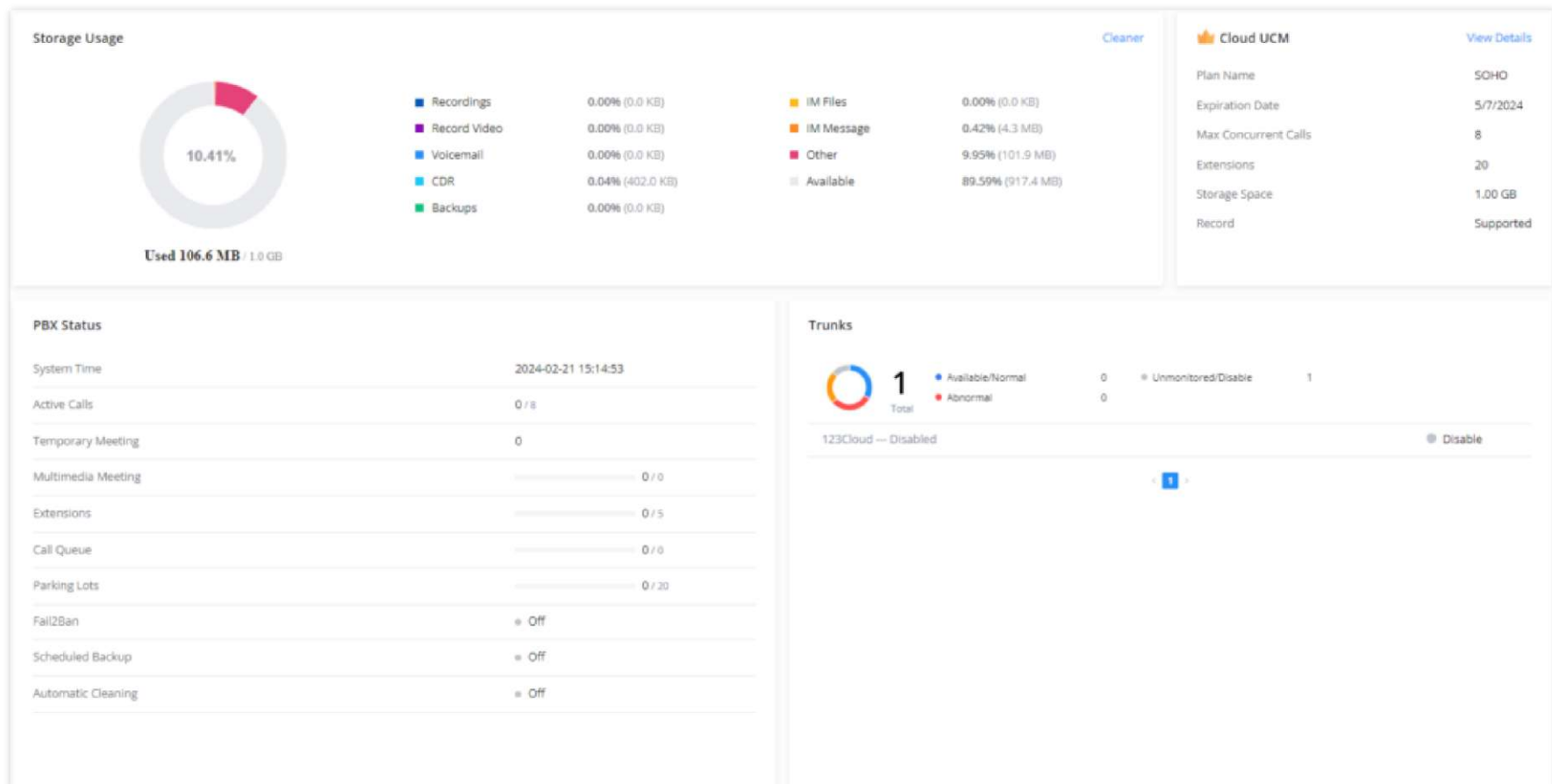
Call Center	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/ work-load, in-queue announcement.
Customizable Auto Attendant	Up to 5 layers of IVR (Interactive Voice Response) in multiple languages
Property Management System (PMS)	Local PMS Supports Integration with Hmobile PMS Systems.
Cloud Storage	Included, varies by plan, additional add-ons available.
CloudUCM App Store	Supports more than 20+ customized applications, with new apps being regularly added. <ul style="list-style-type: none"> • CRM add-ons • Google Drive and Office 365 • Whatsapp and Telegram • Hotdesking (coming soon)
Microsoft Integration	Supports integration with Microsoft Teams (via TeamMate), Outlook, AD Contact, and Office 365
Computer Telephony Integration	CTI Mode to Control GXP, GRP, GXV, GHP Series' IP Phones
Wired and Bluetooth Headset	Supports docking with different types of headphones Supports Microsoft Teams certified Headsets, supports phone call control
High Availability (HA)	Amazon Web Services (AWS) provides 99.99% service guarantee HA between UCM6300 Series IP PBX and CloudUCM (coming soon) HA between multiple CloudUCM systems (coming soon)
Firmware Upgrade and Provision	Supported by Grandstream Device Management System (GDMS), a zero-touch cloud provisioning and management system, GDMS provides a centralized interface to provision, manage, monitor and troubleshoot Grandstream products.
IP Cameras, Intercom and Door Access Integration	Supports Grandstream GSC Series IP Cameras and Intercom/Public Address devices, supports GDS Series Door Access Solutions Supports third party devices, including Hikvision, Dahua, and more
API and SDK	Full CGI API available for third-party platform and application integration Wave add-in SDK Wave Android and iOS SDK Wave H5 Embedded for MAC/Windows application
Multi-Language Support	<ul style="list-style-type: none"> • Web User Interface: 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, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic • Customizable language pack to support any other languages
Security	Frequency Restriction, Fail2ban, Ping Defense, Ping of Death, SYN-Flood, Remote Login Interception, Multi-factor Authentication, SMS Login Authentication
Network Protocols	SIP, TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, NTP, HTTP/HTTPS, STUN, SRTP, TLS, LDAP, IPv4/6

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 4582, RFC 4583, RFC 5245, RFC 5389, RFC 5766, RFC 6347, RFC 6455, RFC 8860, RFC 4734, RFC 3665, RFC 3323, RFC 3550
DTMF Methods	In-band audio, RFC 2833, and SIP INFO
Transmission Encryption	SRTP, DTLS-SRTP, TLS, HTTPS
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
Video Codecs	H.264, H.263, H263+, VP8
QoS	Layer 3 (ToS, DiffServ, MPLS) QoS
Administration	Call Detail Record, event alert and SMS notifications, event logs, export import extensions, feature codes, LDAP, feedback system, PBX monitor, resource monitor, system prompt, user permission, web-based control panel, user portal, trunk cluster, voice prompt customization, firewall, Fail2ban, IP blacklist, Syslog, gateway and endpoint provisioning, Wave permissions (<i>deploy & configure Wave Desktop, installations en masse, pre-install Wave Add-ons for extensions, manage Wave feature, access permissions</i>), local backup

SYSTEM STATUS

Dashboard

- **Storage Usage**
- **CloudUCM**
- **PBX Status**
- **Trunks**



System Information

System Information	
General	Remark
System Information	
Model	CloudUCM
MAC Address	00:0B:82:07:00:37
Serial Number	3420070005
System Time	2024-02-21 15:17:17 UTC-01:00
Up Time	14day (s) 08:32:07
Version Information	
Wave Web	1.0.25.5
Program	1.0.24.23

System Information

System Information	
General	Remark
Remark	<input type="text" value="CloudUCM Boston"/>
<input type="button" value="Cancel"/> <input type="button" value="Save"/>	

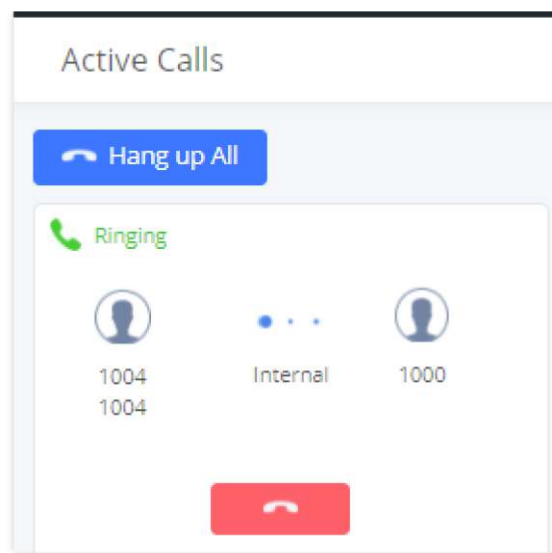
Remark

Active Calls

The active calls on the CloudUCM are displayed in the Web GUI→**System Status**→**Active Calls** page. Users can monitor the status, hang up a call, and barge in the active calls in a real-time manner.

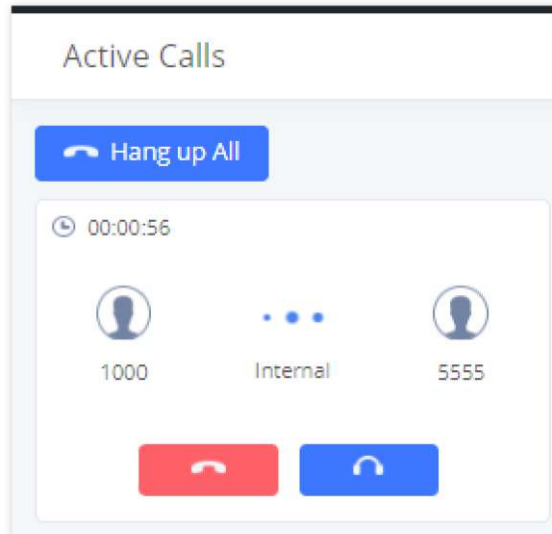
Active Calls Status

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.



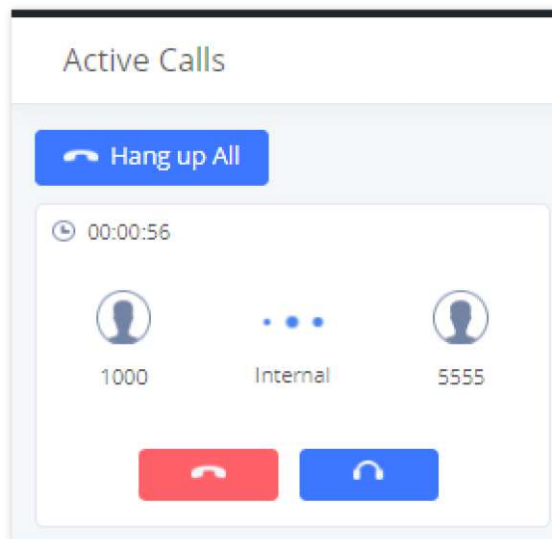
Active Calls – Ringing Status

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

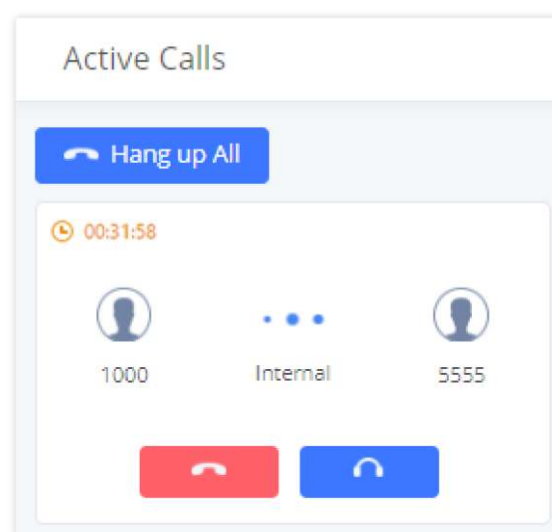


Active Call – Established Status

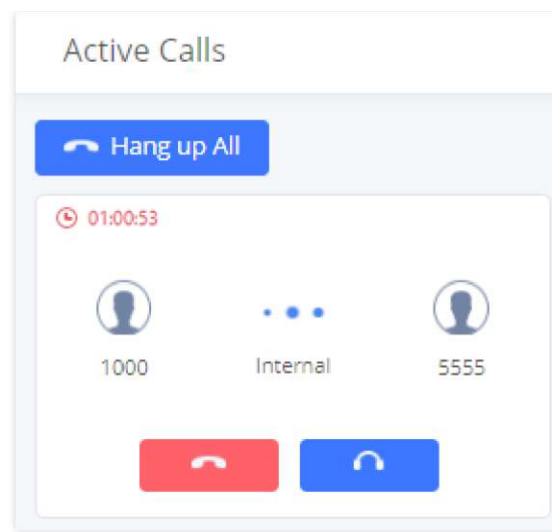
The gray color of the active call means the connection of call time is less than half an hour. It means this call is normal.



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.



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.



EXTENSION/TRUNK

Extensions

Create a New SIP Extension

To manually create a 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.

Extensions > Edit Extension: 1000

Basic Settings | Media | Features | Voicemail | Specific Time | Wave Client | Follow Me | Advanced Settings

General

* Extension: 1000 | CallerID Number: |

* Call Privileges: Local | * SIP Password: |

AuthID: | * Concurrent Registrations: 3 |

Disable This Extension:

User Settings

First Name: | Last Name: |

Email Address: | * User/Wave Password: |

* User Portal/Wave Privileges: Default | Mobile Number: +1 |

Department: | Job Title: |

Contact Privileges

Cancel | Save

Create New Extension

Extension options are divided into five categories:

- Basic Settings
- Media
- Features
- Voicemail
- Specific Time
- Wave Client
- Follow me

- o Advanced Settings

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 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.
Concurrent Registrations	The maximum endpoints which can be registered into this extension. For security concerns, the default value is 3. Note: When this option is set to "1(seize)"
Auth ID	Configure the authentication ID for the user. If not configured, the extension number will be used for authentication.
Disable This Extension	If selected, this extension will be disabled on the CloudUCM. 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/Wave 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 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.
Job Title	Enter the job title of the user of the extension.
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.
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SIP Extension Configuration Parameters → Basic Settings

SIP Settings	
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.
FECC	Configure to enable Remote Camera Management.
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.
QoS	
Jitter Buffer	Select the jitter buffer method. <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	Configure to enable Packet Loss Retransmission. <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.
SRTP	
SRTP	Enable SRTP for the extension. <ul style="list-style-type: none"> ● Disabled ● Enabled and Enforced ● Optional <p>The default setting is disabled.</p>

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
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SIP Extension Configuration Parameters → Media

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. • "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.

	<ul style="list-style-type: none"> 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	<p>If enabled, CloudUCM 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 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>

	The minimum value is 1.
CC Max Monitors	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. The minimum value is 1.
Ring Simultaneously	
Ring Simultaneously	Enable this option to have an external number ring simultaneously along with the extension. If a register trunk is used for outbound, the register number will be used to be displayed for the external number as the caller ID number.
External Number	Set the external number to ring simultaneously. '-' is the connection character that will be ignored. This field accepts only letters, numbers, and special characters + = * #.
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 Seamless Transfer	Any extensions on the CloudUCM 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 can set wakeup service for other extensions	Selected extensions can set a PMS wakeup service for this extension 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 CloudUCM. The valid range is between 5 seconds and 600 seconds. Note: If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.
Auto Record	Enable automatic recording for the calls using this extension. The default setting is disabled. The 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 the password when making outbound calls.
Dial Trunk Password	Configure personal password when making outbound calls via the trunk.
Support Hot-Desking Mode	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.
Enable LDAP	If enabled, the extension will be added to the LDAP Phonebook PBX list. Default is enabled.
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	Specify which Music On Hold class to suggest to the bridged channel when putting them on hold.
Call Settings	
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
The Maximum Number of Call Lines	The maximum number of simultaneous calls that the extension can have. 0 indicates no limit.
Outgoing Call Frequency Limit	If enabled, if the number of outbound calls exceed the configured threshold within the specified period, further outbound calls will be not be allowed.
Send PCPID Header	If enabled, this extension's SIP INVITE messages will contain the P-Called-Party-ID (PCPID) header if the callee is a SIP device.
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 valide tange 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.

SIP Extension Configuration Parameters →Features

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

SIP Extension Configuration Parameters →Specific Time




Normal	
Enable Wave	Enable Wave for the specific extension.
Wave Welcome Email	Wave Welcome Email template.
Wave	
Download Link	https://fw.gdms.cloud/wave/download/

SIP Extension Configuration Parameters →Wave

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

Search and Edit Extension

All the CloudUCM extensions are listed under Web GUI → **Extension/Trunk → Extensions**, with status, Extension, CallerID Name, IP, and Port. Each extension has a checkbox for users to "Edit" or "Delete". Also, options "Edit" , "Reboot"  and "Delete"  are available per extension. Users can search for an extension by specifying the extension number to find an extension quickly.

Extensions									
+ Add Edit Delete Edit All SIP E-mail Notification More <input type="text" value="Extension Number or Name"/> Search									
<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

o 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)

o Edit single extension

Click on to start editing the extension parameters.

o 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 for username and permissions and delete the user voicemail prompt and voice messages.

Note

This is the expected behavior when you reset an extension:

- o 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.
- o 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.
- o 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.

o Reboot the user

Click on  to send NOTIFY reboot event to the device that has a CloudUCM extension already registered. To successfully reboot the user.

- o **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:

- o The system will delete all the data of the extension except the CDR and meetings records. All the data on the user side will be erased.
- o 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.
- o 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.

- o **Modify selected extensions**

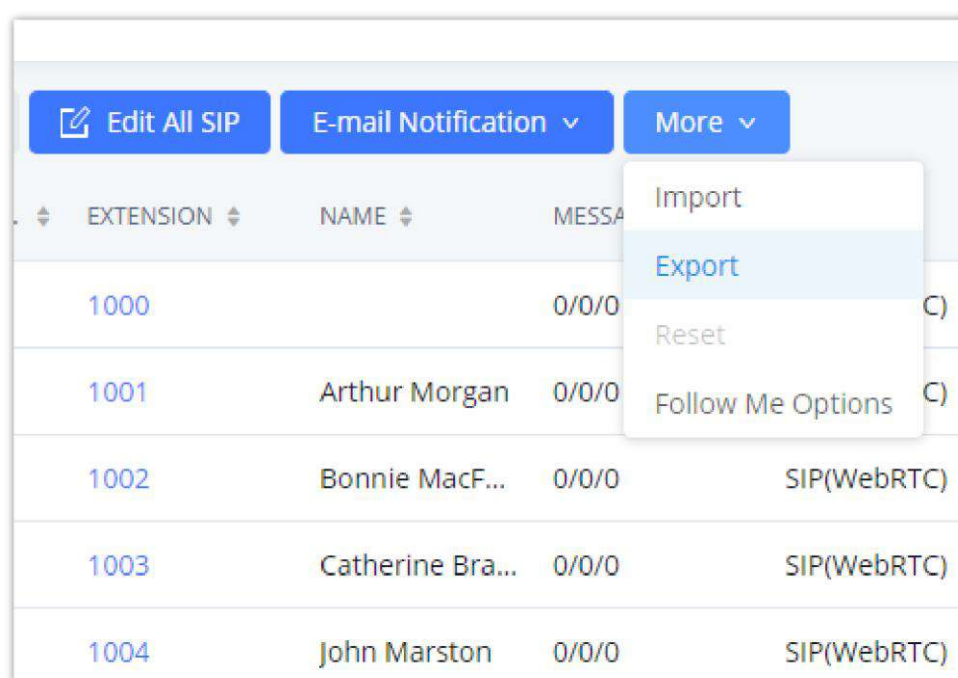
Select the checkbox for the extension(s). Then click on "Edit" to edit the extensions in a batch.

- o **Delete selected extensions**

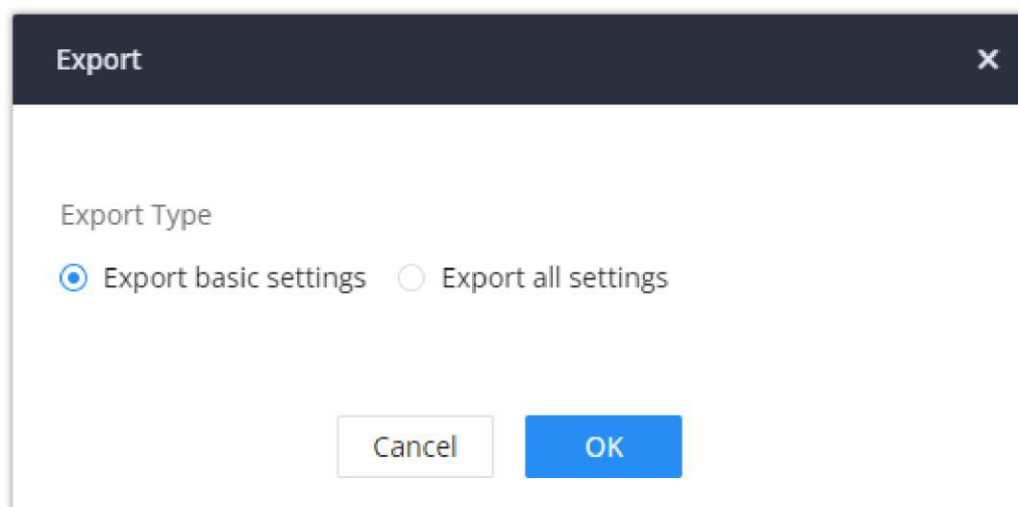
Select the checkbox for the extension(s). Then click on "Delete " to delete the extension(s).

Export Extensions

The extensions configured on the CloudUCM can be exported to a CSV format file. Click on the "Export Extensions" button and select technology in the prompt below.



Export Extensions



Export Basic Settings

Export Basic Information includes:

- Extension
- CallerID Number
- Privilege
- SIP Password
- AuthID
- Voicemail
- Voicemail Password
- Sync Contact
- First Name
- Last Name
- Email Address
- User/Wave Password

If importing extensions with no values for settings, the following will occur:

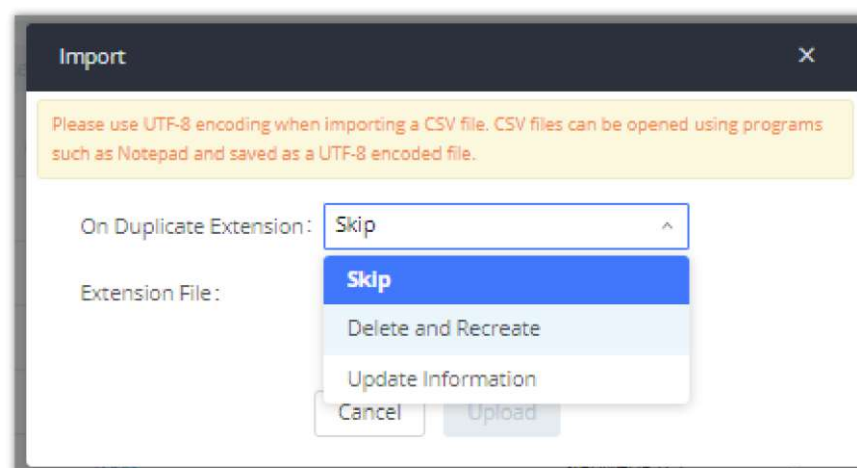
- If importing new extensions, or if **Replace** is selected as the duplicate import option, the default values for those settings will be used.
- If **Update** is selected as the duplicate import option, no changes will be made to the existing settings.

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

Import Extensions

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

1. Export the extension CSV file from the CloudUCM by clicking on the "Export Extensions" button.
2. Fill up the extension information you would like in the exported CSV template.
3. Click on the "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.
 - **Skip:** Duplicate extensions in the CSV file will be skipped. The PBX will keep the current extension information as previously configured without change.
 - **Delete and Recreate:** The current extension previously configured will be deleted and the duplicate extension in the CSV file will be loaded to the PBX.
 - **Update Information:** The current extension previously configured in the PBX will be kept. However, if the duplicate extension in the CSV file has a 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 a CSV file from a local directory on the PC.

6. Click on "Apply Changes" to apply the imported file on the CloudUCM.

Example of a file to import:

1	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	Q	R	S	T	U	V	W
2	1000	Bonnie	MacFarla	SIP(WebR	yes	Pas0123!	660408	no			off	no	None	Allow All									
3	1001			SIP(WebR	yes	Pas0123!	623712	no			off	no	None	Allow All									
4	1002			SIP(WebR	yes	rY3z\$jp	47962	no			off	no	None	Allow All									
5	1003			SIP(WebR	yes	gU2@m*\$	83232	no			off	no	None	Allow All									
6	1004			SIP(WebR	yes	tW8%008C	5187	no			off	no	None	Allow All									
7																							
8																							
9																							
10																							
11																							
12																							
13																							

Import File

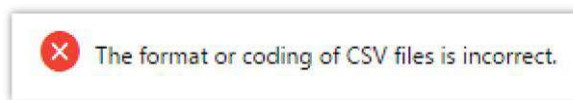
Field	Supported Values
Extension	Digits
Technology	SIP/SIP(WebRTC)
Enable Voicemail	yes/no/remote
CallerID Number	Digits
SIP Password	Alphanumeric characters
Voicemail Password	Digits
Skip Voicemail Password Verification	yes/no
Ring Timeout	Empty/ 3 to 600 (in second)
SRTP	yes/no
Skip Trunk Auth	yes/no/bytime
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
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

Field	Supported Values
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

Field	Supported Values
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

The CSV file should contain all the above fields, if one of them is missing or empty, the CloudUCM 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:** This shows the Extension number.
- **Status:** This shows the status of the extension.
- **Presence status:** Indicates the Presence Status of this extension.
- **Terminal Type:** This shows the type of the terminal using this extension
- **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.

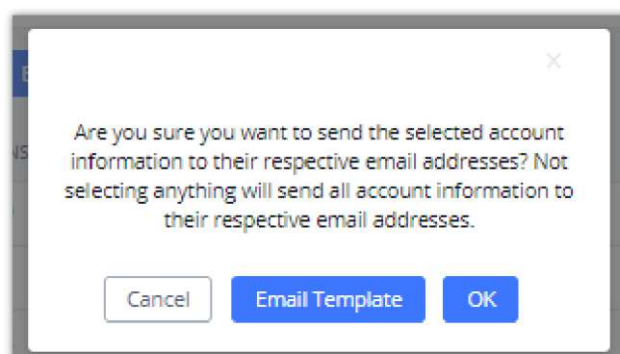
OPTIONS	VALUE
Extension	1000
Status	● Unavailable
Presence Status	Available
Terminal Type	SIP(WebRTC)
CallerID Name	
Message	0/0/0
IP and Port	--
Email Status	To Be Sent
Ring Group	
Call Queue	
Call Queue(Dynamic)	

Extension Details

E-mail Notification

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

When clicking on “More” > “E-mail Notification” button, the following message will be prompted on 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 an Email including account registration information as well as the Wave Settings with the QR code:

Wave Settings and QR Code

Important Note

For security and confidentiality reasons, it is highly advisable for the user to change the Wave login extension after the first time log in.

The CloudUCM admin can also send "Extension Information" mail and "Wave Welcome" mail as the figure below shows

Send Email Notification

Multiple Registrations per Extension

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

This feature can be enabled by configuring the option “Concurrent Registrations” under Web GUI→**Extension/Trunk**→**Edit Extension**. The default value is set to 3 registrations. The maximum is 10. When the option “1(allowed to seize) is selected, the UCM will allow newer registration attempts to seize the extension from a previously registered endpoint. To prevent this behavior, please select the option 1.

Extension – Concurrent Registration

SMS Message Support

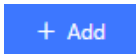


The CloudUCM provides built-in SIP SMS message support. For SIP end devices such as Grandstream GXP or GXV phones that support SIP messages, after a CloudUCM account is registered on the end device, the user can send and receive SMS messages. Please refer to the end device documentation on how to send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

Extension Groups

The CloudUCM extension group feature allows users to assign and categorize extensions in different groups to better manage the configurations on the CloudUCM. For example, when configuring the “Enable Source Caller ID Whitelist”, 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 a business environment.





Configure Extension Groups

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

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

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

Edit Extension Group

Click on     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 Source Caller ID Whitelist". Both single extensions and extension groups will show up for users to select.

Outbound Routes > Create New Outbound Rule

General

<p>* Outbound Rule Name <input style="width: 90%;" type="text" value="Rule_1"/></p> <p>* Pattern <input style="width: 90%;" type="text" value="_9x."/></p> <p>PIN Groups <input style="width: 90%;" type="text" value="None"/></p> <p>Password <input style="width: 90%;" type="text"/></p> <p>Local Country Code <input style="width: 90%;" type="text"/></p>	<p>Disable This Route <input type="checkbox"/></p> <p>Privilege Level <input style="width: 90%;" type="text" value="Disable"/></p> <p>PIN Groups with Privilege Level <input type="checkbox"/></p> <p>Auto Record <input type="checkbox"/></p>
--	--

Enable Source Caller ID Whitelist

<p>Enable Source Caller ID Whitelist <input checked="" type="checkbox"/></p> <p>Source Caller ID Pattern <input style="width: 90%;" type="text"/></p>	<p>Outbound Route CID <input style="width: 90%;" type="text"/></p> <p>Whitelisted Extensions/Extension Groups <input style="width: 90%;" type="text"/></p>
---	--

Call Duration Limit

Select Extension Group in Outbound Route

VoIP Trunks

VoIP Trunk Configuration

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

- Click on "Add SIP 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.

The VoIP trunk options are listed in the table below.

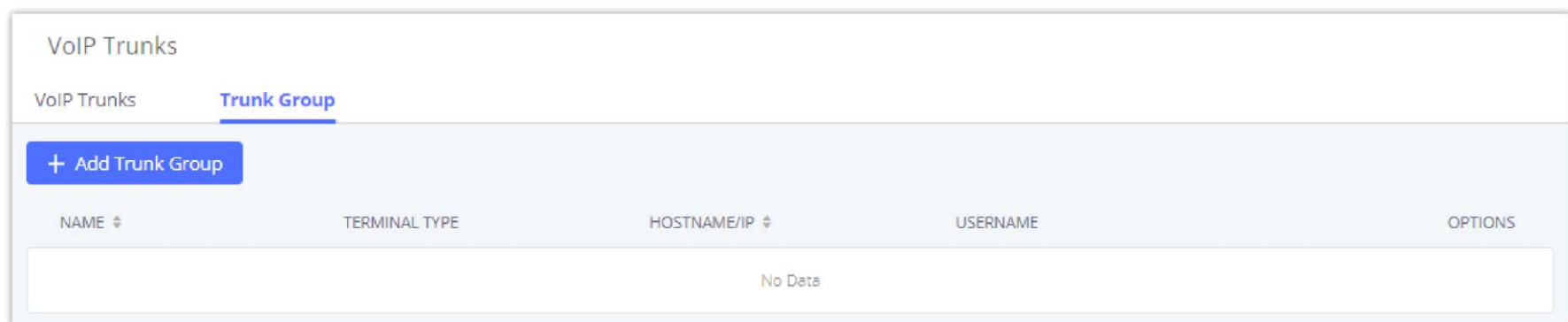
Disable This Trunk	Check this box to disable this trunk.
Type	<p>Select the VoIP trunk type.</p> <ul style="list-style-type: none"> ● Peer SIP Trunk ● Register SIP Trunk

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	Select the transport protocol to use. <ul style="list-style-type: none"> • UDP • TCP • TLS
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".
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 then 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.
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.
Domain Connection Mode	If enabled, the following options will be automatically configured: TLS transport, From Domain , Enable Heartbeat Detection and ICE Support . Please ensure that the trunk host name is a GDMS-assigned address and 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	<p>This setting defines how long until the time allowed for outgoing calls is reset.</p> <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

Trunk Group

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 the **+** 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

Disable This Trunk	Check this box to disable this trunk
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. Only TLS is supported, and TLS service must be enabled on the other end.
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.
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	<ul style="list-style-type: none"> ● Linear: Use lines in the list order for outbound calls. ● Round Robin: Use lines based on rotary line selection for outbound calls. Previously used lines will be remembered.
AuthTrunk	If enabled, the UCM 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.
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.


Direct Outward Dialing (DOD)

The CloudUCM 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.

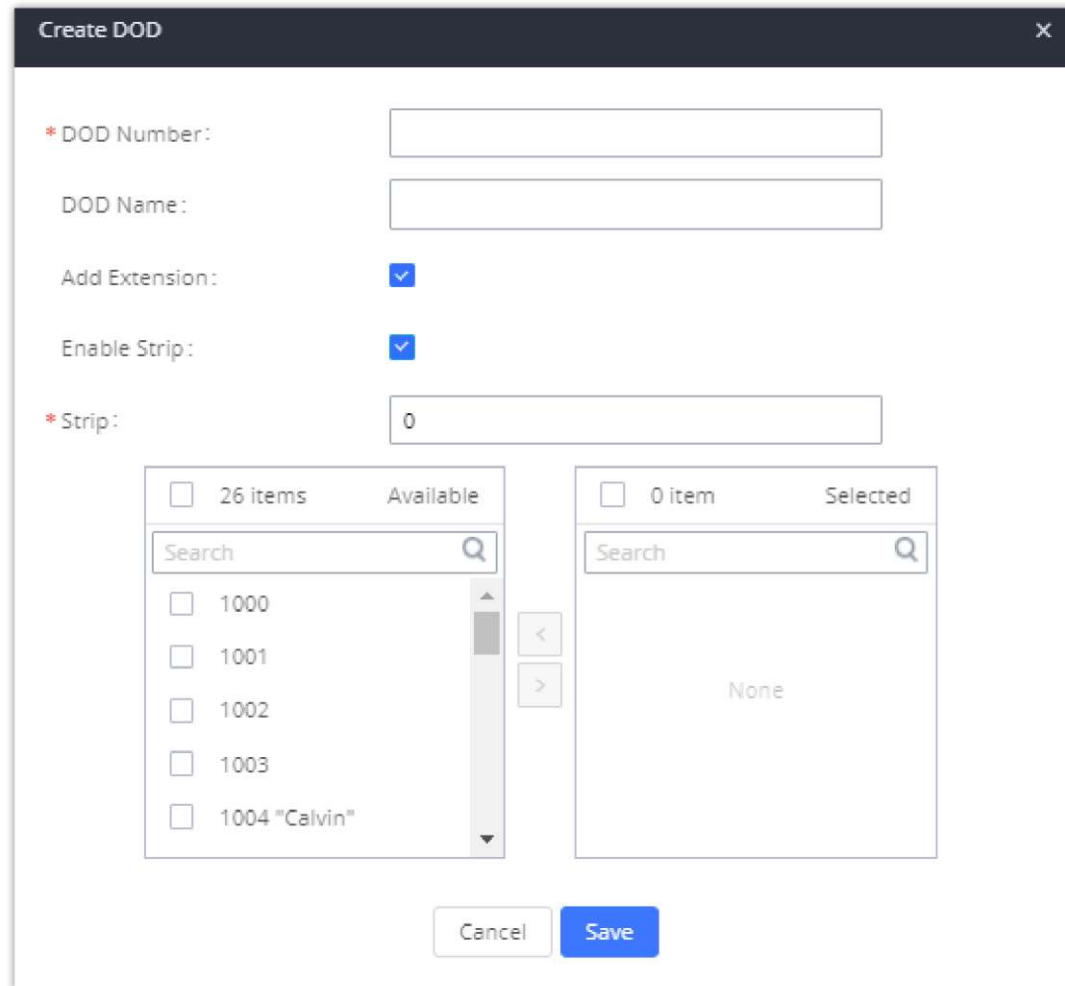
Example of how DOD is used:

Company ABC has a SIP trunk. This SIP trunk has 4 DID numbers associated with 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 CloudUCM:

1. To setup DOD go to CloudUCM 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. Enter a SIP trunk DID number in the "DOD number" field. In this example, ABC company has a total of 4 DID numbers. Enter the phone number used by the CEO here.
5. When adding extensions, you can choose whether to "Enable Strip" according to your needs. If it is enabled, you can configure the number (0-64) that will be stripped from the extension number before being added to the DOD number. For example, if the entered digit is 2, and the DOD number for extension 4002 is 1122, then dialing out from 4002, 112202 will be used as the caller ID (DOD).
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.

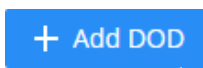
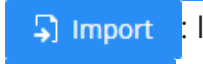




DOD extension selection

7. Click "Save" at the bottom.

Once completed, the user will return to the **EDIT DOD** page which shows all the extensions that are associated with a particular DOD.

Edit DOD

-  Add DOD : Add a DOD.
-  Import : Import DODs using a csv file.
-  Export : Export the DODs using a csv file.
-  Filter : Filter DODs by number or name.

For DOD importing, please refer to the screenshot below for the template used.

	A	B	C	D	E	F	G
1	DOD Number	DOD Name	Add Extension	Local Members	Ldap Members	Enable Strip	Strip Number
2	2.12555E+11	DOD1	no	1,002,100,310,051,000		no	0
3							
4							
5							
6							
7							

DOD CSV file Template

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 CloudUCM supports creating WebRTC trunks to use exclusively with web applications, 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:

Disable This Trunk:

Auto Record:

Enable Concurrent Call Threshold:

* Incoming Concurrent Call Threshold:

WebRTC Inbound Link Address: Automatically generated after saving

Create WebRTC Trunk

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.

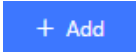


Outbound Routes

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

Configuring Outbound Routes

In the CloudUCM, 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. Users can also set up a fail-over 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 the "Import" button to upload the outgoing route in .CSV format.
- Click the "Export" button to generate outgoing routes in .CSV format.
-  Click to edit the outbound route.
-  Click to delete the outbound route.

On the CloudUCM, the outbound route priority is based on the "Best matching pattern". For example, the CloudUCM has outbound route A with pattern 1xxx and outbound route B with pattern 10xx configured. When dialing 1000 for an 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 Rule Name	Configure the name of the calling rule (e.g., local, long_distance, etc.). Letters, digits, _ and - are allowed.
Pattern	<p>All patterns are prefixed by the "_" 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 the 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 levels can use this rule. • National: Users with National or International levels can use this rule. • International: The highest level required. Only users with the 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 the "Internal" level, which means all users can use this outbound rule to dial out from the trunk.</p>
Enable Filter on Source Caller ID	<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. 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>Note: Multiple patterns can be used. Patterns should be separated by a comma ",". Example: _X, _NNXXNXXXXX, _818X.</p>
Outbound Route CID	Attempt to use the configured outbound route CID. This CID will not be used if DOD is configured.
Send This Call Through Trunk	
Trunk	Select the trunk for this outbound rule.
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: The users will dial 9 as the first digit of long-distance calls. 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.
Use Failover Trunk	

Failover Trunk	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. CloudUCM supports up to 10 failover trunks.
Strip	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. Example: The users will dial 9 as the first digit of long-distance calls. In this case, 1 digit should be stripped before the call is placed.
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	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. 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.

Failover Trunk Toggles

Outbound Routes

An outgoing calling rule associates an extension pattern with a trunk used to dial the pattern. This allows different patterns to be dialed through different trunks. For example, "local" allows 7-digit dialed through FXO port while "long distance" allows 10-digit dialed through a low-cost SIP trunk. A failover trunk can be set up to be used when the primary trunk fails. Note: This panel only manages individual outgoing calling rules.

[+ Add](#)
[Scheduled Sync](#)
[Outbound Blacklist](#)
[PIN Groups](#)
[Failover Trunk Toggles](#)
[Import](#)
[Export](#)

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

Inbound Routes

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

Failover Trunk Toggles [Cancel] [Save]

No-Failover Response Codes :

Available	Selected
<input type="checkbox"/> 403 <input type="checkbox"/> 404 <input type="checkbox"/> 408 <input type="checkbox"/> 480 <input type="checkbox"/> 503 <input type="checkbox"/> 603	<input type="checkbox"/> 486

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Failover Trunk Toggles

If a call receives the selected response codes, the UCM will 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

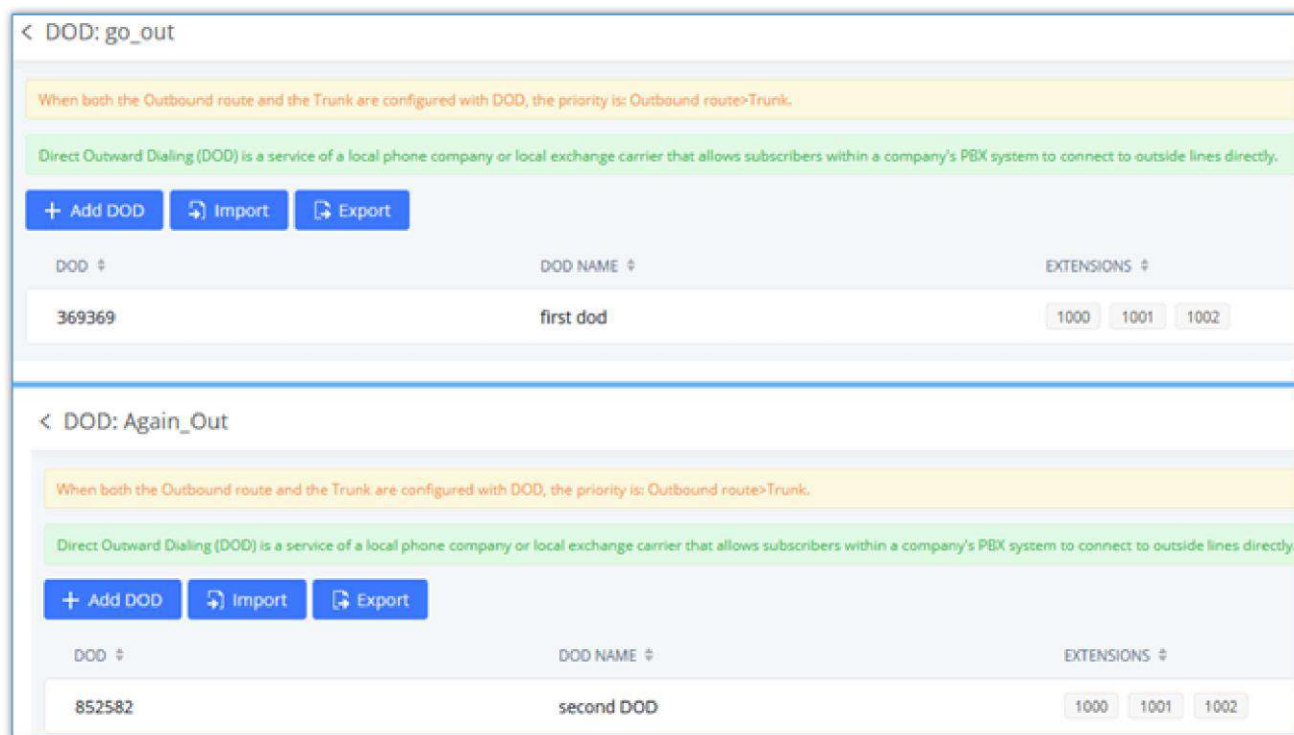
It is possible to specify the DOD number based on the Outbound Route, as displayed in the screenshot below. For each outbound route.



The screenshot shows the 'Outbound Routes' page with a table containing two entries:

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

Outbound Routes Page



The screenshot shows the configuration for DOD (Direct Outward Dialing) for two routes:

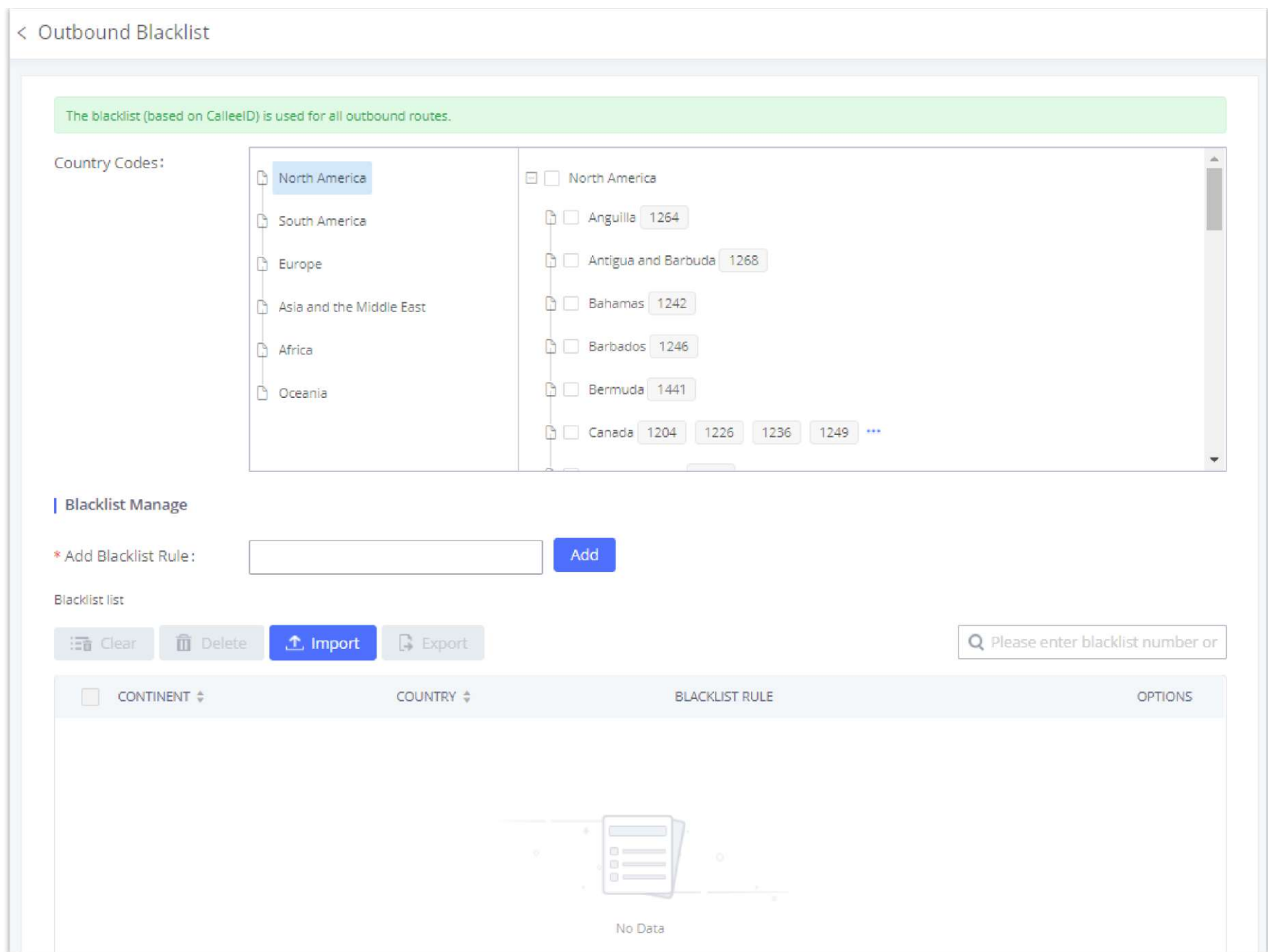
- DOD: go_out**: DOD # 369369, DOD NAME first dod, EXTENSIONS 1000, 1001, 1002.
- DOD: Again_Out**: DOD # 852582, DOD NAME second DOD, EXTENSIONS 1000, 1001, 1002.

DOD Configuration by Outbound Route

Outbound Blacklist

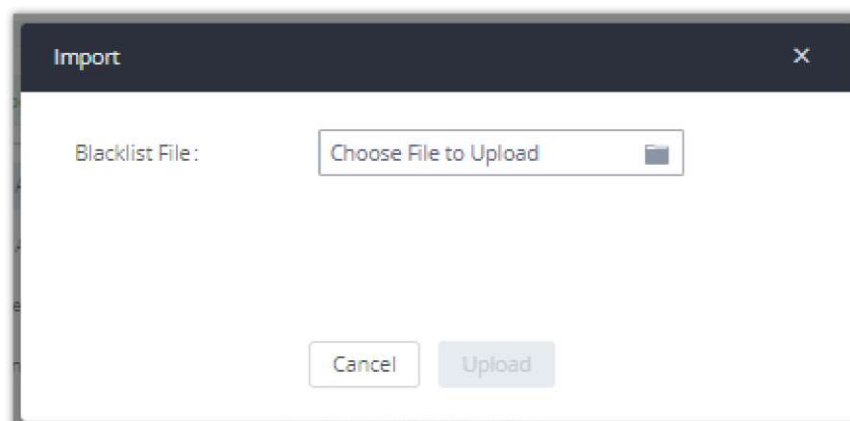
The CloudUCM allows users to configure a blacklist for outbound routes. If the dialing number matches the blacklist numbers or patterns, the outbound call will not be allowed. The outbound blacklist can be configured under UCM Web GUI > **Extension/Trunk > Outbound Routes: Outbound Blacklist**.

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



Country Codes

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



Blacklist Import/Export

PIN Groups

The CloudUCM supports the pin group. Once this feature is configured, users can apply pin groups to specific outbound routes. When placing a call on pin-protected outbound routes, the caller will be asked to input the group PIN, this feature can be found on the Web GUI > **Extension/Trunk** > **Outbound Routes** > **PIN Groups**.

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

Outbound Routes/PIN Group

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

Create New PIN Group

* Name:

Record in CDR:

Members

* PIN Number:

* PIN Name:

Create a New PIN Group

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

< PIN Groups

NAME	RECORD IN CDR	OPTIONS						
GSEMEA	yes	<input type="button" value="Edit"/> <input type="button" value="Download"/> <input type="button" value="Delete"/>						
<table border="1"> <thead> <tr> <th>PIN NUMBER</th> <th>PIN NAME</th> </tr> </thead> <tbody> <tr> <td>125478963</td> <td>Dao</td> </tr> <tr> <td>1596324</td> <td>Emily</td> </tr> </tbody> </table>			PIN NUMBER	PIN NAME	125478963	Dao	1596324	Emily
PIN NUMBER	PIN NAME							
125478963	Dao							
1596324	Emily							

Total: 1

PIN Members

If the PIN group is enabled on the outbound route level, the password, privilege level and enable the filter on source caller ID will be disabled, unless 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:

* Pattern:

PIN Groups:

Password:

Disable This Route:

Privilege Level:

PIN Groups with Privilege Level:

Outbound PIN

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

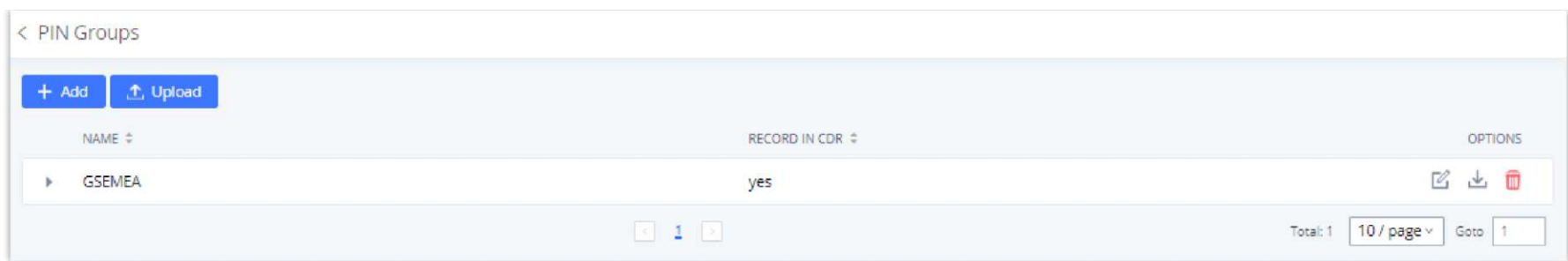
STATUS	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	TALK TIME	ACCOUNT CODE	RECORDING FILE OPTIONS
<input checked="" type="checkbox"/>	5555	99856352 [Tr...	DIAL	2019-12-04 10:57:47	0:00:08	0:00:08	Emily/GSEMEA	-
<input checked="" type="checkbox"/>	1000	*36	DIAL	2019-12-03 10:12:37	0:00:19	0:00:19		-

CDR Record

o **Importing PIN Groups from CSV files:**

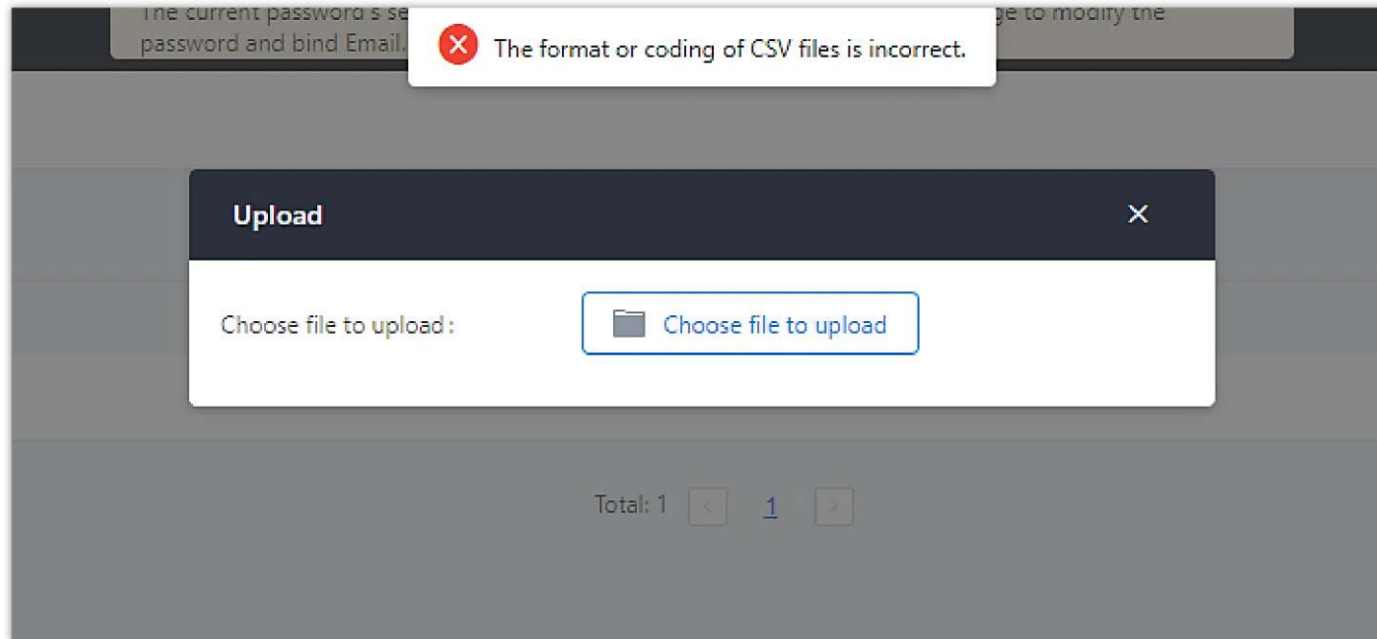
Users can also import PIN Groups by uploading CSV files for each group. To do this:

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



Importing PIN Groups from CSV files

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



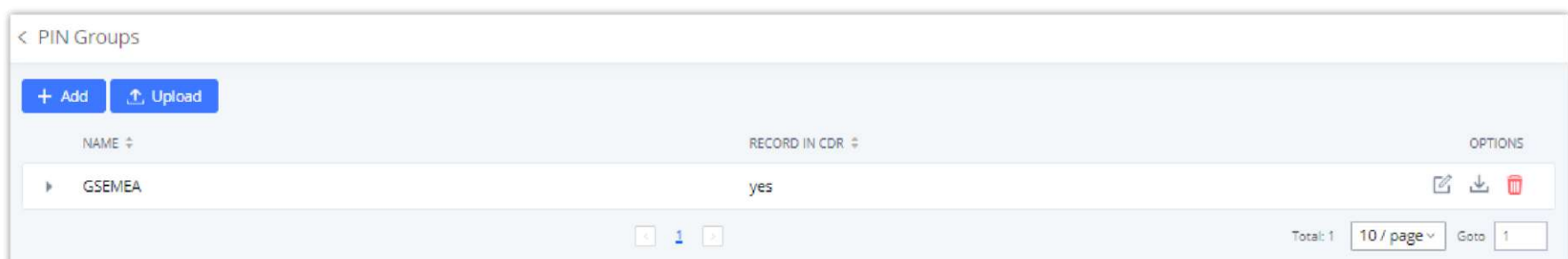
Incorrect CSV File

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

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

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

- o Click on  to delete the inbound route.

Inbound Route Configuration

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.</p> <p>_ 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 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	<p>If enabled, the trunk name will be added to the caller id name as the displayed caller id name.</p>
Set Caller ID Info	<p>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.</p>
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	<p>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.</p>
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	<p>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.</p>
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. • 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.

	<ul style="list-style-type: none"> ● 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

CloudUCM allows users to prepend digits to an inbound DID pattern, with strip taking precedence over prepend. With the ability to prepend digits in the inbound route DID pattern, the 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, UCM will strip the first 8 digits.
3. If **Prepend** is set to 2, UCM will then prepend a 2 to the stripped number, now the number becomes 2163.
4. The UCM will 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.

Inbound Route feature: Prepend

Inbound Route: Multiple Mode

In the UCM, the user can configure an 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

Disable This Route:

Alert-info: None

Fax Detection:

Block Collect Calls:

Set CallerID Info:

Inbound Multiple Mode:

CallerID Pattern: [Empty]

Allowed to seamless transfer: [Empty]

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 UCM can dial feature code *62 to switch to the inbound route “Mode 1” and dial feature code *61 to switch back to “Default Destination”. Switching between different modes can be easily done without a Web GUI login.

For example, the customer service hotline destination has to be set to a different IVR after 7 PM. 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 the “Inbound Routes” page, check the “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 UCM, users can enable Route-Level Inbound Mode to switch between different destinations for each inbound route. The inbound Route-Level mode can be enabled under Inbound Route settings.

Inbound Route – Route-Level Mode

The global inbound mode must be enabled before configuring Route-Level Inbound Mode. Additionally, 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.

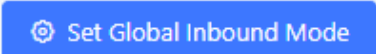
For example, the Inbound Default Mode feature code is set to *61 and the Inbound Mode suffix for route 1 is set to 1010. To switch the mode of route 1 to Default Mode, users can dial *611010.

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

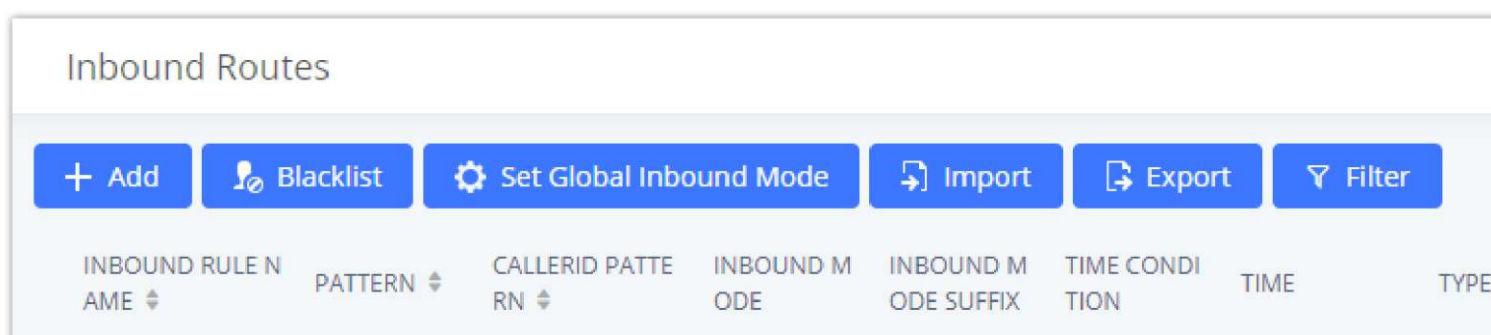
- Pressing the key will toggle the inbound mode to "Mode 1", and the button's color will change to red.



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.



Import/Export Inbound Route



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

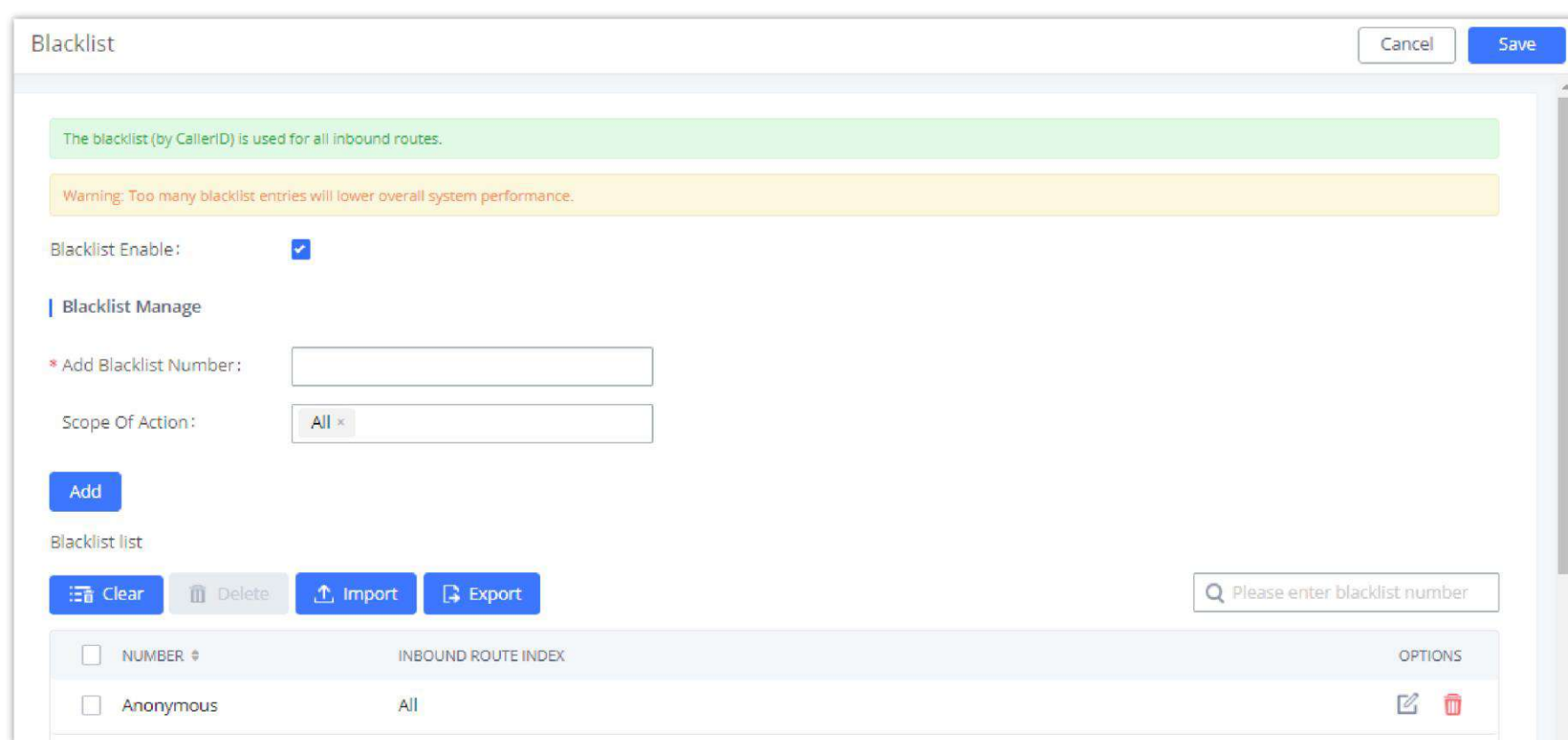
- Disable This Route: Yes/No.
- Pattern: Always prefixed with _
- CallerID Pattern: Always prefixed with _
- Prepend Trunk Name: Yes/No.
- Prepend User Defined Name Enable: Yes/No.
- Prepend User Defined Name: A string.
- Alert-info: None, Ring 1, Ring 2... The user should enter an Alert-info string 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... Users 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... Users 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.

Blacklist Configurations

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

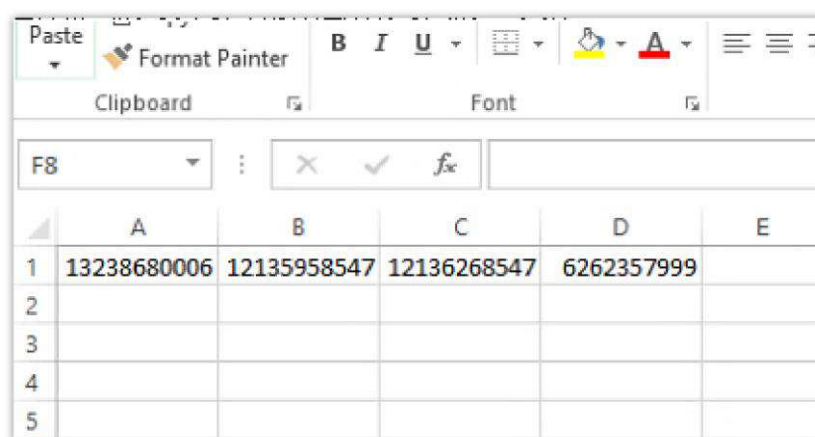
- Select the checkbox for "Blacklist Enable" to turn on the Blacklist feature for all inbound routes. The blacklist is disabled by default.
- Enter a number in the "Add Blacklist Number" field and then click "Add" to add to the list. Anonymous can also be added as a Blacklist Number by typing "Anonymous" in Add Blacklist Number field.
- To remove a number from the Blacklist, select the number in the "Blacklist list" and click on  or click on the "Clear" button to remove all the numbers on the blacklist.
- Users can also export the inbound route blacklist by pressing the  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.' and a yellow warning says '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. The 'Blacklist list' section has buttons for 'Clear', 'Delete', 'Import', and 'Export'. A search bar contains 'Please enter blacklist number'. A table below shows one entry: 'Anonymous' under the 'All' index, with edit and delete icons.

Blacklist Configuration Parameters

- To add blacklisted numbers in batch, click on "Import" to upload the 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

i 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**.

CALL FEATURES

Multimedia Meeting

The UCM supports multimedia meeting room allowing multiple rooms used at the same time.

The multimedia meeting room configurations can be accessed under Web GUI→**Call Features**→ Multimedia Meeting. On this page, users can create, edit, view, invite, manage the participants, and delete multimedia meeting rooms. The multimedia meeting room status and meeting call recordings (if recording is enabled) will be displayed on this web page as well.

For video meeting, which is based on WebRTC, participants can join the meeting from a PC without installing extra plug-ins or software.

The UCM admin can create multiple multimedia meeting rooms for users to dial in.

Meeting room specifications affect user participation to a certain extent. UCM supports the forecasting of meeting resources. There will be corresponding judgments and adjustments in the following scenarios:



1. When meeting resources are used up, scheduled meeting members cannot join the meeting in advance.
2. When a point-to-point call is transferred to a conference, the conference resources are used up.
3. When meeting resources are used up, do not join a group IM chat when you initiate a meeting.
4. When meeting resources are used up, do not join an instant meeting.
5. Close other instant meetings or scheduled meetings that have timed out to ensure that invited members can join the scheduled meeting.
6. In an ongoing meeting, if the number of invited members exceeds the upper limit, members cannot be invited to join the meeting.
7. Enable flow control for videos and presentations in the conference room.

Notes

The multimedia meeting room supports up to 4 video calls and one video presentation.

- The administrator can set the number of videos to 9 parties. The increase in the number of videos will take up more system resources and affect the overall performance of the UCM system. Please set it according to your needs.
- During a meeting, when the system detects that another scheduled meeting is about to be held, it will remind the meeting members that the subsequent meeting room has been reserved, please end the meeting in advance.
- The use of video in the meeting will take up system resources and may cause performance problems when used.
- The maximum meeting duration is 12 hours. If it exceeds 12 hours, the system will remind the current meeting and the host can continue to extend the meeting.

Multimedia Room Configurations

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

Meeting Settings contains the following options:

Extension	The number to dial to reach the meeting room.
Meeting Name	Meeting Name

Privilege	Please select the permission for outgoing calls.
Allow User Invite	If enabled, participants will be able to invite other to the meeting by pressing 1 on their keypad or by clicking the Participants -> Invite option on the Wave bottom bar.
Allowed to Override Most Mute	Allowed to Override Host Mute
Auto Record	Meeting audio and video can be automatically recorded. These recordings can be found under the Meeting Recording or Meeting Video Recordings Page. <ul style="list-style-type: none"> • None: Auto record is disabled. • Record Audio: Record only the meeting Audio. • Record video (Focus Mode): Record the focus screen and all audio of the meeting. When a shared source is present in the meeting, only the shared screen is recorded.
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.

Log in to the UCM Web GUI and open the **Call Features > Multimedia Meeting** page to manage the conference room. Users can create, edit, view, invite, manage meeting members, and delete meeting rooms. The conference room status and conference call recording (if the recording function is enabled) will be displayed on the page. The meeting rooms in the list include public meeting rooms and random meeting rooms. For temporary meeting room administrators, only the “batch kicking people” function is supported. The temporary meeting room has no meeting password or host code. The member who initiates the group meeting is the host, and ordinary members have the right to invite.

Multimedia Meeting

Meetings Settings

To edit the general settings of the meeting rooms created in the UCM, the user can click on “Meetings Settings” button under the **Room** tab.

Meeting Max Concurrent Audio	Maximum number of participants that can be heard simultaneously in multimedia meetings. If the number of participants talking at any given point exceeds this value, the audio of the excess participants will not be heard.
Meeting Voice Indicator Sensitivity	Configures the sensitivity of the talking indicator in multimedia meetings. Setting this higher will make the talking indicator appear more easily for lower volumes of audio. Note: This does not adjust audio input sensitivity itself. Lower volumes of sounds may still be heard even if the talking indicator does not show the source.
Meeting Audio Quality	Audio quality of multimedia meetings
Meeting Record Prompt	If enabled, system will prompt the user before the start of meeting recording that your meeting will be recorded.

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.
Meeting AGC (beta)	Enabling this option will toggle on Automatic Gain Control for meeting audio. AGC is a system that dynamically reduces the variability of sound levels by adjusting high and low volumes based on the average or peak sound level. High volume sounds will be lowered, and low volume sounds will be boosted.
Silence Suppression	Silence suppression for temporary accounts (e.g., meeting participants that joined the meeting via link). 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.
Enable Talk Detection	If enabled, the AMI will send the corresponding event when a user starts or stops talking.
DSP Talking Threshold (ms)	The amount of time(ms) that sound exceeds what the DSP has established as the baseline for silence 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.
DSP Silence Threshold (ms)	The amount of time(ms) that sound falls within what the DSP has established as the baseline for 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.
Max Number of Video Feeds	Set the maximum number of video feeds supported per meeting room.
Audio Codec Preference	Configures the preferred codecs for temporary accounts such as meeting participants who joined via link.
Packet Loss Retransmission	Packet Loss Retransmission configuration for temporary accounts (meeting participants without registered extensions who entered the meeting via link).
Jitter Buffer	<p>Select the jitter buffer method for temporary accounts such as meeting participants who joined via link.</p> <ul style="list-style-type: none"> ● Disabled: 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.

Multimedia Meeting Call Operations


Join a Meeting Call

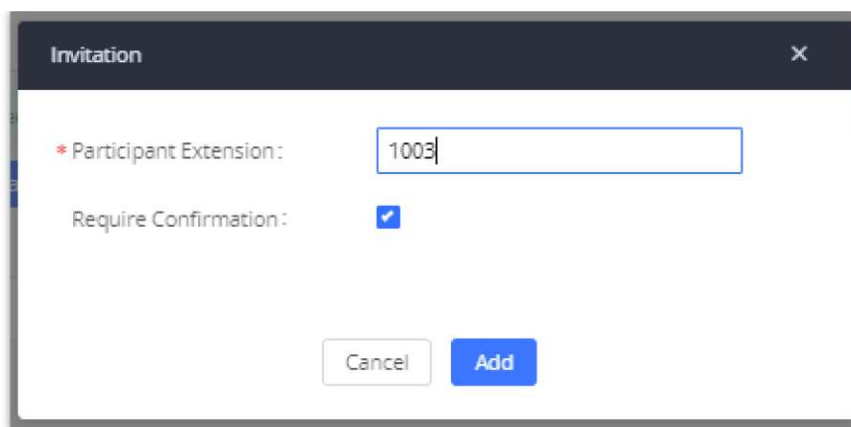
Users could dial the meeting room extension to join the meeting. If the password is required, enter the password to join the meeting as a normal user, or enter the admin password to join the meeting as an administrator.

Invite Other Parties to Join a Meeting

When using the UCM meeting room., there are two ways to invite other parties to join the meeting.

- **Invite from Web GUI.**

For each meeting room in CloudUCM Web GUI→**Call Features**→ **Multimedia Meeting**, there is an icon  for option "Invite a participant". Click on it and enter the number of the party you would like to invite. Then click on "Add". A call will be sent to this number to join the conference.



Meeting Invitation from Web GUI

- **Invite by dialing 0 or 1 during a conference call.**

A meeting participant can invite other parties to the meeting by dialing from the phone during the meeting call. Please make sure the option "Enable User Invite" is turned on for the meeting room first. Enter 0 or 1 during the meeting call. Follow the voice prompt to input the number of the party you would like to invite. A call will be sent to this number to join the meeting.

0: If 0 is entered to invite another party, once the invited party picks up the invitation call, permission will be asked to "accept" or "reject" the invitation before joining the conference.

1: If 1 is entered to invite another party, no permission will be required from the invited party.






i Conference administrators can always invite other parties from the phone during the call by entering 0 or 1. To join a conference room as an administrator, enter the admin password when joining the conference. A conference room can have multiple administrators.

During The Meeting

During the meeting call, users can manage the conference from Web GUI or IVR.

- **Manage the meeting call from Web GUI.**

Log in UCM Web GUI during the meeting call, and the participants in each meeting room will be listed.

1. Click on  to kick a participant from the meeting.
2. Click on  to mute the participant.
3. Click on  to lock this meeting room so that other users cannot join it anymore.
4. Click on  to invite other users into the meeting room.
5. Click on  to Invite meeting rooms or Invite contact groups.

- **Manage the meeting call from IVR.**

Please see the options listed in the table below.

Meeting Administrator IVR Menu	
1	Mute/unmute yourself.
2	Lock/unlock the conference room.
3	Kick the last joined user from the conference.
4	Decrease the volume of the conference call.
5	Decrease your volume.
6	Increase the volume of the conference call.

7	Increase your volume.
8	More options. <ul style="list-style-type: none"> ○ 1: List all users currently in the conference call. ○ 2: Kick all non-administrator participants from the conference call. ○ 3: Mute/Unmute all non-administrator participants from the conference call. ○ 4: Record the conference call. ○ 8: Exit the caller menu and return to the conference.
Meeting User IVR Menu	
1	Mute/unmute yourself.
4	Decrease the volume of the conference call.
5	Decrease your volume.
6	Increase the volume of the conference call.
7	Increase your volume.
8	Exit the caller menu and return to the conference.

Meeting Caller IVR Menu

i When there is a participant in the meeting, the meeting room configuration cannot be modified.

Google Service Settings Support

CloudUCM supports Google OAuth 2.0 authentication. This feature is used for supporting the CloudUCM meeting scheduling system. Once OAuth 2.0 is enabled, the CloudUCM conference system can access Google Calendar to schedule or update conference.

Google Service Settings can be found under Web GUI → **Call Features** → **Multimedia Meeting** → **Google Service Settings** → **Google Service Settings**.

OAuth2.0 Authentication

* OAuth2.0 Client ID:

* OAuth2.0 Client Secret:

Google Service Settings → OAuth2.0 Authentication

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

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

1. Go to the Google Developers page <https://console.developers.google.com/start> Create a New Project on the Google Developers page.

New Project

Project name ?

Your project ID will be animated-surfer-112001 ? [Edit](#)

[Show advanced options...](#)

Please email me updates regarding feature announcements, performance suggestions, feedback surveys and special offers.

Yes No

I agree that my use of any [services and related APIs](#) is subject to my compliance with the applicable [Terms of Service](#).

Google Service → New Project

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

Google Developers Console OAuthTest

Home
Permissions
APIs & auth
APIs
Credentials
Push
Monitoring
Source Code
Deploy & Manage
Compute
Networking
Storage
Big Data

API Library [Enabled APIs \(8\)](#)

Some APIs are enabled automatically. You can disable them if you're not using their services.

[API](#) ^

- BigQuery API
- Calendar API
- Cloud Debugger API
- Debuglet Controller API
- Google Cloud Logging API
- Google Cloud SQL
- Google Cloud Storage
- Google Cloud Storage JSON API

Google Service → Create New Credential

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

Google Calendar Authorization

1. Click "Get Authorization Code". [Get Authorization Code](#)
2. Enter the Google account and password (Note: please make sure the account on authorization page is correct, if you have logged in other account, please log out then log in again).
3. Click "Accept" on authorization page.
4. Copy the string to the Authorization Code input box, click the "authorize" button.

* Authorization Code: [Authorization](#)

Google Service → OAuth2.0 Login

6. Once this has been done, the CloudUCM will connect to Google services.

You can also configure the Status update, which automatically refreshes your Google Calendar with the configured time (m).
Note: Zero means disable.

Meeting Schedule

Log in to the UCM Web GUI, open the **Call Features → Multimedia Meeting → Meeting Schedule** page, and you can manage the reservation management of the meeting room. Users can create, edit, view, and delete conference room reservation records. The following is a set meeting room reservation, which shows the ongoing and pending reservations. Once the conference room is reserved, all users will be removed from the conference room at the start time, and extensions will no longer be allowed to enter the conference room. At the scheduled meeting time, UCM will send invitations to the extensions that have been selected to participate in the meeting. At the same time, it supports users to enter the meeting 10 minutes in advance. If the current meeting is occupied, enter the waiting room and wait (members joining the meeting in advance occupy global member resources, but it will be released after the scheduled meeting starts); otherwise, you can join the meeting directly and the meeting will be held in advance. After the meeting ends, the reservation record is transferred to the historical meeting list. History meeting displays the information of the ended and expired meetings.

- Click the button "Schedule Meeting" to edit the meeting room reservation.

Schedule New Meeting
Cancel [Save](#)

<p>* Meeting Subject: <input type="text"/></p> <p>* Time: <input type="text" value="2021-10-26"/> <input type="text"/></p> <p>Password: <input type="text"/></p> <p>* Host: <input type="text"/></p> <p>Allow User Invite: <input checked="" type="checkbox"/></p> <p>* Email Reminder (m): <input type="text" value="60"/></p> <p>Enable Google Calendar: <input type="checkbox"/> Google Service Settings</p> <p>Invitees: <div style="border: 1px solid #ccc; padding: 5px; min-height: 100px; font-size: 0.8em;">Please enter and submit participants with the Enter key</div></p> <p style="font-size: 0.8em; margin-top: 5px;">For improved voice audio quality of link users, please add Opus to the meeting room's supported codecs. Warning: Opus is a resource-intensive codec.</p>	<p>Meeting Room: <input type="checkbox"/> Use Existing Meeting Room</p> <p>* Time Zone: <input type="text" value="{ UTC+01:00 } Etc/GMT-1"/></p> <p>* Host Password: <input type="text" value="0590"/></p> <p>Repeat: <input type="text" value="No Repeat"/></p> <p>Call Participants: <input checked="" type="checkbox"/></p> <p>Auto Record: <input type="checkbox"/></p> <p>Description: <div style="border: 1px solid #ccc; min-height: 40px;"></div></p>
---	---

Schedule meeting Interface

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	If this option is enabled, the user can: <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. Note: Meeting host is always allowed to access this menu.
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)	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. 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.
Auto Record	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.
Enable Google Calendar	Select this option to sync scheduled meeting with Google Calendar. Note: Google Service Setting OAuth2.0 must be configured on the CloudUCM. Please refer to Google Services configuration section.
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 the Meeting Schedule is configured, the scheduled meeting will be displayed as the below figure.

Multimedia Meeting

Room [Meeting](#) Meeting Recordings Meeting Video Recordings

[Pending Meeting](#) Meeting History

[Schedule Meeting](#) Meeting Su... [Search](#)

Meeting Subject	Meeting Room	Meeting Owner	Start Time	Meeting Duration	Repeat	Options
Weekly_Meeting	6300	admin	2024-03-20 16:00 Etc/GMT+1	01:00:00	No Repeat	i edit delete

Total: 1 [<](#) [1](#) [>](#) 10 / page

Meetings Schedule

- Click the button [i](#) to view the meeting details in the Meeting room. The meeting details of Meeting History include actual participant information.

Meeting Details ✕

Meeting Subject **Scheduled Meeting 1**
[Record Audio](#) | [Record Video](#)

Room Number **50204021**

Session state **Not started**

Start Time **2022-03-08 16:00**

Time Zone **Etc/GMT-1**

Meeting Owner **Ilias**

Password

Host Password **7961**

Enable Google Calendar **No**

Repeat **No Repeat**

[Invitees](#)

STATUS	FIRSTNAME	PHONE NUMBER	EMAIL	LEAVE A MESSAGE
Require Confirmation		1004	i @grandstream.com	

Meeting details

- Click on [edit](#) to edit the Meeting Schedule.
- Click on [delete](#) to delete the Meeting Schedule.

At the scheduled meeting time, CloudUCM will send INVITE to the extensions that have been selected for the conference.

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

[Pending Meeting](#) Meeting History

Host [Schedule Meeting](#)

MEETING SUBJECT	MEETING ROOM	MEETING OWNER	START TIME	MEETING DURATION	REPEAT	OPTIONS
Meeting Ongoing	6300	admin	Today 16:15 Etc/GMT-1	00:15:00	No Repeat	i edit delete

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

MEETING SUBJECT	MEETING ROOM	MEETING OWNER	START TIME	MEETING DURATION	REPEAT	OPTIONS
Meeting by 1004	20205093	1004	2022-03-08 10:08	00:01:00	No Repeat	
Meeting by 1004	30903092	1004	2022-03-08 10:04	00:00:00	No Repeat	
Meeting by 1004	40500041	1004	2022-03-07 09:50	00:04:00	No Repeat	

Meeting Schedule – Completed

- Click the button to download the Meeting Report of the meeting.
- Click the button to reschedule the Meeting.

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

You can also choose to display the meetings that took place in a specific time frame. Please see the screenshot below:

MEETING SUBJECT	MEETING ROOM	MEETING OWNER	START TIME	MEETING DURATION	REPEAT	OPTIONS
Meeting by 1004	20205093	1004	2022-03-08 10:08	00:01:00	No Repeat	
Meeting by 1004	30903092	1004	2022-03-08 10:04	00:00:00	No Repeat	
Meeting by 1004	40500041	1004	2022-03-07 09:50	00:04:00	No Repeat	

Please make sure that the outbound route is properly configured for remote extensions to join the meeting.

Meeting Recordings

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

To record the Meeting call, when the meeting room is idle, enable “Auto Record” 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 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.

Multimedia Meeting

Room Meeting Google Service Settings **Meeting Recordings** Meeting Video Recordings

Download Download All Delete Clear Local 2022-04

NAME	ROOM	DATE	SIZE	OPTIONS
meeting-6300-1651061843.wav	6300	2022-04-27 13:17:23 UTC+01:00	342.23 KB	00:22 [play] [download] [delete]



Total: 1 10 / page Goto 1

Meeting Recordings

Meeting Video Recordings

The CloudUCM allows users to record the audio and video of the meeting call and retrieve the recording from Web GUI → **Call Features** → **Multimedia Meeting** → **Meeting Recordings**.

To record the Meeting call, when the meeting room is idle, enable "Auto Record" 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 .mkv format.

The recording files will be listed 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.

Call Statistics

Meeting reports will now be generated after every conference. These reports can be exported to a .CSV file for offline viewing. The conference report page can be accessed by clicking on the Call Statistics button on the main Conference page.

+ Add Meeting Settings **Call Statistics** Enable CEI Notify

ROOM	ATTENDEE	HOST	START TIME	ACTIVITY	OPTIONS
6301	0	0		--	[download] [delete] [refresh] [print] [share] [help] [close]

USER	CALLER ID	CALLER NAME	CHANNEL NAME	ACTIVITY	OPTIONS

Meeting Call Statistics

Call Statistics

Download Delete Report Scheduled Export 2019-11-01 to 2019-11-07 Access To Rep

CONFERENCE	START TIME	TIME
6301	2019-11-07 16:12:16	00:01:16

1001	1001	Received	Answered
Sales	--	Invitation	0 / 4
Conference invitation	1002	Invitation	Failed
Conference invitation	1005	Invitation	Failed

Meeting Report on Web



	A	B	C	D	E	F	G
1	Room	Start Time	Duration Time				
2	6301	11/7/2019 16:12	0:01:16				
3	Contact Number	Name	Way	Status	Contact Group Name		
4	1002	Conference invitatio	INVITE	FAILURE	Sales		
5	1005	Conference invitatio	INVITE	FAILURE	Sales		
6	1004	Conference invitatio	INVITE	FAILURE	Sales		
7	1003	Conference invitatio	INVITE	FAILURE	Sales		
8	1001	1001	CALLIN	ANSWER			

Meeting Report on CSV

IVR

Configure IVR

IVR configurations can be accessed under the CloudUCM 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

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 the 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 CloudUCM will compare the IVR's permission level with the outbound route's privilege level.</p> <p>If the IVR's permission level is higher than (or equal to) the outbound route's privilege level, the call will be allowed to go through.</p>
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 ○ Conference ○ Call Queue ○ Ring Group ○ Paging/Intercom Groups ○ Voicemail Groups ○ Dial by Name ○ All
IVR Black/Whitelist	<p>If enabled only numbers inside of the Whitelist or outside of the Blacklist can be called from IVR.</p>
Internal Black/Whitelist	<p>Contain numbers, either of Blacklist or Whitelist.</p>
External Black/Whitelist	<p>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...</p>
Replace Display Name	<p>If enabled, the CloudUCM will replace the caller display name with the IVR name.</p>
Return to IVR Menu	<p>If enabled and if a call to an extension fails, the caller will be redirected to the IVR menu.</p>
Alert Info	<p>When present in an INVITE request, the alert-Info header field specifies an alternative ring tone to the UAS.</p>
Prompt	<p>Select an audio file to play as the welcome prompt for the IVR. Click on "Prompt" to add audio file under Web GUI→PBX Settings→Voice Prompt→Custom Prompt.</p>
Digit Timeout	<p>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 CloudUCM will consider the entries complete. The default timeout is 3s.</p>
Response Timeout	<p>After playing the prompts in the IVR, the CloudUCM 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.</p>

Response Timeout Prompt	Select the prompt message to be played when the 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 currently available voice prompt languages on the CloudUCM. To add more languages in the list, please download the voice prompt package by selecting "Check Prompt List" under Web GUI→ PBX Settings → Voice Prompt → Language Settings .
Key Pressing Events	
Key Press Event:	Select the event for each key pressing for 0-9, *, Timeout, and Invalid. The event options are:
Press 0	<ul style="list-style-type: none"> ○ Extension ○ Voicemail ○ Multimedia Meeting ○ Voicemail Group
Press 1	<ul style="list-style-type: none"> ○ IVR ○ Ring Group
Press 2	<ul style="list-style-type: none"> ○ Queues
Press 3	<ul style="list-style-type: none"> ○ Page Group
Press 4	<ul style="list-style-type: none"> ○ Custom Prompt ○ Hangup
Press 5	<ul style="list-style-type: none"> ○ DISA
Press 6	<ul style="list-style-type: none"> ○ Dial by Name ○ External Number
Press 7	<ul style="list-style-type: none"> ○ Callback
Press 8	For each key event, time conditions 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 conditions. Up to 5 time conditions can be added for each key.
Press 9	
Press *	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.

Timeout	When exceeding the number of defined answer timeout, IVR will enter the configured event when timeout. If not configured, then it will hang up.
Invalid	Configure the destination when the Invalid Repeat Loop is done.
Time Condition	Configure the time condition for each key press event, so that it goes to the corresponding destination within a specified time.

IVR Configuration Parameters

Key Pressing Events

Black/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, and some special extensions are used on IP surveillance endpoints which should not be reached from external calls via IVR for privacy reasons. CloudUCM has now added blacklist and whitelist in IVR settings for users to manage this.

! Up to 500 extensions are allowed on the back/whitelist.

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

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

Create New IVR

Basic Settings Key Pressing Events

* Name:

* Extension:

Dial Trunk:

* Permission:

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

* IVR Black/Whitelist:

Internal Black/Whitelist:

28 items Available		2 items Selected	
<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 a 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 the 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 CloudUCM, it will be added to the prompt list options for users to select in different IVR scenarios.

Key Pressing Events

Standard Key Event

CloudUCM 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 supports 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.

Edit IVR: IVR-DISA

Basic Settings **Key Pressing Events** Cancel Save

Press 0

Destination: Time Condition: ⊖

Destination: Time Condition: ⊖

Press 1

Destination: Time Condition: ⊖

⊖

- All Time ⊕
- Office Time
- Out of Office Time
- Holiday**
- Out of Holiday ⊖
- Out of Office Time or Holiday ⊕
- Office Time and Out of Holiday

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.

Specific Time ✕

Time: -

Frequency: By Week By Month

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

Week Day

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
1							
2							
3							
4							
5							
6							

Excluding Holidays:

Cancel OK

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

Basic Settings **Key Pressing Events** Cancel Save

Key Event Type: Standard Custom

Key Events

+ Add Delete Clear

Key: 🗑️

Destination: Time Condition: ⊖ ⊕

Other Settings

Timeout

Destination: Time Condition: ⊖ ⊕

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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

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

Voicemail

Configure Voicemail

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

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

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 the 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 is 10 to 1000. The default setting is 50.
Max Message Time	Select the maximum duration of the voicemail message. The message will not be recorded if the duration exceeds the maximum message time. The default setting is 15 minutes. The available options are: <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, 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 CloudUCM with 2 options:</p> <ul style="list-style-type: none"> ○ Press 1 to leave a message. <p>To leave a message for the extension reached by DID.</p> <ul style="list-style-type: none"> ○ Press 2 to access the voicemail management system. <p>This will allow the caller to access any extension VM after entering the extension number and its VM password.</p> <p>Note: This option applies to inbound calls 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.

ⓘ 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 CloudUCM 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 to access that specific extension's voicemail.

Main Menu	Sub Menu 1	Sub Menu 2
1 – New messages	3 - Advanced options	1 - Send a reply
		2 - Call the person who sent this message
		3 - Hear the message envelop
		4 - Leave a message
		* - Return to the main menu
	5 - Repeat the current message	
	7 - Delete this message	
	8 - Forward the message to another user	
	9 – Save	
	* - Help	
# - Exit		
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 the user is not available, the caller will be automatically redirected to the voicemail to leave a message on which case they can press # to submit the message.

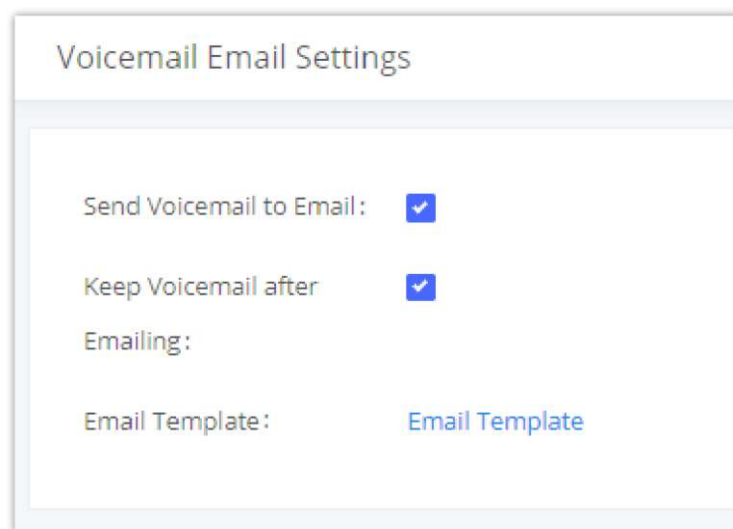
In case the caller is calling from an internal extension, they will be directly forwarded to the extension's voicemail box. But if the caller is calling from outside the system and the incoming call is routed by DID to the destination extension, then the caller will be prompted with the choice to either 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 CloudUCM can be configured to send the voicemail as an attachment to the Email. Click on the "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	Fill in the "Subject:" and "Message:" content, to be used in the Email when sending to the user. The template variables are: <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. (Format: MM/dd/yyyy hh:mm:ss)

Voicemail Email Settings



Voicemail Email Settings

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

Configure Voicemail Group

The CloudUCM 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 > Create New Voicemail Groups

* Extension

* Name

* Method Forwarded Shared

Voicemail Password

Email Address

Shared Voicemail Status

Members

<input type="checkbox"/> 498 Available	<input type="checkbox"/> 0 Selected
<input type="text" value="Search"/> <ul style="list-style-type: none"> <input type="checkbox"/> 1000 "" <input type="checkbox"/> 1003 "Johnny Doe" <input type="checkbox"/> 1004 "John Marston" <input type="checkbox"/> 1005 "Abigail Roberts" <input type="checkbox"/> 1006 "Mary-Beth Gaskill" <input type="checkbox"/> 1007 "Hosea Matthews" 	<input type="text" value="Search"/> <p style="text-align: center;">None</p>

Voicemail prompt will be played when user enters voicemail. Priority: Temporary Prompt > Unavailable Prompt > Name Prompt
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.

Name Prompt

Temporary Prompt

Unavailable Prompt

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 Groups

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

Configure Ring Group

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

EXTENSION	NAME	STRATEGY	MEMBERS	MESSAGE	OPTIONS
6400	TechSupport	Ring in Order	1000 1001 1002 1003 1004	Messages: 0/0/0	[Edit] [Delete]

Ring Group

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

Create New Ring Groups

* Ring Group Name:

* Extension:

Members:

27 items Available

Search

- 1000
- 1001
- 1005
- 1006
- 1007

3 items Selected

Search

- 1002
- 1003
- 1004

LDAP Phonebook:

0 item Available

Search

None

0 item Selected

Search

None

Ring Group Options

Ring Strategy:

Music On Hold:

Custom Prompt: [Upload Audio File](#)

Ring Group Configuration


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	<p>Select the ring strategy. The default setting is “Ring in order”.</p> <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	<p>This option is to set a custom prompt for a ring group to announce to caller. Click on ‘Prompt’, it will direct the users to upload the customized voice prompts.</p> <p>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>

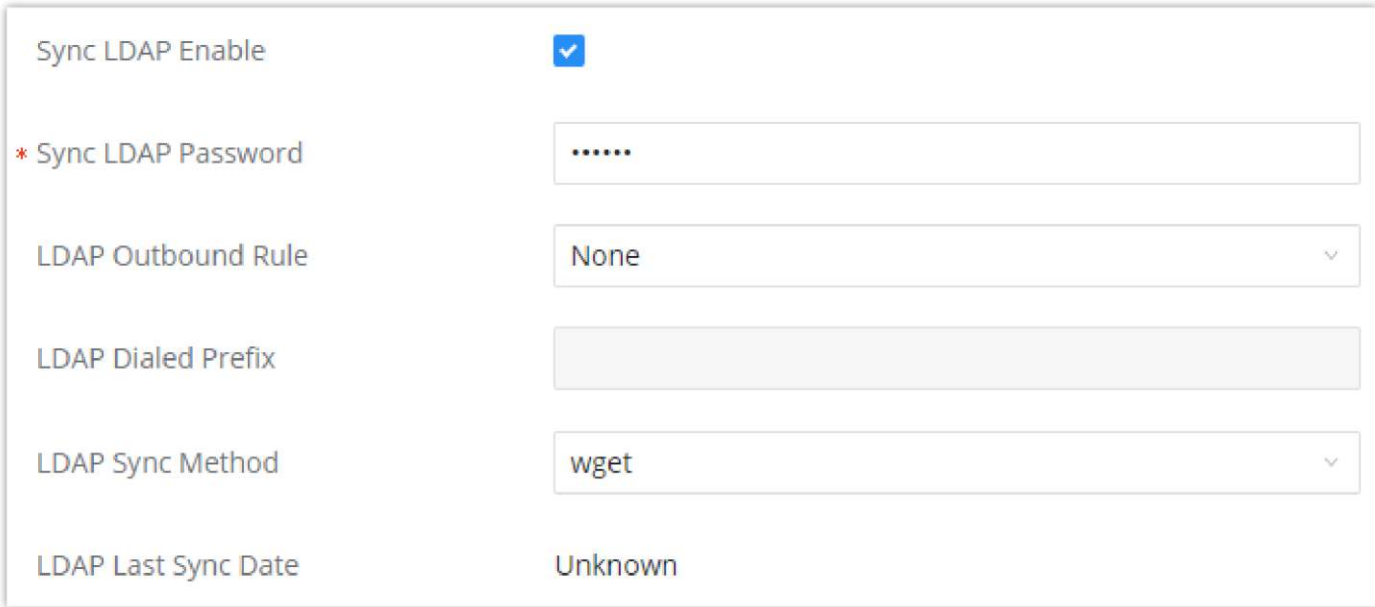
Ring Timeout on Each Member	<p>Configure the number of seconds to ring each member. If set to 0, it will keep ringing. The default setting is 60 seconds.</p> <p>Note: The actual ring timeout might be overridden by users if the phone has ring timeout settings as well.</p>
Auto Record	<p>If enabled, calls on this ring group will be automatically recorded. The default setting is No. The recording files can be accessed from WebGUI→ CDR→ Recording Files.</p>
Endpoint Call Forwarding Support	<p>This allows the UCM to work with endpoint-configured call forwarding settings to redirect calls to ring group. For example, if a member wants to receive calls to the ring group on his mobile phone, he will have to set his endpoint's call forwarding settings to his mobile number. By default, it is disabled. However, this feature has the following limitations:</p> <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	<p>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.</p>
Skip Busy Agent	<p>If enabled, skip busy agents regardless of call waiting settings.</p>
Enable Destination	<p>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.</p>
Default Destination	<p>The call would be routed to this destination if no one in this ring group answers the call.</p> <p>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.</p>
Voicemail	<p>Whether to enable the voicemail for the ring group or not.</p>
Voicemail Password	<p>Configure the voicemail password (only numbers).</p>
Email Address	<p>Fill in the user's Email address (s), the voice message will be sent to this address (s).</p>
Busy Prompt	<p>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</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>
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 only be played when the callee's extension is unregistered. 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>
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Remote Extension in Ring Group

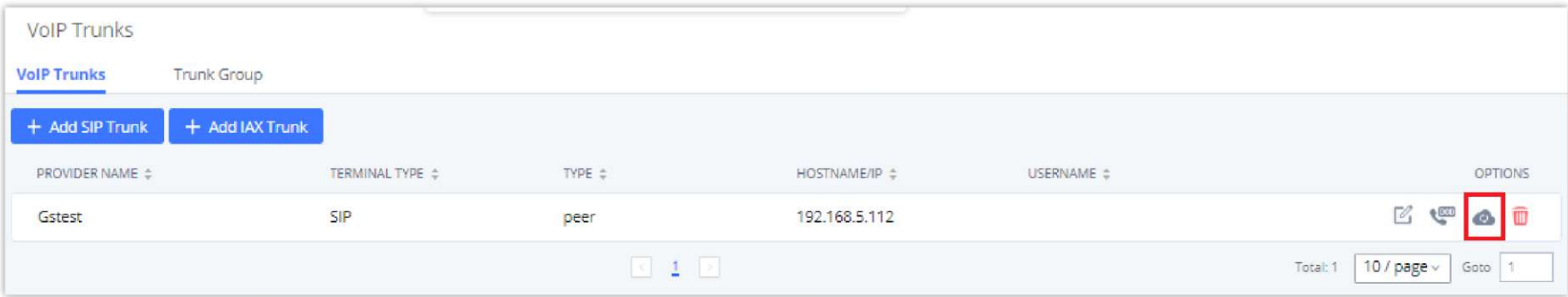
Remote extensions from the peer trunk of a remote UCM can be included in the ring group with local extensions. An example of Ring Group with peer extensions is presented in the following:

1. Creating SIP Peer Trunk between both UCM_A and UCM_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 UCM_A update remote LDAP server automatically from peer UCM_B. In addition, **Sync LDAP Password** must match for UCM_A and UCM_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 Server Options

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 UCM.



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 UCM can be included into that UCM's LDAP contact.

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.

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. By default it's enabled.

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 CloudUCM 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 0 item Selected

- 1000
- 1001

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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 the 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.

2-way Intercom Configuration Parameters

1-way Paging

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 on the [CDR → Recordings → Video Recordings](#) page.

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 -

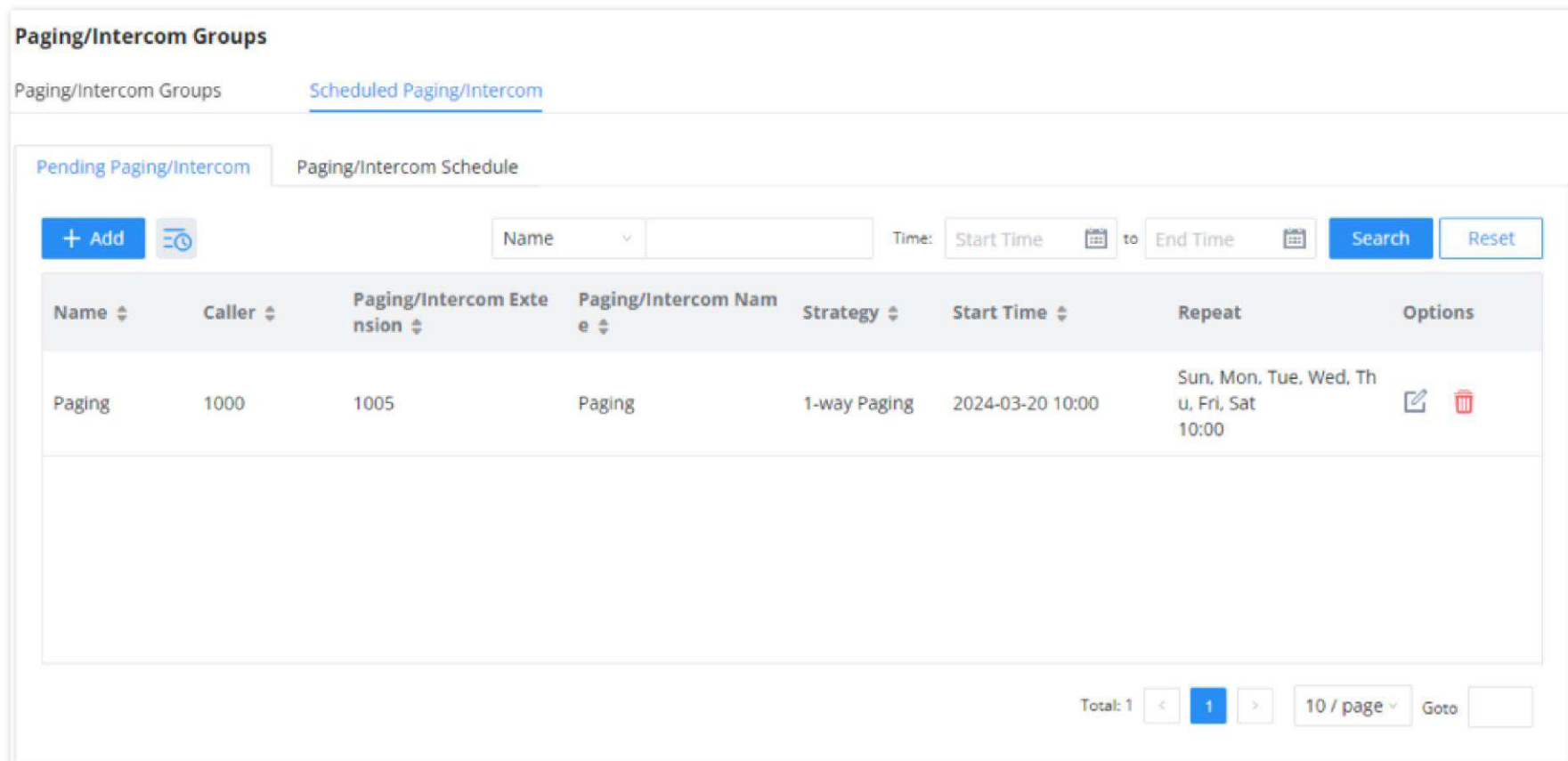
[Add Start Time](#) +

Repeat

Sync to Google Calendar [Google Services](#)

Paging/Intercom	Select existing paging/intercom groups.
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. 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.
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.

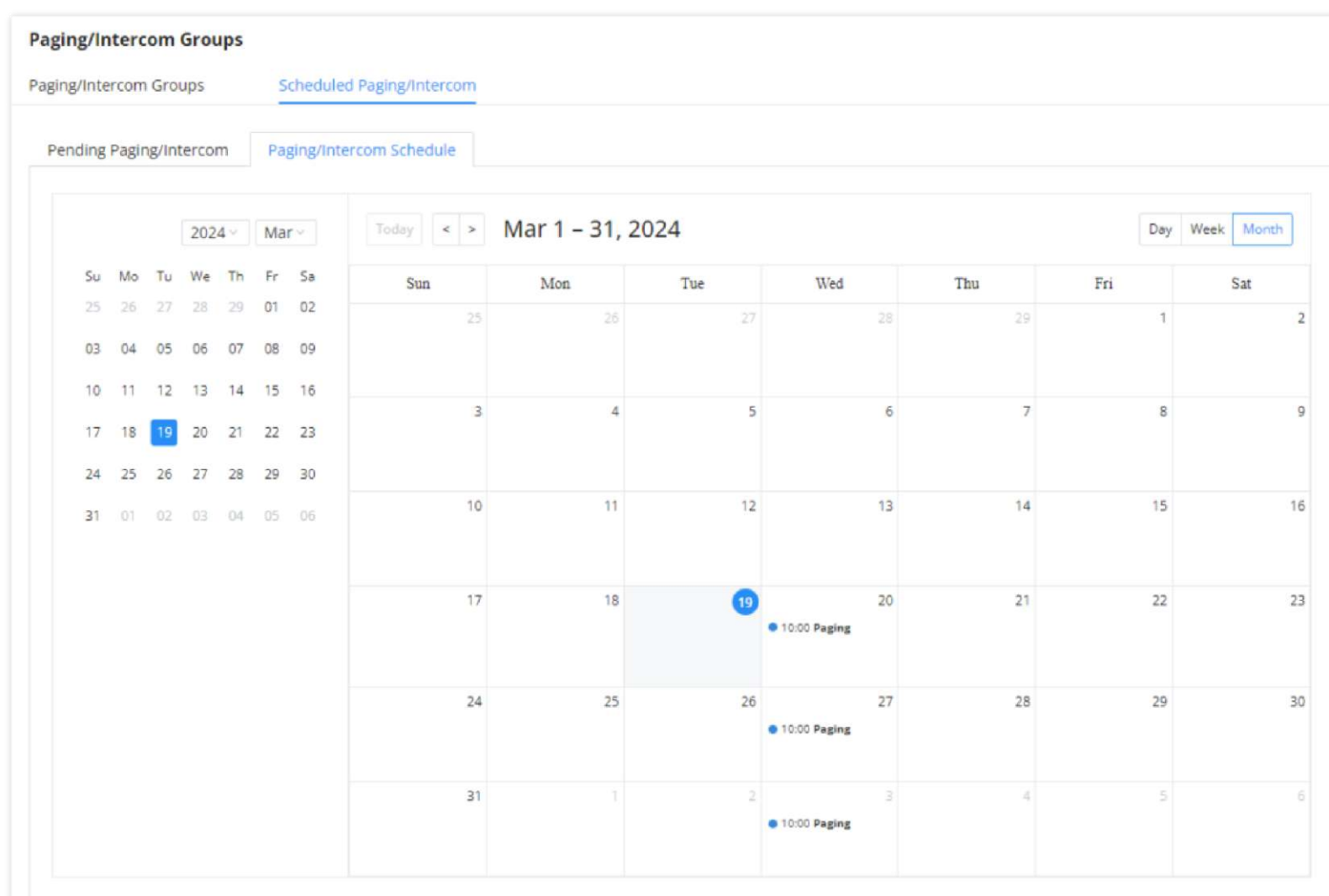
Once the paging and intercom has been created, it can be viewed on the same page.



Pending Paging/Intercom

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.



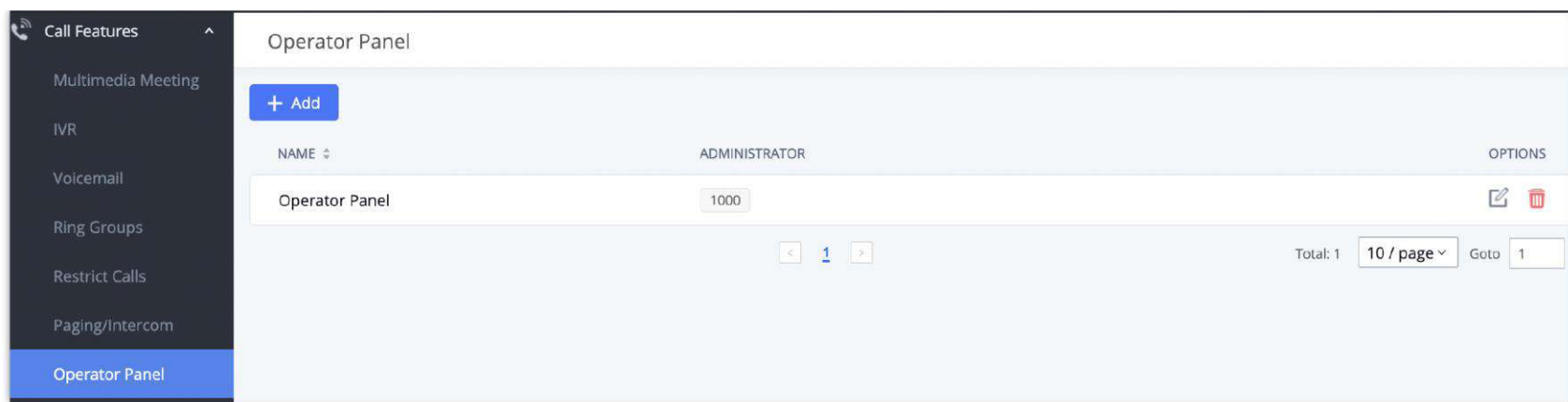
Scheduled Paging/Intercom

Operator Panel



Configure Operator Panel

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

The CloudUCM supports the 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, 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 the operator panel.



Operator Panel Configuration Page

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

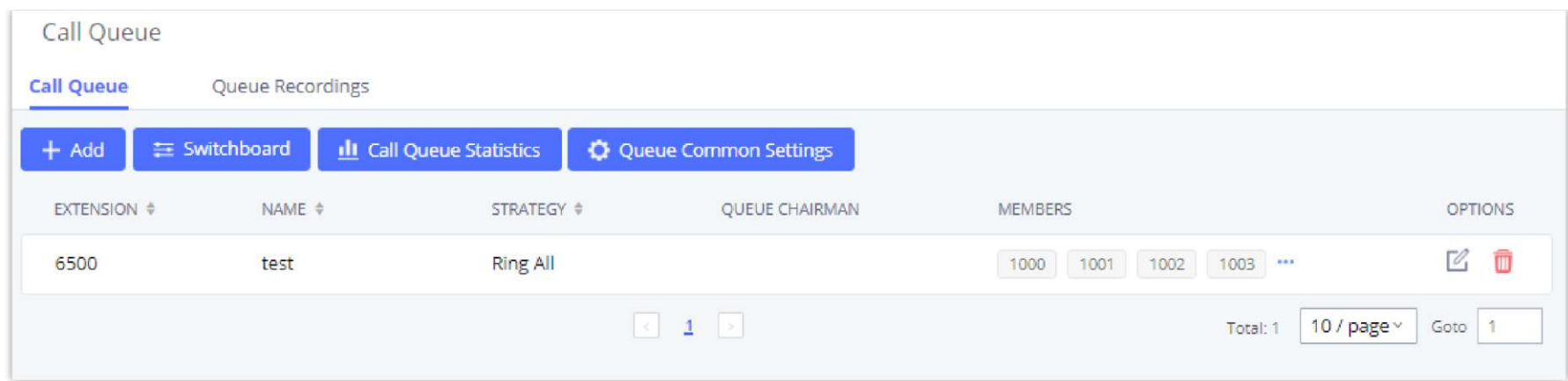
Name	The Operator panel name.
Administrator	The operator of the call console can select extensions, extension groups, and departments. For the selected extension groups and departments, subsequent extensions will automatically become administrators.
Management Module	
Extension	The selected extensions 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 checked Ring Groups will be supervised by the administrator. Select “All”, all Ring Groups and subsequent updates will be automatically supervised by the administrator.
Voicemail Groups	The checked Call Queue will be supervised by the administrator. Select “All”, all Call Queue and subsequent updates will be automatically supervised by the administrator.
Call Queue	The checked Call Queue will be supervised by the administrator. Select “All”, all Call Queue and subsequent updates will be automatically supervised by the administrator.
Parking Lot	The checked Parking Lot will be supervised by the administrator. Select “All”, all Parking Lot and subsequent updates will be automatically supervised by the administrator.

Call Queue

CloudUCM 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

CloudUCM 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.

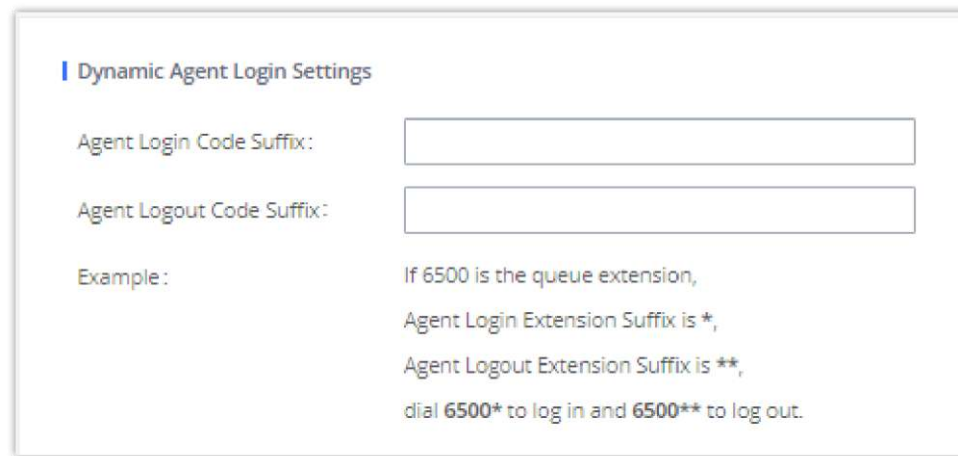
Basic Settings	
General	
Extension	Configure the call queue extension number.
Name	Configure the call queue name to identify the call queue.
Strategy	<p>Select the strategy for the call queue.</p> <ul style="list-style-type: none"> ● Ring All: Ring all available Agents simultaneously until one answers. ● 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 CloudUCM 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.

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.

The 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 the current number as a 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	The 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	Enable the announcement of the caller's position periodically. Note: The queue position will now be announced to the caller upon entering the queue.
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	When enabled, statics agents can conveniently log in and out of a queue by configuring a programmable key on their phones as a shortcut. Notes: <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.

Call Center Parameters

Queue Auto fill enhancement:

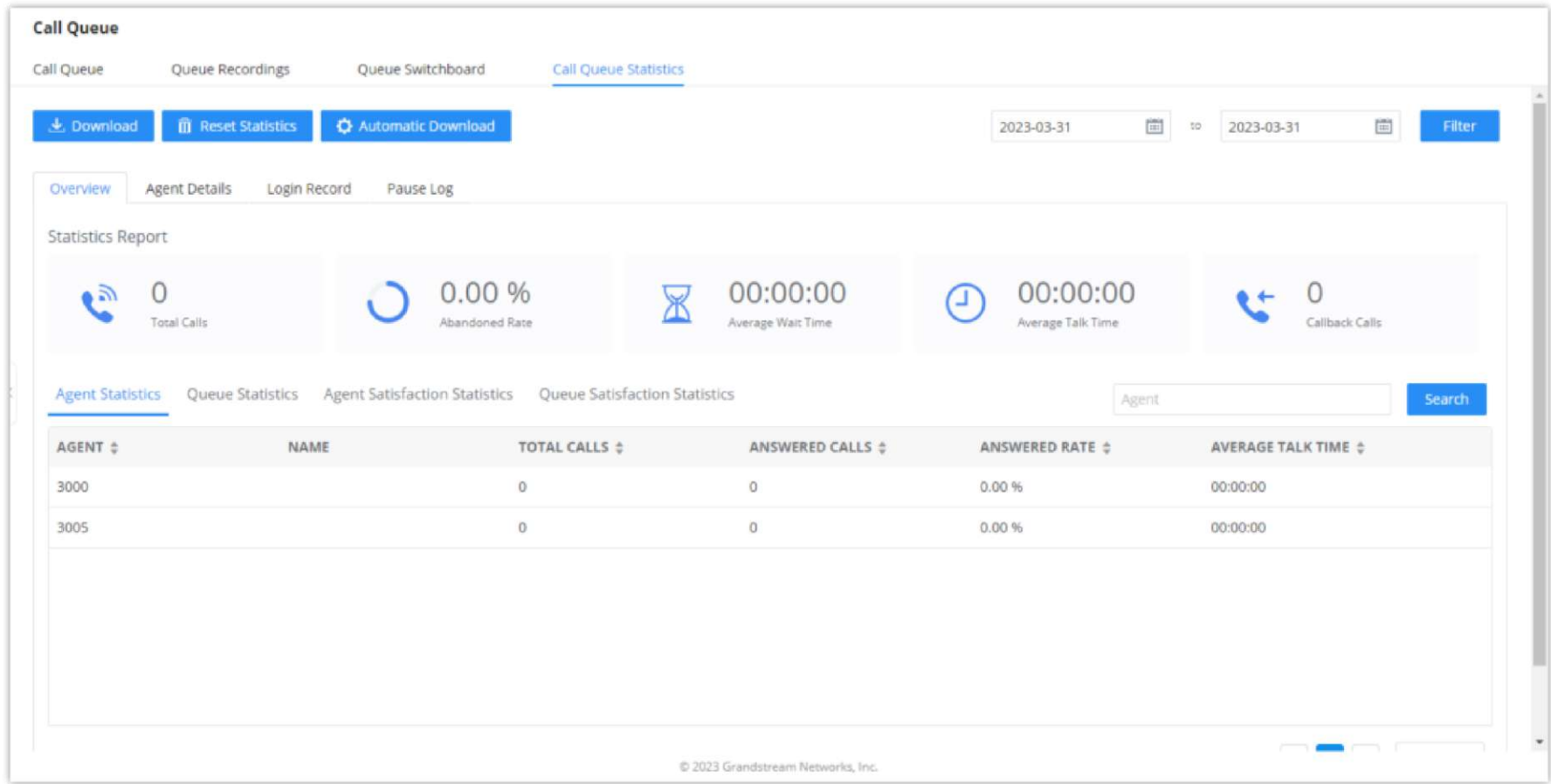
The waiting callers are connecting with available members in a parallel fashion until there are no more available members or no more waiting callers.

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.

Call Queue Statistics

Along with call center features, users can also gather detailed call queue statistics allowing them to make better changes/decisions to manage 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:

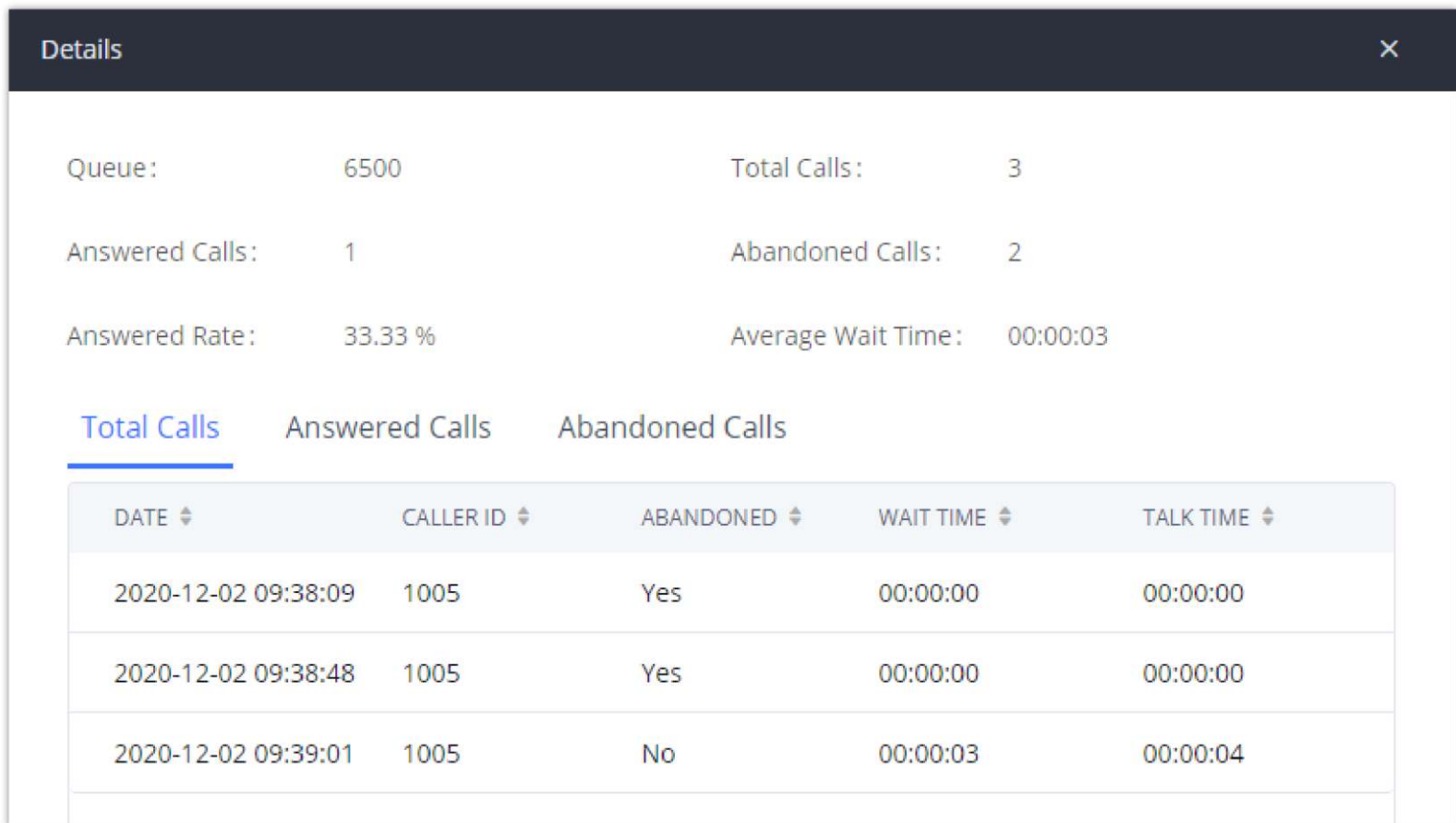


Call Queue Statistics

- 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 count 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:



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 pre-configured Period, this can be done by clicking on **Automatic Download**, and the user will be directed to the 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 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 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 the 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 the 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/unpaused.
- **Name:** The name of the agents that paused/unpaused.
- **Queue:** The queue that the agent is in.
- **Pause Time:** The time when the agent paused.
- **Resume Time:** The time when the agent unpaused.
- **Pause Duration:** The total length of time the agent was paused for.
- **Pause Reason:** The reason of the pause (e.g., lunch, coffee break, etc...)

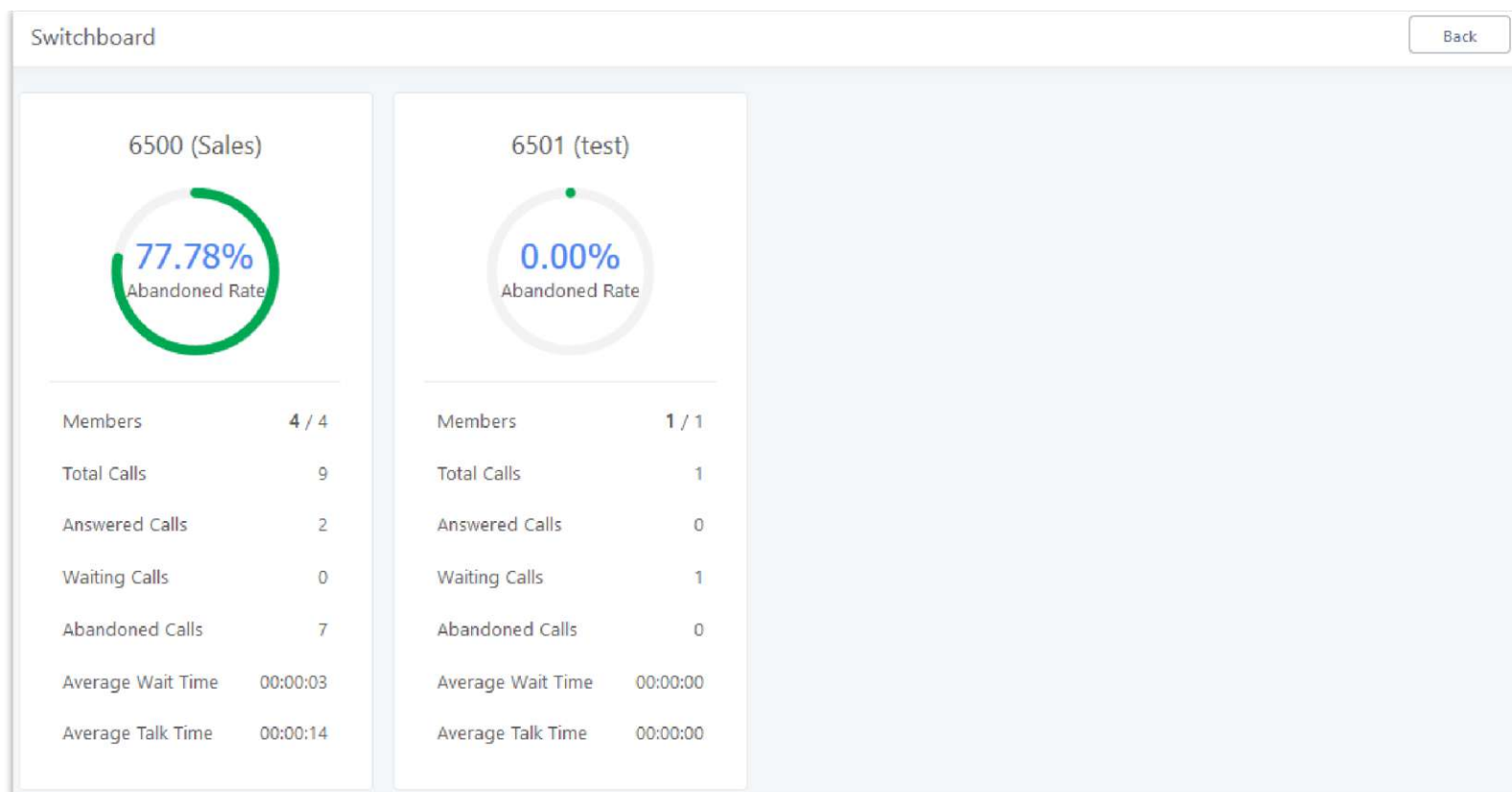
AGENT	NAME	QUEUE	PAUSE TIME	RESUME TIME	PAUSE DURATION	PAUSE REASON
No data						

Pause Log

Switchboard

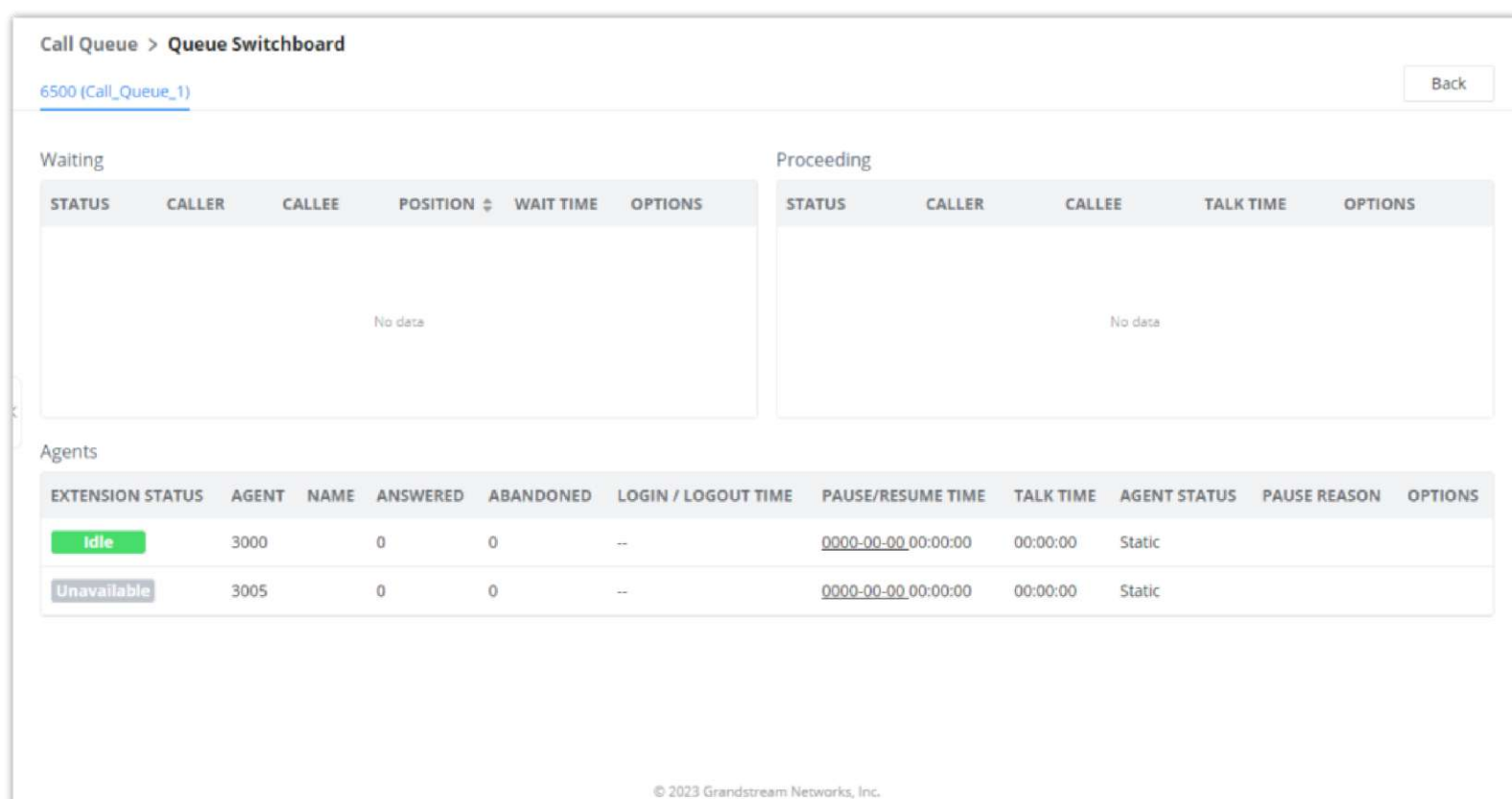
Switchboard is a Web GUI tool for call queue monitoring and management, admin can access to it from the menu **Call Features**→**Call Queue** then press "Switchboard".

Following page will be displayed:




Switchboard Summary

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



Call Queue Switchboard

The table below gives a brief description of 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 currently established calls along with the caller ID and the callee (agent) as well as the option to hang up, transfer, add conference, or barge-in the call.
Agents	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). 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.

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 and their related information (Extension Status, Agent, Name, Answered, Abandoned, Login/Logout Time, Pause/Resume Time, Talk Time, Agent Status, Pause Reason, and Options). The Chairman is also able to log out dynamic agents from call queues.
- **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.

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

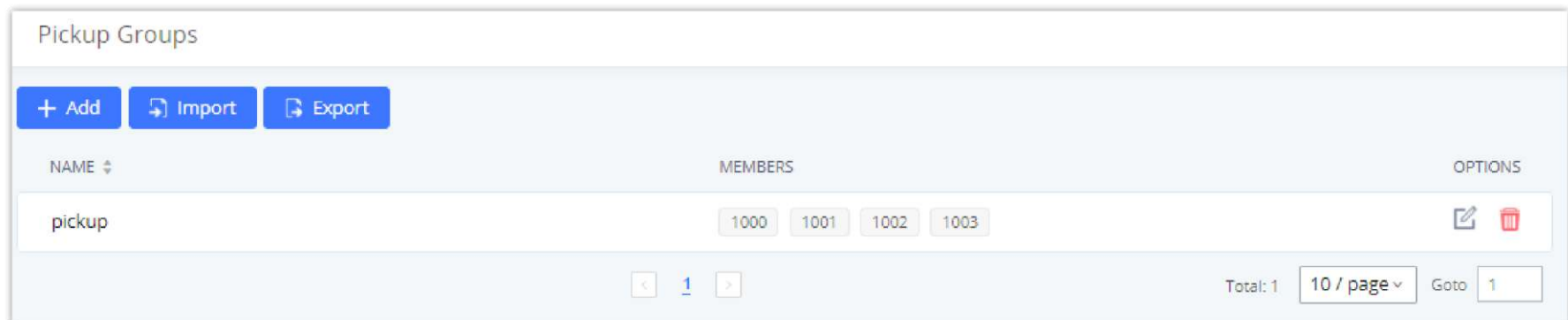
Virtual Queue Callback Key Settings	
Enable	Select whether to enable or disable virtual queue callback feature. By default it's disabled.
Call Back Current Number	Press the feature key configured to set your current number as callback number.
Custom Callback Number	Press these feature keys configured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

Global Queue Settings

Pickup Groups

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

Configure Pickup Groups

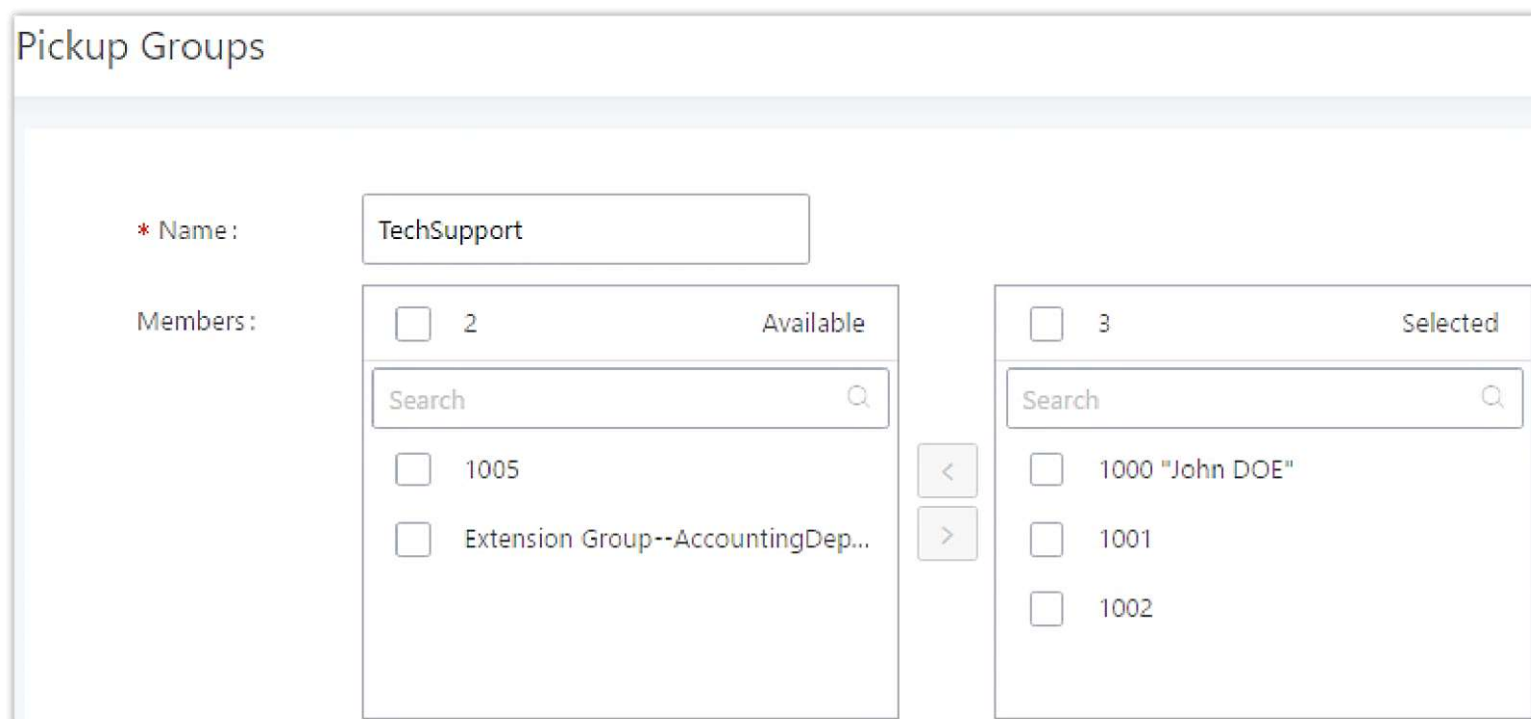


Pickup Groups interface

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

- Click on **+ Add** to create a new pickup group.
- Click the **Import** button to upload the pickup group information in CSV format.
- Click the **Export** button to generate pickup group information in .CSV format.
- 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.



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 to perform the call pickup operation. The following figure shows where you can customize these feature codes.

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

Edit Pickup Feature Code

Dial By Name

Dial by Name is a feature on the PBX that allows the 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 the 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: 30 items Available 0 item Selected

LDAP Phonebook: 1 item Available 0 item Selected

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: Last Name:

Email Address: * User Password:

* Language: * Concurrent Registration...:

Mobile Phone Number:

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.

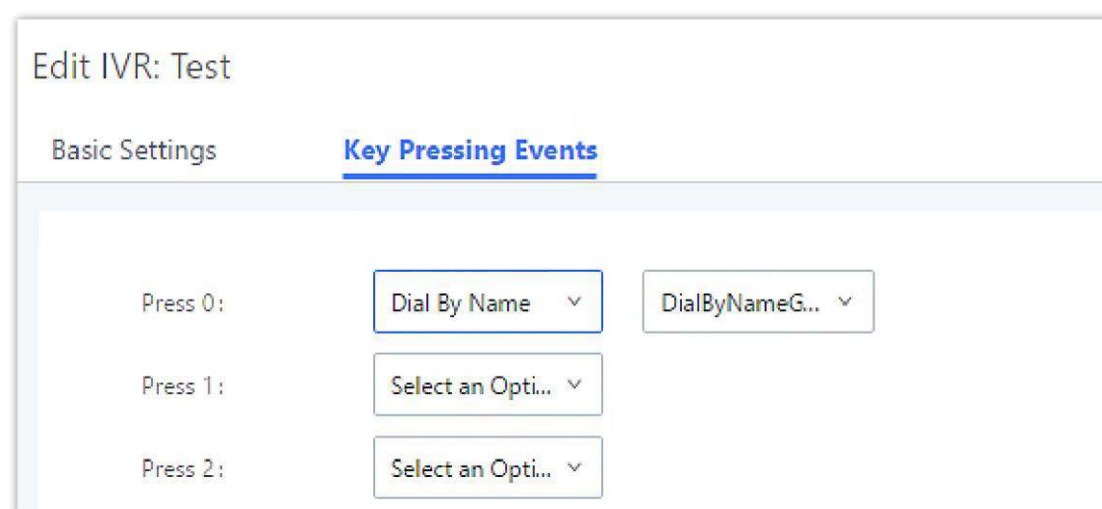
7. Select Type

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.



The screenshot shows a web interface for configuring an IVR. The title is "Edit IVR: Test". There are two tabs: "Basic Settings" and "Key Pressing Events", with the latter being active. Below the tabs, there are three rows of configuration for key presses:

Key Press	Event	Destination
Press 0:	Dial By Name	DialByNameG...
Press 1:	Select an Opti...	
Press 2:	Select an Opti...	

Dial By Name Group In IVR Key Pressing Events

Edit Inbound Rule
Save

<p>* Pattern: <input style="width: 90%;" type="text" value="."/></p> <p>Disable This Route: <input type="checkbox"/></p> <p>Prepend User Defined Nam... <input type="checkbox"/> <input style="width: 80%;" type="text"/></p> <p>Alert-info: None ▼</p> <p>Privilege Level: Internal ▼</p> <p>Allowed to seamless transfe... <input style="width: 90%;" type="text"/></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: 90%;" type="text"/></p>
---	--

Default Mode

* Default Destination: Dial By Name ▼ DialByNameGP1 ▼

Dial by Name Group In Inbound Rule

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

Speed Dial

Add Speed Dial

The CloudUCM 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 CloudUCM.

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

User should first click on + Add . 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, conference room, voicemail group, IVR, ring group, call queue, page group, DISA, Dial by Name and external number.

i **Note**

The maximum number of speed dial entries that can be configured is 1000 speed dial entries.

Create New Speed Dial

Enable

Destination:

* Speed Dial

Extension:

Default Extension ▼ 1007 ▼

Destination:

Speed Dial Destinations

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

 Goto

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

 Goto

Import Dial Speed

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.

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				
+ Add ↶ Import ↷ Export				
SPEED DIAL EXTENSION ↕	SPEED DIAL	DEFAULT DESTINATION	DEFAULT DESTINATION	OPTIONS
123	Enable	Extension	1000	✎ 🗑
999	Enable	Voicemail		✎ 🗑

[<](#)
[1](#)
[>](#)
 Total: 2

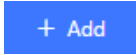


 Goto

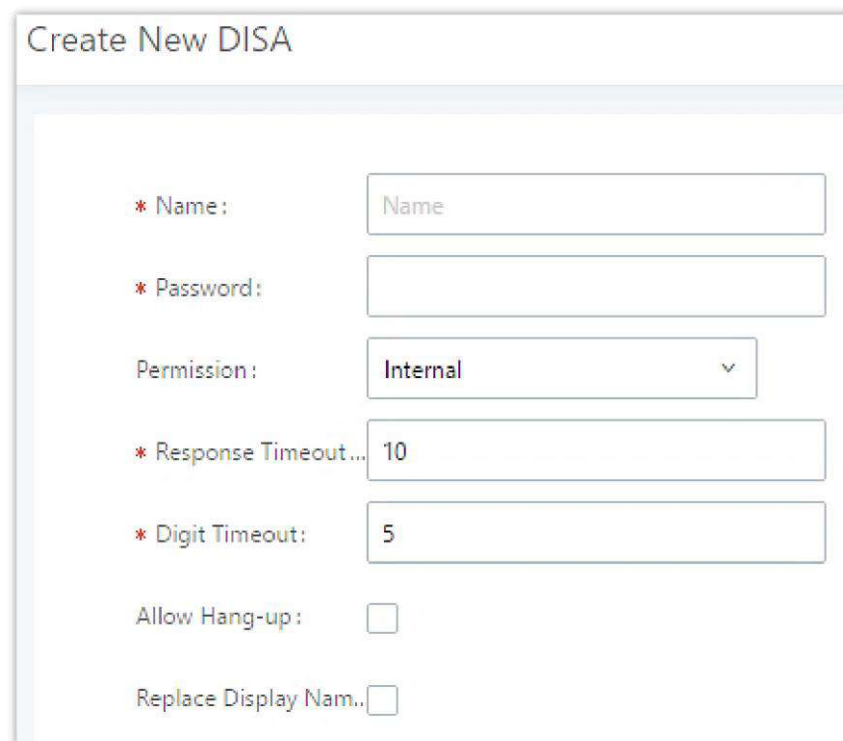
Export Dial Speed

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, etc. After calling into the CloudUCM, the user can then dial out via the SIP trunk connected to the CloudUCM as it is an internal extension.

The CloudUCM 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.



The screenshot shows a 'Create New DISA' form with the following fields and values:

- Name:** Name
- Password:** (empty)
- Permission:** Internal
- Response Timeout:** 10
- Digit Timeout:** 5
- Allow Hang-up:**
- Replace Display Name:**

Create New DISA

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

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

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

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 CloudUCM.
2. On the CloudUCM, configure destination of the inbound route for callback.
3. Save and apply the settings.
4. The user calls number of the CloudUCM 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 CloudUCM 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 CloudUCM 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 CloudUCM, go to Web GUI → **Call Features** → **Callback** page and click on [+ Create New Callback](#). Configuration parameters are listed in the following table.

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 CloudUCM trunk.

Event List

Besides BLF, users can also configure the phones to monitor event list. In this way, both local extensions on the same CloudUCM 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 CloudUCM. The valid characters are letters, digits, _ and -.
Local Extensions	Select the available extensions/Extension Groups listed on the local CloudUCM to be monitored in the event list.
Remote Extensions	If LDAP sync is enabled between the CloudUCM and the peer CloudUCM, 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

The screenshot shows the 'Create New Event List' configuration interface. 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", 1004, 1005 "Will", 1006 "Iala", 1007 "Kiki") and a list of 3 selected extensions (1000 "Mia", 1001 "John", 1002 "Chris").
- Remote Extensions:** Two empty lists for available and selected remote extensions.
- Special Extensions:** An empty text area for manually entering remote extensions.

Create New Event List

Remote extension monitoring works on the UCM via event list BLF, among Peer SIP trunks or Register SIP trunks (register to each other). Therefore, please properly configure SIP trunks on the UCM 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 UCM and remote extensions are added to the list, the UCM will send out SIP SUBSCRIBE to the remote UCM to obtain the remote extension status. When the SIP end points register and subscribe to the local UCM 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.



- To configure LDAP sync, please go to CloudUCM Web GUI→Extension/Trunk→VoIP Trunk. You will see "Sync LDAP Enable" option. Once enabled, please configure password information for the remote peer UCM to connect to the local UCM. Additional information such as port number, LDAP outbound rule, LDAP Dialed Prefix will also be required. Both PBXs need enable LDAP sync option with the same password for successful connection and synchronization.
- Currently LDAP sync feature only works between two UCM.

- (Theoretically) Remote BLF monitoring will work when the remote PBX being monitored is non-UCM PBX. However, it might not work the other way around depending on whether the non-UCM PBX supports event list BLF or remote monitoring feature.

Feature Codes

Feature Maps	
Blind Transfer	<ul style="list-style-type: none"> - Default code: #1 - 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. - Options: <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	<ul style="list-style-type: none"> - Default code: *2 - 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. - Options: <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	<ul style="list-style-type: none"> - Default code: *0 - Enter the code during active call. It will disconnect the call. - Options: <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	<ul style="list-style-type: none"> - Default code: #72 - Enter the code during active call to park the call. - Options: <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 PBX 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	<p>- Default Code: *90</p> <p>- Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.</p>
Call Forward Busy Deactivate	Default Code: *91
Call Forward No Answer Activate	<p>- Default Code: *92</p> <p>- Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.</p>
Call Forward No Answer Deactivate	Default Code: *93
Call Forward Unconditional Activate	<p>- Default Code: *72</p> <p>- Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.</p>
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 Enable	Configures and enables CFU for any extension.
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	- 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	Picks up a ringing call for another extension. 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.
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 dial the appropriate key for the desired room status. Example: The housekeeper with the 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: The 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 CloudUCM also allows user to one click enable / disable specific feature code as shown below:

Enable/Disable Feature codes

Fax/T.38

The CloudUCM supports T.38 Fax 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 on the same web page for users to view, retrieve and delete.

Create New Fax Extension

- 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 CloudUCM. Two modes "Only" and "More" are supported.</p> <ul style="list-style-type: none"> ○ Only <p>Under this mode, the CloudUCM allows only a single user to send a fax at a time.</p> <ul style="list-style-type: none"> ○ More <p>Under this mode, the CloudUCM supports multiple concurrent faxes 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 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 the 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 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. The template variables are:</p> <ul style="list-style-type: none"> ○ \${CALLERIDNUM} : Caller ID Number ○ \${CALLERIDNAME} : Caller ID Name ○ \${RECEIVEEXTEN} : The extension to receive the Fax ○ \${FAXPAGES} : Number of pages in the Fax ○ \${VM_DATE} : The date and time when the Fax is received. (Format: MM/dd/yyyy hh:mm:ss)
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	<p>Enables the fax resend option which allows the UCM to keep attempting to send faxes up to a specified amount of times. Additionally, if fax still fails to send, a <i>Resend</i> button will appear in the File Send Progress list in <i>Call Features > Fax Sending</i> to allow manual resending.</p>
Max Resend Attempts	<p>Configures the number of the maximum attempts to resend the fax.</p> <p>The default value is set to 5.</p>
Fax Resend Frequency	<p>Configures the Fax Resend Frequency.</p> <p>The default value is set to 50.</p>

FAX/T.38 Settings

Parking Lot

User can create parking lots and their related slots under **Call Features > Parking Lot**. In the Parking Lot page, users can create lots of their own. This allows different groups within an organization to have their own parking lots instead of sharing one large parking lot with others. While creating a new parking lot, users can assign it a range that they think is appropriate for the group that will use the parking lot.

Parking Lot			
Parking Lot Settings		Parking Lot Status	
+ Add			
EXTENSION	NAME	SLOTS	OPTIONS
700	DefaultLot	701-720	
7000	Dales	7001-7022	

Total: 2 10 / page Goto 1

Parking Lot

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

Create New Parking Lot
Cancel Save

<p>* Parking Lot Extension: <input type="text"/></p> <p>* Parking Slots: <input type="text"/></p> <p>* Parking Timeout (s): <input type="text" value="300"/></p> <p>Failover Destination: <input type="text"/></p> <p>Forward to Destination on Timeout: <input type="checkbox"/></p>	<p>* Parking Lot Name: <input type="text"/></p> <p>Use parklot as extension: <input type="checkbox"/></p> <p>Music on Hold Playlists: <input type="text" value="Default"/></p> <p>Ring-All Callback on Timeout: <input type="checkbox"/></p>
---	--

New Parking Lot

Parking Lot Extension	<ul style="list-style-type: none"> ○ 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.

Parking Lot

Call Park

The CloudUCM 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.



Monitored Call Park CID name

Emergency Calls

Emergency Calls

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.

The CloudUCM is fully compliant 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**"

To configure emergency numbers, users need to follow below steps:

1. Navigate on the web GUI under "**Call Features → Emergency Calls**"
2. Click on **+ Add** 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.

The screenshot shows a web form titled "Create New Emergency Call". The form contains the following fields and options:

- Name:** Text input field containing "911".
- Emergency Number:** Text input field containing "911".
- Emergency Level:** Dropdown menu with "1 - Not Urgent" selected.
- Disable Hunt on Busy:** Unchecked checkbox.
- Custom Prompt:** Dropdown menu with "None" selected, and a "Prompt" button to the right.
- Use Trunks:** Empty text input field.
- Members Notified:** Two selection panes. The left pane is titled "11 items Available" and contains a search bar and a list of items: "1001 'John Doe'", "1002", "1003", and "1004". The right pane is titled "1 item Selected" and contains a search bar and one item: "1000 'James tuan'".
- Strip:** Empty text input field.
- Prepend:** Empty text input field.
- Auto Record:** Checked checkbox.
- Send Recording File:** Checked checkbox.
- Email Address:** Text input field containing "admin@domain.local" and a "+" button to the right.

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. Note: 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 an 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 prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
Auto Record	If enabled, the calls using this extension or trunk will be automatically recorded, default is disabled Note: Emergency recording will be also available under Emergency Recording Tab.
Send Recording File	If Enabled the recordings will be sent to the configured Email address(es)
Email Address	Used to specify the Email address(es) for Emergency calls recordings.

Emergency Calls

[Emergency Calls](#) [Emergency Recordings](#) [Emergency Location Mapping](#)

[+ Add](#)

NAME	EMERGENCY NUMBER	EMERGENCY LEVEL	DISABLE HUNT ON BUSY	OPTIONS
911	911	1	No	Edit Delete

Total: 1

911 Emergency Sample

Emergency Recordings

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

Emergency Calls

[Emergency Calls](#) [Emergency Recordings](#) [Emergency Location Mapping](#)

[Download](#) [Download All](#) [Delete](#) [Clear](#)

NAME	EMERGENCY CALLS	CALLER NUMBER	DATE	SIZE	OPTIONS
No Data					

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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 exact location of callers. The new options can be found under **Call Features** → **Emergency Calls** → **Emergency Location Mapping**.

Create New Emergency Location Mapping

Cancel Save

• **ELIN:**

• **Subnet:**

• **Location:**

Geolocation Routing:

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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.

Scheduled Call

Call scheduler feature allows the user to schedule a wakeup call to a specific extension. The user can choose the time and date of the call, and when the time arrives, a call will be initiated to designated extension(s). When the call is answered, the chosen prompt will be played.

Scheduled Call > Create New Scheduled Call

Enable Scheduled Call

* Name

Prompt

Custom Date

* Date

* Time

* Members

<input type="checkbox"/> 5 Available	<input type="checkbox"/> 0 Selected
<input type="text" value="Search"/>	<input type="text" value="Search"/>
<input type="checkbox"/> 1000	None
<input type="checkbox"/> 1001	
<input type="checkbox"/> 1002	
<input type="checkbox"/> 1003	
<input type="checkbox"/> 1004	

Scheduled Call

Fax Sending

The CloudUCM supports sending Fax via Web GUI . This feature can be found on **Call Features > Fax Sending** page. The user can enter the number of the destination, then the fax will be sent to a fax gateway on the receiving end.

After entering the fax number, please upload the pdf file that you wish to send as a fax.

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:

[Send](#)

File Send Progress

[Delete](#)
[Clear](#)

	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

! Only A3, A4, and B4 paper sizes are supported for the Fax Sending.

Announcement Center

The CloudUCM supports Announcement Center feature which allows users to pre-record and store voice message into the CloudUCM 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.

UCM6301

Security level of current username or password is too low. Click here to change them. If you have forgotten your password, please enter an email address so that a password reset email may be sent.

admin

- System Status
- Extension/Trunk
- Call Features
- PBX Settings
- System Settings
- Contacts
- Device Management
- Maintenance
- CDR
- RemoteConnect
- Other Features
 - Zero Config
 - API Configuration
 - AMI
 - CTI Server
 - CRM
 - PMS
 - Wakeup Service
 - Fax Sending
 - Announcement Cen...
 - QueueMetrics

Announcement Center

[+ Add Announcement Center](#)

CODE	NAME	OPTIONS
55	Announcement_1	✎ 🗑

1 / 1 Total: 1 10 / page Goto 1

[+ Add Group](#)

NUMBER	NAME	MEMBERS	OPTIONS
33	Group_1	1001 1002 1005	✎ 🗑

1 / 1 Total: 1 10 / page Goto 1

Announcement Center

Announcement Center Settings

Announcement Center > Create New Announcement Center

* Name

* Code

* Custom Prompt

* Ring Timeout (s)

* Auto Answer

Announce Message Caller-ID

Create New Announcement Center

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

Announcement 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.

The screenshot shows a 'Create New Group' dialog box. At the top, the title is 'Create New Group'. Below the title, there are three main sections:

- Name:** A text input field containing 'GStest'.
- Number:** A text input field containing '666'.
- Members:** This section is divided into two panes:
 - Available:** Shows a list of 27 items. A search bar is at the top. The visible items are 1003, 1004, 1005, 1006, and 1007, each with an unchecked checkbox.
 - Selected:** Shows a list of 3 items. A search bar is at the top. The visible items are 1000, 1001, and 1002, each with an unchecked checkbox.

Announcement Center Group Configuration

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.

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

5. Click [+ Add Announcement Center](#) to create a new Announcement Center.
6. Give a name to the newly created Announcement Center.
7. Specify the code which will be used with group number to send the voice message to.
8. 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.

Announcement Center > Create New Announcement Center

* Name

* Code

* Custom Prompt

* Ring Timeout (s)

* Auto Answer

Announce Message Caller-ID

Announcement Center Code Configuration

Code and Group number are used together to direct specified message to the target group. All extensions in the group will receive the message. For example, we can send code 55 to group 666 by dialing 55666 from any extension registered to the CloudUCM. 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				
<input type="button" value="+ Add Announcement Center"/>				
CODE	NAME	OPTIONS		
55	GStest	<input type="button" value="Edit"/> <input type="button" value="Delete"/>		
1		Total: 1	10 / page	Goto 1
<input type="button" value="+ Add Group"/>				
NUMBER	NAME	MEMBERS	OPTIONS	
666	GStest	1000 1001 1002	<input type="button" value="Edit"/> <input type="button" value="Delete"/>	
1		Total: 1	10 / page	Goto 1

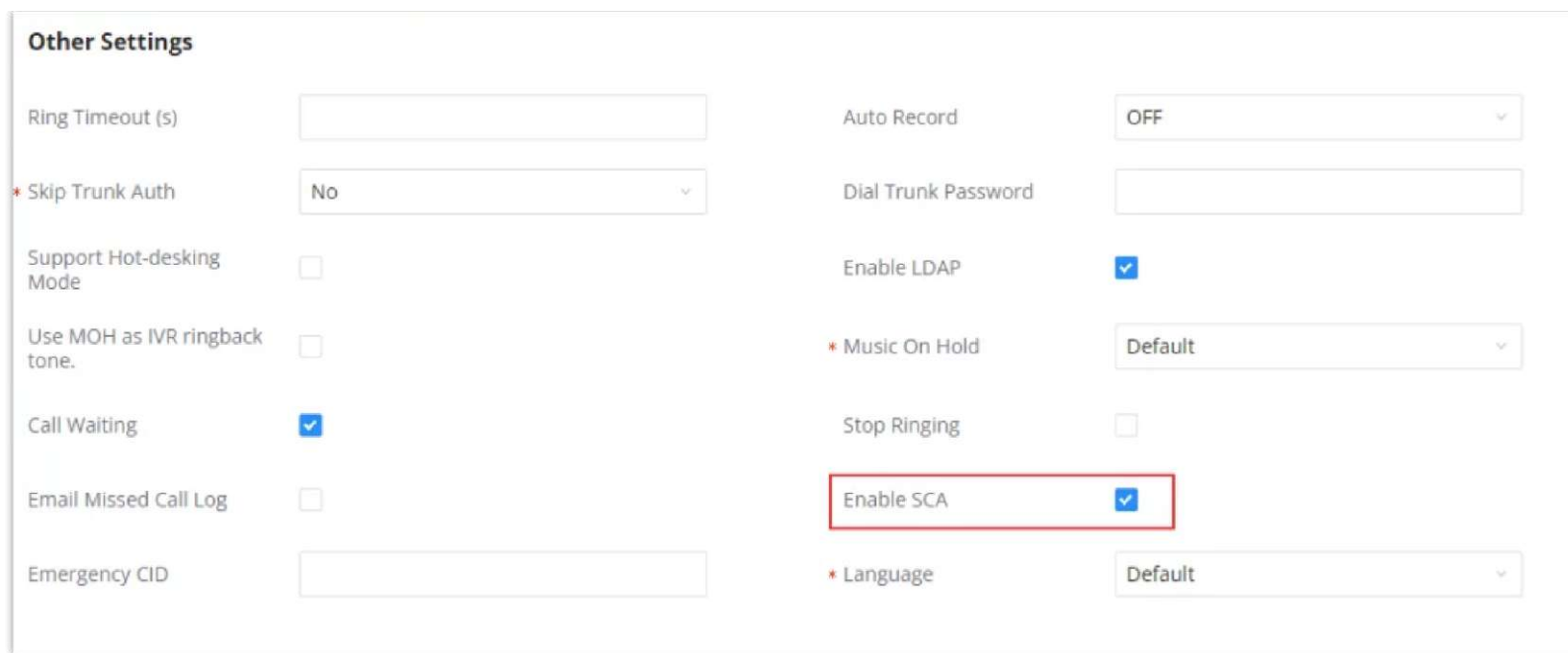
Announcement Center Example

Shared Call Appearance (SCA)

Shared Call Appearance (SCA) functionality has been added to the CloudUCM. 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:

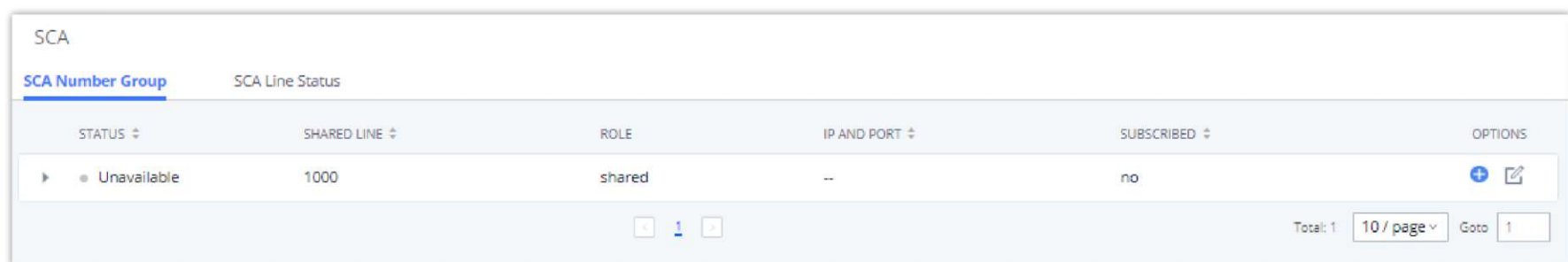
1. Users can enable SCA by navigating to the Extensions page, editing the desired extension, and enabling the option SCA under the "Feature" tab, and in "Other Settings" category.

! With SCA enabled, the Concurrent Registrations field can only have a value of 1.



Enabling SCA option under Extension > Features

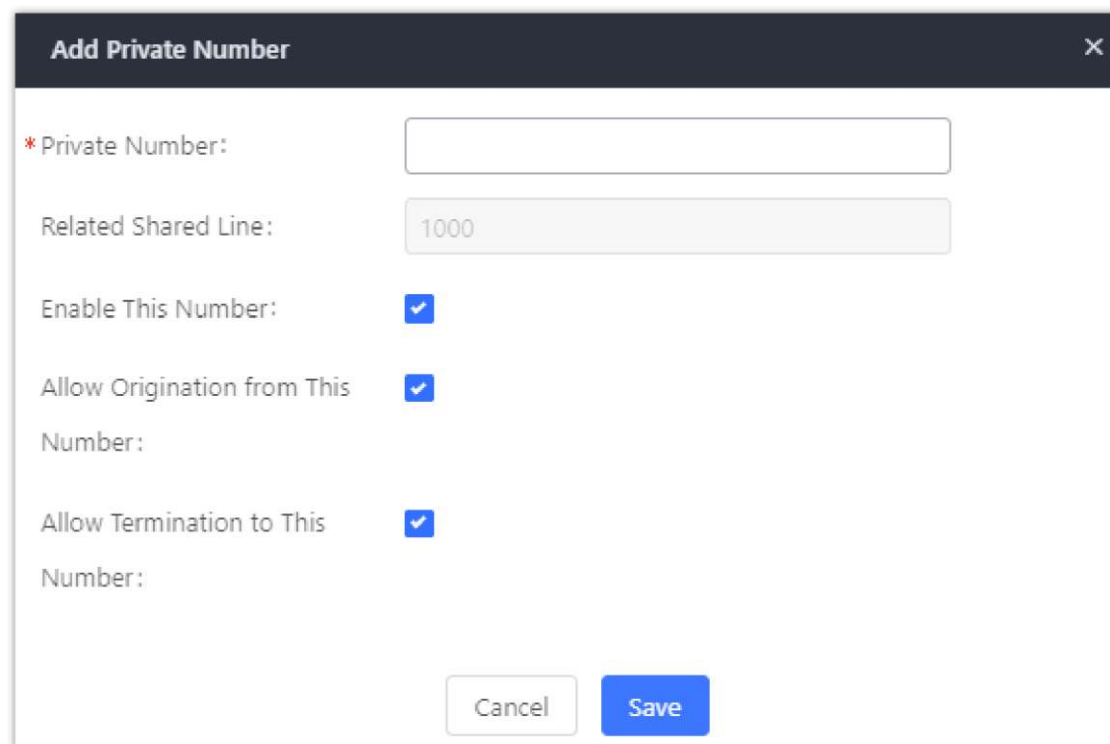
2. After enabling the option, navigate to *Call Features* → *SCA*. The newly enabled SCA extension will be listed. Click the “+” button under the Options column to add a number that will share the main extension’s call appearance, which will be called private numbers.



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

SCA Number Configuration

3. Configure the private number as desired.



SCA Private Number Configuration

4. Once the private number has been created, users must now register a device to it. To properly register a device to the private number, use the configured private number as the SIP User ID. Auth ID and Password will be the same as the main extensions. Once registration is complete, SCA is now configured.

Edit SCA Number Group:

* Shared Line Number:

Allow Call Retrieve from

Another Location:

Alert All Appearances for

Group Paging Calls:

Multiple Call Arrangement:

Allow Bridging between

Locations:

Bridge Warning Tone:

SCA Options

5. 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:

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.

Add SCA Private Number

The following table describes the options available when 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	<p>Configures the notification in the bridge when another party join.</p> <ul style="list-style-type: none"> ○ 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. <p>By default, it is set to "Barge-In Only".</p>
----------------------------	---

Editing the SCA Number

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	Audio file that needs to be uploaded in order to be played for a specific destination.
Default Destination	Select the destination where to play the audio file.

Announcement Parameters

MESSAGING

IM Settings

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

IM Settings

Read Receipts

Notify Inactive Users of New Messages [Email Template](#)

* Max File Size Upload (MB)

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.
New Message Email Notification	<p>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.</p>
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

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

Create New Live Chat

Name	Enter the name of the Live Chat
Destination	Configure the destination of this Live Chat.
Web Page Language	Configure the language of visitor page.
Visitor Information	
Require Visitor Info	Configure the visitor information required to start a live chat session.
Privacy Control	If enabled, visitors must consent to allow the processing of personal data and cookies before entering Live Chat.
Call Settings	
Allow Visitor To Call	If enabled, visitors will have the option to call the configured destination from this Live Chat.
Customer Service Agent Information	
Avatar	Upload the avatar of the agent.

	Please select a png, jpg, or jpeg format file.
Name	Enter the name of the agent.
Show Agent's Real Name	Enable this option to show the agent's real name in the live chat.
Chat Settings	
Welcome Message	Configure the initial message to display to visitors when they first enter the live chat session.
Reply to First Message	Configure the message to send to visitors in response to their first message.
Visitor Chat Log Retention Time (days)	This value determines how long a visitor's chat history will be kept before it's deleted automatically. Note:
Live Chat Link Address	This link can be embedded onto web pages. Clicking it will connect visitors to the configured Live Chat destination. This link can also be entered directly into the browser address bar for testing purposes.

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

Company Contact

All

Technical Support

Quality Control

Sales

IT

Marketing

Human Resources

Finance

1000 " "

Selected(0)

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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.

SMS Settings

SMS Settings

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

SMS Settings

Enable SMS	Tick this box to enable SMS service.
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 Templates

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

PBX SETTINGS

General Settings

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 CloudUCM 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 CloudUCM. 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. ● 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.

Default Extension Segment

Clicking this button will reset the extension range to their default values.

SIP Settings

General

Realm For Digest Authentication	Configure this as the host name or domain name of the PBX. Realms MUST be globally unique according to RFC3261.
MWI From Header	If disabled, the server will not transfer the Diversion Header
Enable Diversion Header	If set to "No", all transfers initiated by the endpoint in the CloudUCM will be disabled (unless enabled in peers or users). The default setting is "Yes".
Send Deflection Diversion	If 'Enable Diversion Header' and this option enabled, the INVITE request will contain Diversion with reason 'deflection' while the inbound call been routed to an external number.
Block Collect Calls	If enabled, collect calls will be blocked. Note: Collect calls are indicated by the header "P-Asserted-Service-Info: service-code=Backward Collect Call, P-Asserted-Service-Info: service-code=Collect Call".
Unavailable Extension Cause	Sets the cause code to respond with when calling an unavailable extension.

Session Timer

Force Timer	Always request and run session timer. By default, this option disabled.
Timer	Run session timer only when requested by other UA.
Session Expiration (s)	Configure the session refresh interval (in seconds). The default setting is 600 seconds.
Min SE (s)	Configure the minimum session refresh interval (in seconds). The default setting is 90 seconds.

Transport Protocol

Choose the supported transport protocol used for SIP packets.

TLS Enable	Allow incoming TLS (secure) connections with the PBX. Note: This option cannot be disabled.
SIP Server (TLS)	Use the following server address on the endpoints to for SIP packet transport over TLS.
TCP Enable	Allow incoming TCP connections with the PBX. Note: This option is disabled by default.
SIP Server (TCP)	Use the following server address on the endpoints to for SIP packet transport over TCP.

Enable UDP	Allow incoming TCP connections with the PBX. Note: This option is disabled by default.
SIP Server (UDP)	Use the following server address on the endpoints to for SIP packet transport over UDP.

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	Configure the default duration (in seconds) of incoming/outgoing registration. The default setting is 120.
Max Registration/Subscription Time	Configure the maximum duration (in seconds) of incoming registration and subscription allowed by the UCM. The default setting is 3600.
Min Registration/Subscription Time	Configure the minimum duration (in seconds) of incoming registration and subscription allowed by the UCM. 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	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. Note: This setting does not apply to calls on hold.
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	This feature can be used to avoid abnormal call drop when the remote provider requires RTP traffic during proceeding. For example, when the call goes into voicemail and there is no RTP traffic sent out from UCM, configuring this option can avoid voicemail drop. When configured, RTP keep-alive packet will be sent to remote party at the configured interval. If set to 0, RTP keep-alive is disabled.
100rel	Configure the 100rel setting on UCM. 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 CloudUCM should generate Inband ringing or not. The default setting is "Never".</p> <ul style="list-style-type: none"> ○ Yes: The CloudUCM will send 180 Ringing followed by 183 Session Progress and in-band audio. ○ No: The CloudUCM 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 UCM 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 CloudUCM.
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/ToS

STIR/SHAKEN

To prevent robocalls, UCM now supports STIR/SHAKE protocols. Related options have been added as a new tab in the **SIP Settings** page.

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

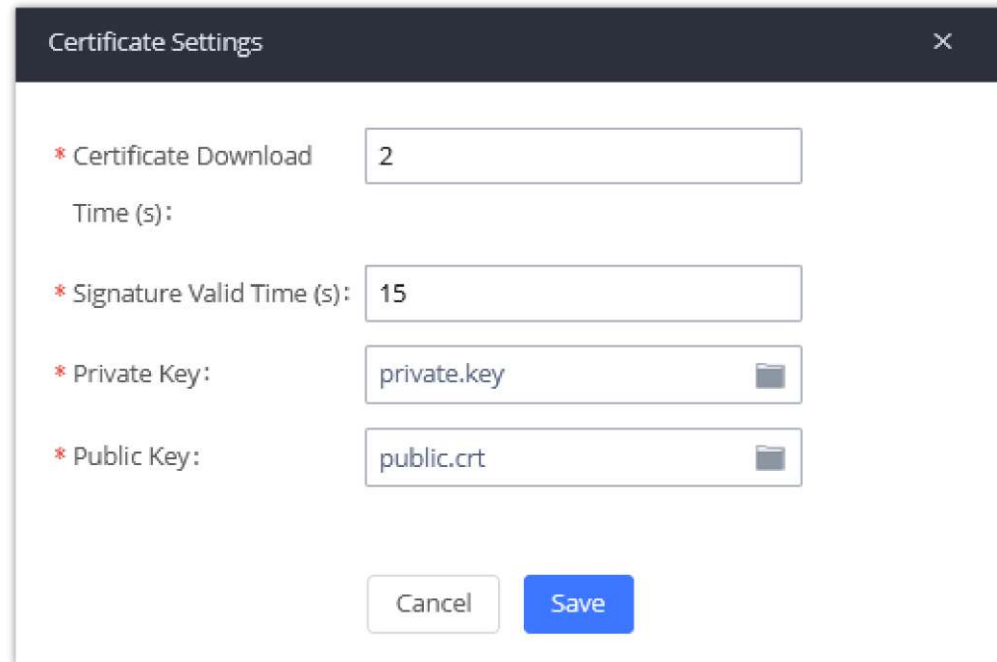
STIR/SHAKEN

STIR/SHAKEN – Add Authorized CID and Attestation level

Authorized CID	Configure the authorized CID.
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Attestation	<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.
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Clicking on the **Certificate Settings** button will bring up the following window:



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	Configure the Private key. 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".
Public Key	Configure the Public Key. 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".

SIP Settings/STIR/SHAKEN – Certificate 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 CloudUCM gives up. The default setting is 0, which means the CloudUCM 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	
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 CloudUCM 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.
Transfer	
Allow Transfer	Configure to allow transfer. When disallowed, all transfers initiated by the endpoint digit will be disabled.
Allow callback to transferer when blind transfer fails	If enabled, the UCM will call back the transferer when a blind transfer fails (i.e., no answer or busy).
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.

RTP Settings

RTP Settings

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.

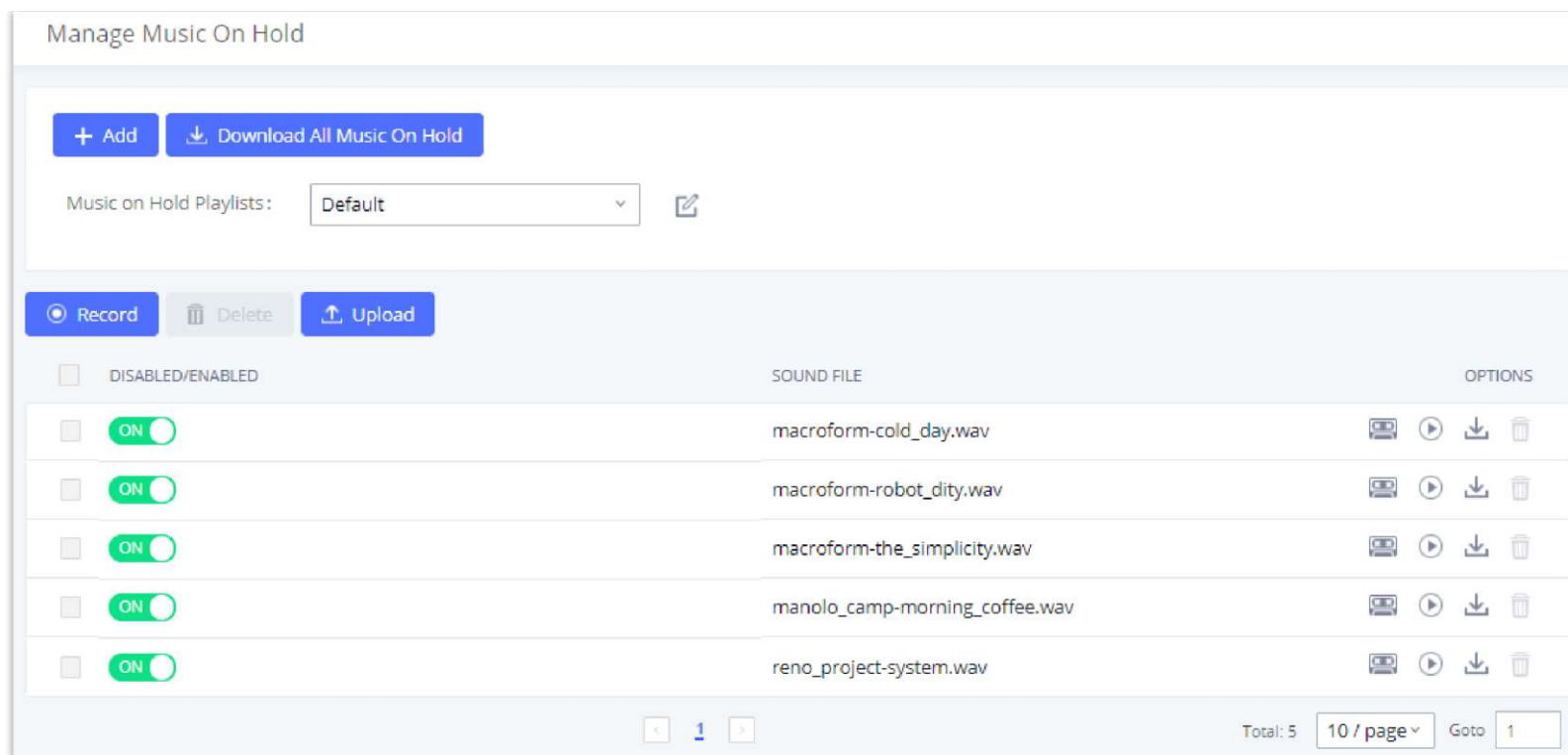
Payload Type Settings

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








Audio Codecs	
AAL2-G.726	ADPCM (G.726, 32kbps, AAL2 codeword packing).
DTMF	Dual-tone Multi-frequency.
G.721 Compatible	G.721 Compatible
G.726	ADPCM (G.726, 32kbps, RFC3551 codeword packing).
ILBC	ILBC Free Compression.
Opus	Opus
G.722.1	G.722.1: Low-complexity coding, 24kbps
G.722.1C	G.722.1C: Low-complexity coding, 48kbps
Audio FEC Payload Type	Audio FEC Payload Type
Audio RED Payload Type	Audio RED Payload Type
Video Coding	
H.264	H.264 Video.
H.265	H.265 Video.
H.263P	H.263+ Video.
VP8	VP8 Video.
Other Settings	
Main Video FEC	Main Video FEC.
RTP FECC	RTP FECC
RTX	Used for packet retransmission. UCM supports only video RTP retransmission.

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.





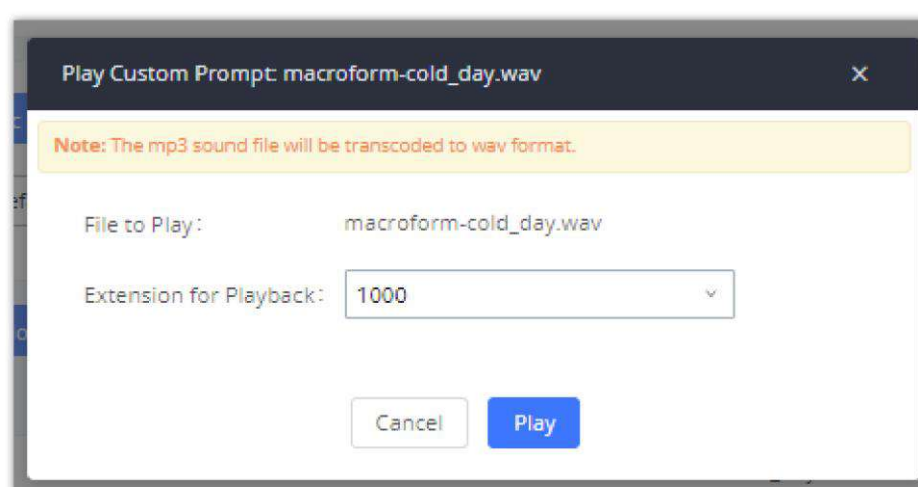
Music On Hold Default Class

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

The CloudUCM allows Users to select the Music On Hold file from WebGUI to play it. The CloudUCM 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 .

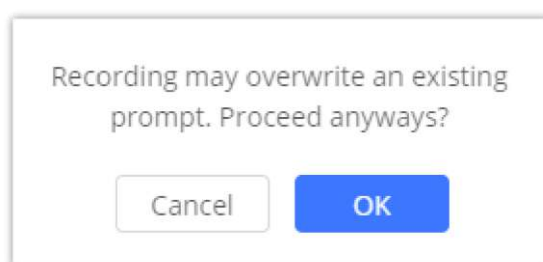


Play Custom Prompt



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

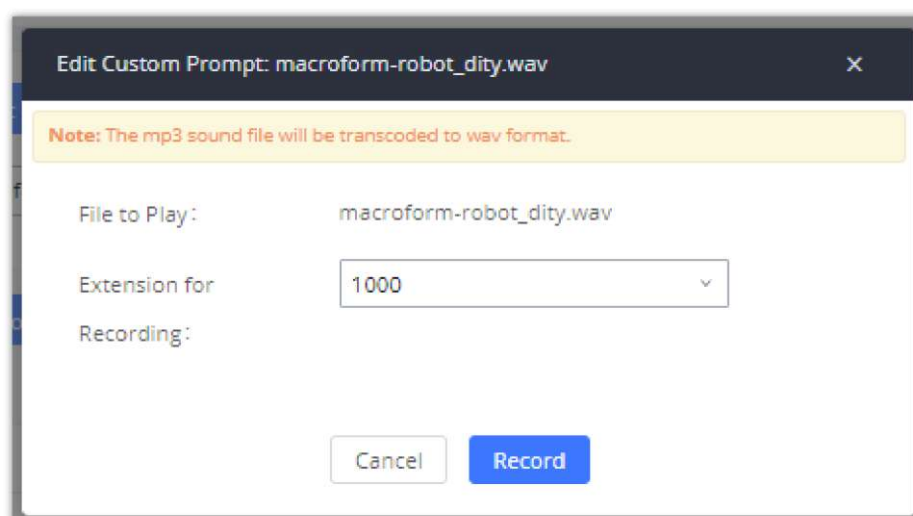
Users could also record their own Music On Hold to override an existing custom prompt, this can be done by following those steps:

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



Information Prompt

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



Record Custom Prompt

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



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.

Voice Prompt

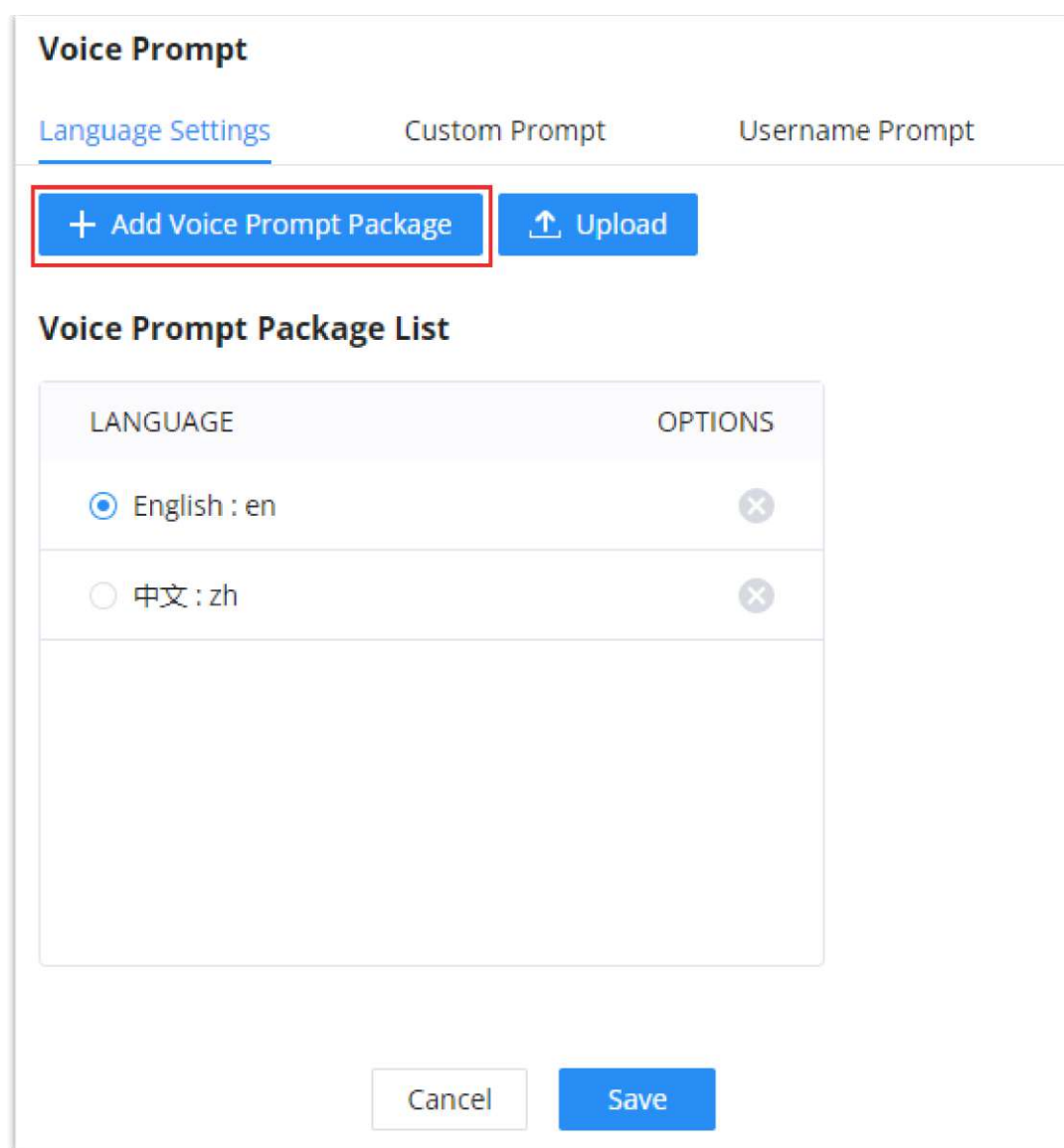
The CloudUCM supports multiple languages in Web GUI as well as system voice prompt. Currently, there are 16 languages supported in system voice prompt: **English (United States), Arabic, Chinese, Dutch, English (United Kingdom), French, German, Greek, Hebrew, Italian, Polish, Portuguese, Russian, Spanish,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

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

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 CloudUCM Web GUI, click on "Add Voice Prompt Package" button.



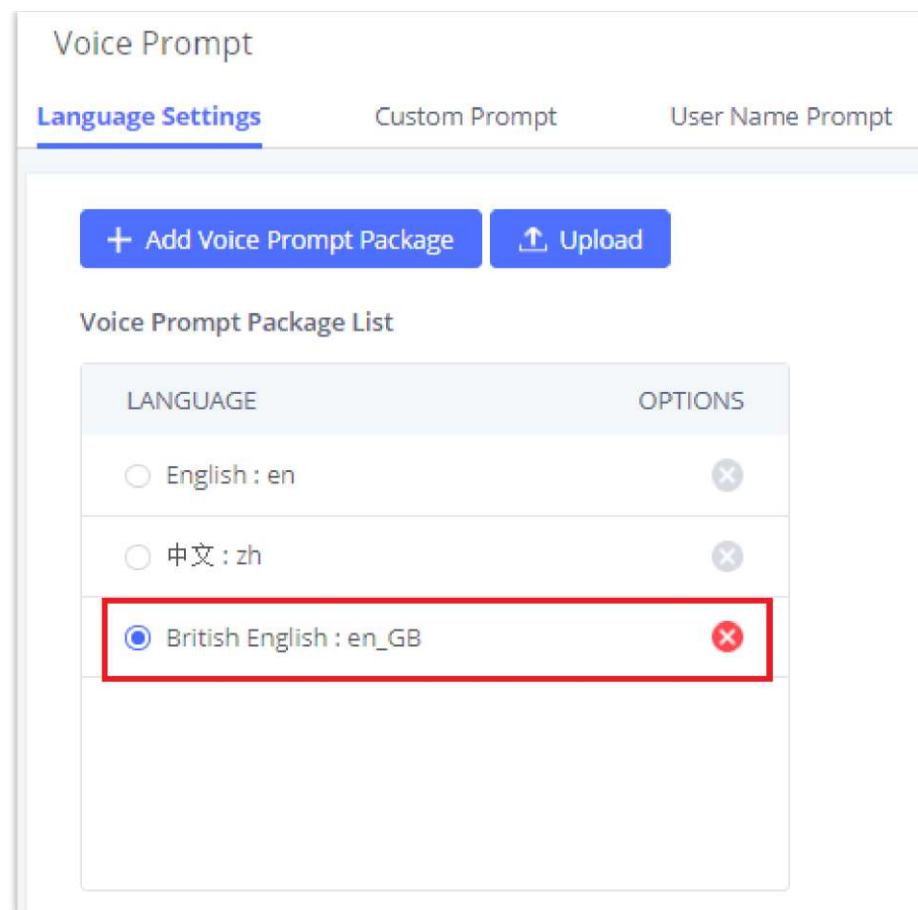
Language Settings for Voice Prompt

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

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

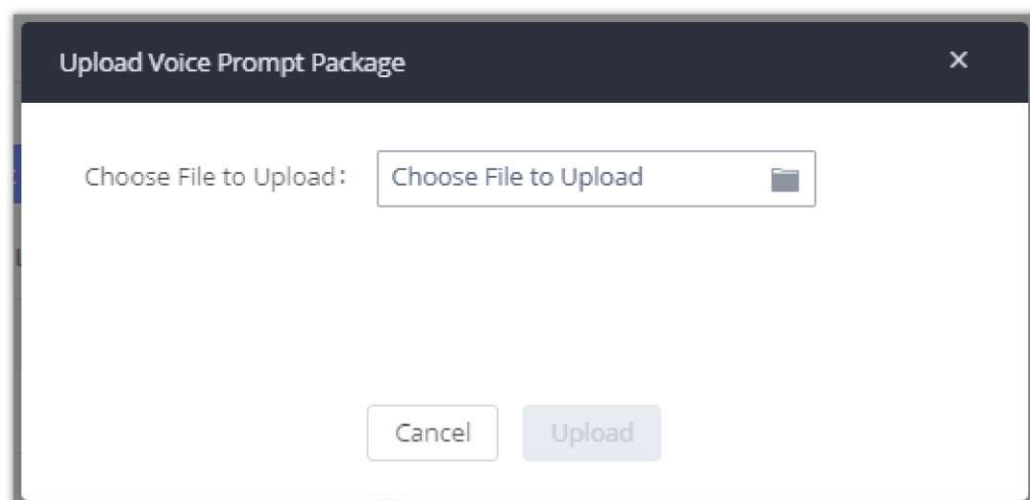


New Voice Prompt Language Added

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

Custom Prompt

On the CloudUCM, 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

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

Upload Username Prompt File from Web GUI

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

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

Record Username via Voicemail Menu

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

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

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

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

Call Prompt Tones

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-wrang-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 Prompt Tones → 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 Prompt Tones

Alert-Info Prompt

The user can change upload the ringtone files which can be used with Alert-info to play various ringtones depending on the configured behaviour. In this setting, the user can view all the ringtones uploaded, play them, download them, or delete them. If the user wishes to upload new ringtones, he/she can click on "Upload" and choose the ringtone files to upload.

Alert-info Prompt

Uploaded greeting files cannot have the same name even if the file format is different.

<input type="checkbox"/>	Name ↕	Type ↕	Format ↕	Options
<input type="checkbox"/>	Ring-IVR.wav	Audio	wav	<input type="button" value="Play"/> <input type="button" value="Download"/> <input type="button" value="Delete"/>
<input type="checkbox"/>	Ring-Queue.wav	Audio	wav	<input type="button" value="Play"/> <input type="button" value="Download"/> <input type="button" value="Delete"/>
<input type="checkbox"/>	Ring-Extension.wav	Audio	wav	<input type="button" value="Play"/> <input type="button" value="Download"/> <input type="button" value="Delete"/>
<input type="checkbox"/>	Ring-External.wav	Audio	wav	<input type="button" value="Play"/> <input type="button" value="Download"/> <input type="button" value="Delete"/>

Total: 4 10 / page

Alert-info Prompt

SYSTEM SETTINGS

This section will explain the available system-wide parameters and configuration options on the CloudUCM series. This includes settings for the following items: General Settings, HTTP server, Security Settings, LDAP Server, Time Settings, and Email Settings.

General Settings

On general settings, the user can configure the **Device Name** and **Enterprise Contact Number**.

General Settings

Device Name

Enterprise Contact Number

[Add](#)

CloudUCM General Settings

Device Name	Enter the device's name.
Enterprise Contact Number	Enter the contact number of the enterprise.

HTTP Server

In this section, the administrator can modify the settings regarding the HTTP server of the CloudUCM. The settings include restricting access to specific public IP addresses.

HTTP Server

UCM Web Settings

Enable IP Address Whitelist

Permitted IP (s) / -

[Add IP Address](#) +

Wave Settings

Cross-origin Address Whitelist

HTTP Server Settings

Enable IP Address Whitelist	IP Whitelist restricts all IP addresses except for those in the whitelist from accessing the device via HTTP(S) (i.e., web portal, API).
Permitted IP (s)	Enter a list of IP addresses permitted by the device (e.g., "xxx.xxx.xxx.xxx/255.255.255.255" or "xxxx:xxxx:xxxx:xxxx:xxxx:xxxx:xxxx:xxxx/64").
Cross-origin Address Whitelist	The UCM will accept cross-server requests from addresses in the whitelist, which should be formatted as https://domain, https://ip:port or *. Entering * will allow cross-server requests from all addresses.

Security Settings

Fail2Ban

Fail2ban allows to protect the CloudUCM from brute-force attacks by sending multiple SIP register attempts or login attempts using a dictionary of passwords. By limiting the attempts, the administrator can ensure that the password guessing attempts are limited in a period of time.

Security Settings

[Fail2Ban](#)

Basic Protection

SIP Defense

Login Attack Defense

Customer Service Protection


Customer Service System Call Defense


Live Chat Protection

Global Protection Settings

Banned Duration (s)

Fail2ban Whitelist

Fail2ban Whitelist Description / Comment 

[Add Fail2ban Whitelist](#) 

Blacklist

Banned Type	IP	Time	Operation
<input type="button" value="Cancel"/> <input type="button" value="Save"/>			

Fail2Ban

SIP Defense	If enabled, the UCM will monitor failed SIP registration attempts and heartbeat timeouts based on the UCM's security logs. Once the number of failed attempts from an IP address has reached SIP Defense's configured <i>Max Retries</i> value, the IP address will be banned for the amount of time configured in the <i>Banned Duration</i> field.
Login Attack Defense	Once the number of login attempts from an IP address has reached the <i>Max Retries</i> value configured for this defense, the IP address will be banned for the amount of time configured in the <i>Banned Duration</i> field.
Customer Service System Call Defense	Once the number of call attempts from an IP address has reached the <i>Max Retries</i> value configured for this defense, the IP address will be banned for the amount of time configured in the <i>Banned Duration</i> field.
Live Chat Protection	Once the number of chat attempts from an IP address has reached the <i>Max Retries</i> value configured for this defense, the IP address will be banned for the amount of time configured in the <i>Banned Duration</i> field.
Banned Duration (s)	Configure the duration (in seconds) that an IP address is banned. 0 indicates a permanent ban.
Fail2ban Whitelist	Configure IP address, CIDR mask or DNS host in the whitelist. Fail2ban will not ban the host with matching address in this list.
Blacklist	Users will be able to view the IPs that have been blocked by UCM.

LDAP Server

The CloudUCM has an embedded LDAP/LDAPS server for users to manage the 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 CloudUCM 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 CloudUCM 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 CloudUCM, 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 CloudUCM. If the CloudUCM 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 CloudUCM 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 an LDAP client to download phonebook entries from another LDAP server.

To access the 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 CloudUCM.

LDAP Server

[LDAP Server Configurations](#) LDAP Phonebook

Enable LDAP Server

LDAP Server Address

LDAPS Server Port

* Base DN

PBX DN .ou=070037,dc=pbx,dc=com

Root DN .ou=070037,dc=pbx,dc=com

* Root Password 👁

* Confirm Root Password 👁

LDAP Server Configurations

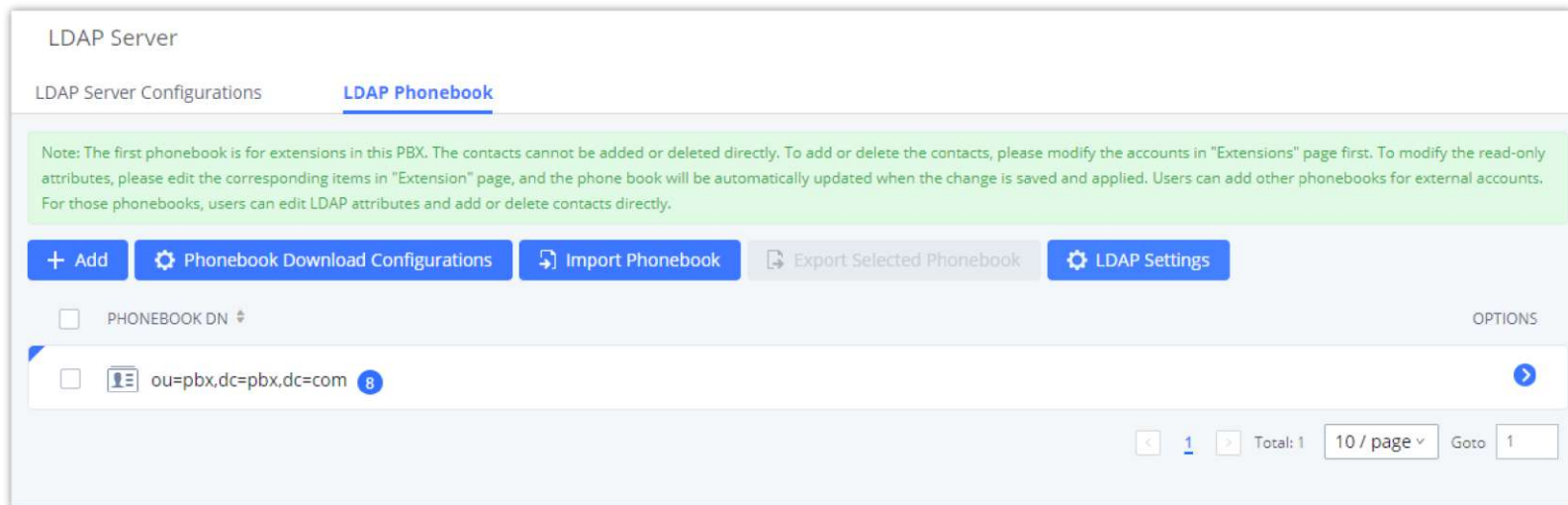
Enable LDAP Server	Enable LDAP Server
---------------------------	--------------------

LDAP Server Address	LDAP server address is predefined by the datacenter chosen for the CloudUCM. The user cannot modify this option.
LDAPS Server Port	LDAPS server port is predefined and cannot be set by the user. The port number is 636.
Base DN	Specifies the location in the directory where the search is requested to begin. It is predefined and cannot be modified.
PBX DN	Specifies the location in the directory where the search for PBX entry is requested to begin. It narrows the search scope and decreases directory lookup time.
Root DN	Specifies the location in the directory where the search for the admin user entry is requested to begin. It narrows the search scope and decreases directory lookup time.
Root Password	Defines the root password for authentication.
Confirm Root Password	Confirms the root password for authentication.

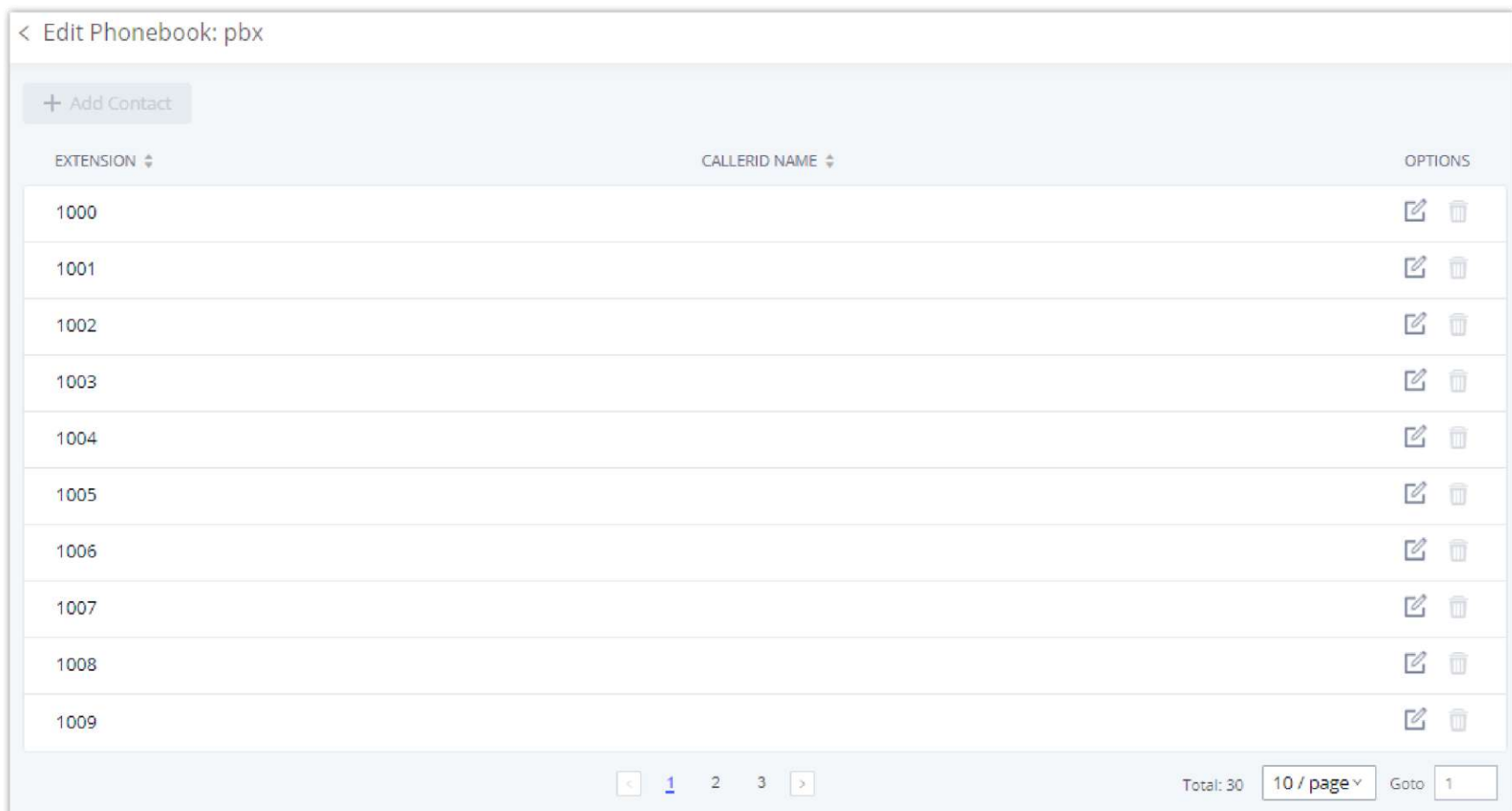
The CloudUCM LDAP server supports anonymous access (read-only) by default. Therefore, the LDAP client does not have to configure a username and password to access the phonebook directory. The "Root DN" and "Root Password" here are for LDAP management and configuration where users will need to provide for authentication purposes 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 CloudUCM support secure LDAP (LDAPS) where the communication is encrypted and secure.

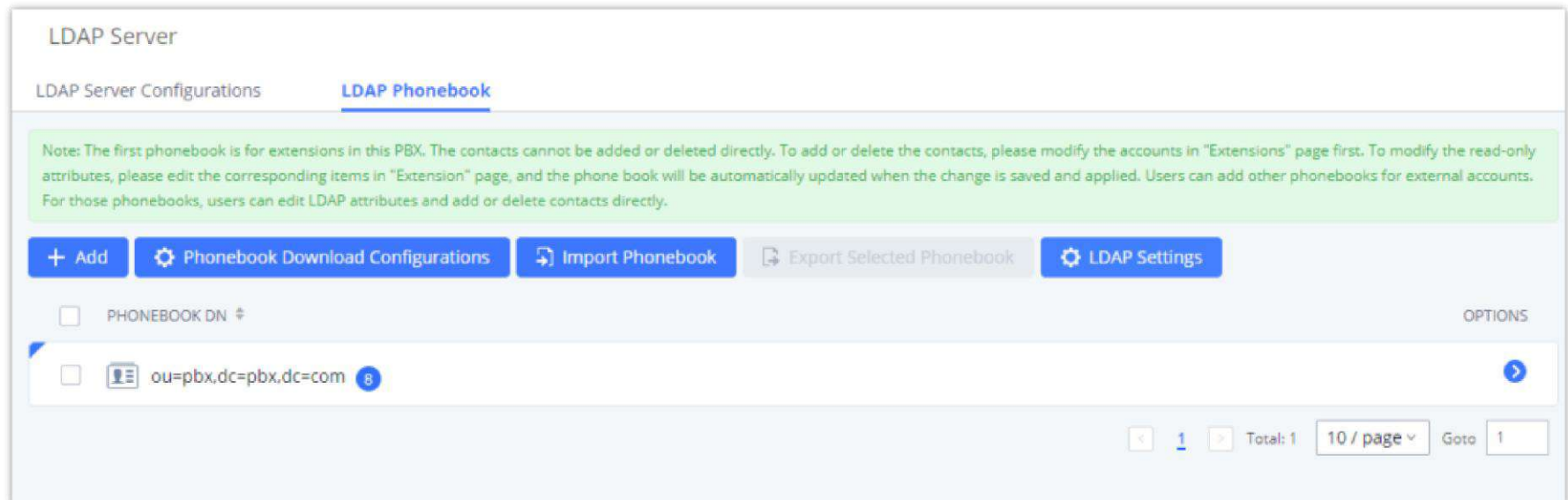


Default LDAP Phonebook DN



LDAP Phonebook

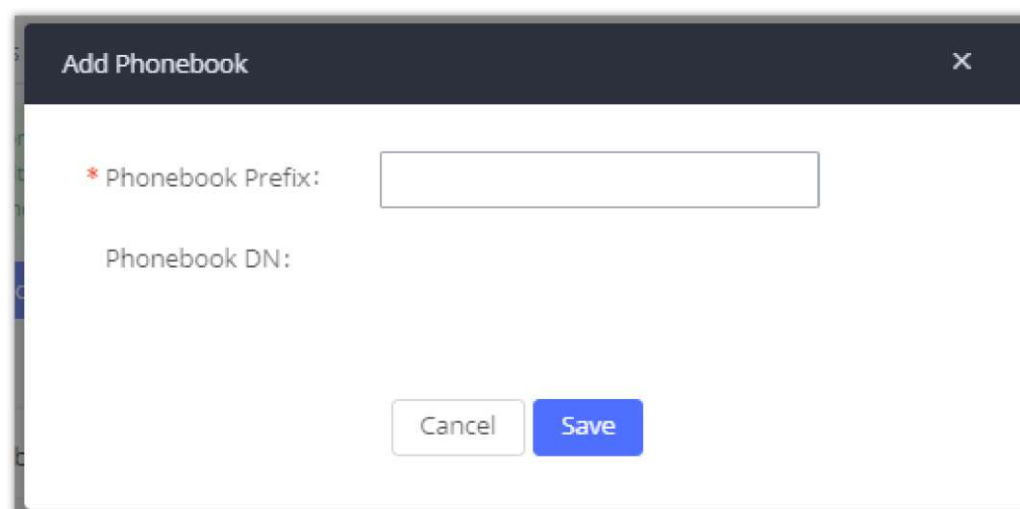
Users could use the default phonebook, edit the default phonebook, add new phonebook, import phonebook on the LDAP server as well as export phonebook from the LDAP server. The first phonebook with default phonebook dn "ou=pbx,dc=pbx,dc=com" displayed on the LDAP server page is for extensions in this 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**

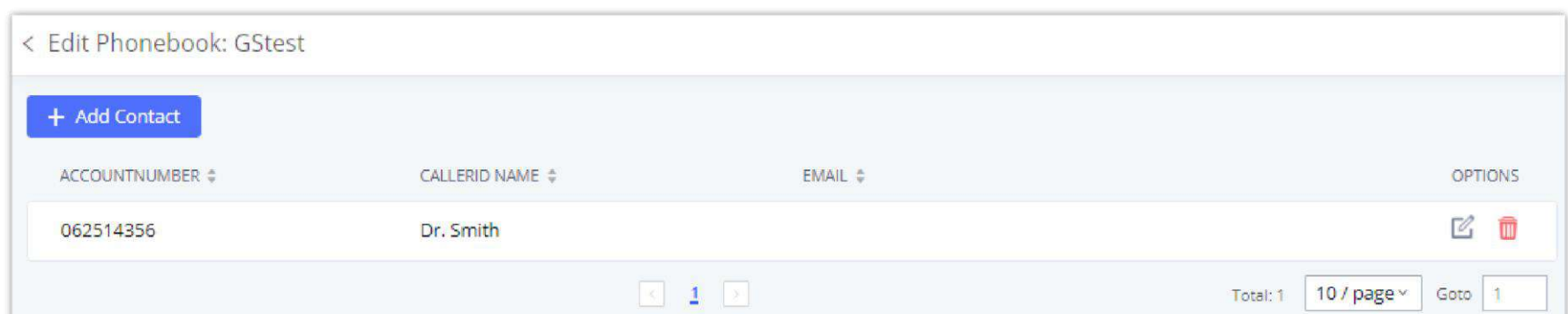
A new sibling phonebook of the default PBX phonebook can be added by clicking on "Add" under "LDAP Phonebook" section.



Add 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".

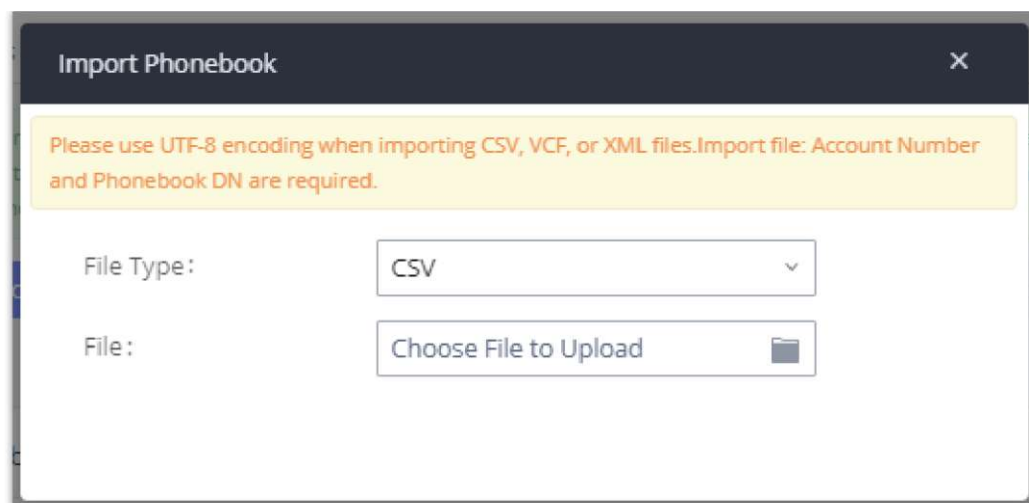
Once added, users can select  to edit the phonebook attributes and contact list (see figure below) or select  to delete the phonebook.



Edit 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 CloudUCM LDAP phonebook section first and use it as a sample to start with.

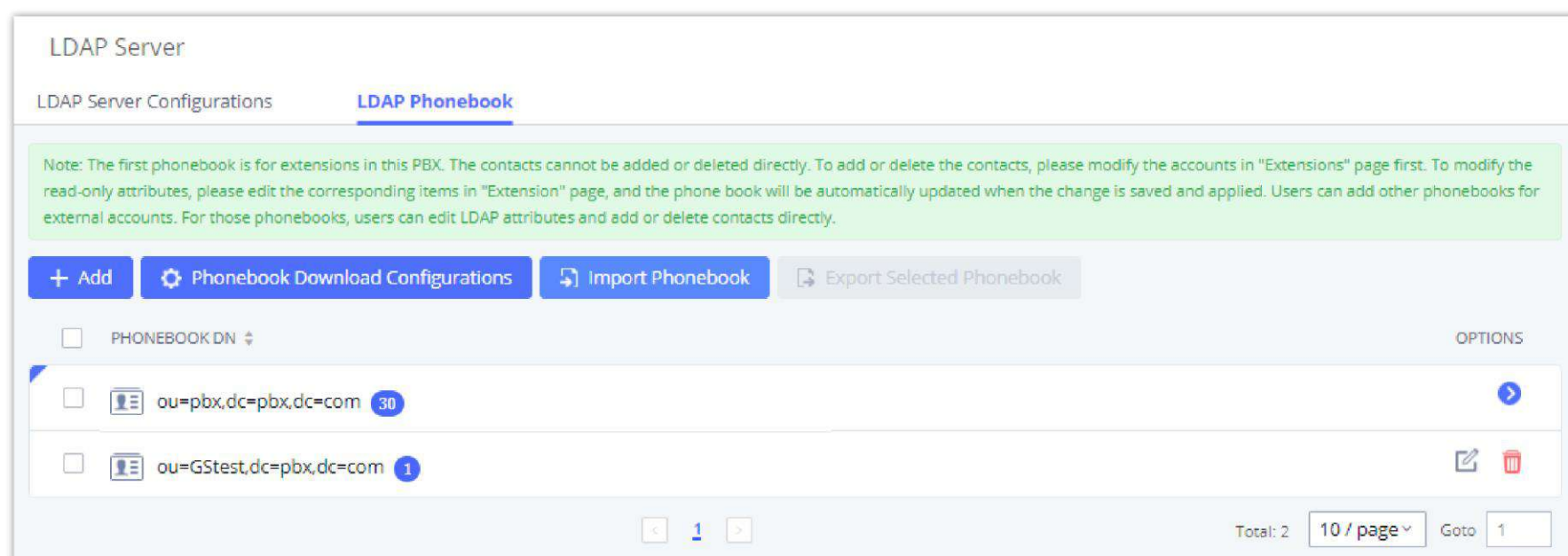
	A	B	C	D	E	F	G	H	I	J
1	First Name	Last Name	Account Number	CallerID Name	Email	Department	Mobile Number	Home Number	Fax	Phonebook DN
2	John	Doe	1001	1001		IT	1001000000			phonebook
3	Jane	Doe	1002	1002		Sales	1002000000			phonebook
4	William	Chung	1003	1003		Marketing	1003000000			phonebook
5	Linda	Kuo	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 CloudUCM 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 CloudUCM LDAP Phonebook, the contacts in the CSV file will be added into the existing phonebook. If the phonebook DN does not exist on the CloudUCM 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 CloudUCM.

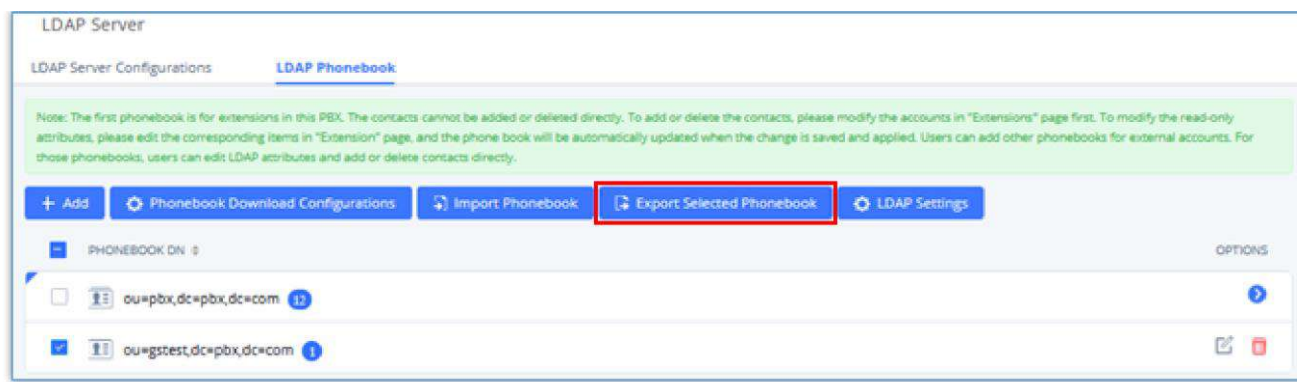


LDAP Phonebook After Import

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.

- o **Export phonebook to your computer from CloudUCM LDAP server**

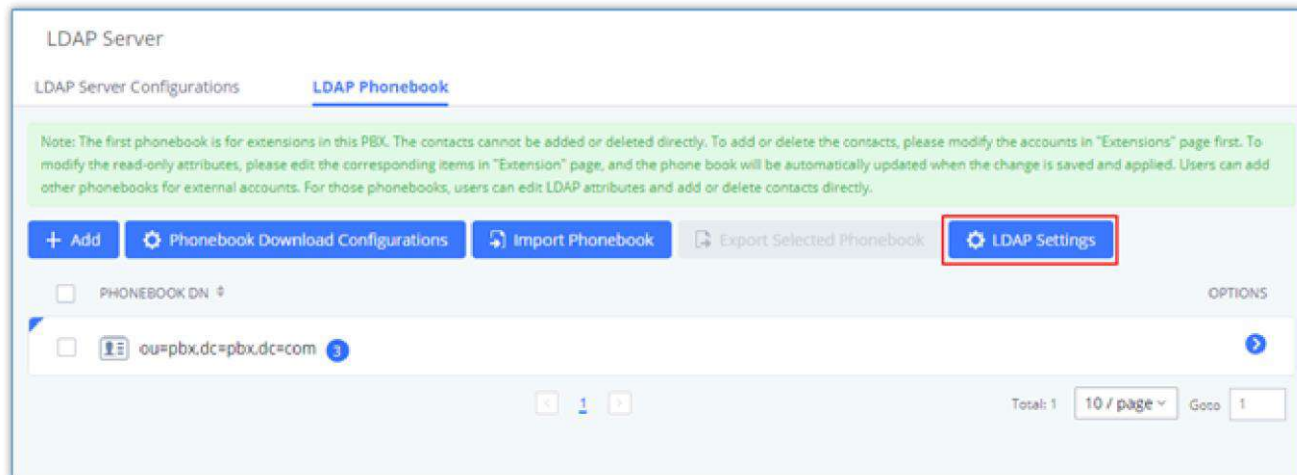
Select the checkbox for the LDAP phonebook and then click on "Export Selected Phonebook" to export the selected phonebook. The exported phonebook can be used as a record or a sample CSV, VCF or XML file for the users to add more contacts in it and import to the CloudUCM again.



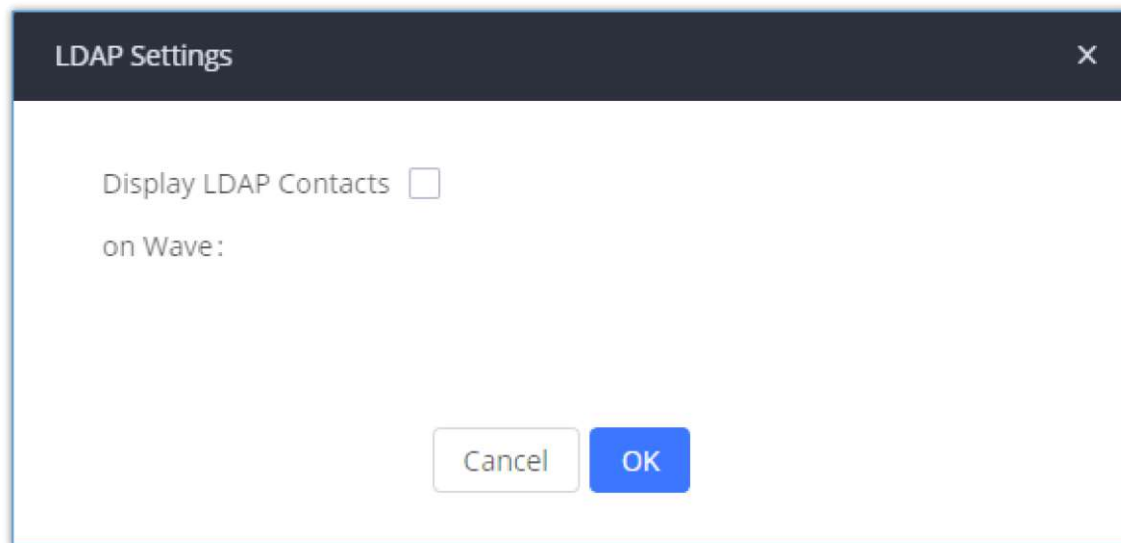
Export Selected LDAP Phonebook

LDAP Settings

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



Display LDAP Contacts on Wave

LDAP Client Configuration

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

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

LDAP Client Type

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 Server > Phonebook Download Configurations

Server Type LDAP AD

* Server Address: 192.168.1.1

* Username: cn=admin,dc=pbx,dc=com

* Filter: (objectClass=*)

* LDAP Number Attributes: AccountNumber MobileNumber HomeN

LDAP Name Attributes: CallerIDName Email Department FirstNa

* Phonebook Name: LdapClient

* Base DN: dc=pbx,dc=com

* Password:

* Port: 389

Automatic Update Cycle: None

Client Type: LDAP

Example

1. Client Configuration Examples
2. LDAP Configurations examples on Grandstream IP Phones
3. Examples of Endpoint Phonebook Download Configuration

Cancel Save

LDAP Client Configurations

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.

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

- o **Server Address:** The IP address or domain name of the UCM
- o **Base DN:** dc=pbx,dc=com
- o **Username:** cn=admin,dc=pbx,dc=com
- o **Password:** admin (by default)
- o **LDAP Name Attribute:** CallerIDName Email Department FirstName LastName
- o **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 **[LDAP Server Configurations]**.

LDAP

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

GXP2170 LDAP Phonebook Configuration

i The CloudUCM LDAP server is no longer supporting the anonymous binding of the LDAP client.

AD Client Type

LDAP Server > Phonebook Download Configurations

Server Type LDAP AD

* Phonebook Name

* Server Address * Base DN

* Username * Password

* Filter * Port

* AD Attributes Automatic Update Cycle

* Host Name

Example

1. Client Configuration Examples
2. LDAP Configurations examples on Grandstream IP Phones
3. Examples of Endpoint Phonebook Download Configuration

AD Phonebook Server 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

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

Time Zone Settings

To configure the CloudUCM timezone, go to Web GUI→**System Settings** → **Time Settings** → **Time Zone Settings**

Time Settings

Time Zone Settings Set Date and Time Office Time Holiday

* Time Zone (UTC-01:00) Etc/GMT+1

Update Time Zone List

Cancel Save

Time Zone Settings

Set Date and Time

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

Time Settings

Time Zone Settings Set Date and Time Office Time Holiday

Date Format yyyy-mm-dd

Time Format Use 24-hour Format

Cancel Save

Time Auto Updating

Office Time

On the CloudUCM, the system administrator can define "office time" which can be used to configure time condition for extension call forwarding and inbound rules. To configure office time, log in to the Web GUI, enter the **System Settings**→**Time Settings**→**Office Time**, and click the "Add" button to see the following configuration page.

Time Settings

Time Zone Settings Set Date and Time Office Time Holiday

+ Add Delete Import Export

<input type="checkbox"/>	Index	Time	Week	Month	Day	Options
No data						

Office Time

Time Settings > Create New Office Time

Time: 00:00 - 23:59

Week: Sun Mon Tue Wed Thu Fri Sat

Show Advanced Options:

Cancel Save

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.

i The office hour feature support import/export CSV files.

Time Settings

Time Zone Settings Set Date and Time Office Time Holiday

+ Add Delete Import Export

<input type="checkbox"/>	Index	Time	Week	Month	Day	Options
<input type="checkbox"/>	1	09:00-18:00	Mon Tue Wed Thu Fri	Default	Default	

Total: 1 < 1 > 10 / page Goto

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 UCM, the system administrator can define "holidays" which can be used to configure time condition for extension call forwarding and inbound rules. To configure office time, log in to the Web GUI, enter the **System Settings→Time Settings→Holiday**, and click the "Add" button to see the following configuration page.

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	

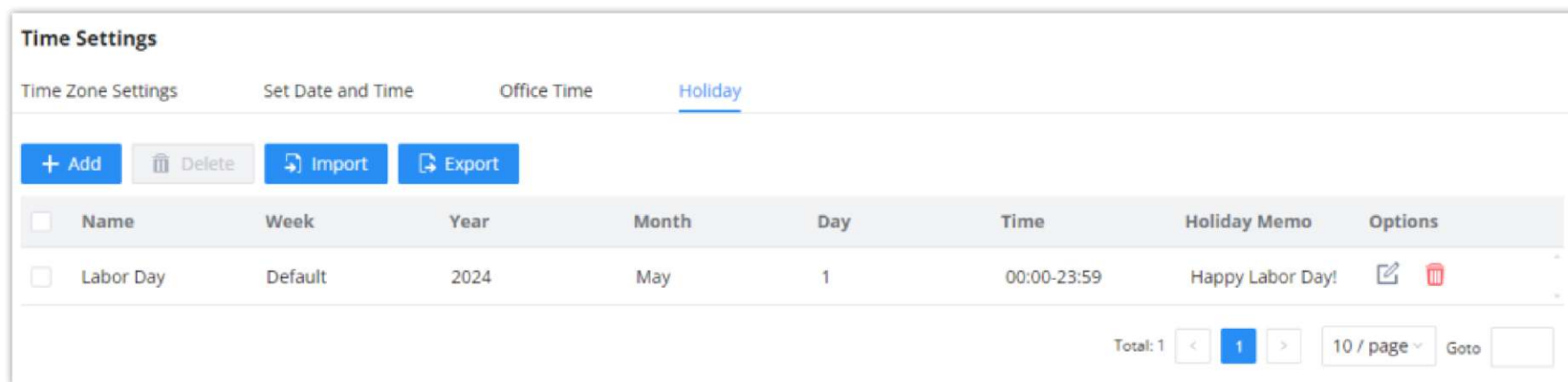
Time : -

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", "Day" and the exact "Hour". 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.

 The Holiday feature support import/export CSV files.



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 feature on the CloudUCM 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**.

Parameter	Description
TLS Enable	If enabled, TLS will be used when forwarding emails to the SMTP server.
Domain	Configures the domain of the UCM's internal email server. This should not be the name of known and existing email servers (e.g. Gmail, Outlook, etc.).
Email Template Sending Format	Select the email template format to be sent. The "HTML" format is compatible with most mail clients and is recommended. If the mail client does not support the "HTML" format, please select the "Plain Text" format.
SMTP Server	Enter the SMTP server. For example, smtp.mydomain.com:465. Port number cannot be 25.
Enable SASL Authentication	Toggles SASL authentication. If disabled, UCM will not use the username and password for email client authentication. Most email servers require login authentication while private email servers may allow anonymous login. If using Microsoft Exchange Server or if credentials are not required, please disable this option.
Username	Enter the username of the email account.
Password	Enter the password of the email account. It is highly recommended to use HTTPS when saving and applying password changes.
Enable Email-to-Fax	Monitors the inbox of the configured email address for the specified subject. If detected, 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	<ul style="list-style-type: none"> • Disable: This option is disabled. • Enable Blacklist: Select the blacklist to apply. • Enable Whitelist: Select the whitelist to apply. <p>Note: This option only appears when "Enable Email-to-Fax" is enabled.</p>
Email-to-Fax Subject Format	<p>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.</p> <p>Note: This option only appears when "Enable Email-to-Fax" is enabled.</p>
Fax Sending Success/Failure Confirmation	<p>If enabled, the UCM will send an email notification to the sender about the fax sending result.</p> <p>Note: This option only appears when "Enable Email-to-Fax" is enabled.</p>
POP/POP3 Server Address	<p>Enter the IP address of the POP/POP3 server.</p> <p>Note: This option only appears when "Enable Email-to-Fax" is enabled.</p>
POP/POP3 Server Port	<p>Enter the port of the POP/POP3 server.</p> <p>Note: This option only appears when "Enable Email-to-Fax" is enabled.</p>
Display Name	<p>Enter the name of the PBX that will be displayed in sent emails.</p>
Sender	<p>Enter the email used to send the emails.</p>

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

Email Settings

[Email Settings](#) | [Email Template](#) | [Email Footer Hyperlink](#) | [Email Send Log](#)

TLS Enable

Email Template Sending Format: HTML

* SMTP Server: Example: smtp.mydomain.com:465

* Enable SASL Authentication

* Username:

The email server must allow 3rd party email clients to use the SMTP service.

* Password:

Enable Email-to-Fax

* Display Name: PBX

* Sender: anonymous@example.com

[Test](#)


Email Settings

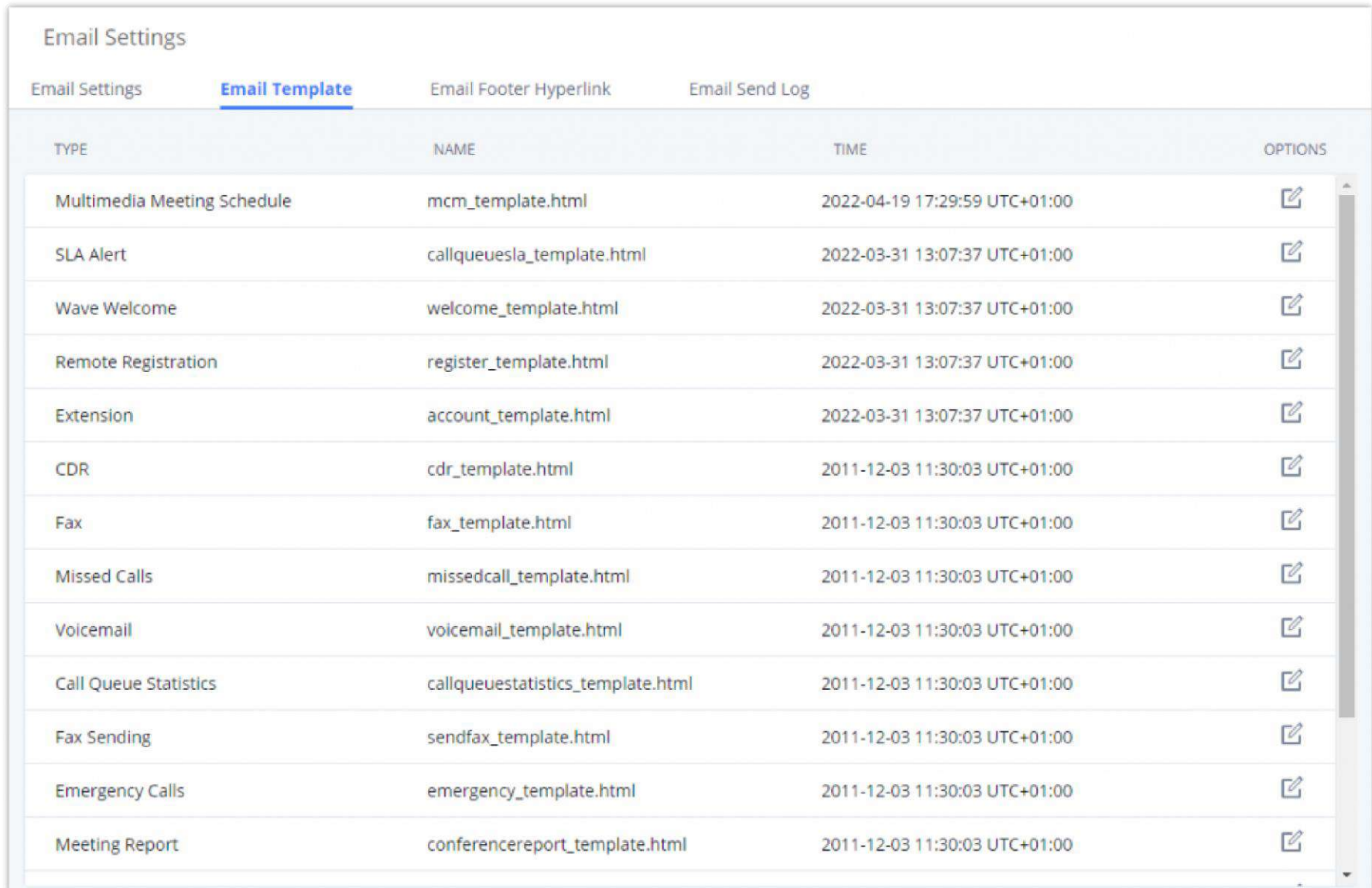
Once the configuration is finished, click on "Test". In the prompt, fill in a valid Email address to send a test email to verify the Email settings on the CloudUCM.














Email Templates

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



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

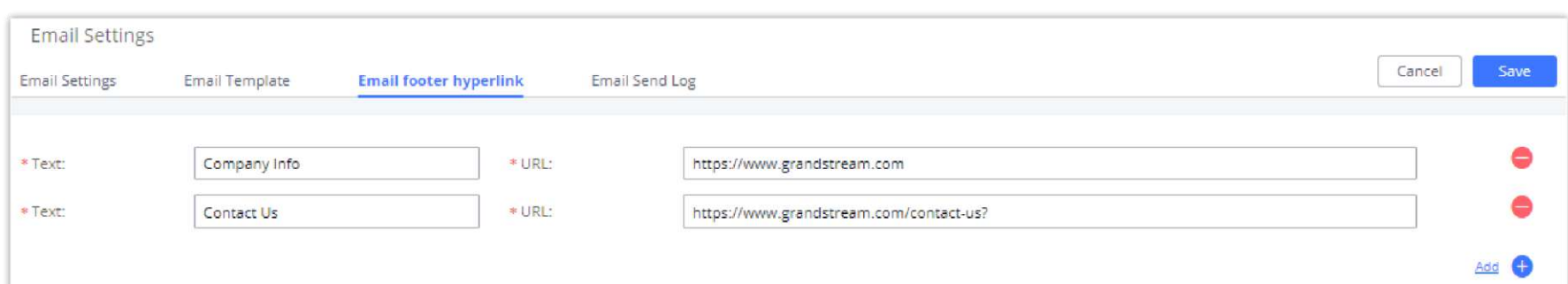
Note

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

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

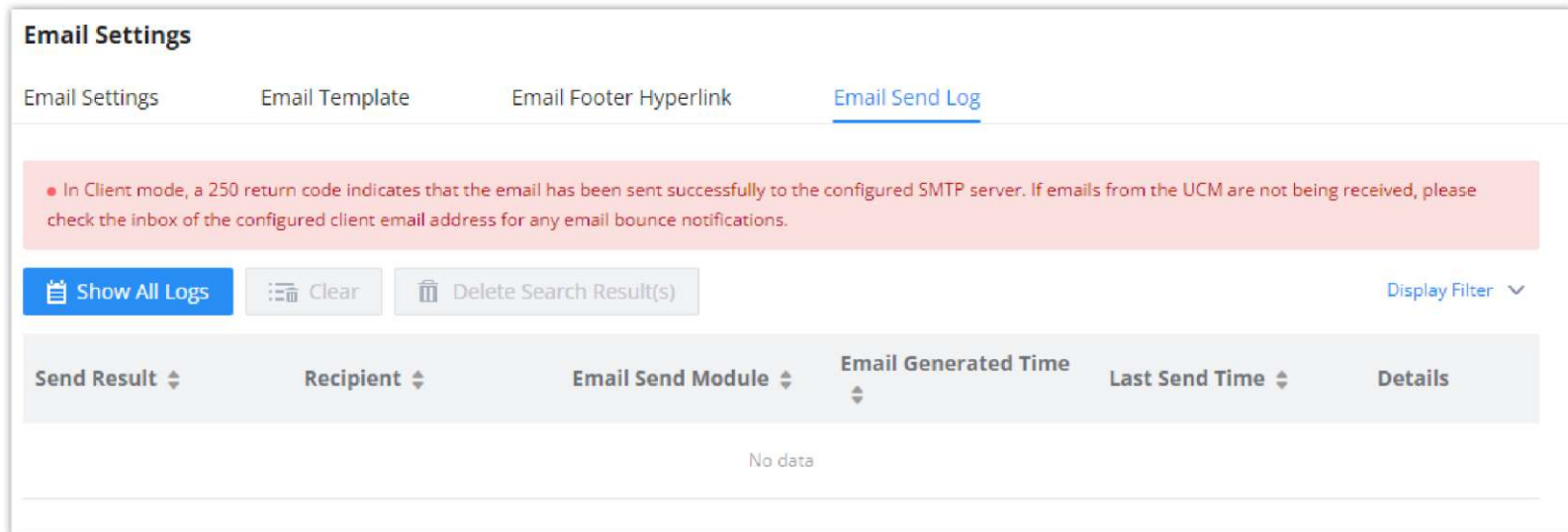


Text	URL
Company Info	https://www.grandstream.com
Contact Us	https://www.grandstream.com/contact-us?

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

Start Time	Enter the start time for the filter
End Time	Enter the end time for the filter
Receivers	Enter the email recipient, while searching for multiple recipients, please separate them with a 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	<p>Select the email module to filter with from the drop-down list, which contains:</p> <ul style="list-style-type: none"> ● All Modules ● Extension ● Voicemail ● Conference Schedule ● User Password ● Alert Events ● CDR ● Test

Email Log – Display Filter

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

Email Settings								
Email Settings	Email Template	Email Footer Hyperlink	Email Send Log					
<p>• In MTA mode, SPF authentication cannot be received. Therefore, even if an email is sent successfully, the return code of 550 will still be returned. Many email servers will place non-SPF-certified mail into the trash or spam folder. If the recipient has not received a sent email, please check to see if the sent email was placed in the recipient's trash or spam folder. In Client mode, a 250 return code indicates that the email has been sent successfully to the configured SMTP server. If emails from the UCM are not being received, please check the inbox of the configured client email address for any email bounce notifications.</p> <p>• In Client mode, a 250 return code indicates that the email has been sent successfully to the configured SMTP server. If emails from the UCM are not being received, please check the inbox of the configured client email address for any email bounce notifications.</p>								
<a>Show All Logs	<a>Clear	<a>Delete Search Result(s)						Display Filter ▾
EMAIL GENERATED TIME	EMAIL SEND MODULE	RECEIVERS	LAST SEND TIME	LAST SEND ADDRESS	SEND RESULT	RETURN CODE	OPTIONS	
2022-04-21 09:59:25	Wave Welcome	*****@grandstream.com	04-21 09:59:32	*****@grandstream.com	sent	250	ⓘ	
2022-04-21 09:59:06	Remote Registration	*****@grandstream.com	04-21 09:59:08	*****@grandstream.com	sent	250	ⓘ	
2022-04-20 12:39:32	SLA Alert	*****@gmail.com	04-20 12:39:34	*****@gmail.com	sent	250	ⓘ	
2022-04-19 17:45:07	SLA Alert	*****@gmail.com	04-19 17:45:09	*****@gmail.com	sent	250	ⓘ	
2022-04-19 17:39:59	SLA Alert	*****@grandstream.com	04-19 17:40:01	*****@grandstream.com	sent	250	ⓘ	
2022-04-19 17:34:43	Test	*****@gmail.com	04-19 17:34:45	*****@gmail.com	sent	250	ⓘ	
2022-04-19 17:19:30	Test	*****@gmail.com	04-19 17:19:31	*****@gmail.com	sent	250	ⓘ	
						Total: 18	10 / page	Go to 1

Email Logs

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

Email Codes

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 username and password verification in the client mode is incorrect. Please check whether the username and password are configured correctly.
550	<p>Possible reasons:</p> <ol style="list-style-type: none"> 1. The recipient's mailbox username 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

Code	Description
none	<p>This 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>

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.

Contact Management

Extension Contacts External Contacts

Change Department Contact Privilege Settings Delete

Extension Number or Name Search

EXTENSION	NAME	DEPARTMENT	EMAIL ADDRESSES	CONTACT PRIVILEGES	OPTIONS
<input type="checkbox"/> 1000		---		All Contacts(Same as Department)	
<input type="checkbox"/> 1001	John Doe	---		All Contacts(Same as Department)	
<input type="checkbox"/> 1002	Jane Doe	---		All Contacts(Same as Department)	
<input type="checkbox"/> 1003	Arthur Morgan	---		All Contacts(Same as Department)	
<input type="checkbox"/> 1004		---		All Contacts(Same as Department)	
<input type="checkbox"/> 1005		---		All Contacts(Same as Department)	
<input type="checkbox"/> 2000		---		All Contacts(Same as Department)	
<input type="checkbox"/> 2001		---		All Contacts(Same as Department)	
<input type="checkbox"/> 2002		---		All Contacts(Same as Department)	
<input type="checkbox"/> 2003		---		All Contacts(Same as Department)	

Total: 11 10 / page Goto 1

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.

* Extension:

First Name:

Last Name:

Department:

Job Title:

Email Address:

Mobile Phone Number:

Home Number:

Fax:

| Contact Privileges

Same as Department

Contact Privileges:

* Contact View [Add / Edit Privileges](#)

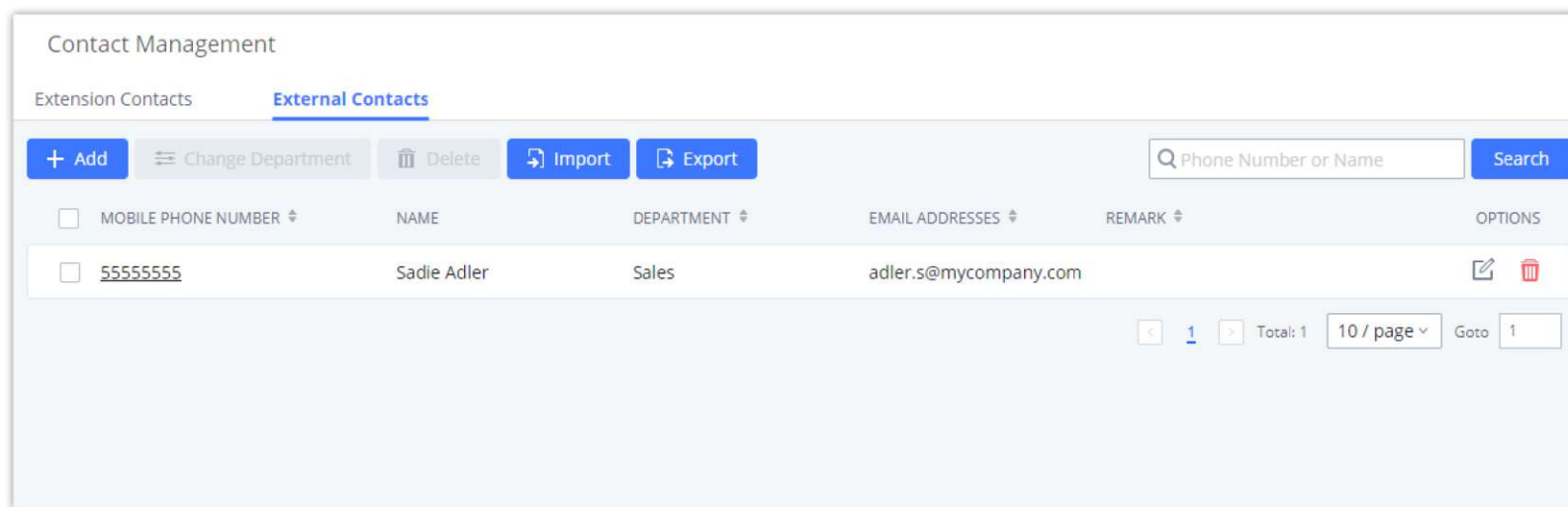
Privileges:

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.

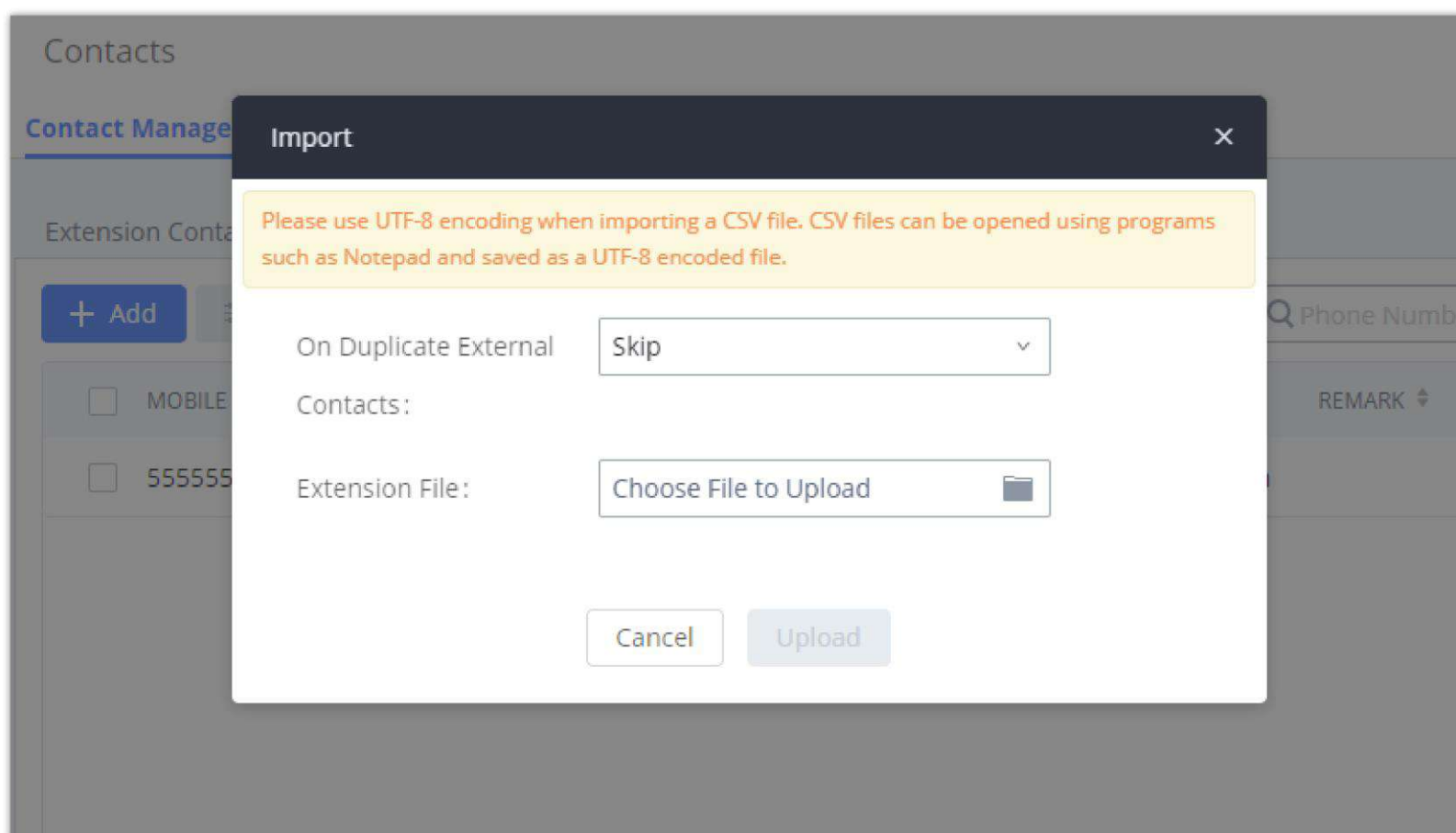


External Contacts

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:

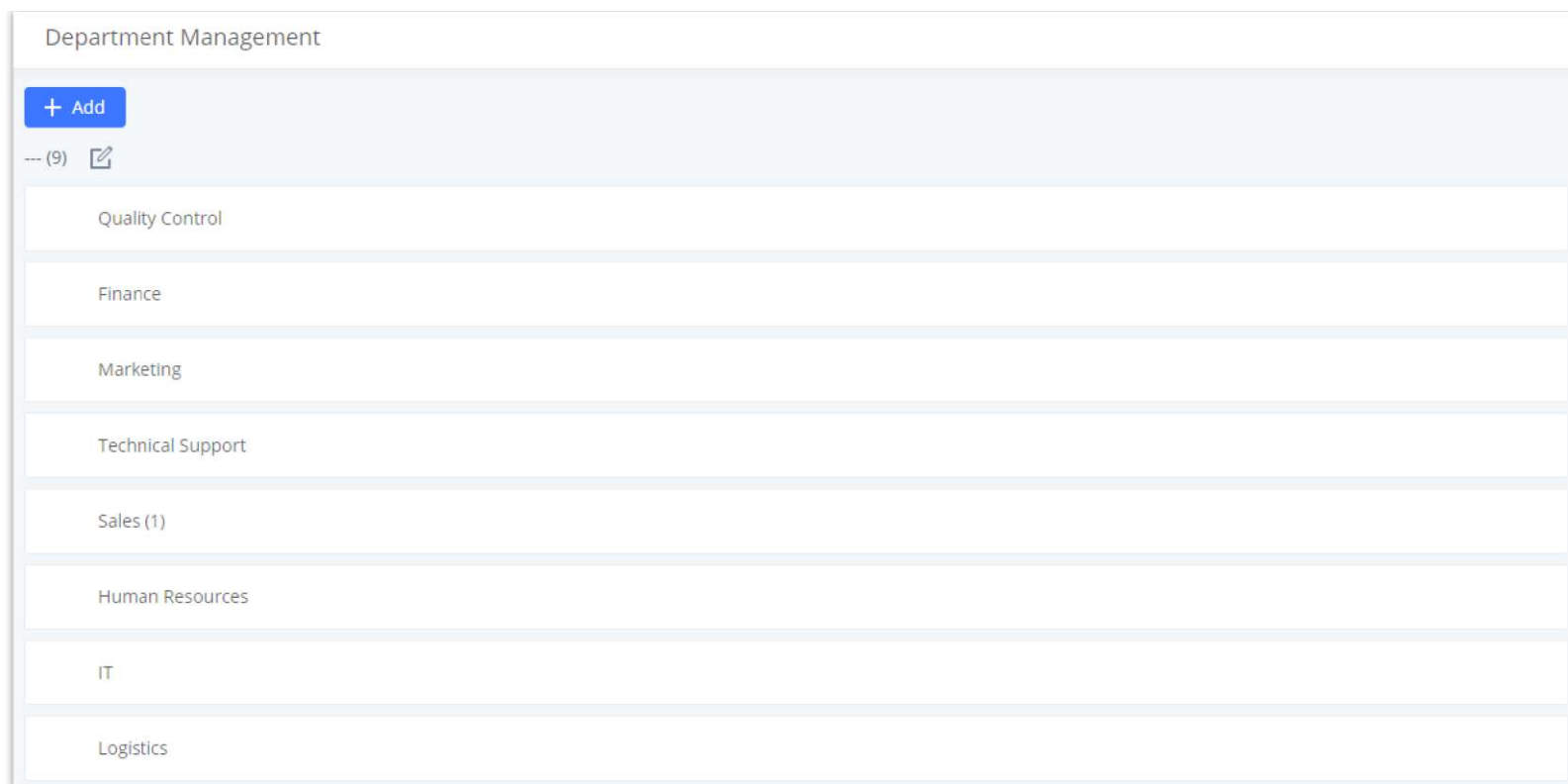
- **Step 1:** For option "On Duplicate External Contacts", select whether to skip duplicate contact on the imported CSV file or update the duplicate UCM contact with the information in the CSV.
- **Step 2:** Choose file from local PC to upload.
- **Step 3:** Click on "Upload".
- **Step 4:** Click on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

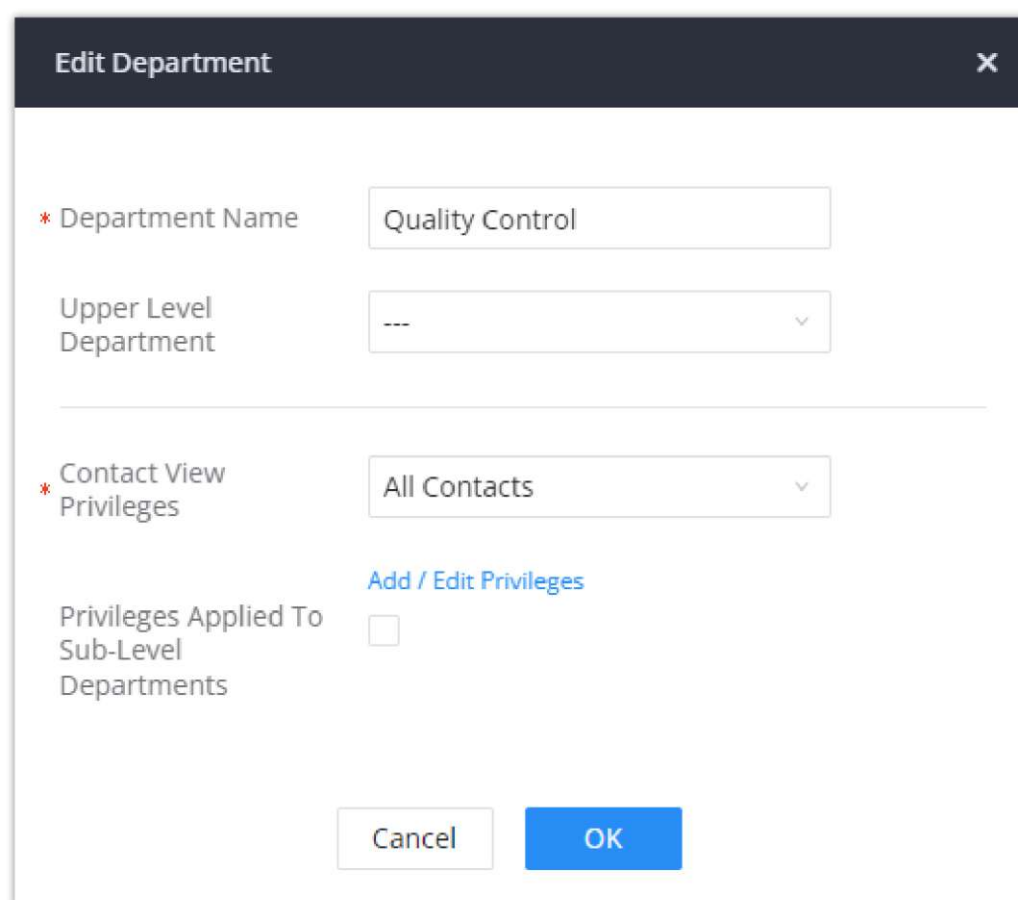


Department Management

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.

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

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.

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

Total: 2 10 / page Goto 1

Privilege Management

DEVICE MANAGEMENT

Onsite Meeting

For workplaces that require employees to return to physical offices for work, Grandstream UCM offers the Onsite Meetings 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.

Schedule Onsite Meetings

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 meetings.

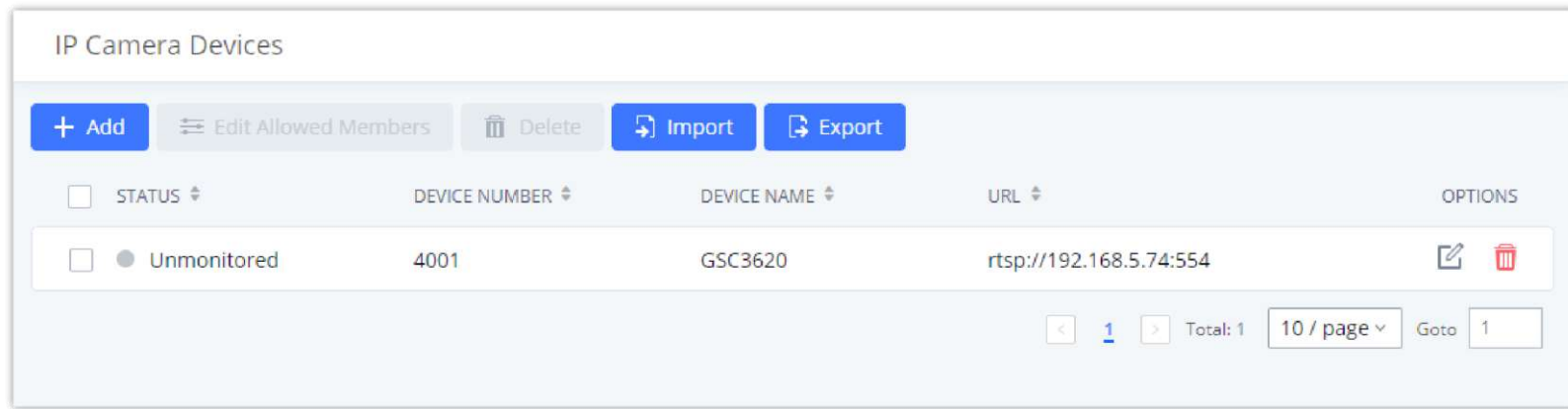
MEETING SUBJECT	MEETING ROOM	MEETING OWNER	START TIME	MEETING DURATION	REPEAT	OPTIONS
Quick Meeting	BackupRoom		Today 20:00 Etc/GMT+8	00:30:00	No Repeat	
Recent Events	BackupRoom		2022-11-24 07:00 Etc/GMT+8	01:00:00	No Repeat	
Yearly Reports	Primary-Meeting-Room		2022-12-29 13:00 Etc/GMT+8	01:00:00	No Repeat	

Pending Onsite Meetings

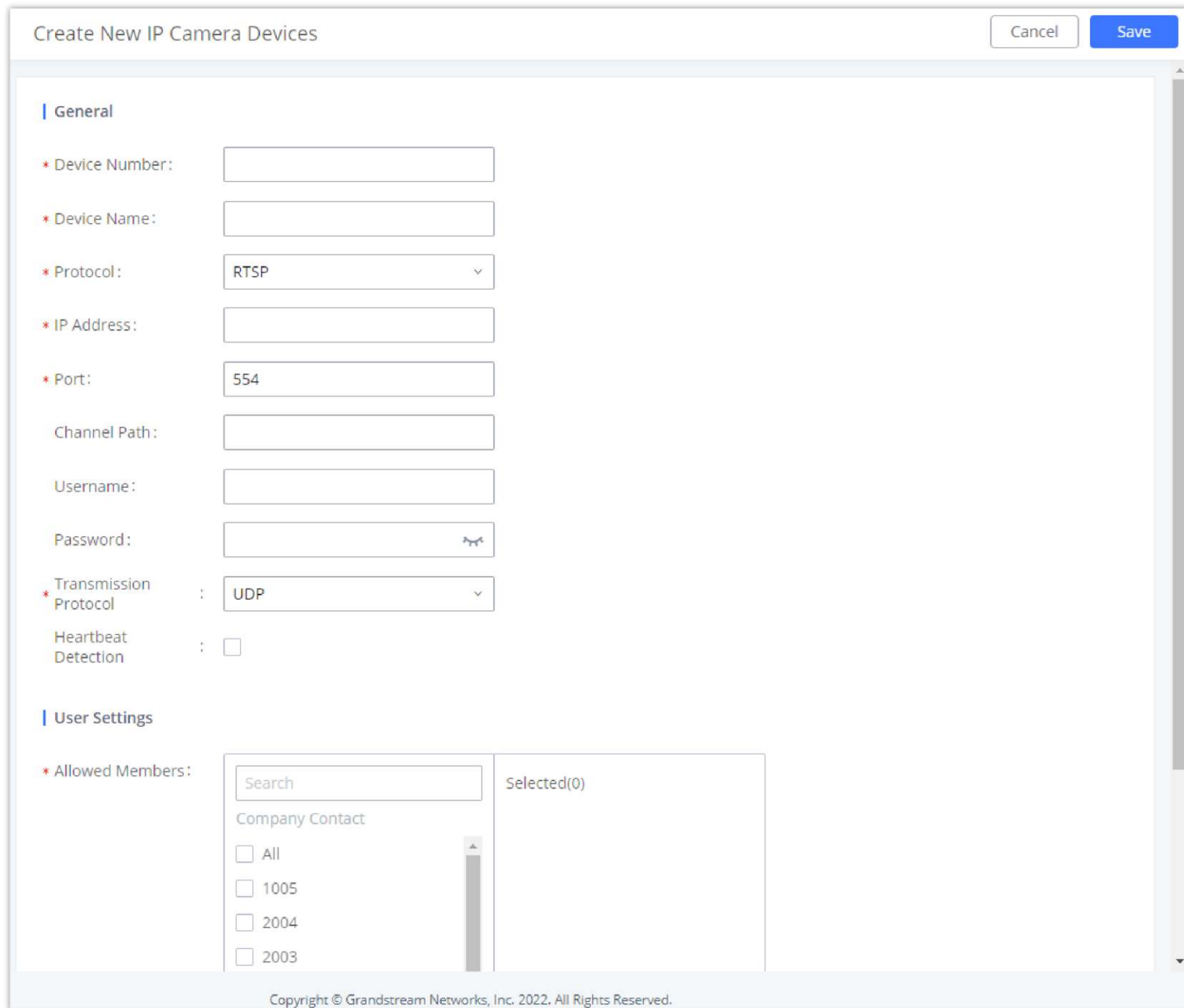
IP Camera Devices

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.



IPC Devices



IP Camera Devices Settings

Edit the IPC device settings in the table below.

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 if 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.

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 CloudUCM 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			
User Information		Custom Privilege	
USERNAME	PRIVILEGE	LAST OPERATION TIME	OPTIONS
admin	Super Administrator	2021-11-05 13:02:11	

User Management Page Display

Login Settings

Change Password / Email

Change Password

After logging in the CloudUCM 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](#) [Remote Login](#)

To remove the security warning message at the top of the page, both username and password must be changed.

Change Password

Change Password

Change Username

Change Username

Change Email Address

* Email Address [Email Template](#)

Change Phone Number

Mobile Number [SMS Template](#)

Enter Current Password

* Enter Current Password

Change Password

Enter Old Password	Enter the Old Password for CloudUCM
Change Password	Enable Change Password
Enter New Password	Enter the New Password for CloudUCM
Re-enter New Password	Retype the New Password for CloudUCM
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 Username

CloudUCM allows users to change Super Administrator username.

Change Username

Change Username:

* Please enter the username:

Change Email Address

CloudUCM allows user to configure binding email in case login password is lost. CloudUCM 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 CloudUCM 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 CloudUCM 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.

Login Timeout Settings

User Login Timeout (m)	Set login timeout (in minutes) for user. If there is no activity within the specified amount of time, the user will be logged out, and the system will jump to the login page automatically. If set to 0, the user will not be logged out automatically.
Maximum number of login attempts	The maximum number of consecutive failed login attempts. When exceeded, the user will not be able to log in for the amount of time specified in "User ban period". A value of 0 means unlimited attempts.
User Ban Period (m)	The amount of minutes that a user will be banned for after exceeding the maximum allowed number of consecutive failed login attempts. A value of 0

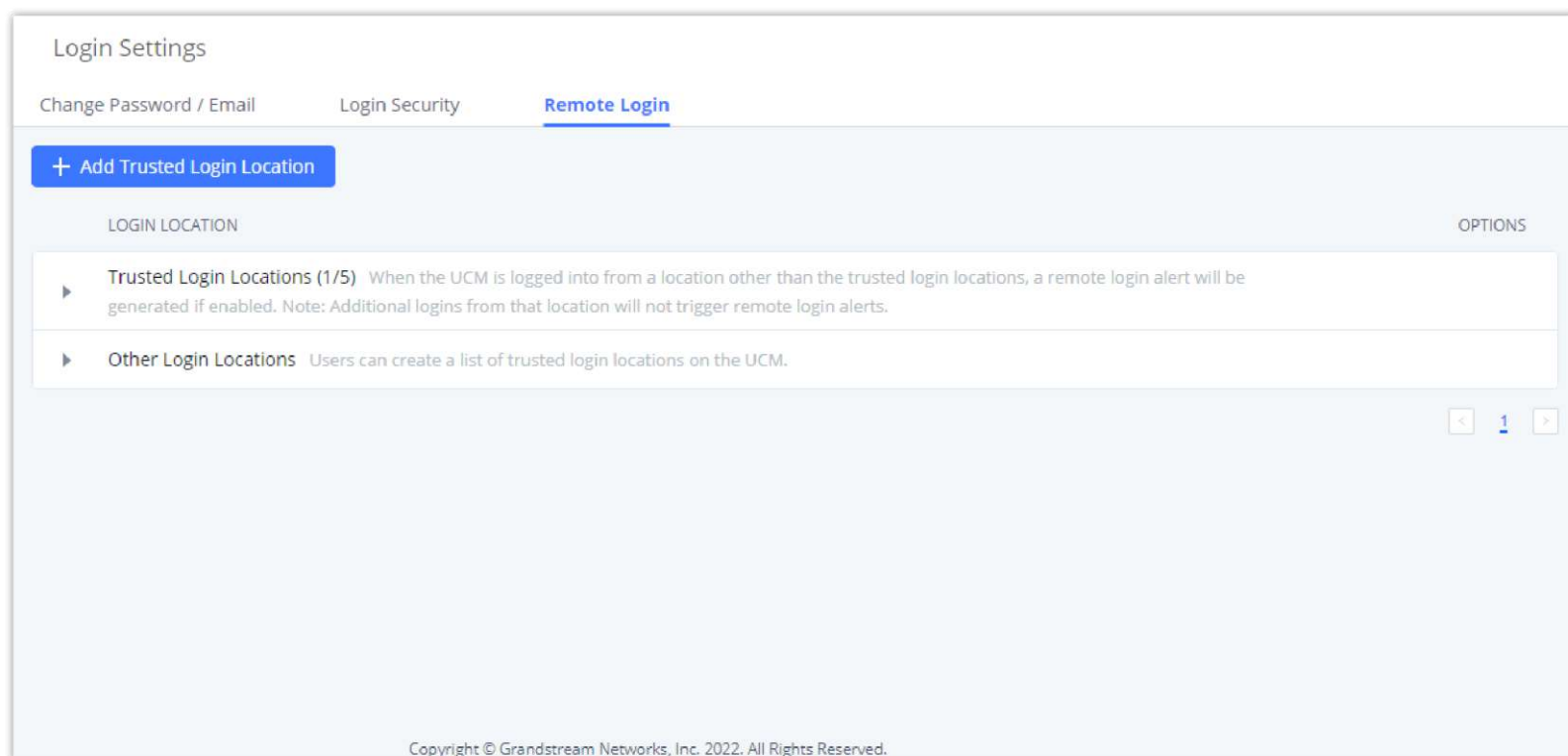
	indicates a permanent ban.
SMS OTP Login	If enabled, users will be able to log in and reset password via SMS verification code. Mobile phone numbers will need to be configured for administrators and extensions.
Login Banned User List	List of IP addresses which are banned from making any further login attempts.
Login Whitelist	List of IP addresses which can make unlimited login attempts.

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 CloudUCM 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.



Remote Login

To add a trusted login location, the user must click on [+ Add Trusted Login Location](#)

Add Trusted Login Location
×

If a public IP address is entered as the trusted login location, the UCM will automatically detect the location of the IP address.

IP Address: [Check Location](#)

Format: xxx.xxx.xxx.xxx

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 CloudUCM 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 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”.

The screenshot shows the 'Operation Log' interface. At the top right, there is a 'Hide Filter' link. Below it, there are filter fields: 'Start Time' (2019-12-10 00:00), 'End Time' (2019-12-11 00:00), 'IPv4/IPv6 Address' (empty), and 'User Name' (admin). There are 'Filter' and 'Reset' buttons. Below the filters, there are buttons for 'Delete Search Result (s)', 'Clear', 'Download Search Result (s)', and 'Download All Log'. At the bottom, there is a table header with columns: DATE, USER NAME, IP ADDRESS, RESULTS, PAGE OPERATION, SPECIFIC OPERATION, and REMARK. The table content is empty, showing 'No Data'.

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.

Syslog

On the CloudUCM, 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 CloudUCM settings because it can be helpful to locate the issues when errors happen.

Some typical modules for CloudUCM 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.



- 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.

System Events

The CloudUCM 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 CloudUCM which will have alert and/or Email generated if occurred:

System Events

Alert Log [Alert Events List](#) Alert Contact Cancel Save

Some alerts will display the following icons when toggled on: indicates that the alert is associated with other settings that must be configured; indicates that the alert is active for the enabled settings, and other related settings can be enabled.

When alert events with the GDMS tag are triggered, alert information will be synced to GDMS.

Delivery Method:

Alert Sending Interval:

Alert On Alert Off Email Notification On Email Notification Off HTTP Notification On HTTP Notification Off SMS Notification On SMS Notification Off

<input type="checkbox"/>	EVENT NAME	ALERT	EMAIL NOTIFICATION	SMS NOTIFICATION	HTTP NOTIFICATION	PARAMETER SETTINGS
<input type="checkbox"/>	Fail2ban Blocking	<input type="radio"/> OFF	<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="radio"/> OFF	
<input type="checkbox"/>	Network Traffic Storm	<input type="radio"/> OFF	<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="radio"/> OFF	
<input type="checkbox"/>	Remote Login	<input type="radio"/> OFF	<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="radio"/> OFF	
<input type="checkbox"/>	User Login Failed	<input type="radio"/> OFF	<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="radio"/> OFF	
<input type="checkbox"/>	Restore Config	<input type="radio"/> OFF	<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="radio"/> OFF	
<input type="checkbox"/>	System Reboot	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	<input type="radio"/> OFF	

Alert Event List

Alert Events
Fail2ban Blocking
User Login Banned
Remote Login
User Login Success
User Login Failed
Restore Config
System Update
System Reboot
Modify Super Admin Password
Data Sync Backup
Local Disk Usage
Emergency Calls
SIP Outgoing Call through Trunk Failure
SIP Internal Call Failure
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

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. **User login banned:** If user login is blocked, the event will be recorded in the alert log.
3. **Remote Login:** An alert will be generated upon a remote login.
4. **System Update**
5. **Restore Config:** Once the system configuration is restored, the configuration restoration event will be recorded in the alert log.
6. **System Update:** Once the system is upgraded, the system upgrade event will be recorded in the alarm log.
7. **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. 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.
8. **Modify Super Admin Password:** Once the super administrator password is modified, the system will record the password modification event in the alarm log.
9. **Emergency Calls:** If the system generates an emergency call, the event will be recorded in the alert log.
10. **Register SIP trunk failed**

Alert Settings: Register SIP trunk failed

Detect Cycle: 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.

13. **SIP peer trunk status:** If the SIP peer trunks status is abnormal, the event will be recorded in the alert log.
14. **SIP Outgoing Call through Trunk Failure:** If the system SIP trunk outgoing call fails, the event will be recorded in the alert log.

15. **Register SIP failed**

Alert Settings: Register SIP failed

Register SIP Failed Email Send Cycle: minute(s)

Time:

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.

- 16. **SIP lost registration:** If System SIP extension registration is lost, the event will be recorded in the alert log. **SIP Internal Call Failure:** If the system SIP extension call fails within the office, the event will be recorded in the alert log.
- 17. **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.
- 18. **Remote concurrent calls:** If the remote concurrent call fails, the event will be recorded in the alert log.
- 19. **Trunk Outbound Call Duration Usage:**
- 20. **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.
- 21. **User login success:** Successful user login events will be recorded in the alert log.
- 22. **User login failed:** User login failure events will be recorded in the alert log.
- 23. **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 **Filter**. Alert logs are classified into two types by the system:

- 1. **Generate Alert:** Generated when alert events happen, for example, alert logs for disk usage exceeding the alert threshold.
- 2. **Restore to Normal:** Generated when alert events being cleared, for example, logs for disk usage dropping back below the alert threshold.

User could filter out alert logs of “Generate Alert” or “Restore to Normal” by specifying the type according to need. The following figure shows an example of filtering out alert logs of type of “Restore to Normal”.

Start Time: 2020-11-03 00:00
End Time: 2020-11-10 14:31
Event Name: All
Type: All
Reset 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.

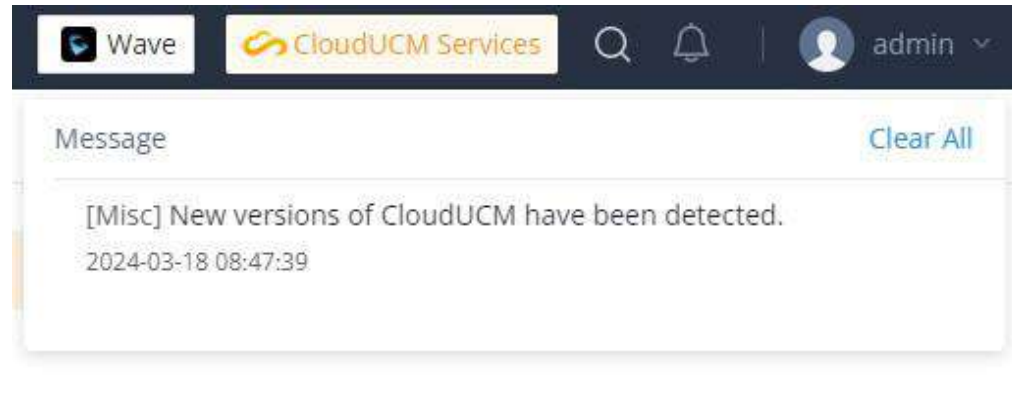
Super Admin Email	Configure the email addresses to send alert notifications to. Up to 10 email addresses can be added.
Admin Email	Configure the email addresses to send alert notifications to. Up to 10 email addresses can be added.
Email Template	Please refer to section [Email Templates]
Protocol	Protocol 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	Modifies the frequency at which notifications are sent in seconds. No notifications will be sent if the value is "0". Default value: 20
Template Variables	<code>\${MAC}</code> : MAC Address <code>\${WARNING_MSG}</code> : Warning message <code>\${TIME}</code> : Current System Time <code>\${CPU_USED}</code> : CPU Usage <code>\${MEM_USED}</code> : Memory Usage <code>\${ACTION}</code> : Message Type. Refer to [Alert Events] <code>\${DISK_USED}</code> : Disk Usage <code>\${EXTERNAL_DISK_USED}</code> : Disk Usage

Alert Contact

Upgrade

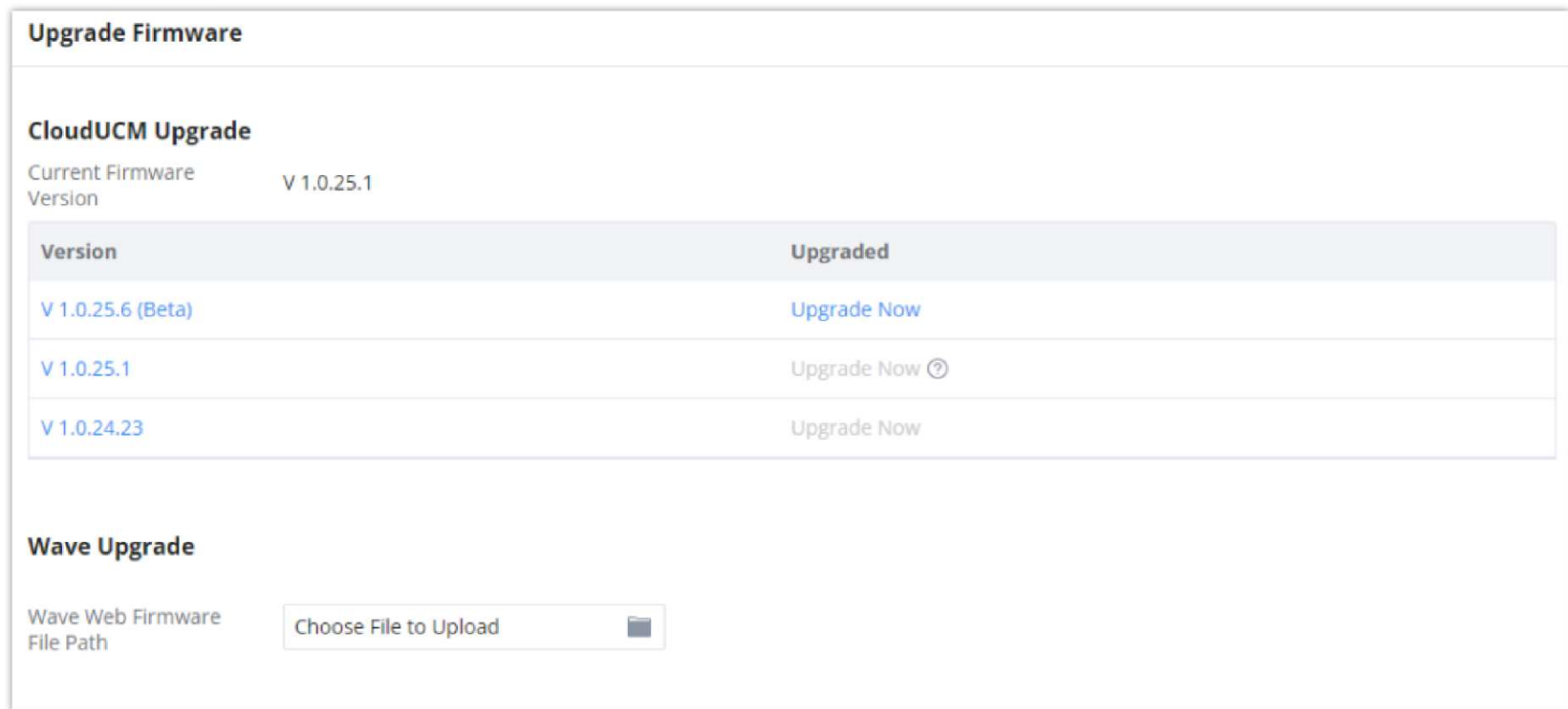
During its lifetime, the CloudUCM Series models will receive regular updates. These updates include new features and functionalities, bug fixes, security patches, and general improvements to the security and performance of the device. It is highly recommended that you keep the software of the CloudUCM up to date.

When a new CloudUCM software is detected, the administrator will receive a notification as shown in the screenshot below.



New CloudUCM Software Version

The administrator can either click on the notification or navigate to **Maintenance** → **Upgrade** then click on "Upgrade Now".



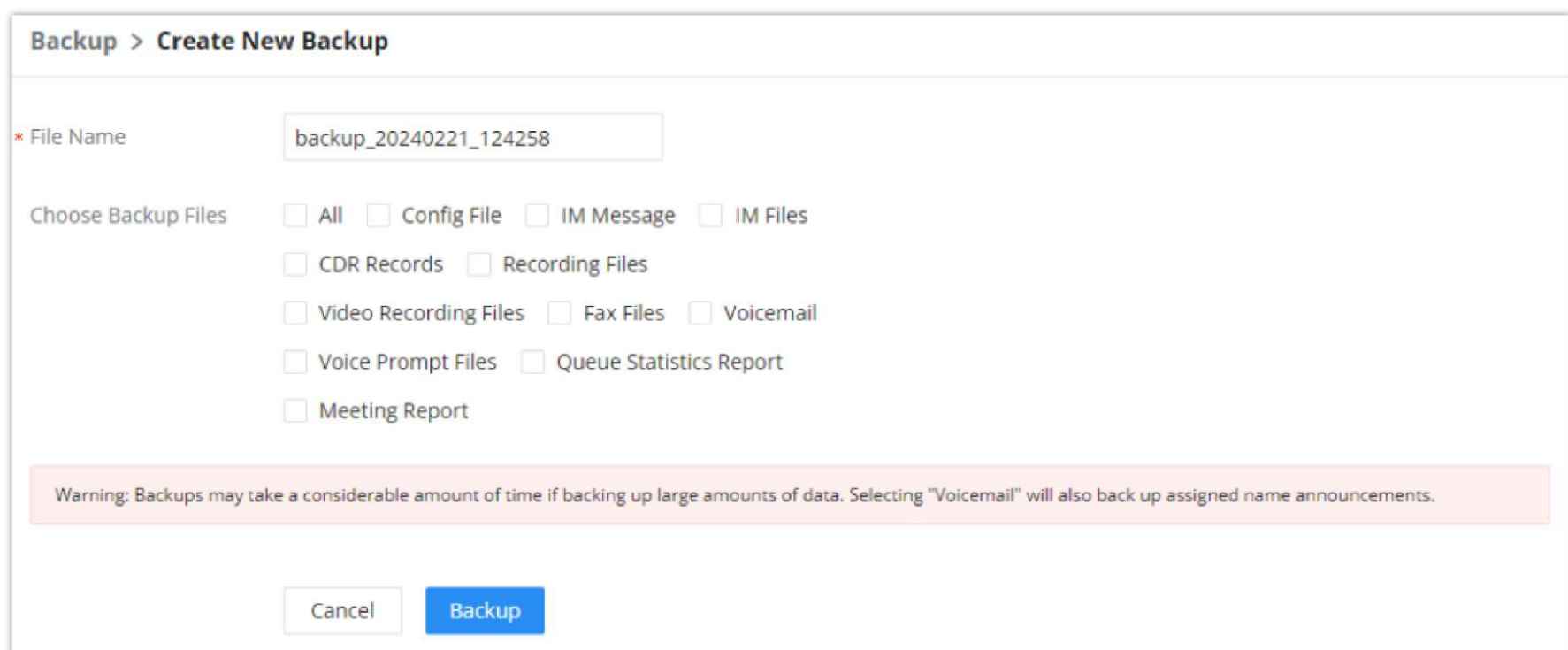
Upgrade Firmware

Backup

Backup/Restore

Users could backup the CloudUCM 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

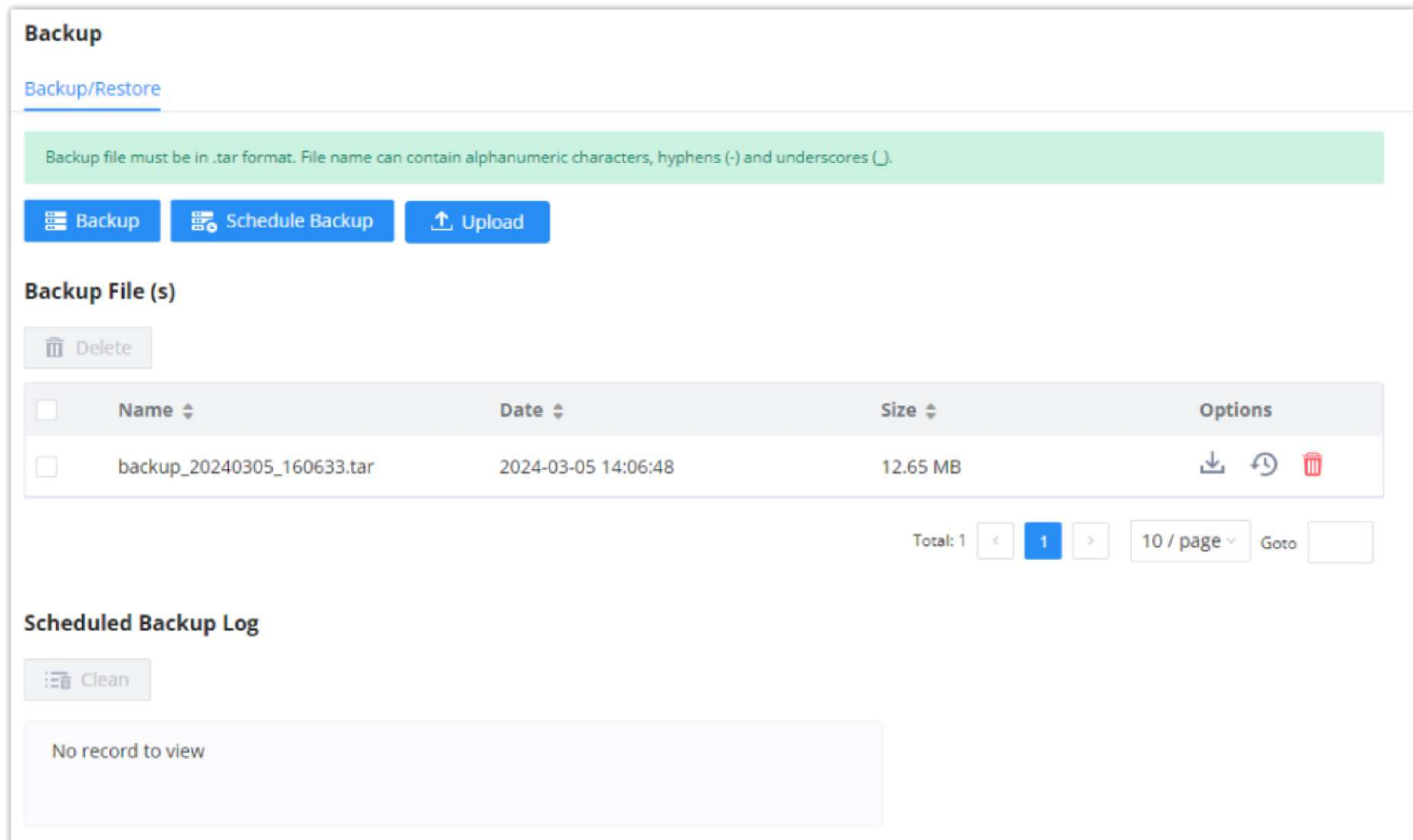
1. Choose the type(s) of files to be included in the backup.
2. Name the backup file.

3. 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 CloudUCM internal storage or the external device.


Click on  to upload backup file from the local device to CloudUCM. The uploaded backup file will also be displayed in the web page and can be used to restore the CloudUCM.

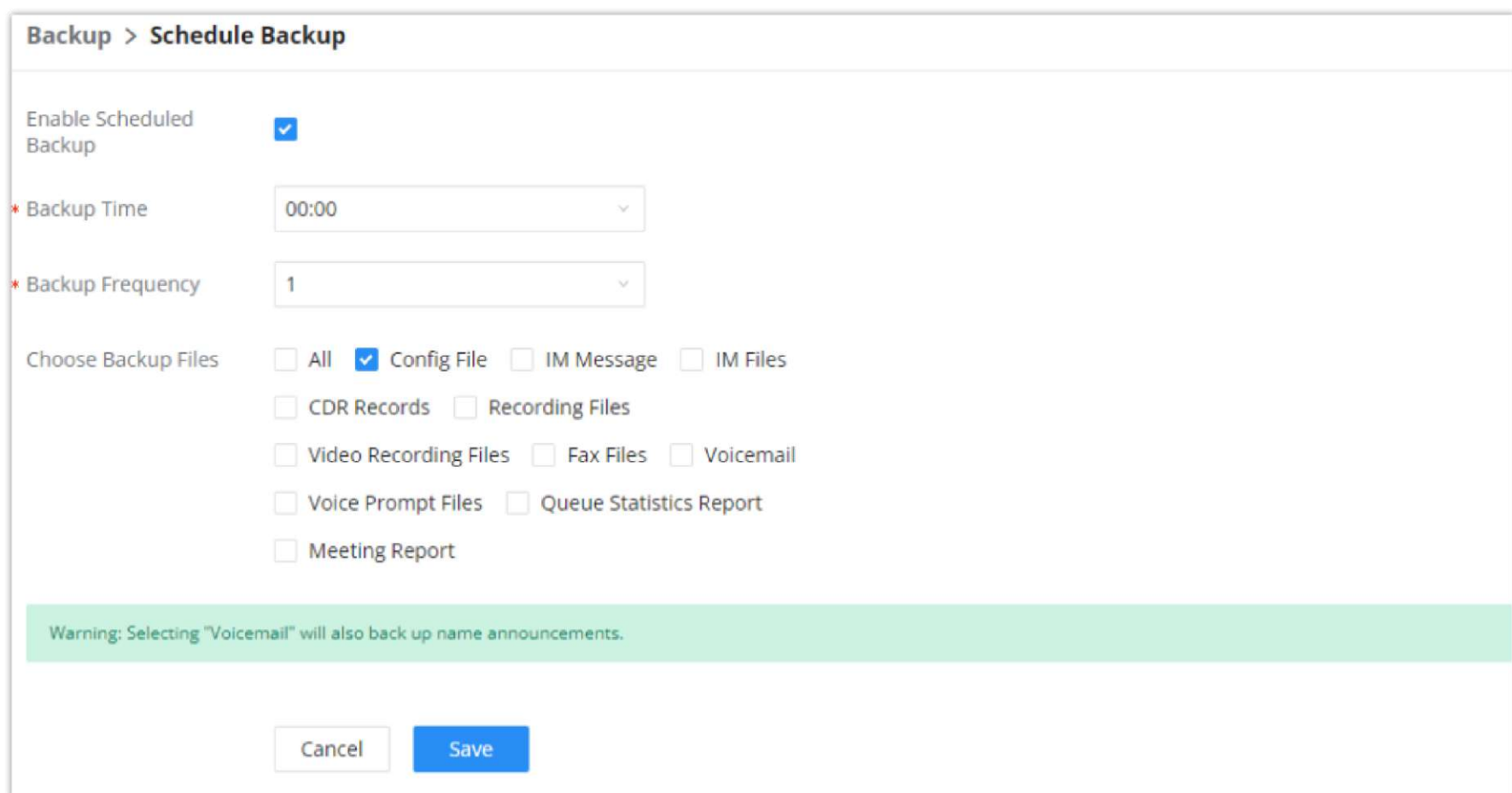
Please make sure the total number of extensions and total number of conference rooms are compatible before restoring to another UCM model. Otherwise it will prompt a warning and stop the restore process.



The screenshot shows the 'Backup' section of the interface. At the top, there is a breadcrumb 'Backup/Restore' and a green warning box stating: 'Backup file must be in .tar format. File name can contain alphanumeric characters, hyphens (-) and underscores (_).' Below this are three buttons: 'Backup', 'Schedule Backup', and 'Upload'. The main section is titled 'Backup File (s)' and contains a 'Delete' button. A table lists the backup files with columns for Name, Date, Size, and Options. One file is listed: 'backup_20240305_160633.tar' with a date of '2024-03-05 14:06:48' and a size of '12.65 MB'. The Options column for this file contains icons for download, refresh, and delete. Below the table is a pagination control showing 'Total: 1' and '10 / page'. The 'Scheduled Backup Log' section has a 'Clean' button and a message 'No record to view'.

Backup / Restore

The  option allows UCM to perform automatically backup on the user specified time. User is allowed to set backup time from 0-23 and how frequent the backup will be performed.



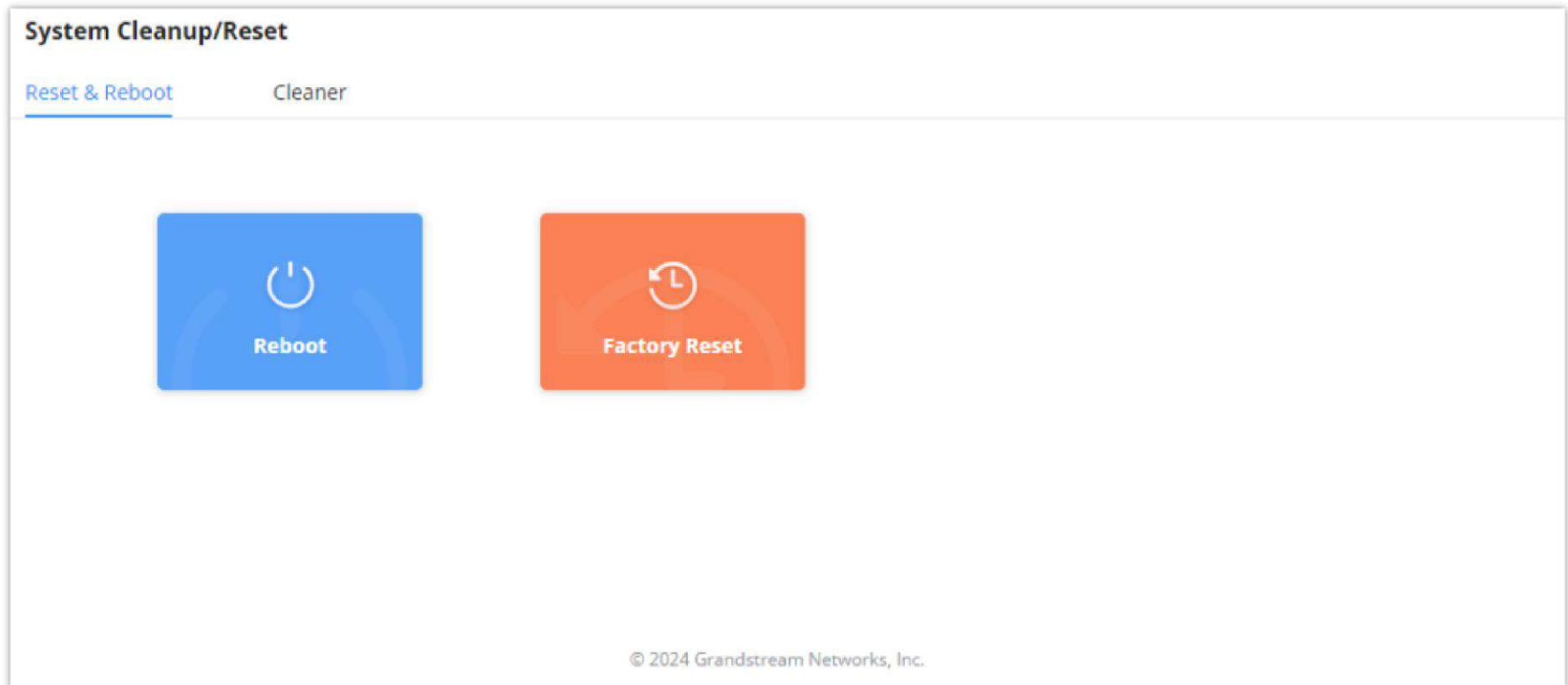
The screenshot shows the 'Schedule Backup' configuration page. It has a breadcrumb 'Backup > Schedule Backup'. The 'Enable Scheduled Backup' checkbox is checked. The 'Backup Time' is set to '00:00' and the 'Backup Frequency' is set to '1'. Under 'Choose Backup Files', several checkboxes are shown: 'All' (unchecked), 'Config File' (checked), 'IM Message' (unchecked), 'IM Files' (unchecked), 'CDR Records' (unchecked), 'Recording Files' (unchecked), 'Video Recording Files' (unchecked), 'Fax Files' (unchecked), 'Voicemail' (unchecked), 'Voice Prompt Files' (unchecked), 'Queue Statistics Report' (unchecked), and 'Meeting Report' (unchecked). A green warning box at the bottom states: 'Warning: Selecting "Voicemail" will also back up name announcements.' At the bottom are 'Cancel' and 'Save' buttons.

Local Backup

System Cleanup/Reset

Reset and Reboot

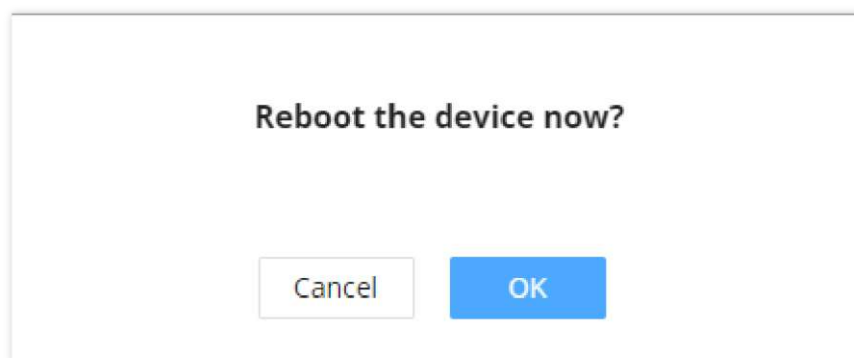
Users could perform reset and reboot under Web GUI→**Maintenance**→**System Cleanup/Reset**→**Reset and Reboot**.



Reset and Reboot

Reboot

When the user clicks on reboot, a confirmation prompt will appear. To proceed with rebooting the device, please click "OK".

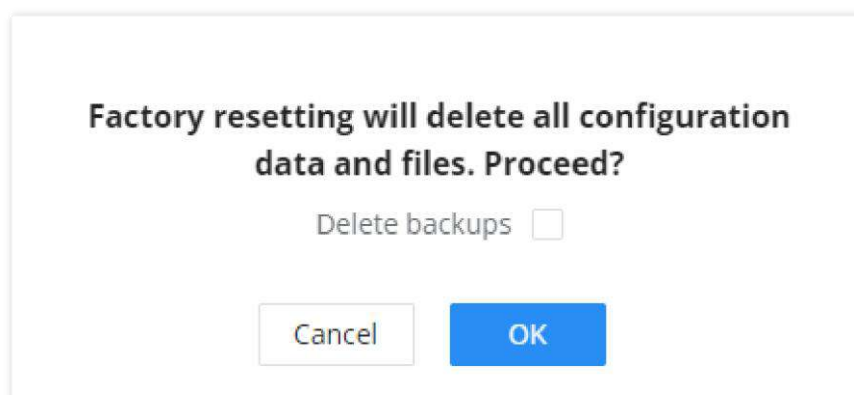


Reboot Confirmation Prompt

Reset

Reset function will delete all the configuration and data store in the CloudUCM device. After reset, the data and configuration cannot be recovered, please use this function with caution and always make sure that you have done a data and configuration backup before resetting the device to its initial configuration.

When performing a reset, the user can choose to remove the backup files. Once the user has chosen to remove the backup files, all the configuration, data, and backup files will be deleted and cannot be recovered. If you have taken a backup that you intend to restore after the reset, please do not tick the option "Delete Backups" as shown in the screenshot below.



It is highly recommended that you download all the backup files and store the files locally before resetting the device, to avoid accidental removal of the backup files..

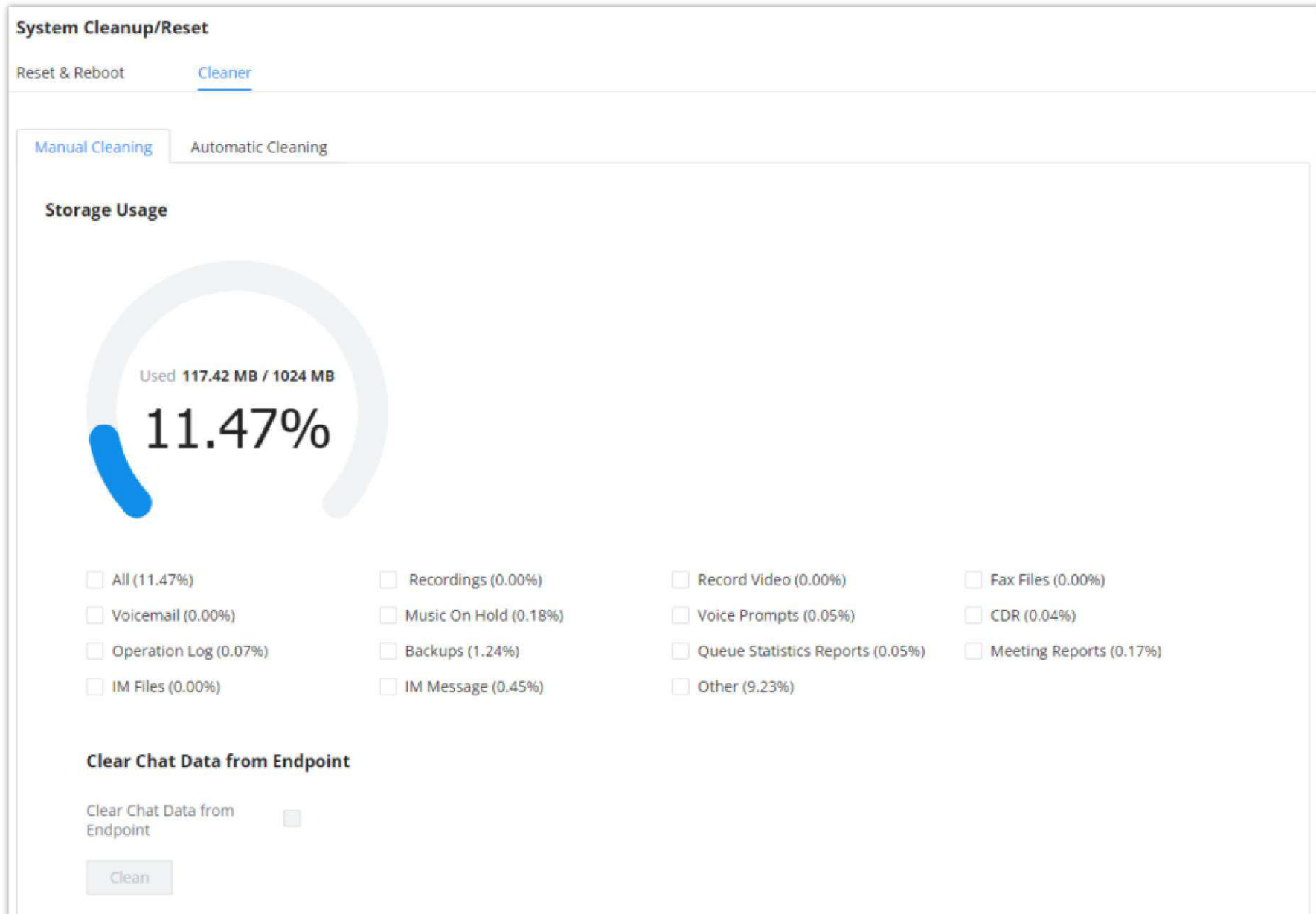
Warning

- Resetting the device will remove all the configuration and data of the device. Please proceed with caution.
- Ticking the option "Delete Backups" will result in deleting all the backup files store in the device. Please proceed with caution.

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 CloudUCM.



System Cleanup/Reset

Reset & Reboot [Cleaner](#)

[Manual Cleaning](#) [Automatic Cleaning](#)

Storage Usage

Used **117.42 MB / 1024 MB**

11.47%

<input type="checkbox"/> All (11.47%)	<input type="checkbox"/> Recordings (0.00%)	<input type="checkbox"/> Record Video (0.00%)	<input type="checkbox"/> Fax Files (0.00%)
<input type="checkbox"/> Voicemail (0.00%)	<input type="checkbox"/> Music On Hold (0.18%)	<input type="checkbox"/> Voice Prompts (0.05%)	<input type="checkbox"/> CDR (0.04%)
<input type="checkbox"/> Operation Log (0.07%)	<input type="checkbox"/> Backups (1.24%)	<input type="checkbox"/> Queue Statistics Reports (0.05%)	<input type="checkbox"/> Meeting Reports (0.17%)
<input type="checkbox"/> IM Files (0.00%)	<input type="checkbox"/> IM Message (0.45%)	<input type="checkbox"/> Other (9.23%)	

Clear Chat Data from Endpoint

Clear Chat Data from Endpoint

[Clean](#)

Manual Cleaning

UCM regularly cleans up CDRs, report data, chat data, recording files, historical appointment meeting records, voice mail, backup files, and fax files. The report data includes queue statistics report and conference room call statistics report; chat data includes chat messages and chat shared files; historical appointment conferences include audio and video conference appointment records. Automatic cleanup is not enabled by default and supports regular cleanup of database data based on dimensions such as cleanup time, cleanup conditions, and cleanup interval.

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 CloudUCM.

System Cleanup/Reset

Reset & Reboot [Cleaner](#)

Manual Cleaning [Automatic Cleaning](#)

Clean CDR, recordings, voicemail, fax, statistics report and IM data automatically.

CDR Cleaner

Enable Cleaner

Clean Time

Cleaning Conditions

Clean Interval (d)

Report Cleaner

Enable Cleaner

Data Type Queue Statistics Meeting Call
 Report Statistics Report
 Multimedia Meeting
 History

Clean Time

Cleaning Conditions

Clean Interval (d)

IM Data Cleaner

Enable Cleaner

Clear Chat Data from Fingerprint

Automatic Cleaning

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 Report Cleaner	Enable scheduled queue log cleaning. By default, is disabled.
Cleanup Type	<ul style="list-style-type: none"> ○ Queue Statistics Report ○ Meeting Call Statistics Report ○ Scheduled Audio Meeting History ○ Scheduled Video Meeting History
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 Conference Statistics Report Cleaner	Enable scheduled Conference 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 Conference Statistics Report has been reached, Conference 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 Conference 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.
Choose Cleaner File	Select the files for system automatic clean. <ul style="list-style-type: none"> ○ Basic Call Recording Files. ○ Conference 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 the cleaner's log.

Automatic Cleaning Configuration

All the cleaner logs will be listed on the bottom of the page.

Network Troubleshooting

Ethernet Capture

Ethernet Capture tool allows to capture the ethernet packets from the CloudUCM to inspect them for troubleshooting purposes.

- **SRTP Debugging:** Enabling this option allows saving SRTP keys which will be used to decrypt the SRTP packets. This is useful when troubleshooting issues related to RTP packets.

The screenshot shows the 'Network Troubleshooting' interface. Under the 'Ethernet Capture' section, there are two main areas: 'SRTP Debugging' and 'Regular Debugging'. In the 'SRTP Debugging' section, the 'Enable SRTP Debugging' checkbox is unchecked, and there is a 'Download' button. In the 'Regular Debugging' section, there is a 'Start' button. Below these sections is a table with columns for 'Capture File', 'Date', 'Size', and 'Options'. The table is currently empty, showing 'No data'. At the bottom, a note states: 'Note: The file will be deleted automatically after 30 days. Please share the capture file with Grandstream Technical Support for file decryption and analysis.'

Ethernet Capture

Service Check

Enable Service Check to periodically check CloudUCM service status. 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 CloudUCM. The default setting is 3. If there is no response from CloudUCM after 3 attempts (default) to check, current status will be stored and the internal service in CloudUCM will be restarted.

The screenshot shows the 'Service Check' configuration interface. It includes the following settings:

- 'Enable Server Check': checked (checkbox with a blue checkmark).
- '* Check Cycle': input field containing the value '60'.
- '* Check times': input field containing the value '3'.

Service Check

CDR

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 CloudUCM, 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.

CDR Filter

<p>Call Type</p>	<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.
<p>Status</p>	<p>Filter with the call status, the available statuses are the following:</p> <ul style="list-style-type: none"> ○ Answered ○ No Answer ○ Busy ○ Failed
<p>Source Trunk Name</p>	<p>Select source trunk(s) and the CDR of calls going through inbound the trunk(s) will be filtered out.</p>
<p>Destination Trunk Name</p>	<p>Select destination trunk(s) and the CDR of calls going outbound through the trunk(s) will be filtered out.</p>

Action Type	Filter calls using the Action Type, the following actions are available: <ul style="list-style-type: none">○ Announce○ Announcement page○ Dial○ Announcements○ Callback○ Call Forward○ Conference○ DISA○ Follow Me○ IVR○ Page○ Parked Call○ Queue○ Ring Group○ Transfer○ VM○ VMG○ Video Conference○ VQ_Callback○ Wakeup○ Emergency Call○ Emergency Notify○ SCA
Extension Group	Specify the Extension Group name to filter with.

<p>Export File Data</p>	<p>Select the fields that will be exported, the following fields are available:</p> <ul style="list-style-type: none"> ○ Account Code ○ Session ○ Premier caller ○ Action type ○ Source trunk name ○ Destination trunk name ○ Caller number ○ Caller ID ○ Caller name ○ Callee number ○ Answer by ○ Context ○ Start time ○ Answer time ○ End time ○ Call time ○ Talk time ○ Source channel ○ Dest channel ○ Call status ○ Dest channel extension ○ Last app ○ Last data ○ AMAFLAGS ○ UIQUEID ○ Call type ○ NAT
<p>Account Code</p>	<p>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.</p>
<p>Start Time</p>	<p>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.</p>
<p>End Time</p>	<p>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.</p>
<p>Caller Number</p>	<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	Enter the caller name to filter the CDR report. CDR with the matching caller name will be filtered out.
Callee Number	Enter the callee number to filter the CDR report. CDR with the matching callee number will be filtered out. Note: The "Callee Number" filter field supports specifying Pattern (example: 3XXX) or using Leading digits (example: 3.) as filtering options.

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

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

- o **Account Code**

Example format:

Grandstream/Test

- o **Status**

Answered, Busy, No answer or Failed.

Users could perform the following operations on the call report.

- o **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  **Delete All** button to remove all the call report information.




- **Delete Search Result**

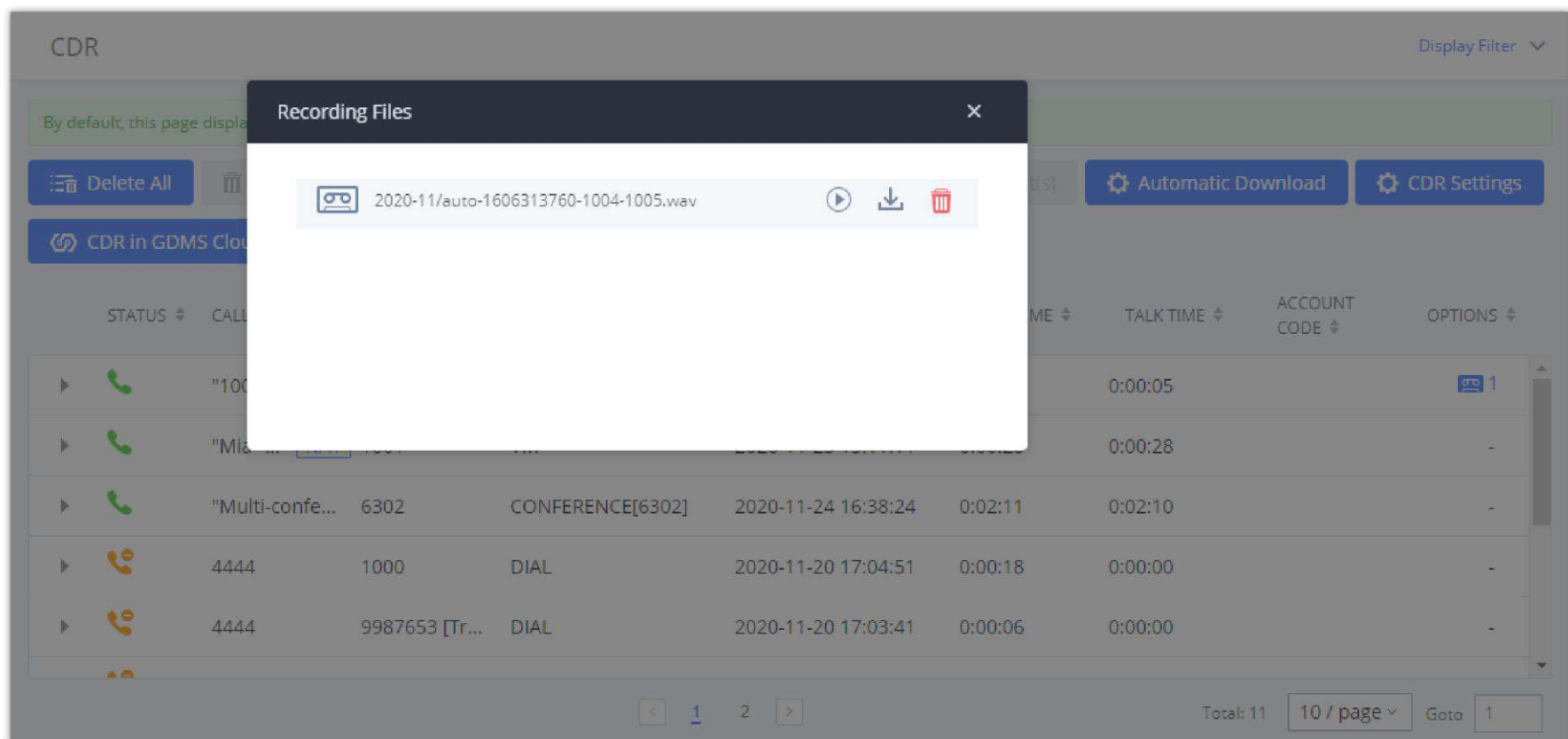
On the bottom of the page, click on  **Delete Search Result (s)** button to remove CDR records that appear on search results.

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

- **Automatic Download CDR Records**

User could configure the CloudUCM 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
✕

Automatically send the new CDR records to the configured Email at a certain period. If you want to upload the CDR records to an FTP/TFTP server, please go to the [Data Sync](#) page to configure.

Automatic

Download:

Delete Sent

Records:

Automatic By Day 0

Download

Period:

Email: GStes@gmail.com [Email](#)
[Template](#)

Cancel
OK

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.

users have the option to delete the sent records "Delete Sent Records"

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

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

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.

- o **Source Channel, Dest Channel**

Example:

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.

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**.

NAME	DATE	SIZE	OPTIONS
No Data			

CDR in GDMS Cloud

Statistics

UCM supports the function of concurrent call statistics. This function provides users with statistics on the number of concurrent calls of all VOIP trunks (SIP trunks). Users can set search criteria to generate custom charts. Select the trunk and time to view the chart of the maximum number of concurrent calls corresponding to the trunk in a certain day or month.

Statistics

CDR Statistics

Concurrent Call Statistics

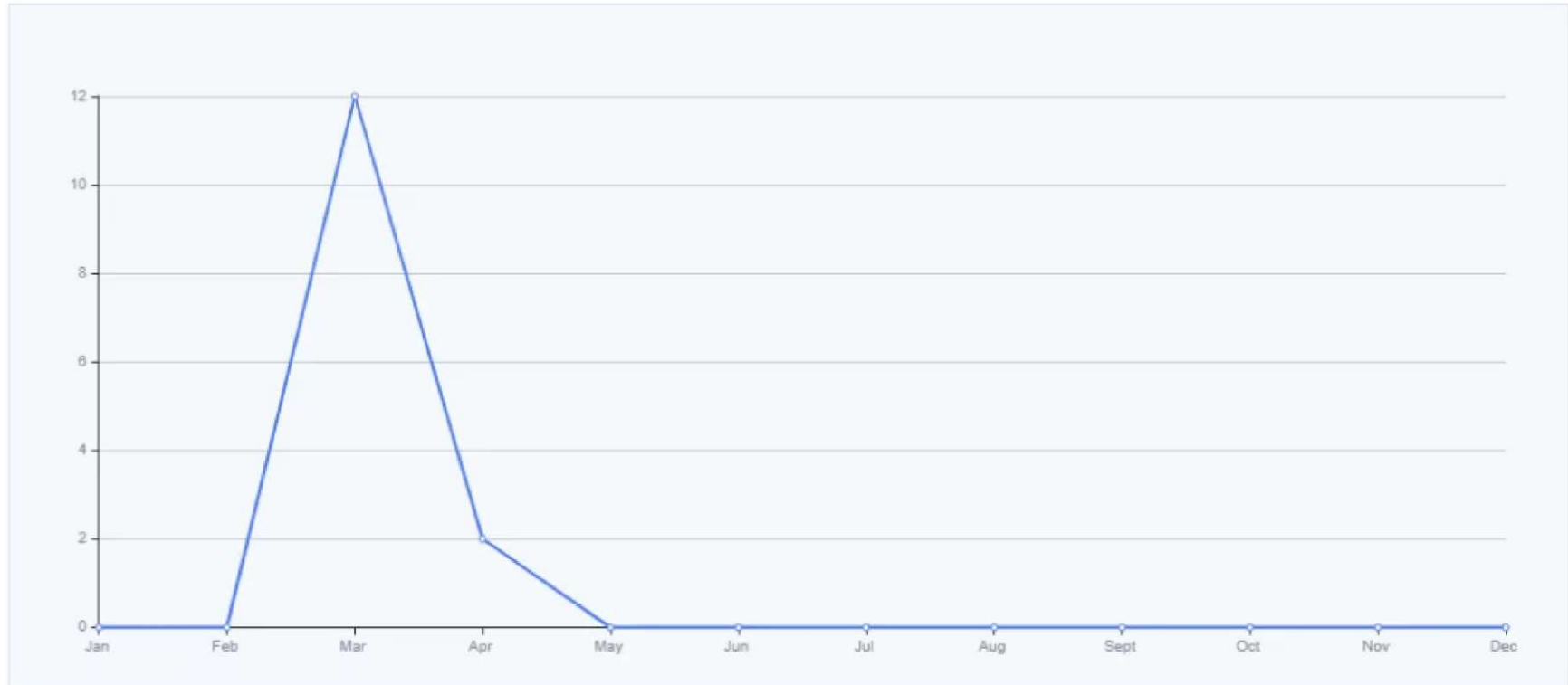
Time

By Month By Week By Day By Hour By Range

2024

CDR Statistics

All Calls Inbound Calls Outbound Calls Internal Calls External Calls



CDR Statistics

Trunk Type	Select one of the following trunk type. <ul style="list-style-type: none"><input type="radio"/> All<input type="radio"/> SIP Calls
Call Type	Select one or more in the following checkboxes. <ul style="list-style-type: none"><input type="radio"/> Inbound calls<input type="radio"/> Outbound calls<input type="radio"/> Internal calls<input type="radio"/> External calls<input type="radio"/> All calls
Time Range	<ul style="list-style-type: none"><input type="radio"/> By month (of the selected year).<input type="radio"/> By week (of the selected year).<input type="radio"/> By day (of the specified month for the year).<input type="radio"/> By hour (of the specified date).<input type="radio"/> By range. For example, 2016-01 To 2016-03.

CDR Statistics Filter Criteria

Recordings

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".



Recordings

[Recording Files](#) Video Recording Files

Basic Call Recordings ▾ 2024-04 ▾

<input type="checkbox"/>	Caller ▾	Caller Name ▾	Callee ▾	Callee Name ▾	Call Time ▾	Size ▾	Options
No data							

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.
- 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.

CLOUDUCM SERVICES

Plan

In this tab, the user can view information about the plan of the CloudUCM all the settings related.

Service Description

In this section, the user can view all the services related to the CloudUCM plan.

CloudUCM Services

Plan

Integrated Customer Service

Enterprise UI Customization

Statistics

Service Description

My Plan

Plan Settings

Current Plan: SOHO (Trial) 

[Renewal/Upgrade](#)

[Learn about CloudUCM](#)

CloudUCM is a cloud audio/video communications and collaboration solution with a rich and powerful suite of features such as audio/video calling, conferencing, Live Chat online customer service, and more. Experience official integrations with software such as CRMs, Whatsapp, Office365, etc., or create custom add-ins with Wave API and SDK to suit your own business needs.



Communicate Across The World

CloudUCM is a comprehensive cloud UC solution that any endpoint can seamlessly connect to regardless of physical location or network. Enjoy features like audio/video calling, conferencing, call centers, instant messaging, Live Chat online customer service and more!

[How to configure endpoints](#)

[Notify Employees to Use](#)



Wave — Communicate & Collaborate From Anywhere

Available on desktop, web, Android and iOS, Wave gives your team the power to communicate and collaborate from anywhere and anytime on any device. Whether they're in the office, traveling or working from home, Wave provides easy access to voice and video calling, meetings, instant messaging and more.

[Learn more](#)

[How to Use Wave](#)

[Notify Employees to Use](#)






My Plan

In "My Plan" tab, the user can view information related to the CloudUCM plan such as, plan name, plan expiration date, plan status, extensions, max concurrent calls, storage space, and admin portal. In addition to the SIP server address.

CloudUCM Services

Plan Integrated Customer Service Enterprise UI Customization Statistics

Service Description **My Plan** Plan Settings

Plan Name	SOHO 
Plan Expiration Date	5/7/2024
Plan Status	Trial
Extensions	5 / 20
Max Concurrent Calls	8
Storage Space	107.21 MB / 1.00 GB Cleaner
Admin Portal	https://myucm.cloud:8443 
 For improved security, it is recommended to prevent direct link access to Cloud UCM by going to GDMS and enabling "Only allow access via GDMS".	
SIP Server	myucm.cloud 
SIP Server	myucm.cloud:5061 
Wave 3rd Party Add-ins	Supported
Record	Supported

Bound Enterprise: Grandstream Networks, Inc. Service Site: EU Organization: Default

Plan Settings

In "Plan Settings" tab, the user can configure SIP extension synchronization between the CloudUCM and the GDMS platform. The user can also set alert events synchronization, configure remote login settings and passwordless remote access.

The screenshot shows the 'Plan Settings' tab within the 'CloudUCM Services' interface. The top navigation bar includes 'Plan', 'Integrated Customer Service', 'Enterprise UI Customization', and 'Statistics'. Below this, there are sub-tabs for 'Service Description', 'My Plan', and 'Plan Settings'. A header bar displays 'Bound Enterprise: Grandstream Networks, Inc.', 'Service Site: EU', and 'Organization: Default'. The main content area is divided into two sections: 'General' and 'Passwordless Remote Access'. Under 'General', there are three items: 'SIP Extension Sync' with a checked checkbox, 'Alert Events Sync' with a checked checkbox, and 'Remote Login Settings' with a 'Go to Page' link. Under 'Passwordless Remote Access', there is one item: 'Enable Passwordless Remote Access' with an unchecked checkbox.

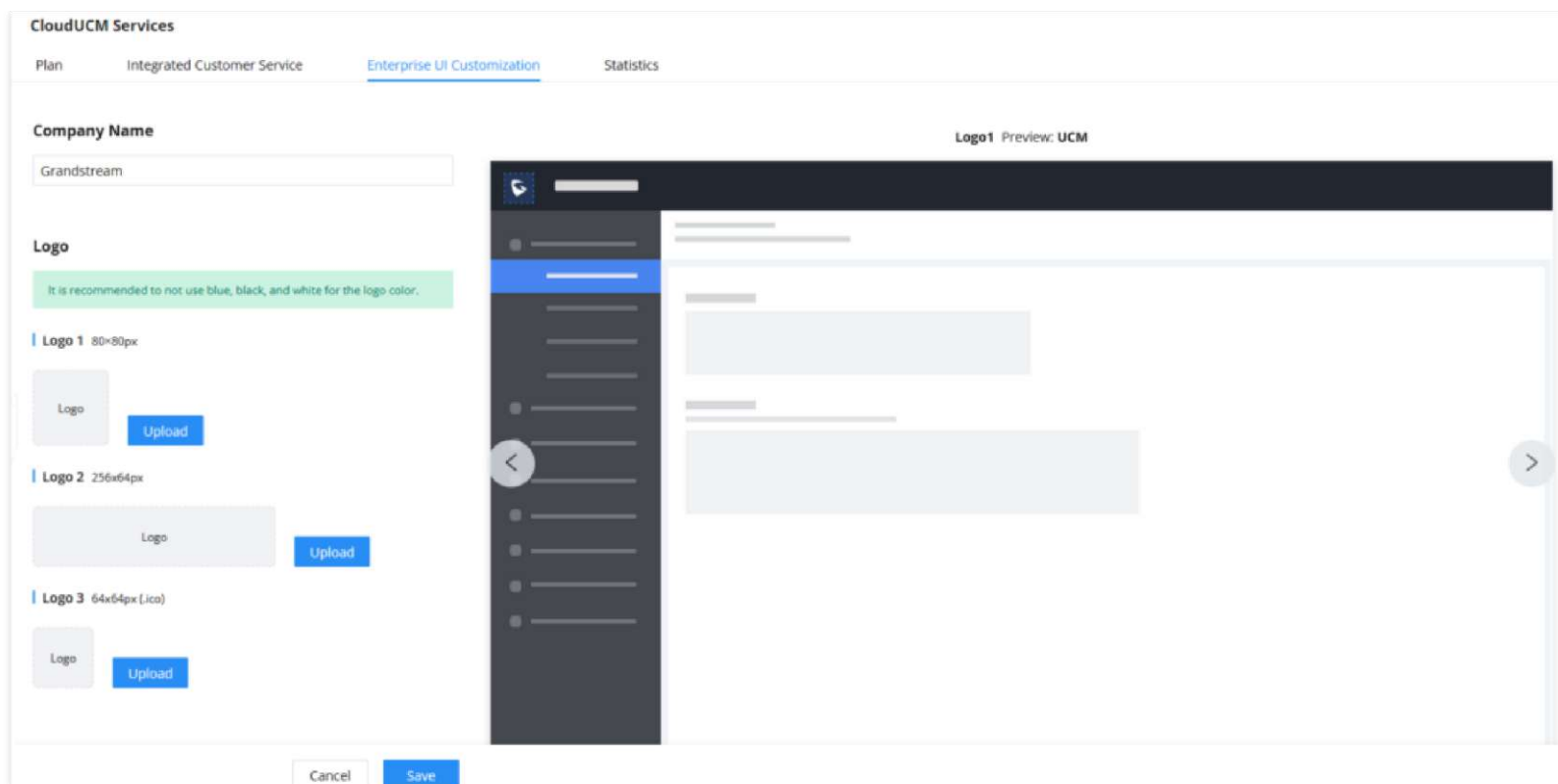
Integrated Customer Service

In "Integrated Customer Service", the user can access the sections for WebRTC Trunks configuration and Live Chat Customer Service.

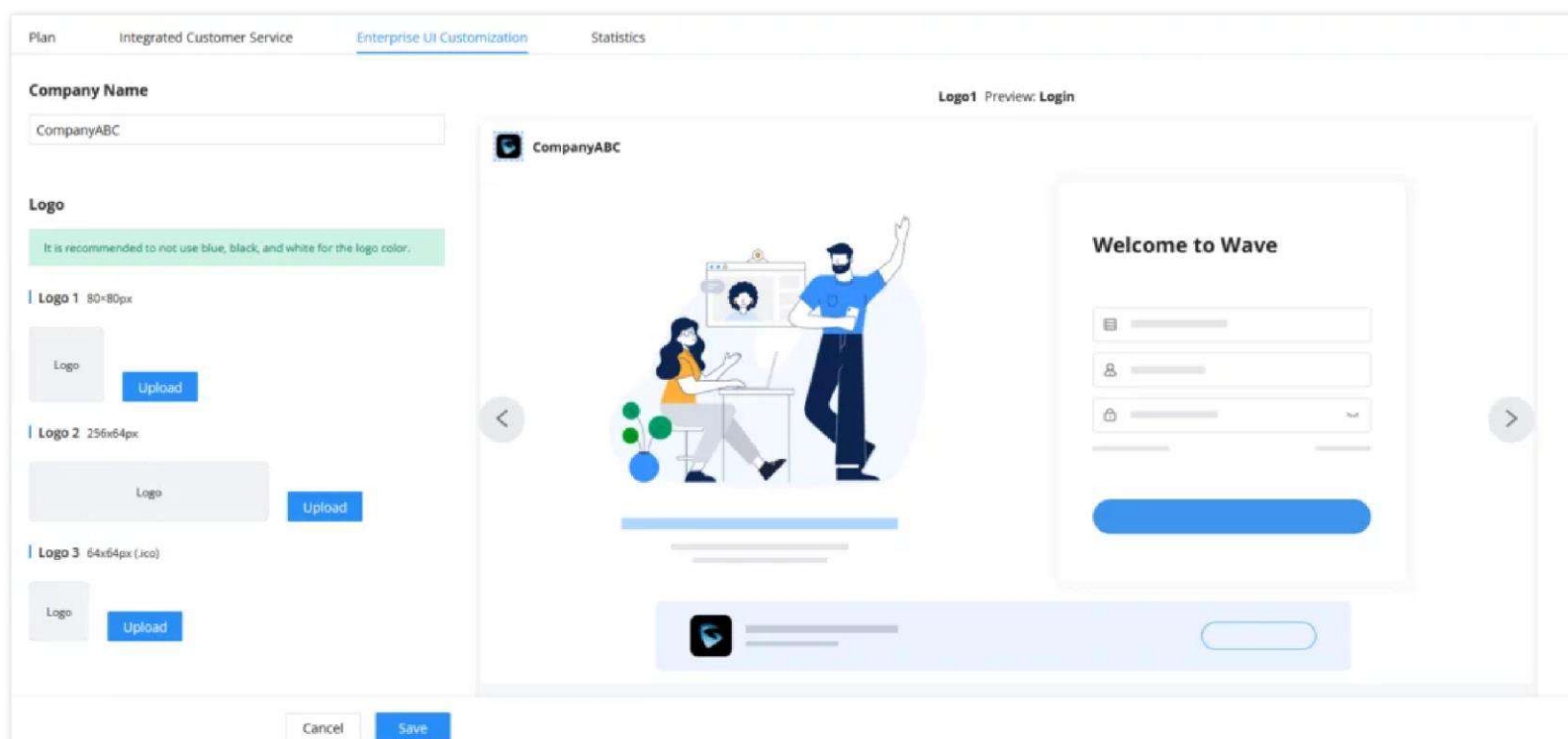
The screenshot shows the 'Integrated Customer Service' tab within the 'CloudUCM Services' interface. The top navigation bar includes 'Plan', 'Integrated Customer Service', 'Enterprise UI Customization', and 'Statistics'. Below this, there are sub-tabs for 'Call Customer Service', 'WebRTC Trunks', and 'Live Chat Customer Service'. The 'Call Customer Service' section contains two items: 'WebRTC Trunks' with a 'Go to Page' link, and 'Live Chat Customer Service' with a 'Go to Page' link. At the bottom of the page, there are two buttons: 'Cancel' and 'Save'.

Enterprise UI Customization

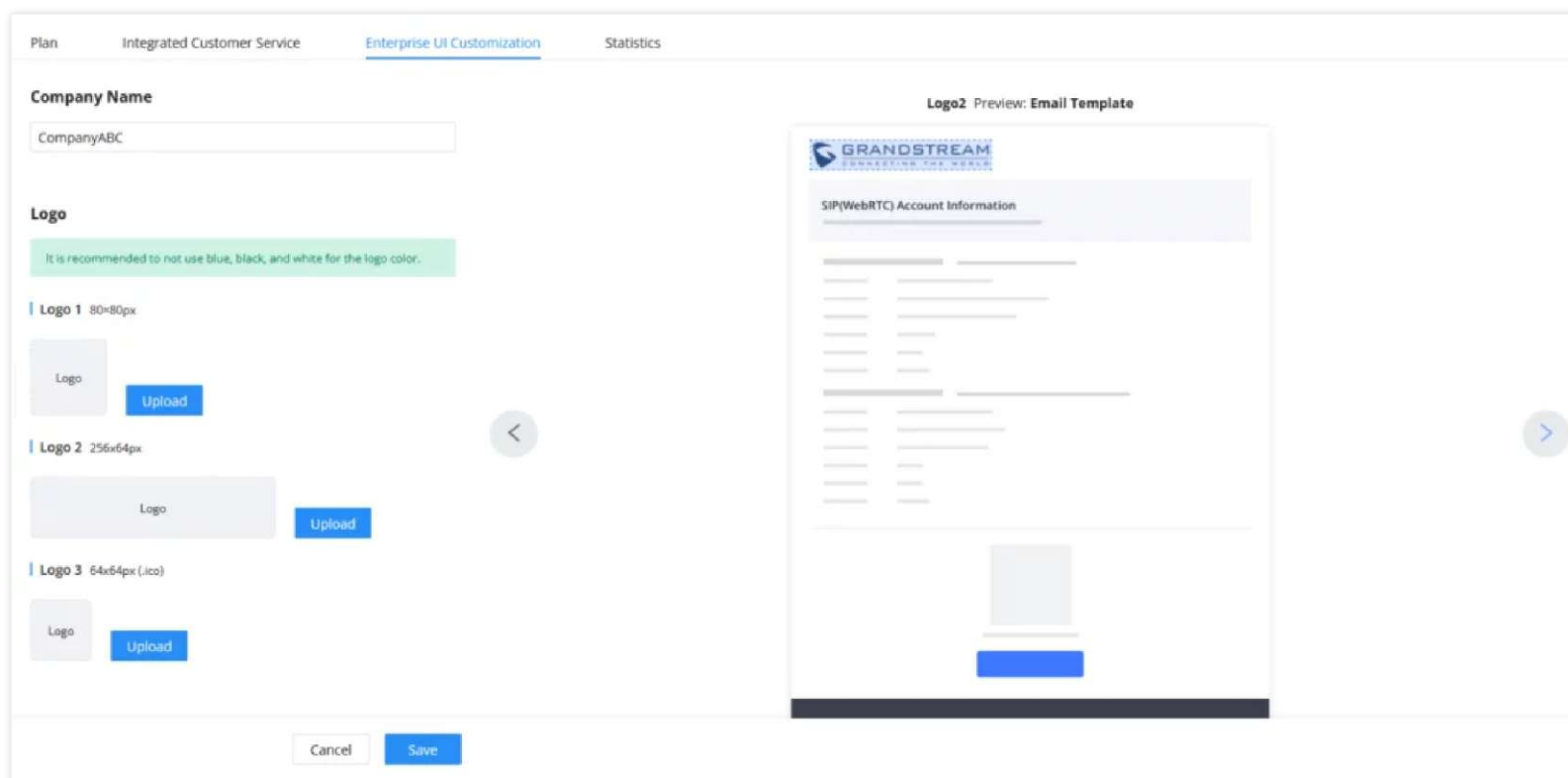
On the **Web GUI → RemoteConnect → Enterprise UI Customization** page, users can edit the company name and select a local image file as the new logo. The company name acts on the text part with the logo, and the pictures are in different formats and sizes according to the logo position, which are 64*64px (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.



CloudUCM UI Customization

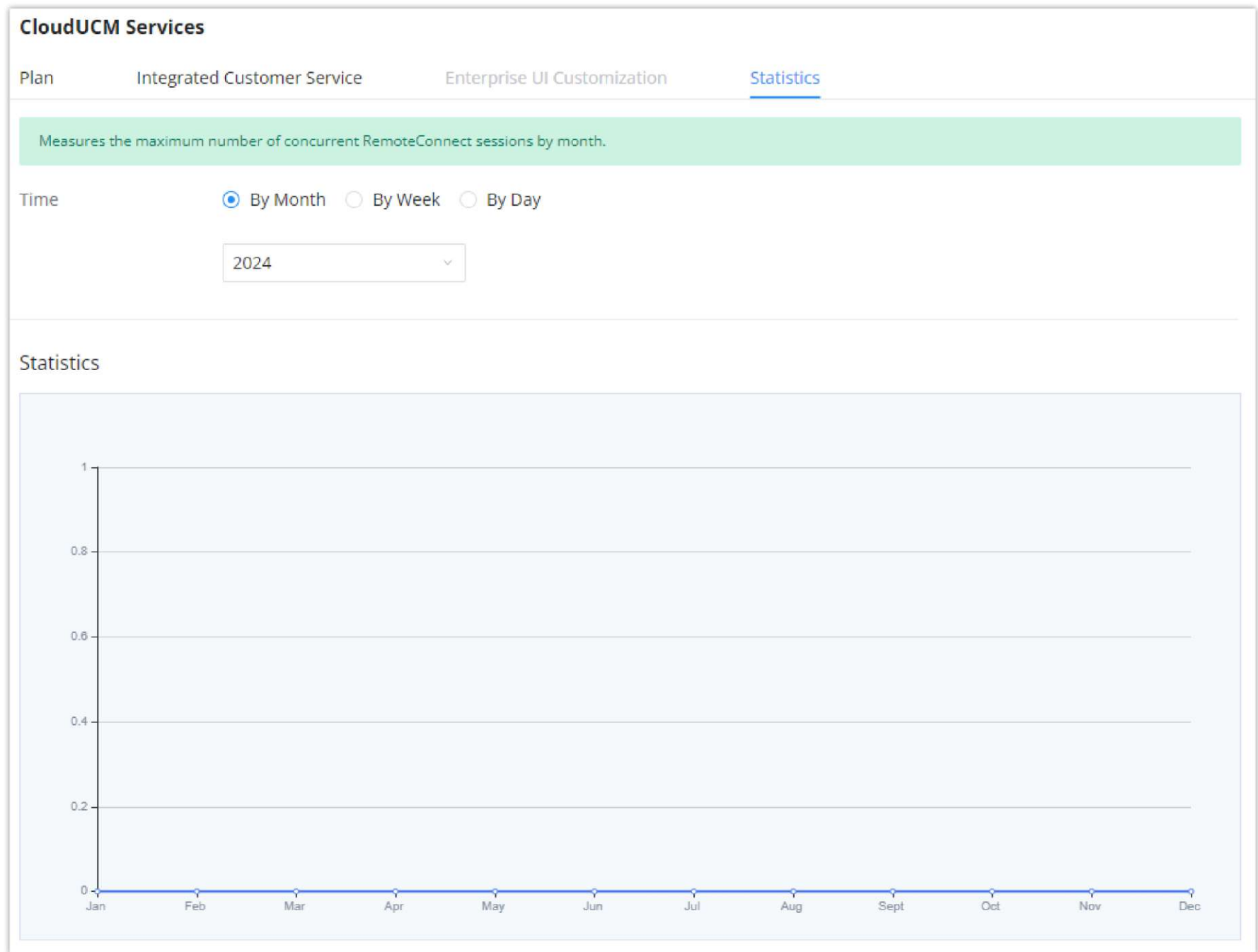


Wave UI Customization



Email Customization

Statistics



INTEGRATIONS

API Configuration

CloudUCM API allows creating an interface between the UCM and an application to send HTTP requests to the UCM, either to configure the UCM to get information from it.

API Settings

Before accessing the API, the administrators need to enable API and configure the access/authentication information on the CloudUCM first under **Integrations → API Configuration**. The API configuration parameters are listed in the tables below.

Username	The username for API Authentication.
Password	The password for API Authentication.
Call Control	If enabled, 3rd 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 and register as this user. By default, this is not configured, allowing all IP addresses to register to this extension. The format is: "xxx.xxx.xxx.xxx/255.255.255.255"

You can create multiple users with different login credentials to access to the HTTPS API.

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 now new queries listed below which will accomplish certain requests and get data about different modules on CloudUCM.

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
listIVR
addIVR
getIVR
updateIVR
deleteIVR
cdrapi
recapi
pmsapi
queueapiget
getPinSets
addPinSets
updatePinSets
deletePinSets
cleanTerminalChatInformation
getSIPAccountQR
getCallQueuesMemberMessage
getQueueCalling

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.

API Configuration Parameters

Upload Voice Prompt via API

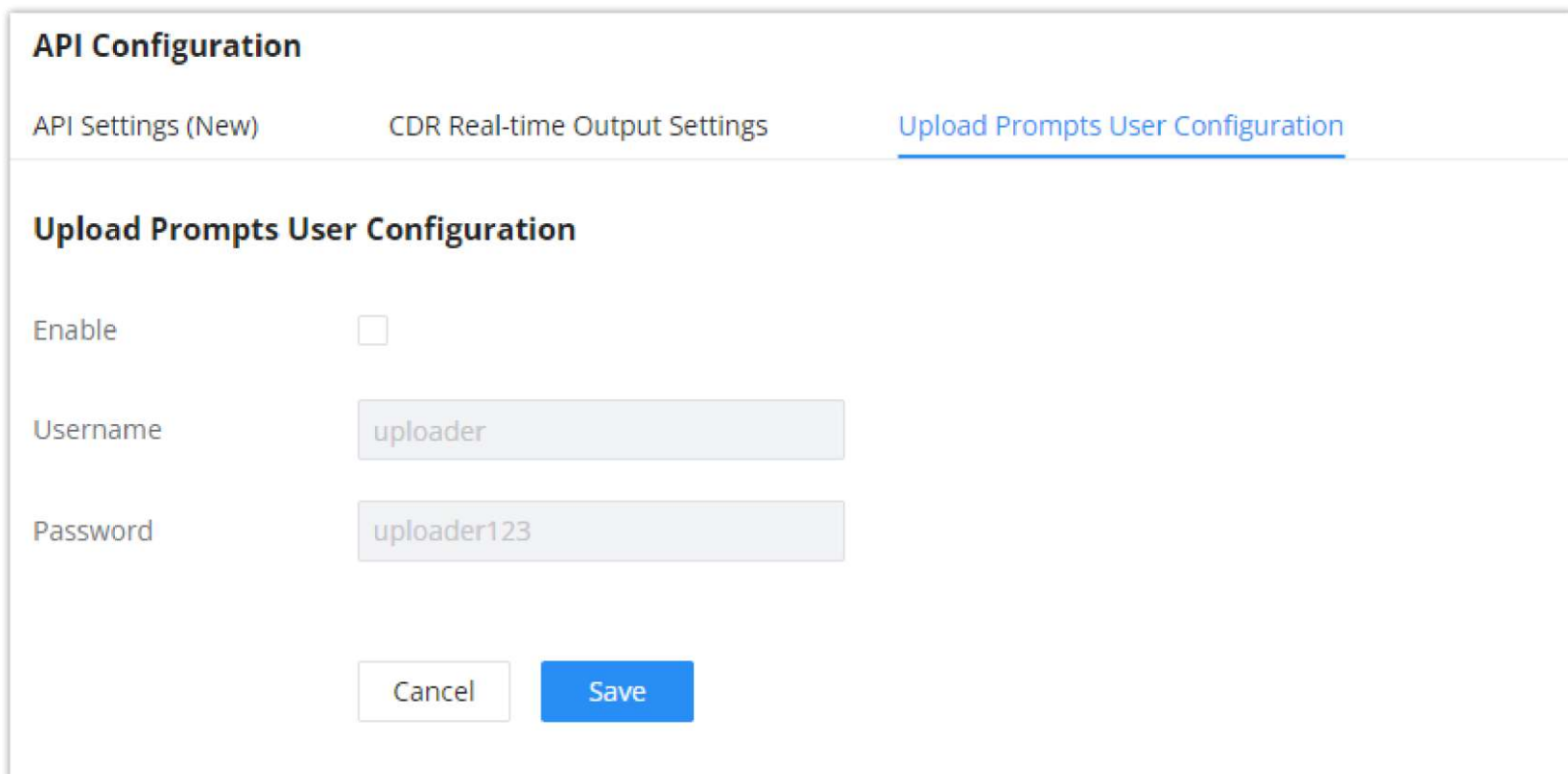
Customers now can use the "Upload Prompts User Configuration" to upload/replace voice prompt files as an alternative method to the manual upload method on UCM **PBX Settings**→ **Voice Prompt**→ **Custom Prompt**.

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 **Integrations → API Configuration → Upload Prompts User Configuration**. By default, the username and password for voice prompt user are "Username: uploader; Password: uploader123".



API Configuration

API Settings (New) CDR Real-time Output Settings Upload Prompts User Configuration

Upload Prompts User Configuration

Enable

Username

Password

Upload Prompt User Configuration

2. Hash the password of the user configured to an MD5 hash format.
3. Set the permission on the FTP server to Anonymous on the local computer hosting the FTP server and make sure that the default FTP port 21 is used.
4. Send an HTTP/HTTPS command to trigger the Prompt file upload on the UCM. If UCM's HTTP server is set to HTTPS, the example of the request sent to the UCM is:

```
https://192.168.124.89:8089/cgi?
action=uploadprompt&username=uploader&password=9191a6394c21b3aabd779213c7179462&filename=test.mp3
```

If UCM's HTTP server is set to HTTP, the example of the request sent to the UCM is:

```
https://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.

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

<https://documentation.grandstream.com/knowledge-base/cdr-rec-api/>

CRM

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 CloudUCM support the following CRMs: SugarCRM, VtigerCRM, 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 **Features**→**CRM**.

SugarCRM Basic Settings

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

CRM System	Select a CRM system from the dropdown menu.
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 to select where the UCM can perform the lookups on the CRM tables, Leads, Accounts, and Contacts.

SugarCRM Settings

Once settings on admin access are configured:

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

Vtiger CRM

Configuration page of the VtigerCRM can be accessed via admin login, on the UCM WebGUI go to **Integrations > CRM**.

VtigerCRM Basic Settings

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

CRM System	Select a CRM system from the dropdown menu.
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 to select where the UCM can perform the lookups on the CRM tables, Leads, Organizations, and Contacts.

VtigerCRM Settings

Once settings on admin access are configured:

2. Click on and .
3. Logout from admin access.
4. Login to the UCM as user and navigate under "**Other Features > 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.

Salesforce CRM

Configuration page of the Salesforce CRM can be accessed via admin login, on the UCM Web GUI **Integrations > CRM**.

The screenshot shows the 'CRM' configuration interface. It includes a dropdown for 'CRM System' set to 'Salesforce', a dropdown for '* Add Unknown Number' set to 'Contacts', and a 'Contact Lookups' section with two panes: '1 item Available' containing 'Look up in Leads table' and '2 items Selected' containing 'Look up in Contacts table' and 'Look up in Accounts table'. Navigation arrows are visible between the panes.

Salesforce Basic Settings

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

CRM System	Select a CRM system from the dropdown menu, four CRM systems are available: SugarCRM, VtigerCRM, 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.

Salesforce Settings

Once settings on admin access are configured:

2. Click on and .
3. Logout from admin access.
4. Login to the UCM as user and navigate under "**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 and . The status will change from "Logged Out" to "Logged In". User can start then using Salesforce CRM features.

The screenshot shows the 'CRM User Settings' form. It has a checked checkbox for 'Enable CRM'. Below are input fields for '* Username' (user@domain), '* Password' (pjdajdlka123@!), and '* Security Token' (mkjhamjkhndjkeFZEfljxwa!@jkjhbamklcel). A 'Login Status' field is also present at the bottom.

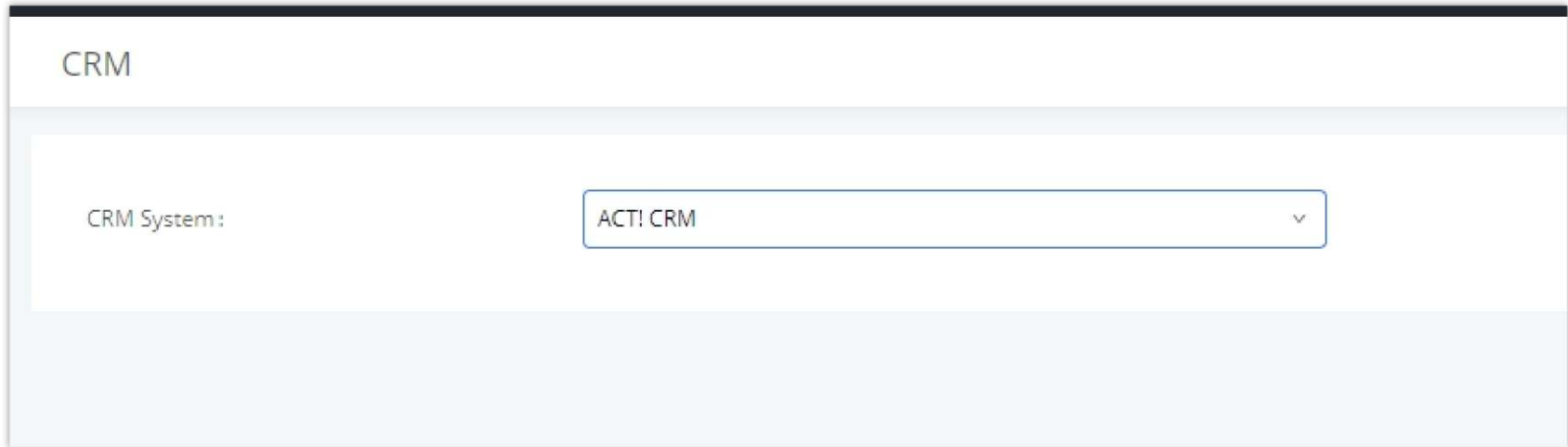
Salesforce User Settings

ACT! CRM

Configuration page of the ACT! CRM can be accessed via admin login, on the UCM Web GUI under **Integrations > CRM**

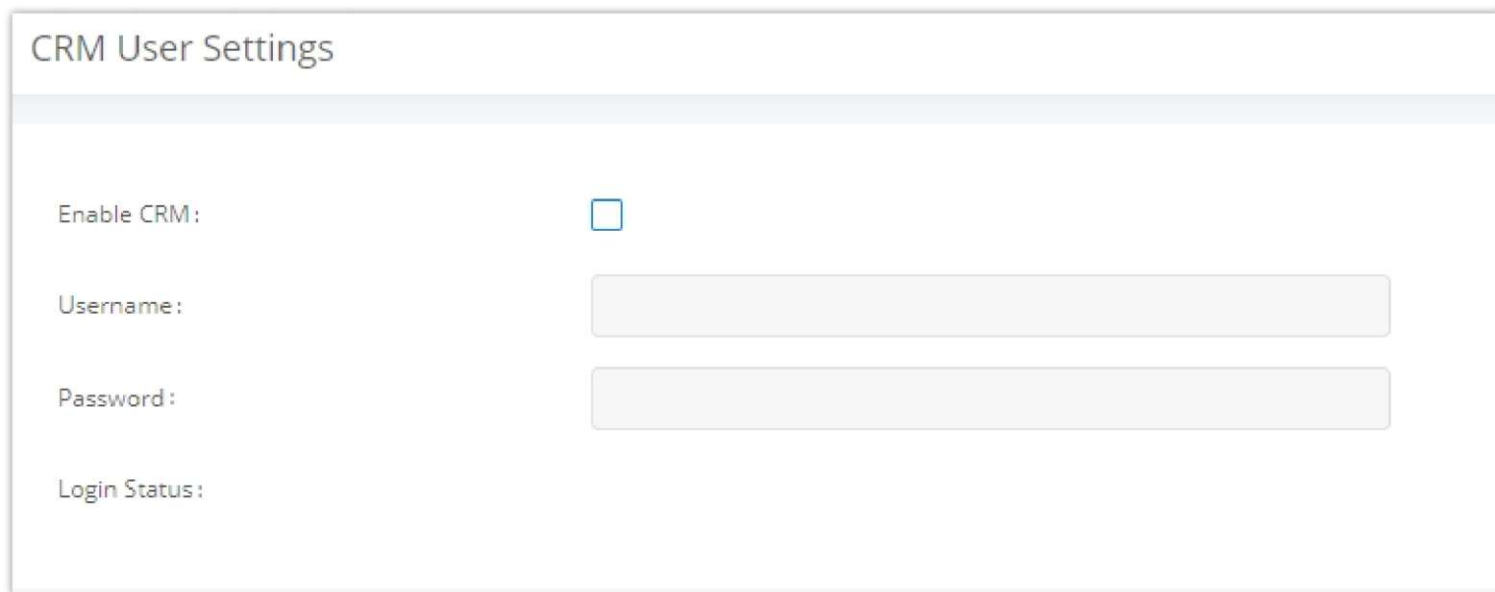
The configuration steps of the ACT! CRM are as follows:

1. Navigate to **Integrations > CRM** and select the "ACT! CRM" option.



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>



Enabling CRM on the User Portal

PMS

CloudUCM 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→**Integrations→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 > **Integrations > PMS > Basic Settings** set the connection information for the PMS platform.

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 • PMS API • Local PMS
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
Username	This username is used to authenticate into the PMS API.
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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Create New Room

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

Room Status

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

* Expected Departure Date

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		Wakeup Service		Maid
Check-in/Check-out History			Custom Room Status Codes					
ROOM NUMBER	CHECK-IN STATUS	CHECK IN / CHECK OUT	ROOM STATUS	CUSTOMER NAME	GUEST CATEGORY CODE	ARRIVAL DATE	EXPECTED DEPARTURE DATE	OPTIONS
1000	● Checked in	Check Out	Available			2023-07-04 11:40:00	2023-07-07 09:40:00	🏠
1001	● Not checked in	Check In	Not checked in	Arthur Morgan				🏠
1002	● Not checked in	Check In	Not checked in	Bonnie MacFarlan				🏠
1003	● Not checked in	Check In	Not checked in	Catherine Braitwaite				🏠
1004	● Not checked in	Check In	Not checked in	John Marston				🏠
1005	● Not checked in	Check In	Not checked in	Abigail Roberts				🏠
1006	● Not checked in	Check In	Not checked in	Mary-Beth Gaskill				🏠
1007	● Not checked in	Check In	Not checked in	Hosea Matthews				🏠
1008	● Not checked in	Check In	Not checked in					🏠
1009	● Not checked in	Check In	Not checked in					🏠

Room Check-in

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.

Custom Room Status Codes

The user can customize the existing room statuses or add more statuses along with the corresponding name. The user can customize the status code to up to 16-digit code. To customize room status, please click on [Custom Room Status Codes](#).

Press	Status Code	Room Status
Press 1	1	Available
Press 2	2	Cleaning
Press 3	3	Repairing
Press 4	4	Vacant
Press 5	5	Dirty
Press 6	6	Closed

Custom Room Status Codes

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.

Sequence	Prefix	Starting Cost	Starting Time (seconds)	Rate	Billing Unit (sec)	Options
No data						

Add Call Rate
✕

Call Charge = Starting Cost + Rate x Billing Unit

Prefix

Starting Cost

Starting Time (seconds)

* Rate

* Billing Unit (sec)

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:

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

PMS Wake up Service

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

This call has been executed but has been rejected, that why we can see the **"Busy"** status.

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 by the housekeeper.

Enable Mini Bar

Increase Mini Bar Usage Code

Decrease Mini Bar Usage Code

Global Tax Rate (%)

Prompt [Upload Audio File](#)

Skip Housekeeper and Password Authentication

Enable Multi-Item Billing

[+ Add Purchasable Items](#)

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 (*).

Housekeeper

In order to create a new housekeeper, click on [+ Add](#) under UCM WebGUI→**Integrations**→**PMS**→**Housekeeper**.

PMS > Create New Housekeeper

* Housekeeper Code

* Password

Create New Housekeeper

Housekeeper Code	Enter the Code to use when the housekeeper wants to change the status of the room.
Password	Enter the password associated with the housekeeper.

Create New Housekeeper

Note

Please note that you can dial the **"Update PMS Room Status"** feature code, the **"Housekeeper"** feature code, and the **"Room Status"** feature code all at once to change the room status.

Local PMS

CloudUCM 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, and Housekeeper functions, the CloudUCM allows a number of 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

Room Management
Room Status
Wakeup Service
Mini Bar

Cancel
Save

PMS Module:

Local PMS

Wakeup Prompt:

Wake Call

Upload Audio File

Back Up Voicemail Recordings :

Automatically Clear Wakeup Calls :

None

Automatically Clear Wave Chat History :

None

Automatically Reset User/Wave Password :

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Local PMS

Field	Description
PMS Module	Users can select the desired PMS module from the drop-down list. <ul style="list-style-type: none"> Hmobile Mitel HSC IDS PMS API
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
Username	This username is used to authenticate into the PMS API.
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	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.

Google Services

Google Services integration allows integrating the CloudUCM with Google Calendar to automatically synchronize the created Multimedia and Onsite Meetings schedule with Google Calendar of the host and the participants. In order to use this integration the user needs to enable API on Google Cloud Console and obtain the Client ID and the Client Secret, then enter the authorized redirect URIs.

Google Calendar Authorization

In Google Calendar Authorization configuration page, please enter the generated OAuth2.0 Client ID, OAuth2.0 Client Secret, and Authorized redirect URIs to integrate the CloudUCM with Google Calendar.

Google Services

[Google Calendar Authorization](#) [Google Calendar Settings](#)

OAuth2.0 Authentication

* OAuth2.0 Client ID

* OAuth2.0 Client Secret

Authorized redirect URIs

Google Calendar Authorization

1. Click "Get Authorization Code".
2. Enter the Google account and password (Note: Please make sure that the account information on the authorization page is correct. If you are not logged into the correct account, please log out and log back into the correct one.).
3. Click "Accept" on authorization page.
4. Copy the string to the Authorization Code input box, then click the "Authorize" button.

* Authorization Code

Google Calendar Settings

On Google Calendar Settings configuration page the user can enable "Google Calendar Auto Refresh Synchronization" to synchronize the meetings created automatically with Google Calendar. The user can also create a calendar label to mark different events on the calendar with customized labels.

To learn more about labels, please refer to the following link: <https://support.google.com/calendar/answer/12377581>


Google Services

Google Calendar Authorization [Google Calendar Settings](#)

Google Calendar Auto Refresh Synchronization

Calendar Label

[+ Add](#)


Name ↕	Description ↕	Options
 No data		

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

Wave Integration & Expansion

Wave integrations include further expansion of the UC features offered by our solutions. These expansions includes integrating popular CRM solutions, like Odoo CRM, Oracle Netsuite, and Salesforce CRM with Wave to easily manage customer relationship directly through the calls received on Wave. Other add-ins include Whatsapp Business, Office356, Google Drive, and IPVideoTalk.


Wave Integration & Expansion



Wave Add-ins — Connect Your Work

Wave can be integrated with a wide variety of services and applications, including popular CRM platforms, Whatsapp Business, Office365, Wave for Outlook, Microsoft Teams and more.

[View all Wave plugins](#)



Integrate Wave with your applications

Wave provides an SDK that opens up integration with other 3rd party software, allowing for the incorporation of audio/video calling functionality into various types of applications.

[View Wave SDK](#)

CHANGELOG

Firmware 1.0.25.11

- This is the initial release.