CALL ROUTES

OUTBOUND ROUTES

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

Go to Web GUI->**PBX**->**Basic/Call Routes**->**Outbound Routes** to add and edit outbound rules.

- Click on "Create New Outbound Rule" to add a new outbound route.
- Click on to edit the outbound route.
- Click on to delete the outbound route.
- On the UCM6200, the outbound route priority is based on “Best matching pattern”. For example, the UCM6200 has outbound route A with pattern 1xxx and outbound route B with pattern 10xx configured. When dialing 1000 for outbound call, outbound route B will always be used first. This is because pattern 10xx is a better match than pattern 1xxx. Only when there are multiple outbound routes with the same pattern configured, users can click on to move the outbound route up/down to arrange the priority among those outbound routes.

Table 56: Outbound Route Configuration Parameters

Calling Rule Name	Configure the name of the calling rule (e.g., local, long_distance, and etc). Letters, digits, _ and - are allowed.
Pattern	• All patterns are prefixed with the "_". • Special characters: 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.
Password	Configure the password for users to use this rule when making outbound calls.
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.
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.
Privilege Level	Select privilege level for the outbound rule. • Internal: The lowest level required. All users can use this rule. • Local: Users with Local, National, or International level are allowed to use this rule. • National: Users with National or International level are allowed to use this rule. • International: The highest level required. Only users with 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. Please be aware of the potential security risks when using "Internal" level, which means all users can use this outbound rule to dial out from the trunk.
Enable Filter on Source Caller ID	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". The following two methods can be used at the same time to define the extensions as the source caller ID. 1. Select available extensions/extension groups from the left to the right. 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. • All patterns are prefixed with the "_". • Special characters: 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.
Send This Call Through Trunk	
Use Trunk	Select the trunk for this outbound rule.
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 a long distance calls. However, 9 should not be sent out via analog lines and the PSTN line. In this case, 1 digit should be stripped before the call is placed.
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. Example: The user's primary trunk is a VoIP trunk and the user would like to use the PSTN when the VoIP trunk is not available. The PSTN trunk can be configured as the failover trunk of the VoIP trunk.
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 a long distance calls. However, 9 should not be sent out via analog lines and the PSTN line. In this case, 1 digit should be stripped before the call is placed.
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.

INBOUND ROUTES

Inbound routes can be configured via Web GUI->**PBX**->**Basic/Call Routes**->**Inbound Routes**.

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

- Click on to delete the inbound route.

INBOUND RULE CONFIGURATIONS

Table 57: Inbound Rule Configuration Parameters

Trunks	Select the trunk to configure the inbound rule.
DID Pattern	• All patterns are prefixed with the "_". • Special characters: 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. • The pattern can be composed of two parts, divided by a '/' character. The first part is used to specify the dialed number the second part is used to specify the caller ID and it is optional, if set it means only the extension with the specific caller ID is allowed to call in or call out. For example, patter '_2XXX/1234' means the only extension with the caller ID '1234' is allowed to use this rule.
Prepend Trunk Name	Prepend trunk name to display
Alert-Info	Configure the Alert-Info, when UCM6200 receives an INVITE request, the Alert-Info header field specifies an alternative ring tone to the UAS.
Inbound Multiple Mode	Multiple mode allows user to switch between destinations of the inbound rule by feature codes. Configure related feature codes in the "**Feature Codes**" page. If this option is enabled, user can use feature code to switch between different destinations.
Default Destination	Select the default destination for the inbound call. • Extension • Voicemail • Conference Room • Queue • Ring Group • Paging/Intercom • Voicemail Group • Fax • DISA • IVR • Dial By Name

	• External Number • By DID When "By DID" is used, the UCM6200 will look for the destination based on the number dialed, which could be local extensions, conference, call queue, ring group, paging/intercom group, IVR, voicemail groups and Fax extension as configured in "DID destination". If the dialed number matches the DID pattern, the call will be allowed to go through.
Strip	Configure the number of digits to be stripped from the beginning of the DID. This option shows up only when "By DID" is selected.
Prepend	Configure the number of digits to be prepended to an inbound DID pattern, with strip taking precedence over prepend.
Dial Trunk	This option shows up only when "By DID" is selected. If enabled, the external users dialing in to the trunk via this inbound route can dial outbound call using the UCM6200's trunk.
DID Destination	This option shows up only when "By DID" is selected. This controls the destination that can be reached by the external caller via the inbound route. The DID destination are: • Extension • Conference • Call Queue • Ring Group • Paging/Intercom Groups • IVR • Voicemail Groups • Fax Extension • Dial By Name • All
Time Condition	
Time Conditions	Select the time condition for the inbound rule.
Destination	Select the destination for the inbound call during the specified time condition.

INBOUND ROUTE: PREPEND EXAMPLE

UCM6200 now allows user to prepend digits to an inbound DID pattern, with strip taking precedence over prepend. With the ability to prepend digits in inbound route DID pattern, user no longer needs to create multiple routes for the same trunk in order to route calls to different extensions.

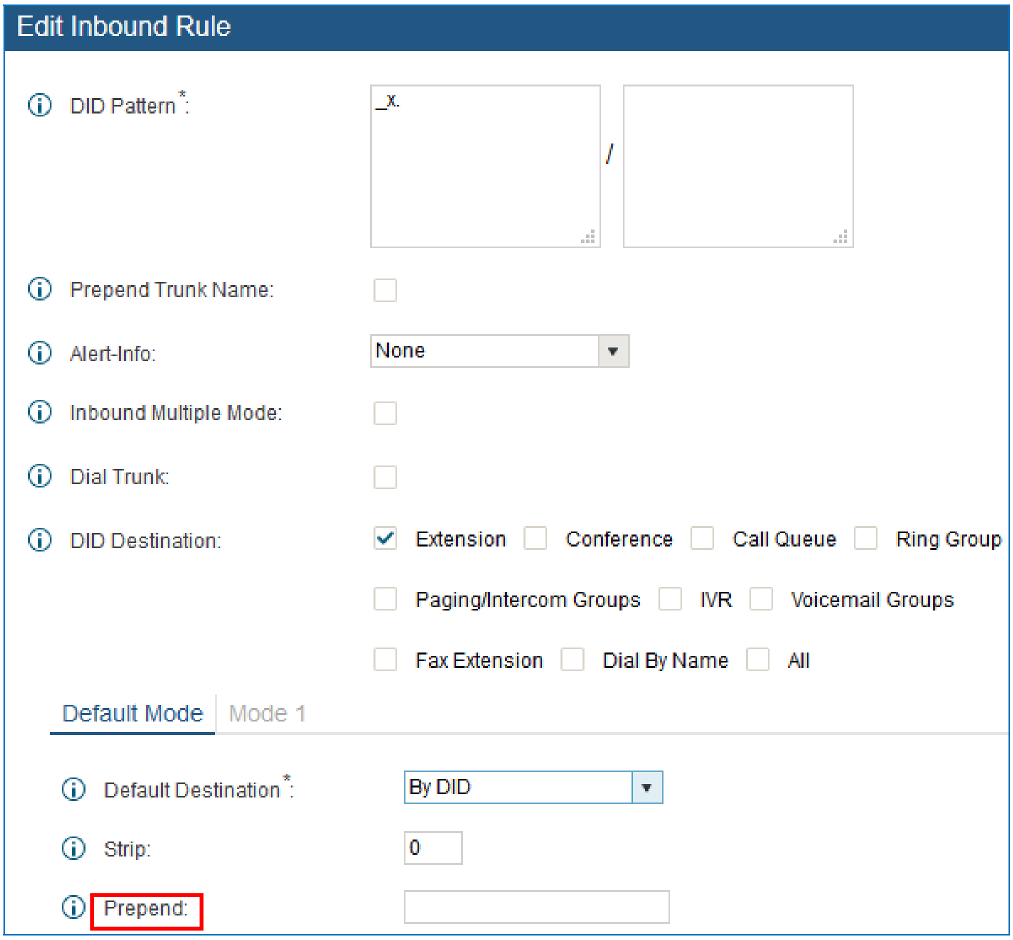


Figure 104: Inbound Route feature: Prepend

The following example demonstrates the process,

1. If Trunk provides a DID pattern of 18005251163.
2. If **Strip** is set to 8, UCM6200 will strip the first 8 digits.
3. If **Prepend** is set to 2, UCM6200 will then prepend a 2 to the stripped number, now the number become 2163.
4. UCM6200 will now forward the incoming call to extension 2163.

INBOUND ROUTE: MULTIPLE MODE

In the UCM6200, the user can configure inbound route to enable multiple mode to switch between different destinations. The inbound multiple mode can be enabled under Inbound Route settings.

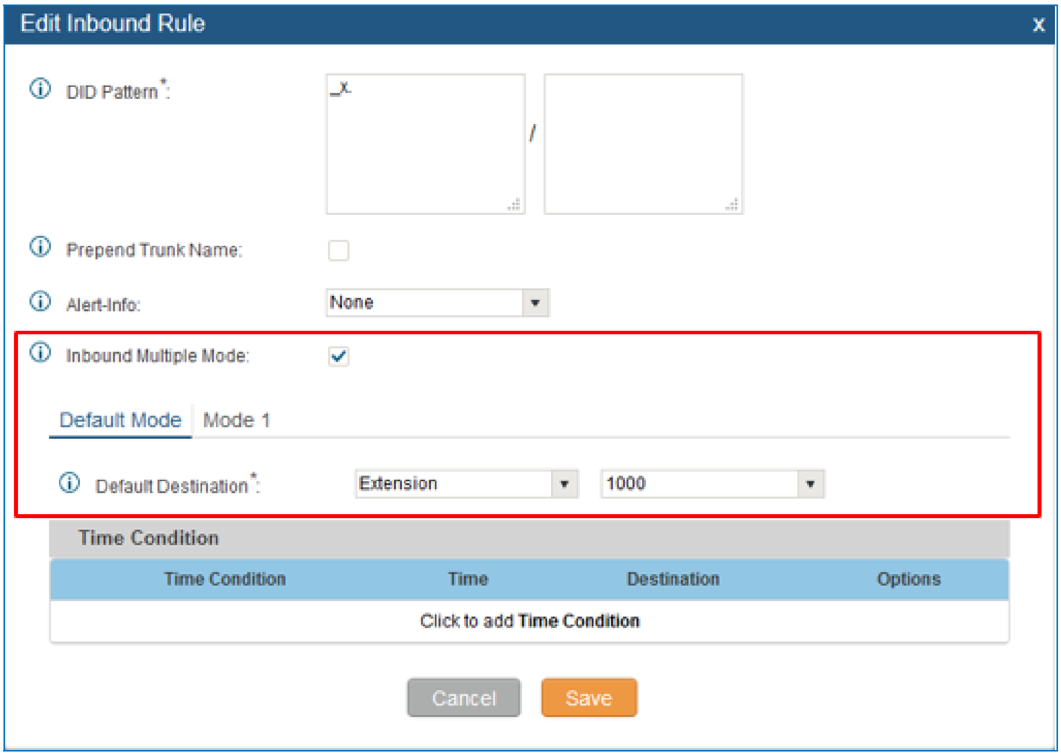


Figure 105: Inbound Route - Multiple Mode

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

SIP end devices that have registered on the UCM6200 can dial feature code *62 to switch to inbound route "Mode 1" and dial feature code *61 to switch back to "Default Destination". Switching between different mode can be easily done without web UI login.

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

FAX INTELLIGENT ROUTE

The UCM6200 can automatically detect Fax and phone signal coming from the FXO port, and then forward Fax or phone signal to the right destination. For example, when a regular phone call is coming, the UCM6200 will be able to detect the phone signal and forward it through the correct inbound route to the destination; if Fax signal is coming, the UCM6200 will be able to forward it to the FXS extension where the Fax machine is connected.

FAX WITH TWO MEDIA

The UCM6200 supports Fax re-invite with multiple codec negotiation. If a Fax re-invite contains both T.38 and PCMA/PCMU codec, UCM6200 will choose T.38 codec over PCMA/PCMU.

BLACKLIST CONFIGURATIONS

In the UCM6200, Blacklist is supported for all inbound routes. Users could enable the Blacklist feature and manage the Blacklist by clicking on "Blacklist".

- Select the checkbox for "Blacklist Enable" to turn on Blacklist feature for all inbound routes. Blacklist is disabled by default.
- Enter a number in "Add Blacklist Number" field and then click ⊕ to add to the list.
- To remove a number from the Blacklist, select the number in "Blacklist list" and click on .

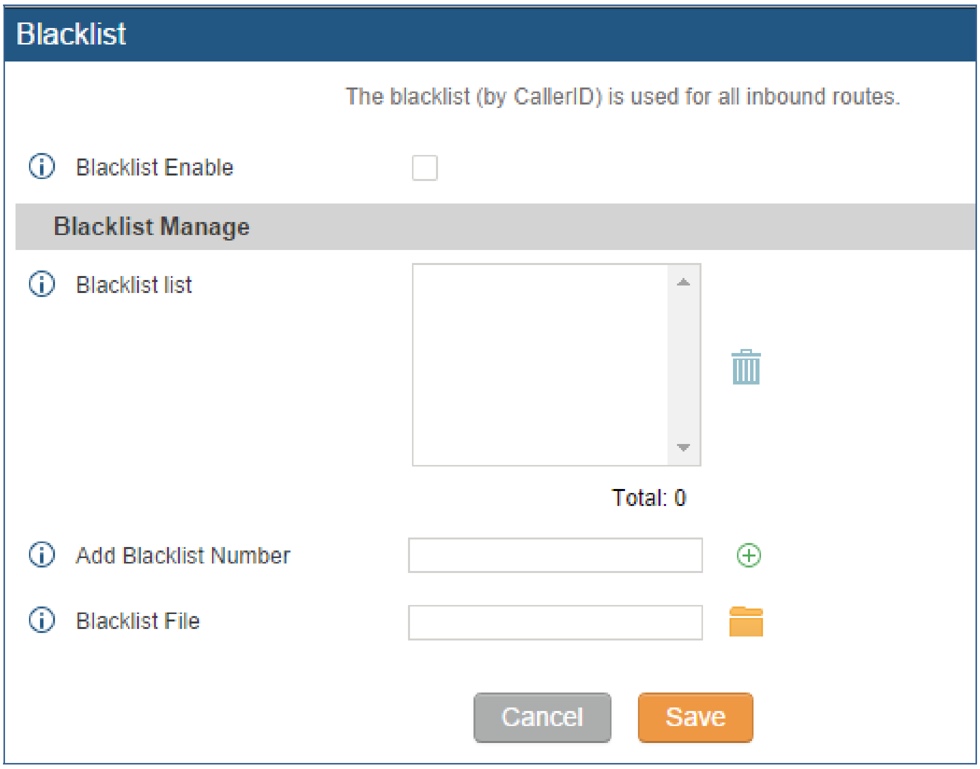


Figure 106: Blacklist Configuration Parameters

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

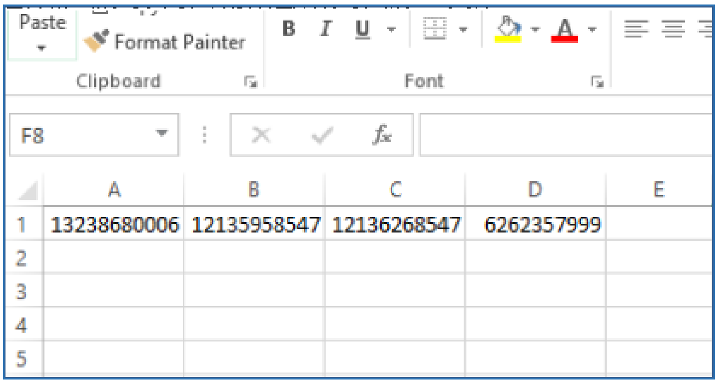


Figure 107: Blacklist csv File

⚠ **Note:**

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

CONFERENCE BRIDGE

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

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

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

CONFERENCE BRIDGE CONFIGURATIONS

- Click on "Create New Conference Room" to add a new conference bridge.
- Click on to edit the conference bridge.
- Click on to delete the conference bridge.

Table 58: Conference Bridge Configuration Parameters

Extension	Configure the conference number for the users to dial into the conference.
Password	When configured, the users who would like to join the conference call must enter this password before accessing the conference bridge. **Note:** • If "Public Mode" is enabled, the password is not required to join the conference bridge thus this field is invalid. • The password has to be at least 4 characters.
Admin Password	Configure the password to join the conference bridge as administrator. Conference administrator can manage the conference call via IVR (if "Enable Caller Menu" is enabled) as well as invite other parties to join the conference by dialing "0" (permission required from the invited party) or "1" (permission not required from the invited party) during the conference call.

	Note: • If "Public Mode" is enabled, the password is not required to join the conference bridge thus this field is invalid. • The password has to be at least 4 characters.
Enable Caller Menu	If enabled, conference participant could press the * key to access the conference bridge menu. The default setting is "No".
Record Conference	If enabled, the calls in this conference bridge will be recorded automatically in a .wav format file. All the recording files will be displayed and can be downloaded in the conference web page. The default setting is "No".
Quiet Mode	If enabled, if there are users joining or leaving the conference, voice prompt or notification tone won't be played. The default setting is "No". **Note:** "Quiet Mode" and "Announce Callers" cannot be enabled at the same time.
Wait For Admin	If enabled, the participants will not hear each other until the conference administrator joins the conference. The default setting is "No". **Note:** If "Quiet Mode" is enabled, the voice prompt for "Wait For Admin" will not be announced.
Enable User Invite	If enabled, users could press 0 to invite other users (with the users' permission) or press 1 to invite other users (without the user's permission) to join the conference. The default setting is "No". **Note:** Conference administrator can always invite other users without enabling this option.
Announce Callers	If enabled, the caller will be announced to all conference participants when there the caller joins the conference. The default setting is "No". **Note:** "Quiet Mode" and "Announce Callers" cannot be enabled at the same time.
Public Mode	If enabled, no authentication will be required when joining the conference call. The default setting is "Yes".
Play Hold Music	If enabled, the UCM6200 will play Hold music when there is only one user in the conference. The default setting is "No".

Music On Hold	Select the music on hold class to be played in conference call. Music On Hold class can be set up under web UI->**PBX**->**Internal Options**->**Music On Hold**.
Skip Authentication When Inviting User via Trunk from Web GUI	If enabled, the invitation from Web GUI for a conference bridge with password will skip the authentication for the invited users. The default setting is "No".

JOIN A CONFERENCE CALL

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

INVITE OTHER PARTIES TO JOIN CONFERENCE

When using the UCM6200 conference bridge, there are two ways to invite other parties to join the conference.

- Invite from Web GUI.

For each conference bridge in UCM6200 Web GUI->**PBX**->**Call Features**->**Conference**, 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 it into the conference.

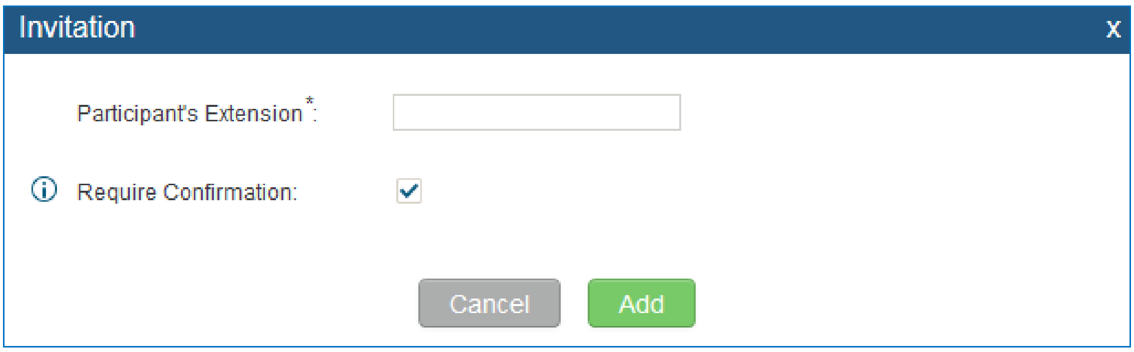


Figure 108: Conference Invitation From Web GUI

- Invite by dialing 0 or 1 during conference call.

A conference participant can invite other parties to the conference by dialing from the phone during the conference call. Please make sure option "Enable User Invite" is turned on for the conference bridge first. Enter 0 or 1 during the conference 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 it into the conference.

0: If 0 is entered to invite other party, once the invited party picks up the invitation call, a permission will be asked to "accept" or "reject" the invitation before joining the conference.
1: If 1 is entered to invite other party, no permission will be required from the invited party.

⚠ **Note:**

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

DURING THE CONFERENCE

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

- Manage the conference call from Web GUI.

  Log in UCM6200 web GUI during the conference call, the participants in each conference bridge will be listed.

  1. Click on to kick a participant from the conference.
  2. Click on to mute the participant.
  3. Click on to lock this conference bridge so that other users cannot join it anymore.
  4. Click on to invite other users into the conference bridge.

- Manage the conference call from IVR.

  If "Enable Caller Menu" is enabled, conference participant can input * to enter the IVR menu for the conference. Please see options listed in the table below.

Table 59: Conference Caller IVR Menu

Conference Administrator IVR Menu	
1	Mute/unmute yourself.
2	Lock/unlock the conference bridge.
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. • 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.
Conference 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.

Note:

When there is participant in the conference, the conference bridge configuration cannot be modified.

RECORD CONFERENCE

The UCM6200 allows users to record the conference call and retrieve the recording from web GUI->**PBX**->**Call Features**->**Conference**.

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

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

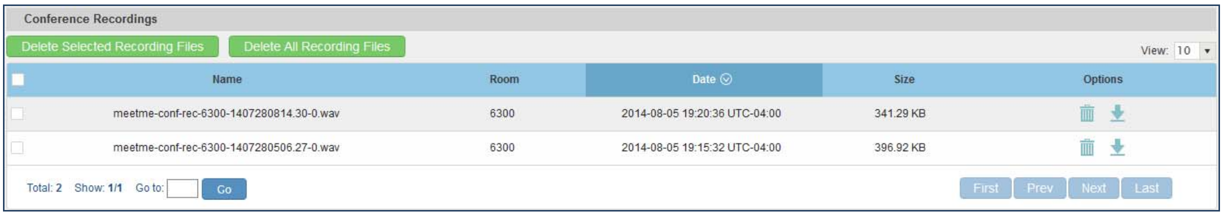


Figure 109: Conference Recording

CONFERENCE SCHEDULE

CONFERENCE SCHEUDLE CONFIGURATION

Conference Schedule can be found under UCM6200 web **UI->PBX->Call Features->Conference Schedule.** Users can create, edit, view and delete a Conference Schedule.

- Click on "Create New Conference Schedule" to add a new Conference Schedule.
- Click on the scheduled conference to edit or delete the event.

After the user configures UCM6200 with Google Service Settings ***[GOOGLE SERVICE SETTINGS SUPPORT]*** and enables Google Calendar for Conference Schedule, the conference schedule on the UCM6200 can be synchronized with Google Calendar for authorized Google account.

Table 60: Conference Schedule Parameters

Schedule Options	
Conference Topic	Configure the name of the scheduled conference. Letters, digits, _ and - are allowed.
Conference Room	Select a conference room for this scheduled conference.
Kick Time(m)	Set kick time before conference starts. When kick time is reached, a warning prompt will be played for all attendees in the conference room. After 5 minutes, this conference room will be cleared and locked for the scheduled conference to begin. **Note:** Kick Time cannot be less than 6 minutes in order to clear the conference room.
Description	The description of scheduled conference.
Repeat	Repeat interval of scheduled conference. By default it's set to single event.
Schedule Time	Configure the beginning date and duration of scheduled conference. **Note:** Please pay attention to avoid time conflict on schedules in the same conference room.
Enable Google Calendar	Select this option to sync scheduled conference with Google Calendar. **Note:** Google Service Setting OAuth2.0 must be configured on the UCM6200. Please refer to section ***[GOOGLE SERVICE SETTINGS SUPPORT]***.

Conference Administrator	Select the administrator of scheduled conference from selected extensions. **Note:** "Public Mode" must be disabled from Conference Room Options tab.
Local Extension	Select available extensions from the list to attend scheduled conference.
Remote Extension	Select available extensions from the remote peer PBX. **Note:** "LDAP Sync" must be enabled on the UCM6200 in order to view remote extensions here.
Special Extension	Add extensions that are not in the list (both local and remote list). If the user wishes to add the special extension, please match the pattern on the outbound route.
Remote Conference	Invite a remote conference.
Conference Room Options	
Password	Configure conference room password. Please note that if "Public Mode" is enabled, this option is automatically disabled.
Admin Password	Configure the password to join as conference administrator. Please note that if "Public Mode" is enabled, this option is automatically disabled.
Enable Caller Menu	If this option is enabled, conference participants will be able to access conference bridge menu by pressing the * key.
Record Conference	If this option is enabled, conference call will be recorded in .wav format. The recorded file can be found from **Conference** page.
Quiet Mode	If this option is enabled, the notification tone or voice prompt for joining or leaving the conference won't be played. **Note:** Option "Quiet Mode" and option "Announce Caller" cannot be enabled at the same time.
Wait For Admin	If this option is enabled, the participants in the conference won't be able to hear each other until conference administrator joins the conference. **Note:** If "Quiet Mode" is enabled, voice prompt for this option won't be played.
Enable User Invite	If this option is enabled, the user can: • Press '0' to invite others to join the conference with invited party's permission • Press '1' to invite without invited party's permission • Press '2' to create a multi-conference bridge to another conference room

	• Press '3' to drop all current multi-conference bridges **Note:** Conference Administrator is always allowed to access this menu.
Announce Callers	If this option is enabled, when a participant joins the conference room, participant's name will be announced to all members in the conference room. **Note:** Option "Quiet Mode" and option "Announce Caller" cannot be enabled at the same time.
Public Mode	If this option is enabled, no authentication is required for entering the conference room. **Note:** Please be aware of the potential security risks when turning on this option.
Play Hold Music	If this option is enabled, UCM6200 will play Hold Music while there is only one participant in the conference room or the conference is not yet started.
Skip Authentication When Inviting Users via Trunk from Web GUI	If this option is enabled, the invitation from Web GUI via a trunk with password won't require authentication. **Note:** Please be aware of the potential security risks when turning on this option.

- Cleaner Options

Cleaner Options	
Enable Conference Schedules Cleaner	If this option is enabled, conference schedules will be automatically cleaned as configured.
Conference Schedules Clean Time	Enter the clean time (in hours). The valid range is from 0 to 23.
Clean Interval	Enter the clean interval (in days). The valid range is from 1 to 30.

- Show/hide Conference Schedule Table

  Enable this option will allow web UI to display scheduled conference in Conference Schedule Table. Please see figure below.

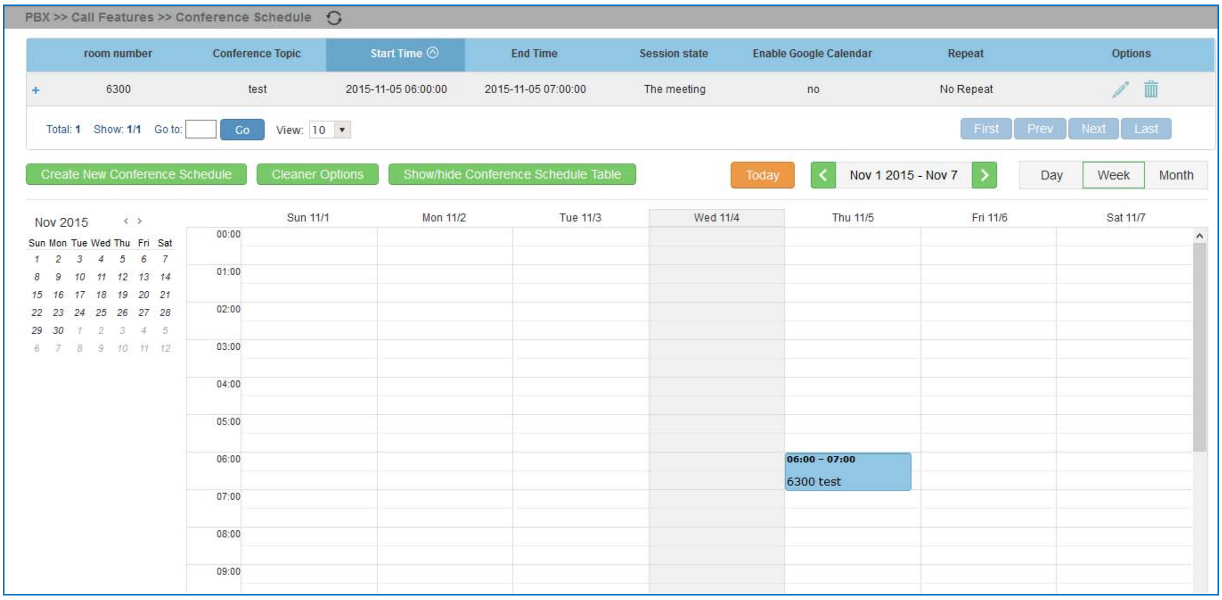


Figure 110: Conference Schedule

Once the conference room is scheduled, at the kick time, all users will be removed from conference room and no extension is allowed to join the conference room anymore. At the scheduled conference time, UCM6200 will send INVITE to the extensions that have been selected for conference.

Note:

- Please make sure that outbound route is properly configured for remote extensions to join the conference.
- Once Kick Time is reached, Conference Schedule is locked and cannot be modified.

IVR

CONFIGURE IVR

IVR configurations can be accessed under the UCM6200 Web GUI->**PBX**->**Call Features**->**IVR**. Users could create, edit, view and delete an IVR.

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

Table 61: IVR Configuration Parameters

Basic Settings	
Name	Configure the name of the IVR. Letters, digits, _ and - are allowed.
Extension	Enter the extension number for users to access the IVR.
DID Destination	This option shows up only when "By DID" is selected. This controls the destination that can be reached by the external caller via the inbound route. The DID destination are: • Extension • Conference • Call Queue • Ring Group • Paging/Intercom Groups • Voicemail Groups • Fax Extension • Dial By Name • All
Dial Trunk	If enabled, all callers to the IVR is allowed to use trunk. The permission must be configured for the users to use the trunk first. The default setting is "No".
Permission	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. The default setting is "Internal". If the user tries to dial outbound calls after dialing into the IVR, the UCM6200 will compared the IVR's permission level with the outbound route's privilege level. 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.
Welcome Prompt	Select an audio file to play as the welcome prompt for the IVR. Click on "Prompt" to add additional audio file under web GUI->**Internal Options**->**IVR Prompt**.
Digit Timeout	Configure the timeout between digit entries. After the user enters a digit, the user needs to enter the next digit within the timeout. If no digit is detected within the timeout, the UCM6200 will consider the entries complete. The default timeout is 3 seconds.
Response Timeout	After playing the prompts in the IVR, the UCM6200 will wait for the DTMF entry within the timeout (in seconds). If no DTMF entry is detected within the timeout, a timeout prompt will be played. The default setting is 10 seconds.
Response Timeout Prompt	Select the prompt message to be played when timeout occurs.
Invalid Prompt	Select the prompt message to be played when an invalid extension is pressed.
Response Timeout Repeat Loops	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 Repeat Loops	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**->**Internal Options**->**Language**. The dropdown list shows all the current available voice prompt languages on the UCM6200. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under web GUI->**PBX**->**Internal Options**->**Language**.
Key Pressing Events	
Key Press Event: **Press 0** **Press 1** **Press 2** **Press 3** **Press 4** **Press 5** **Press 6** **Press 7** **Press 8**	Select the event for each key pressing for 0-9, *, Timeout and Invalid. The event options are: • Extension • Voicemail • Conference Rooms • Voicemail Group • IVR • Ring Group • Queues • Page Group

Press 9 **Press *** **Timeout** **Invalid**	• Fax • Custom Prompt • Hangup • DISA • Dial By Name • External Number • Callback

CREATE CUSTOM PROMPT

To record new IVR prompt or upload IVR prompt to be used in IVR, click on “Prompt” next to the “Welcome Prompt” option and the users will be redirected to Custom Prompt page. Or users could go to Web GUI->**PBX**->**Internal Options**->**Custom Prompt** page directly.

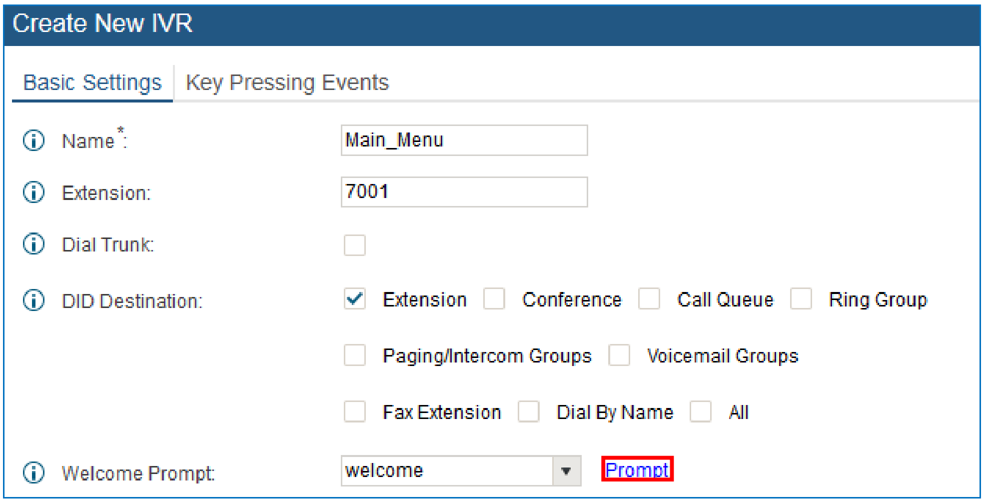


Figure 111: Click on Prompt to Create IVR Prompt

Once the IVR prompt file is successfully added to the UCM6200, it will be added into the prompt list options for users to select in different IVR scenarios.

RECORD NEW CUSTOM PROMPT

In the UCM6200 web UI->**PBX**->**Internal Options**->**Custom Prompt** page, click on “Record New Custom Prompt” and follow the steps below to record new IVR prompt.

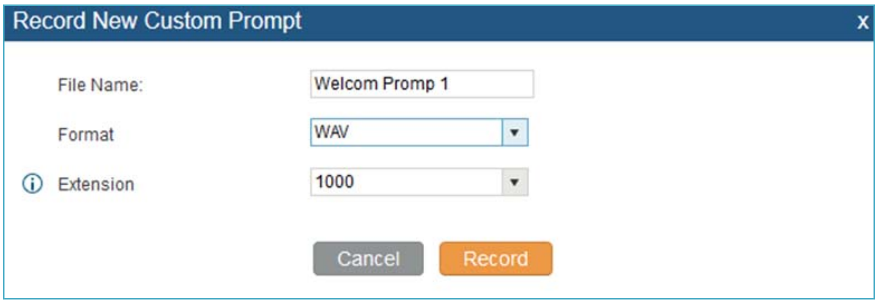


Figure 112: Record New Custom Prompt

- Specify the IVR file name.
- Select the format (GSM or WAV) for the IVR prompt file to be recorded.
- Select the extension to receive the call from the UCM6200 to record the IVR prompt.
- Click the "Record" button. A request will be sent to the UCM6200. The UCM6200 will then call the extension for recording the IVR prompt from the phone.
- Pick up the call from the extension and start the recording following the voice prompt.
- The recorded file will be listed in the IVR Prompt web page. Users could select to re-record, play or delete the recording.

UPLOAD CUSTOM PROMPT

If the user has a pre-recorded IVR prompt file, click on "Upload Custom Prompt" in Web GUI->**PBX**->**Internal Options**->**Custom Prompt** page to upload the file to the UCM6200. The following are required for the IVR prompt file to be successfully uploaded and used by the UCM6200:

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

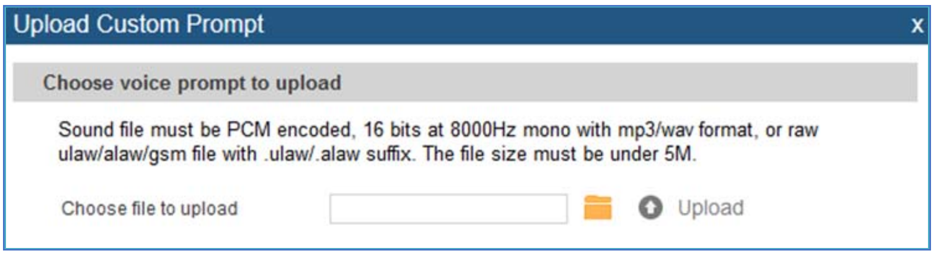


Figure 113: Upload Custom Prompt

Click on to select audio file from local PC and click on to start uploading. Once uploaded, the file will appear in the Custom Prompt web page.

LANGUAGE SETTINGS FOR VOICE PROMPT

The UCM6200 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, Swedish and Turkish.***

English (United States) and Chinese voice prompts are built in with the UCM6200 already. The other languages provided by Grandstream can be downloaded and installed from the UCM6200 web GUI directly. Additionally, users could customize their own voice prompts, package them and upload to the UCM6200.

Language settings for voice prompt can be accessed under Web GUI->**PBX**->**Internal Options**->**Language**.

DOWNLOAD AND INSTALL VOICE PROMPT PACKAGE

To download and install voice prompt package in different languages from UCM6200 web GUI, click on "Check Prompt List" button.

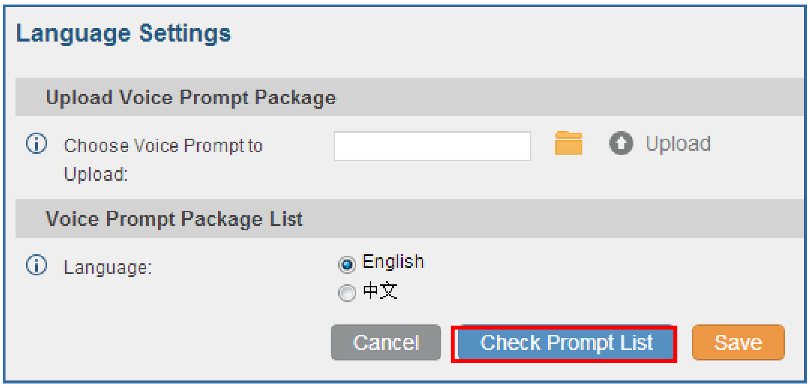
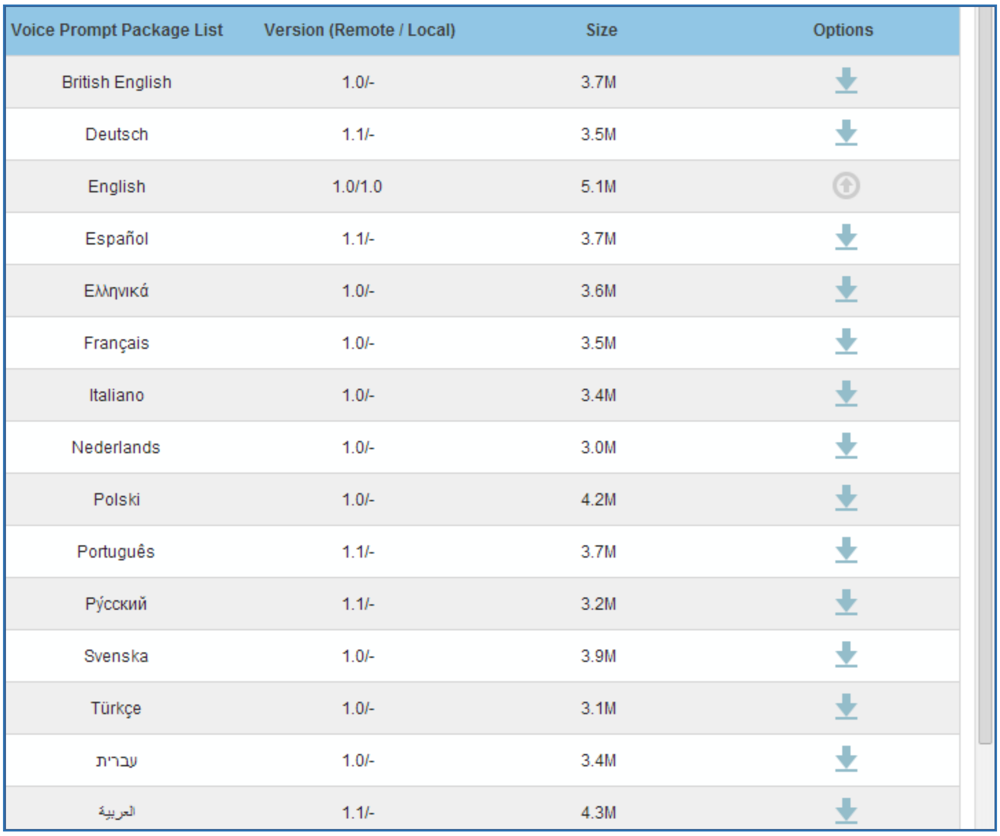


Figure 114: Language Settings for Voice Prompt

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



Voice Prompt Package List	Version (Remote / Local)	Size	Options
British English	1.0/-	3.7M	
Deutsch	1.1/-	3.5M	
English	1.0/1.0	5.1M	
Español	1.1/-	3.7M	
Ελληνικά	1.0/-	3.6M	
Français	1.0/-	3.5M	
Italiano	1.0/-	3.4M	
Nederlands	1.0/-	3.0M	
Polski	1.0/-	4.2M	
Português	1.1/-	3.7M	
Pýсский	1.1/-	3.2M	
Svenska	1.0/-	3.9M	
Türkçe	1.0/-	3.1M	
עברית	1.0/-	3.4M	
العربية	1.1/-	4.3M	

Figure 115: Voice Prompt Package List

Click on to download the language to the UCM6200. The installation will be automatically started once the downloading is finished.

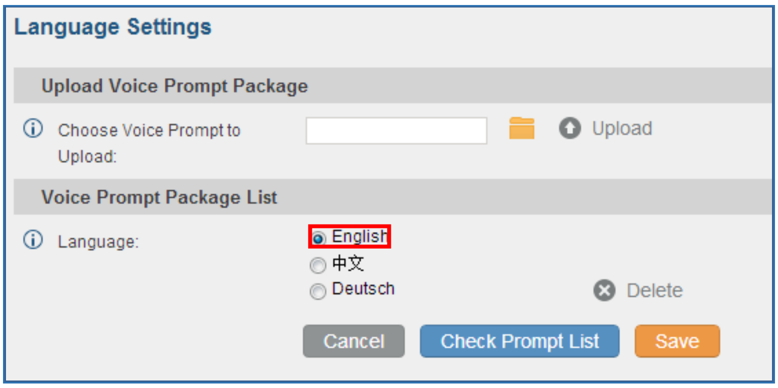


Figure 116: New Voice Prompt Language Added

A new language option will be displayed after successfully installed. Users then could select it to apply in the UCM6200 system voice prompt or delete it from the UCM6200.

CUSTOMIZE SPECIFIC PROMPT

On the UCM6200, if the user needs to replace some specific customized prompt, the user can upload a single specific customized prompt from web **UI->PBX->Internal Options->Language** instead of the entire language pack.

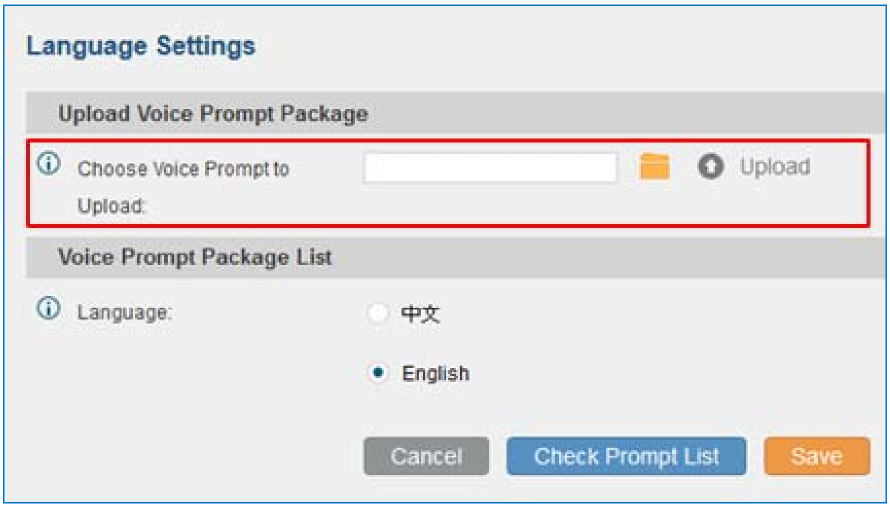


Figure 117: Upload Single Voice Prompt for Entire Language Pack

VOICEMAIL

CONFIGURE VOICEMAIL

If the voicemail is enabled for UCM6200 extensions, the configurations of the voicemail can be globally set up and managed under Web GUI->**PBX**->**Call Features**->**Voicemail**.

Table 62: Voicemail Settings

Max Greeting	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. The operator extension can be configured under web GUI->**PBX**->**Internal Options**->**General**.
Max Messages Per Folder	Configure the maximum number of messages per folder in users’ voicemail. The valid range 10 to 1000. The default setting is 50.
Max Message Time	Select the maximum duration of the voicemail message. The message will not be recorded if the duration exceeds the max message time. The default setting is 15 minutes. The available options are: • 1 minute • 2 minutes • 5 minutes • 15 minutes • 30 minutes • Unlimited
Min Effective Message Time	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: • No minimum • 1 second • 2 seconds • 3 seconds • 4 seconds • 5 seconds **Note:** Silence and noise duration are not counted in message time.
Announce Message Caller-ID	If enabled, the caller ID of the user who has left the message will be announced at the beginning of the voicemail message. The default setting

	is "No".
Announce Message Duration	If enabled, the message duration will be announced at the beginning of the voicemail message. The default setting is "No".
Play Envelope	If enabled, a brief introduction (received time, received from, and etc) of each message will be played when accessed from the voicemail application. The default setting is "Yes".
Play from Last	If enabled, UCM will play from the voice message left most recently; if disabled, UCM will play from the earliest left voice message
Allow User Review	If enabled, users can review the message following the IVR before sending the message out. The default setting is "No".

ACCESS VOICEMAIL

If the voicemail is enabled for UCM6200 extensions, the users can dial the voicemail access feature code (by default *98 or *97) to access the extension's voicemail. The users will be prompt to enter the voicemail password and then can enter digits from the phone keypad to navigate in the IVR menu for different options.

Table 63: Voicemail IVR Menu

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	

VOICEMAIL EMAIL SETTINGS

The UCM6200 can be configured to send the voicemail as attachment to Email. Click on "Voicemail Email Settings" button to configure the Email attributes and content.

Table 64: Voicemail Email Settings

Attach Recordings to E-Mail	If enabled, voicemails will be sent to user's Email address. The default setting is "Yes".
Keep Recordings	If enabled, voicemail will be stored in the UCM6200 after the email is sent. The default setting is “Yes”.
Template For Voicemail Emails	Fill in the "Subject:" and "Message:" content, to be used in the Email when sending to the user. The template variables are: • \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

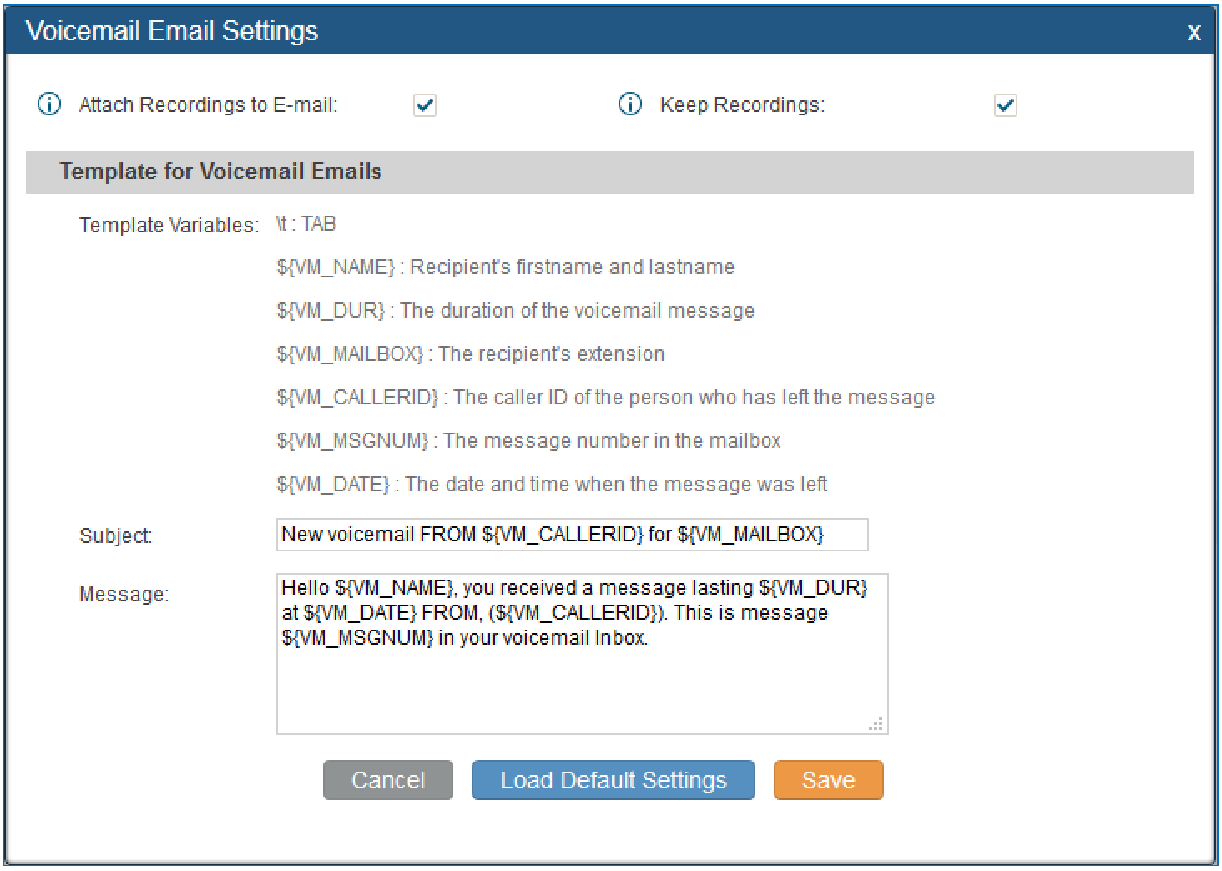


Figure 118: Voicemail Email Settings

Click on "Load Default Settings" button to view the default template as an example.

CONFIGURE VOICEMAIL GROUP

The UCM6200 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->**PBX**->**Call Features**->**Voicemail Group**. Click on "Create New Voicemail Group" to configure the group.

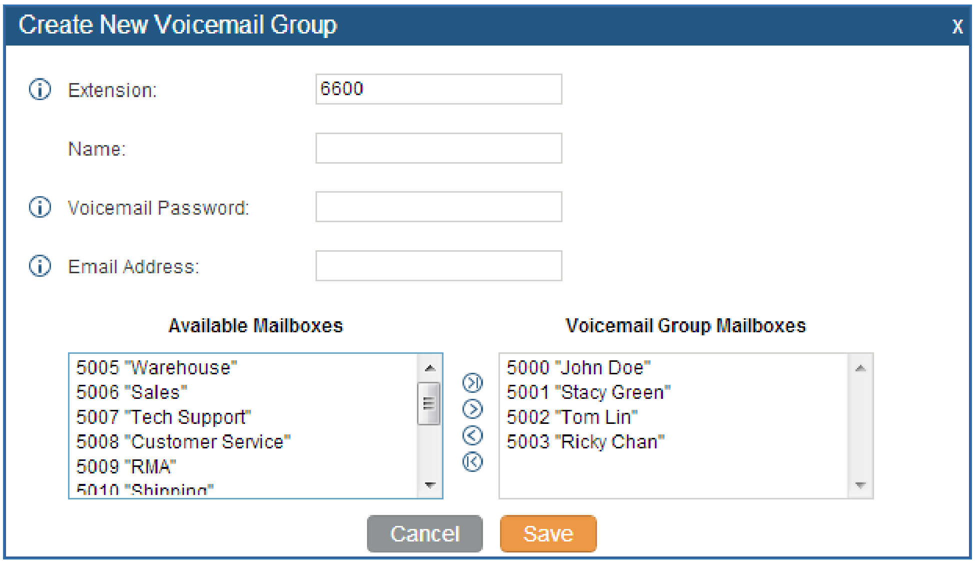


Figure 119: Voicemail Group

Table 65: Voicemail Group Settings

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.
Voicemail Password	Configure the voicemail password for the users to check voicemail messages.
Email Address	Configure the Email address for the voicemail group extension.
Voicemail Group Mailboxes	Select available mailboxes from the left list and add them to the right list. The extensions need to have voicemail enabled to be listed in available mailboxes list.

RING GROUP

The UCM6200 supports ring group feature with different ring strategies applied to the ring group members. This section describes the ring group configuration on the UCM6200.

CONFIGURE RING GROUP

Ring group settings can be accessed via Web GUI->**PBX**->**Call Features**->**Ring Group**.

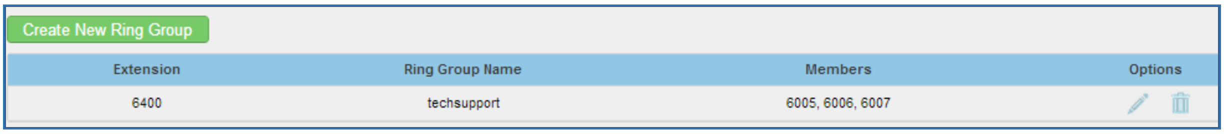


Figure 120: Ring Group

- Click on "Create New Ring Group" to add ring group.
- Click on to edit the ring group. The following table shows the ring group configuration parameters.
- Click on to delete the ring group.

Table 66: Ring Group Parameters

Ring Group Name	Configure ring group name to identify the ring group. Letters, digits, _ and – are allowed.
Extension	Configure the ring group extension.
Ring Group Members	Select available users from the left side to the ring group member list on the right side. Click on to arrange the order.
Selected LDAP Numbers	Select available remote users from the left side to the ring group member list on the right side. Click on to arrange the order. Note: LDAP Sync must be enabled first.
Ring Strategy	Select the ring strategy. The default setting is "Ring in order". • 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 doesn't answer the call, it will stop ringing the first member and start ringing the second member.

Custom Prompt	This option is to set a custom prompt for a ring group to announce to caller. Click on ‘Prompt’, it will direct to the page **PBX->Internal Options->Custom Prompt**, where users could record new prompt or upload prompt files.
Ring Timeout on Each Member	Configure the number of seconds to ring each member. If set to 0, it will keep ringing. The default setting is 30 seconds. **Note:** The actual ring timeout might be overridden by users if the phone has ring timeout settings as well.
Auto Record	If enabled, calls on this ring group will be automatically recorded. The default setting is No. The recording files can be accessed from web GUI->**CDR**->**Recording Files**.
Enable Destination	If enabled, users could select extension, voicemail, ring group, IVR, call queue, voicemail group as the destination if the call to the ring group has no answer. Secret and Email address are required if voicemail is selected as the destination.
Secret	Configure the password to access the ring group extension's voicemail. **Note:** The password has to be at least 4 characters.
Email Address	Configure the Email address of the ring group extension's voicemail. If "Attach Recordings to E-mail" is enabled from Web GUI->**PBX**->**Voicemail**->**Voicemail Email Settings**, the voicemail can be sent to the ring group's Email address as attachment.

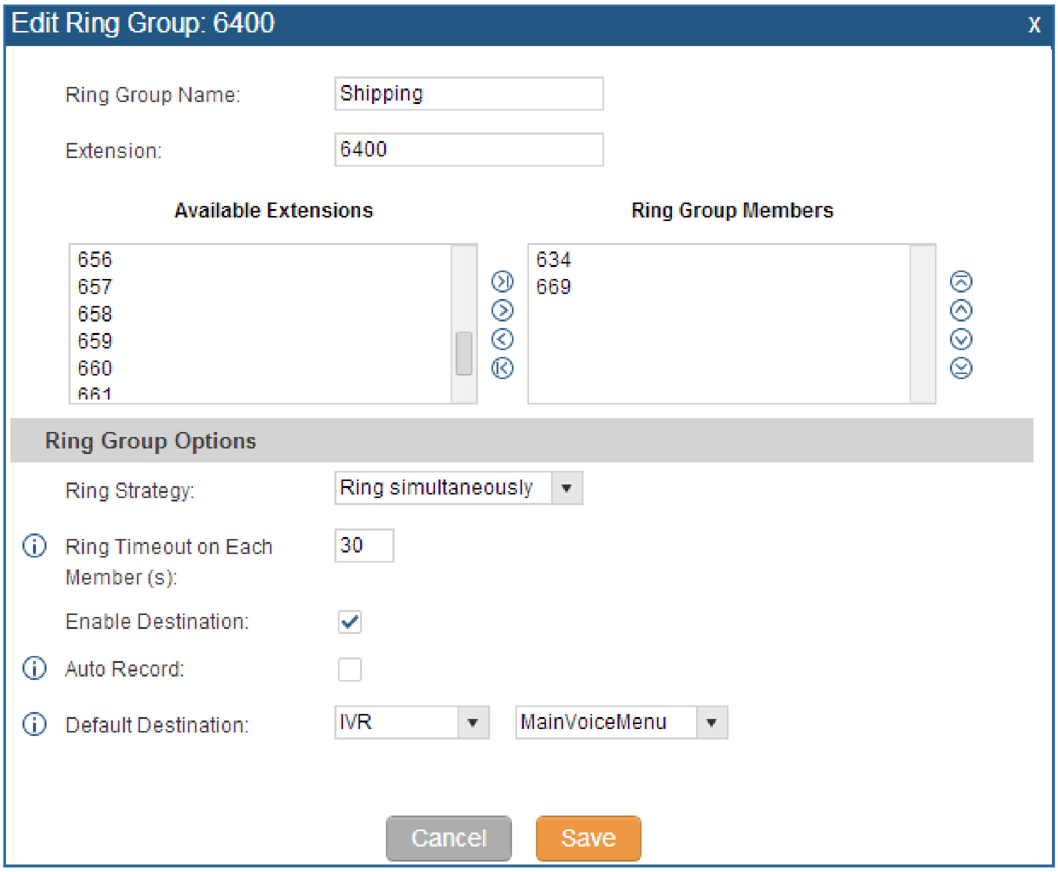
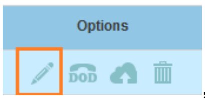


Figure 121: Ring Group Configuration

REMOTE EXTENSION IN RING GROUP

Remote extensions from the peer trunk of a remote UCM6200 can be included in the ring group with local extension. An example of Ring Group with peer extensions is presented in the following:

1. Creating SIP Peer Trunk between both UCM6200_A and UCM6200_B. **SIP Trunk** can be found under web **UI-> PBX-> Basic/Call Routes-> 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 UCM6200_A update remote LDAP server automatically from peer UCM6200_B. In addition, **Sync LDAP Password** must match for UCM6200_A and UCM6200_B in order 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**.**

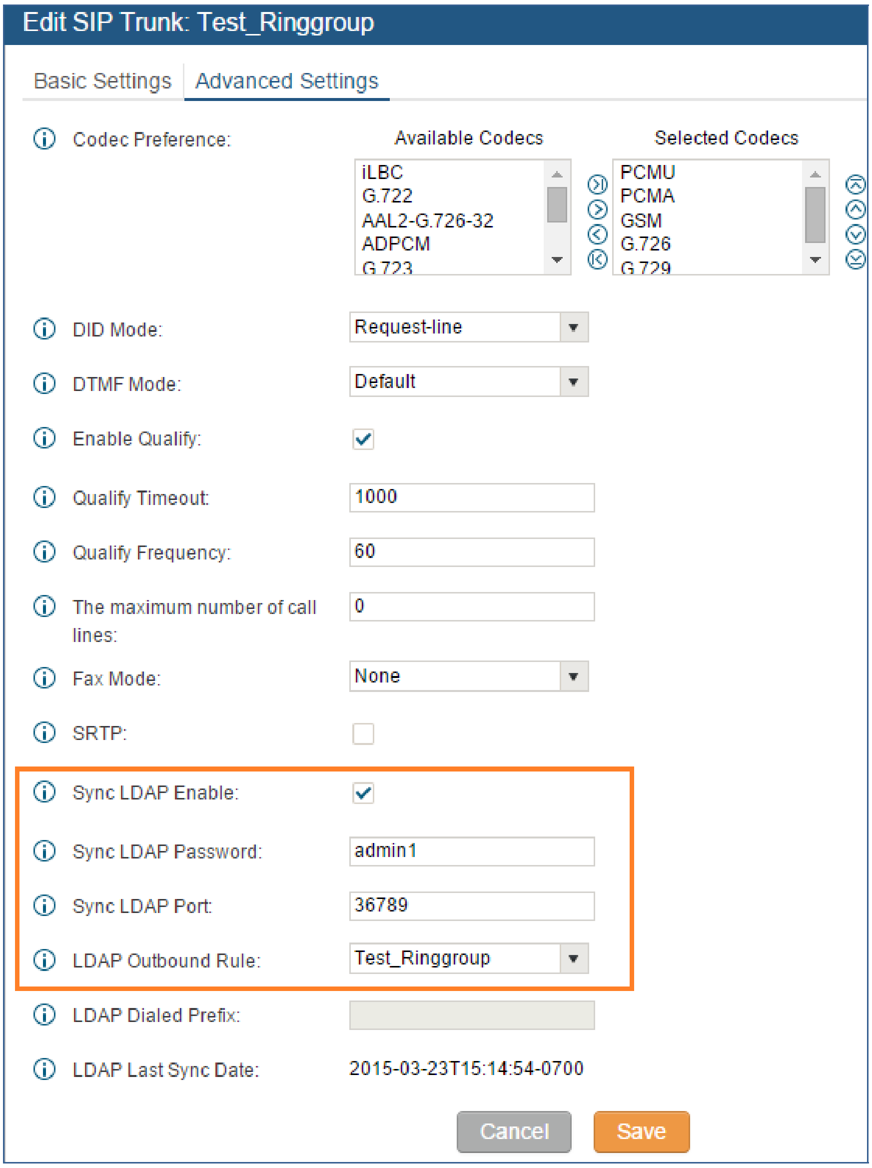


Figure 122: Sync LDAP Server option

3. In case if LDAP server doesn't 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 UCM6200.

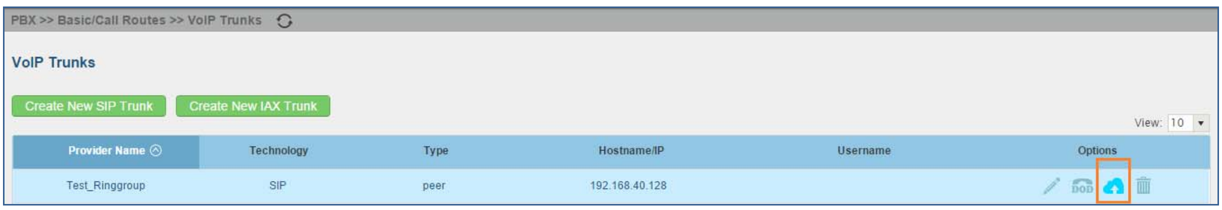


Figure 123: Manually Sync LDAP Server

4. Under **Ring Groups** setting page, click 

. **Ring Groups** can be found under web **UI-> PBX-> 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 UCM6200 can be included into that UCM6200's LDAP contact.

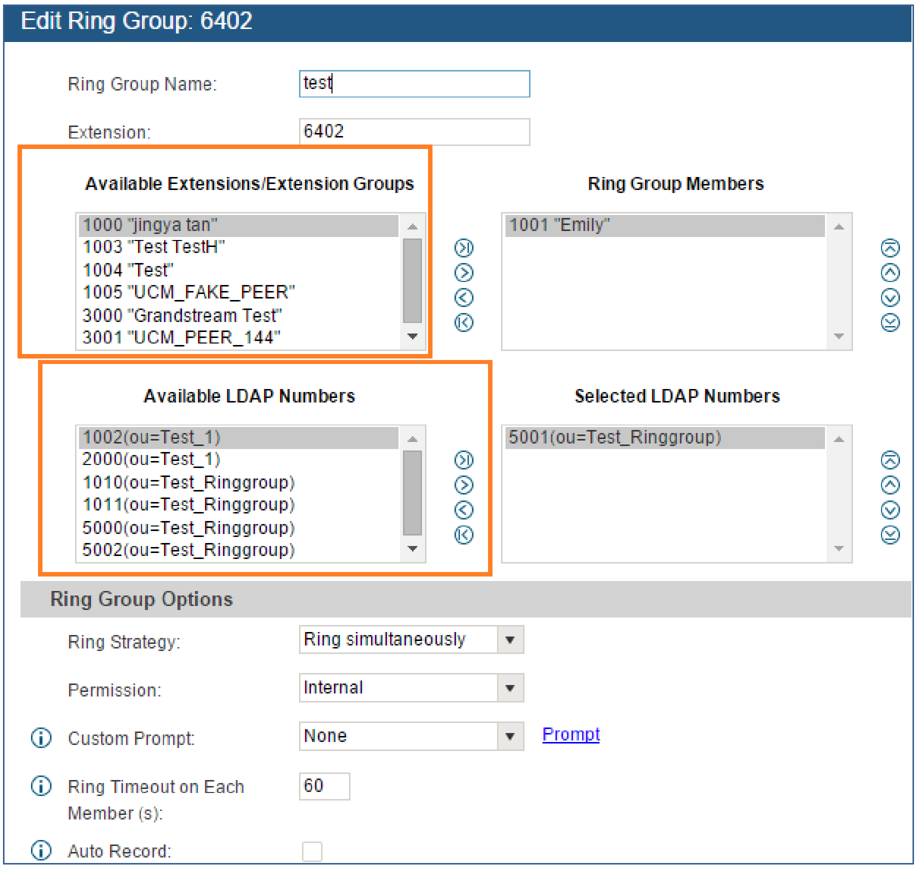


Figure 124: Ring Group Remote Extension

PAGING AND INTERCOM GROUP

Paging and Intercom Group can be used to make an announcement over the speaker on a group of phones. Targeted phones will answer immediately using speaker. The UCM6200 paging and intercom can be used via feature code to a single extension or a paging/intercom group. This sections describes the configuration of paging/intercom group under Web GUI->**PBX**->**Call Features**->**Paging/Intercom**.

CONFIGURE PAGING/INTERCOM GROUP

- Click on "Create New Paging/Intercom Group" to add paging/intercom group.

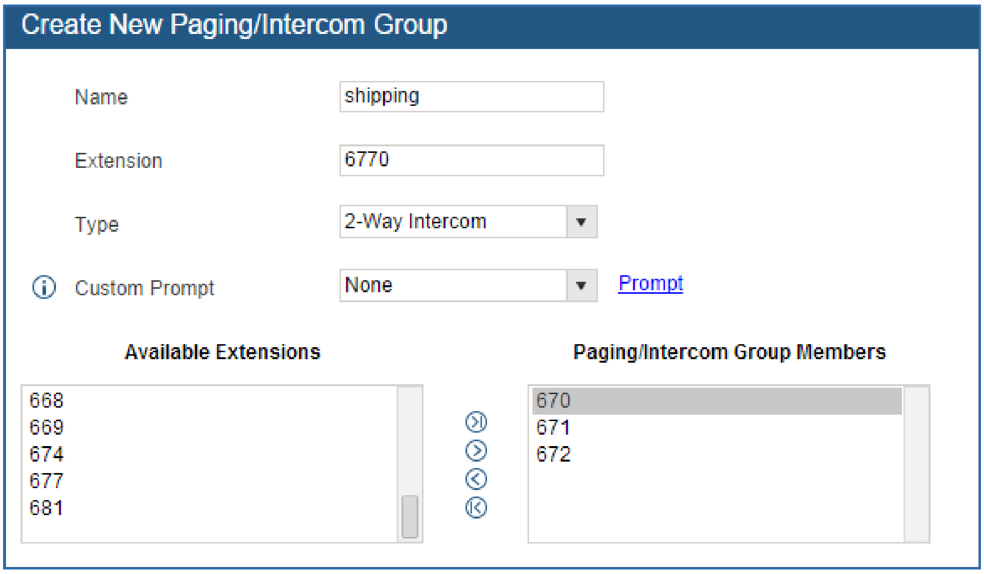


Figure 125: Paging/Intercom Group

Table 67: Paging/Intercom Group Configuration Parameters

Name	Configure paging/intercom group name.
Extension	Configure the paging/intercom group extension.
Type	Select "2-way Intercom" or "1-way Page".
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 to the page **PBX->Internal Options->Custom Prompt**, where users could record new prompt or upload prompt files.
Page/Intercom Group Members	Select available users from the left side to the paging/intercom group member list on the right.

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

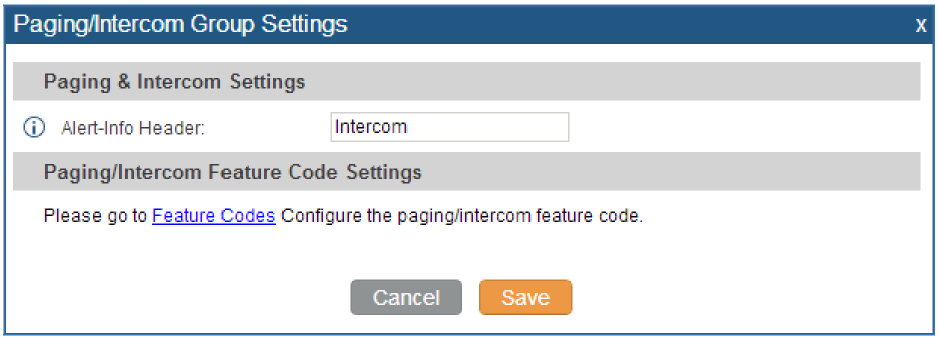


Figure 126: Page/Intercom Group Settings

- The UCM6200 has pre-configured paging/intercom feature code. By default, the Paging Prefix is *81 and the Intercom Prefix is *80. To edit page/intercom feature code, click on "Feature Codes" in the "Paging/Intercom Group Settings" dialog. Or users could go to Web GUI->**PBX**->**Internal Options**->**Feature Codes** directly.

CALL QUEUE

The UCM6200 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->**PBX**->**Call Features**->**Call Queue**.

CONFIGURE CALL QUEUE

Call queue settings can be accessed via Web GUI->**PBX**->**Call Features**->**Call Queue**.

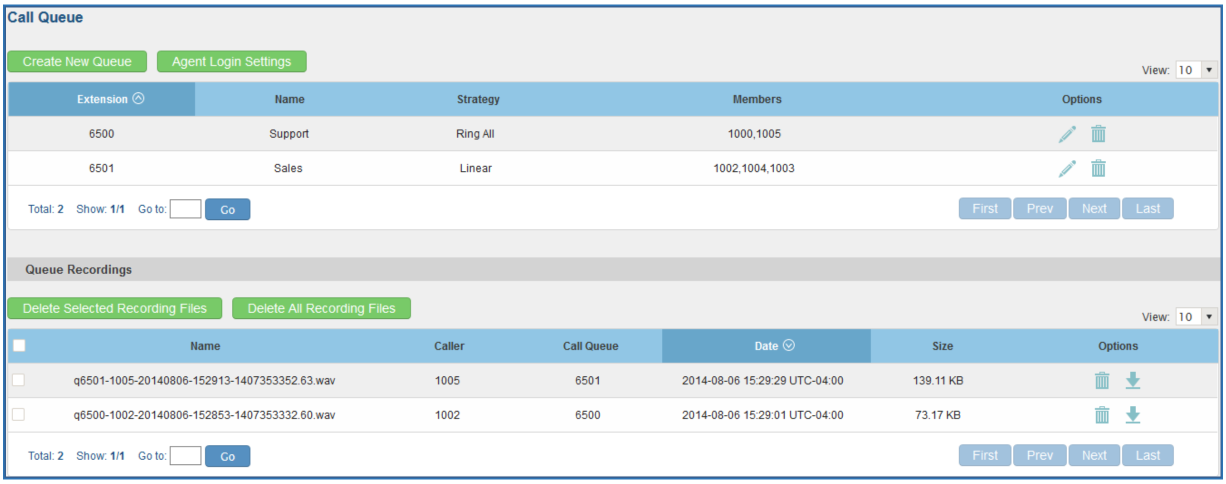


Figure 127: Call Queue

- Click on "Create New Queue" to add call queue.
- Click on to edit the call queue. The call queue configuration parameters are listed in the table below.

Table 68: Call Queue Configuration Parameters

Extension	Configure the call queue extension.
Name	Configure the call queue name to identify the call queue.
Strategy	Select the strategy for the call queue. • 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. The default setting is "Ring All".
Music On Hold	Select the Music On Hold class for the call queue. **Note:** Music On Hold classes can be managed from Web GUI-> **PBX**->**Internal Options**->**Music On Hold**.
Leave When Empty	Configure whether the callers will be disconnected from the queue or not if the queue has no agent anymore. The default setting is "Strict". • 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	Configure whether the callers can dial into a call queue if the queue has no agent. The default setting is "No". • 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.
Dynamic Login Password	If enabled, the configured PIN number is required for dynamic agent to log in. The default setting is disabled.
Ring Time Out	Configure the number of seconds an agent will ring before the call goes to the next agent. The default setting is 15 seconds.
Wrapup Time	Configure the number of seconds before a new call can ring the queue

	after the last call on the agent is completed. If set to 0, there will be no delay between calls to the queue. The default setting is 15 seconds.
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. It only includes calls not connected yet. The default setting is 0, which means unlimited. When the maximum value is reached, the caller will be treated with busy tone followed by the next calling rule after attempting to enter the queue.
Report Hold Time	If enabled, the UCM6200 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".
Wait Time	If enabled, users will be disconnected after the configured number of seconds. The default setting is "No". **Note:** It is recommended to configure "Wait Time" longer than the "Wrapup Time".
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->**PBX**->**Call Features**->**Call Queue**.
Enable Destination	If enabled, the incoming call for the call queue will be routed to the destination configured in the next field if none of the agents answers the call after ringing for a time of “Ring Timeout”.
Queue Timeout	Configure the global timeout (in seconds) of call queue. It must be bigger than the value of ring timeout. The call in the queue will be transferred to the failover destination directly if this time is exceeded.
Failover Destination	Configure the call destination for the call to be routed to if no agent in this call queue answers the call.
Enable Feature Codes	Enable feature codes option for call queue. For example, *83 is used for “Agent Pause”
Agents	Select the available users to be the static agents in the call queue. Choose from the available users on the left to the static agents list on the right. Click on to arrange the order.

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

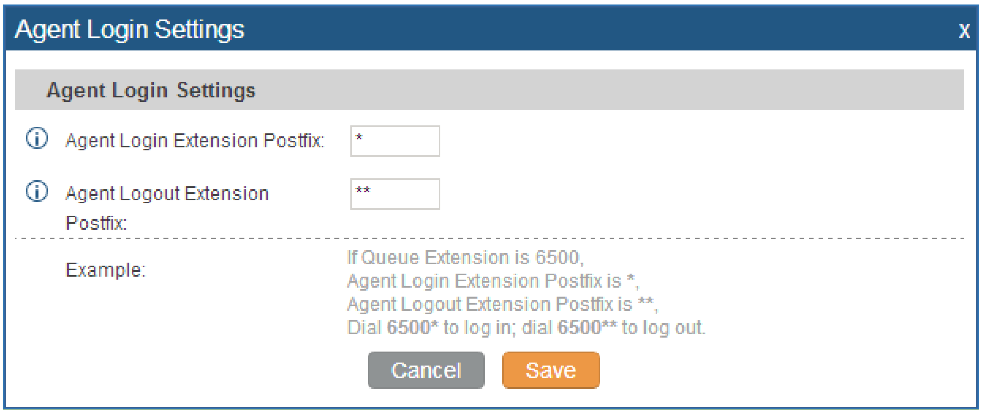


Figure 128: Agent Login Settings

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 doesn't 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->**PBX**->**Internal Options**->**Feature Codes**. The default feature code is *83 for "Agent Pause" and *84 for "Agent Unpause".
- Queue recordings are shown on the Call Queue page. 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.

EXTENSION GROUPS

The UCM6200 extension group feature allows users to assign and categorize extensions in different groups to better manage the configurations on the UCM6200. For example, when configuring "Enable Filter on Source Caller ID", users could select a group instead of each person's extension to assign. This feature simplifies the configuration process and helps manage and categorize the extensions for business environment.

CONFIGURE EXTENSION GROUPS

Extension group can be configured via Web GUI->**PBX**->**Call Features**->**Extension Groups**.

- Click on "Create New Extension Group" to create a new extension group.
- Click on to edit the extension group.

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

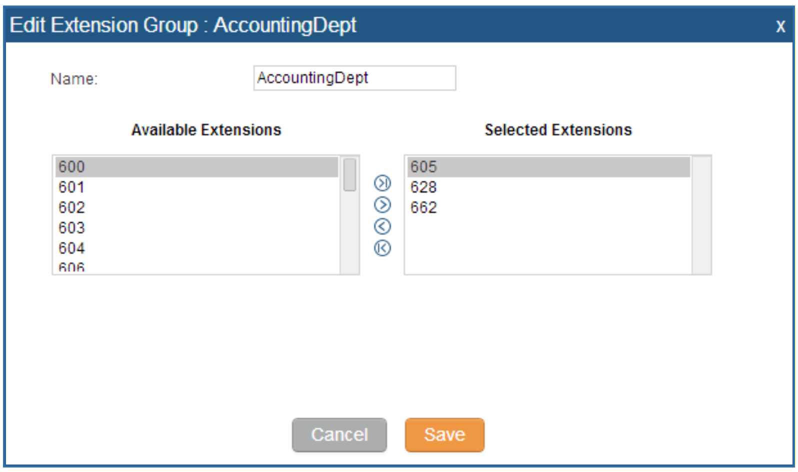


Figure 129: Edit Extension Group

- Click on to delete the extension group.

USING EXTENSION GROUPS

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

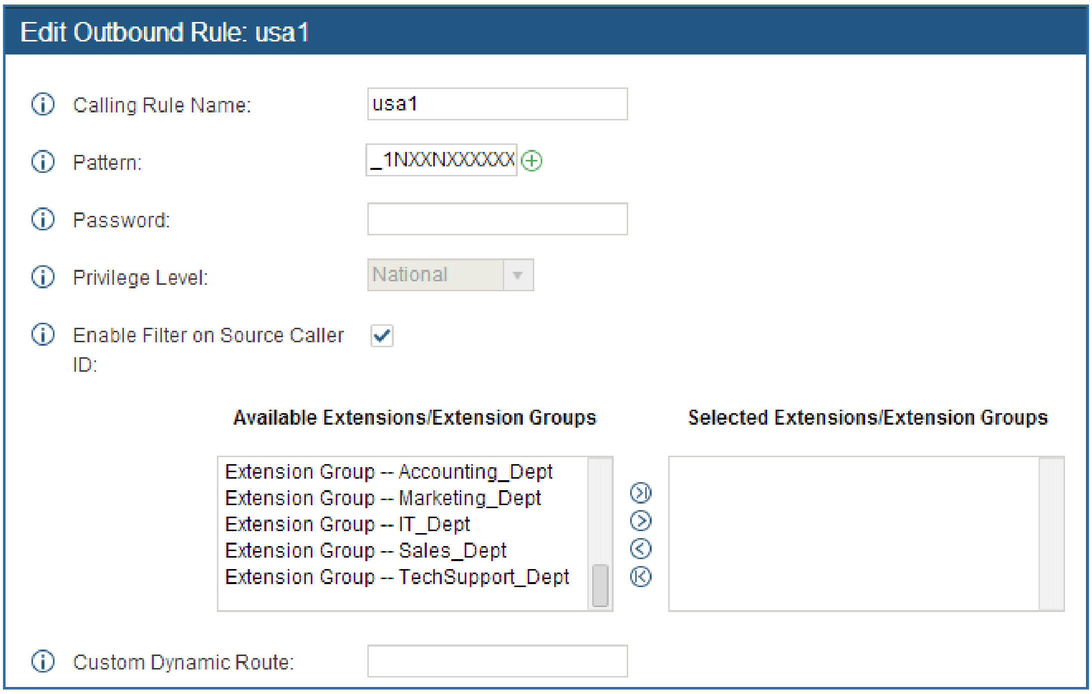


Figure 130: Select Extension Group in Outbound Route

PICKUP GROUPS

The UCM6200 supports pickup group feature which allows users to pick up incoming calls for other extensions if they are in the same pickup group, by dialing "Pickup Extension" feature code (by default *8).

CONFIGURE PICKUP GROUPS

Pickup groups can be configured via Web GUI->**PBX**->**Call Features**->**Pickup Groups**.

- Click on "Create New Pickup Group" to create a new pickup group.
- Click on to edit the pickup group.

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

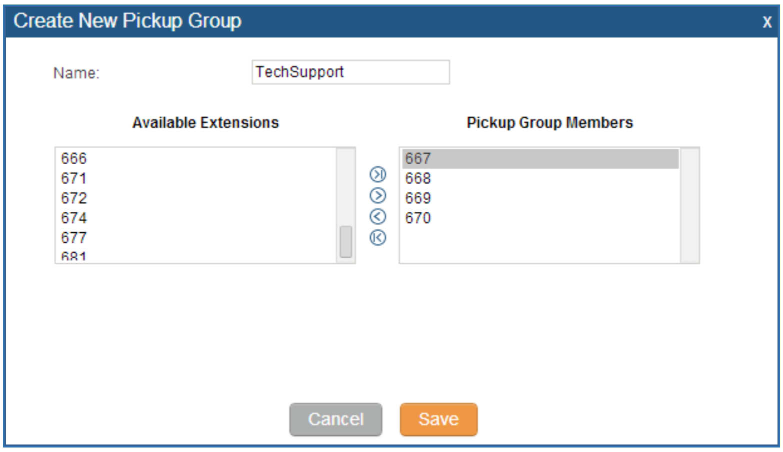


Figure 131: Edit Pickup Group

- Click on to delete the 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's not necessary to add the extension number after the pickup feature code. The pickup feature code is configurable under Web GUI->**PBX**->**Internal Options**->**Feature Codes**.

The default pickup feature code is *8.

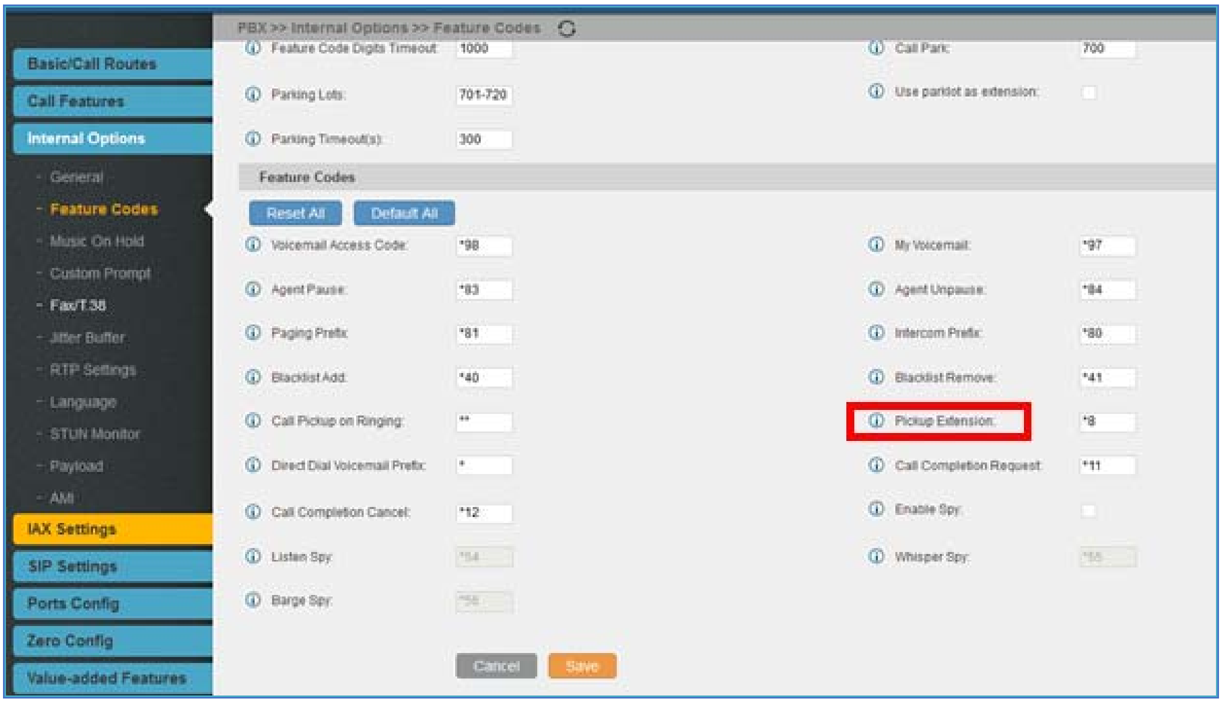


Figure 132: Edit Pickup Feature Code

MUSIC ON HOLD

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

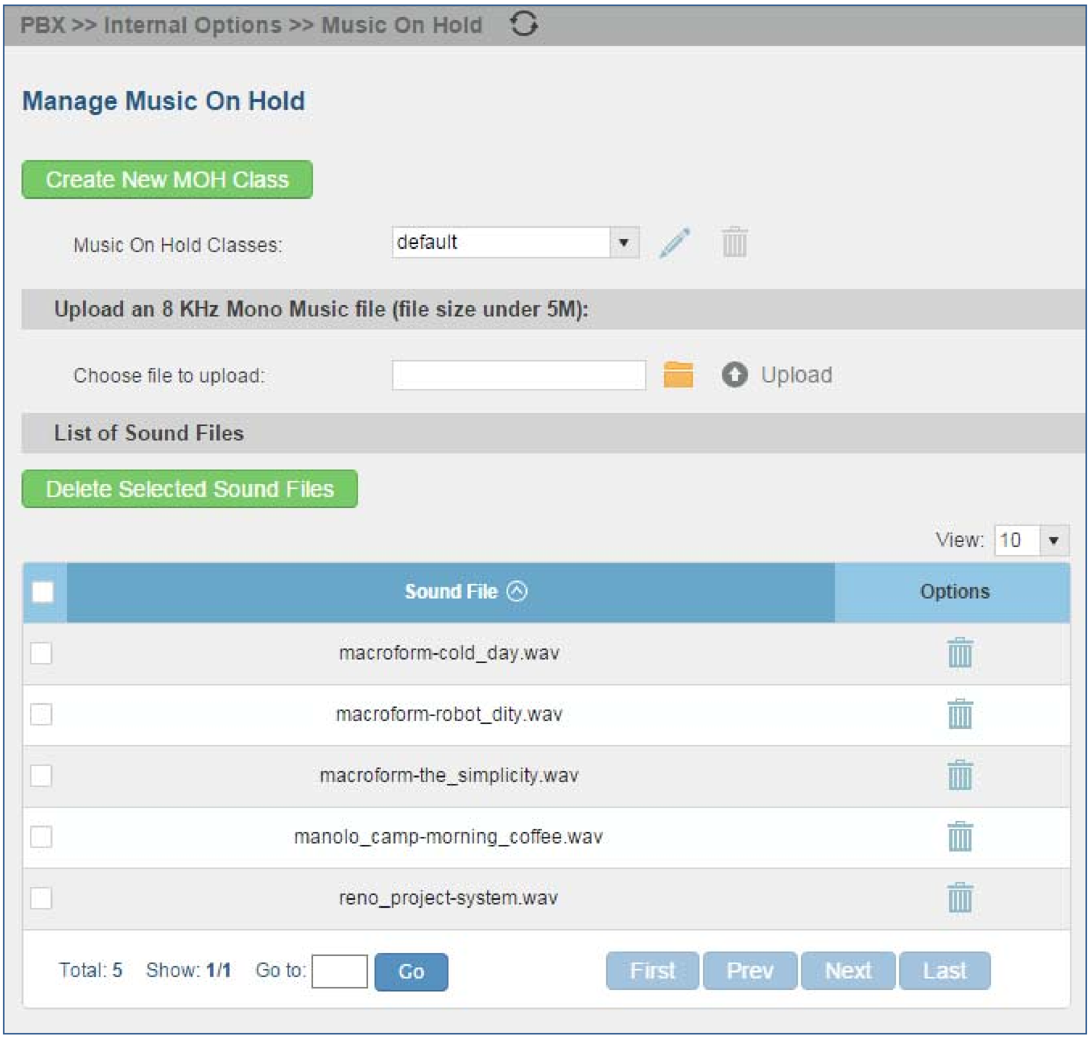


Figure 133: Music On Hold Default Class

- 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 [folder icon] to select music file from local PC and click on [upload icon] to start uploading. The music file uploaded has to be 8 KHz Mono format with size smaller than 5M.
- Click on [delete icon] next to the sound file to delete it from the selected Music On Hold Class.
- Select the sound files and click on **Delete Selected Sound Files** to delete all selected music on hold files.

⚠ **Note:**

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

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

FAX/T.38

The UCM6200 supports T.30/T.38 Fax and Fax Pass-through. It can convert the received Fax to PDF format and send it to the configured Email address. Fax/T.38 settings can be accessed via Web GUI->**PBX**->**Internal Options**->**FAX/T.38**. The list of received Fax files will be displayed in the same web page for users to view, retrieve and delete.

CONFIGURE FAX/T.38

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

Table 69: FAX/T.38 Settings

Enable Error Correction Mode	Configure to enable Error Correction Mode (ECM) for the Fax. The default setting is "Yes".
Maximum Transfer Rate	Configure the maximum transfer rate during the Fax rate negotiation. The possible values are 2400, 4800, 7200, 9600, 12000 and 14400. The default setting is 14400.
Minimum Transfer Rate	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.
Max Concurrent Sending Fax	Configure the concurrent fax that can be sent by UCM6200. Two mode “Only” and “More” are supported. • Only Under this mode, the UCM6200 allows only single user to send fax at a time. • More Under this mode, the UCM6200 supports multiple concurrent fax sending by the users. By default, this option is set to “only”.
Fax Queue Length	Configure the maximum length of Fax Queue from 6 to 10. The default setting is 6.
Default Email Address	Configure the Email address to send the received Fax to if user's Email address cannot be found. **Note:**

	The extension's Email address or the Fax's default Email address needs to be configured in order to receive Fax from Email. If neither of them is configured, Fax will be not be received from Email.
Template Variables	Fill in the "Subject:" and "Message:" content, to be used in the Email when sending the Fax to the users. The template variables are: • ${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

- Click on to edit the Fax extension.
- Click on to delete the Fax extension.

SAMPLE CONFIGURATION TO RECEIVE FAX FROM PSTN LINE

The following instructions describe how to use the UCM6200 to receive Fax from PSTN line on the Fax machine connected to the UCM6200 FXS port.

1. Connect Fax machine to the UCM6200 FXS port.
2. Connect PSTN line to the UCM6200 FXO port.
3. Go to web GUI->**PBX**->**Analog Trunks** page.
4. Create or edit the analog trunk for Fax as below.

   Fax Mode: Make sure "Fax Mode" option is set to "None".

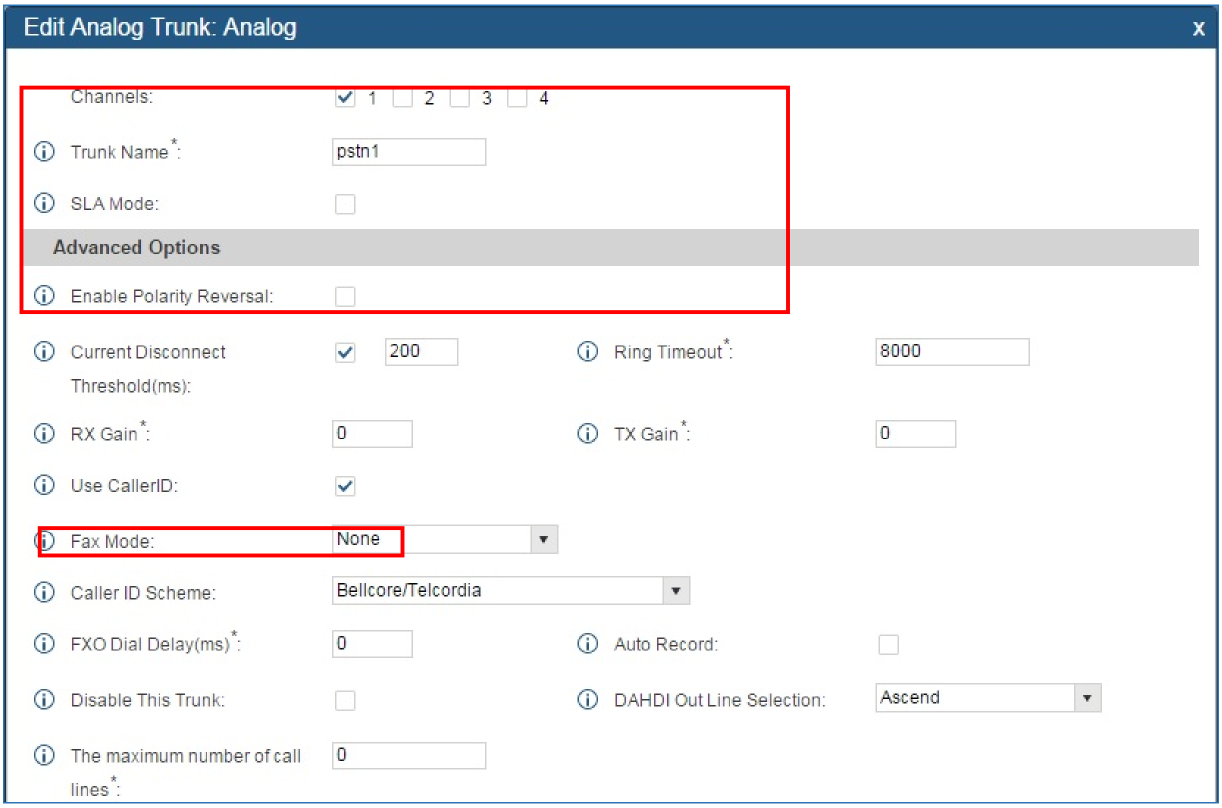


Figure 134: Configure Analog Trunk without Fax Detection

5. Go to UCM6200 web GUI->**PBX**->**Basic/Call Routes**->**Extensions** page.
6. Create or edit the extension for FXS port.
   - **Analog Station**: Select FXS port to be assigned to the extension. By default, it's set to "None".
   - Once selected, analog related settings for this extension will show up in "**Analog Settings**" section.

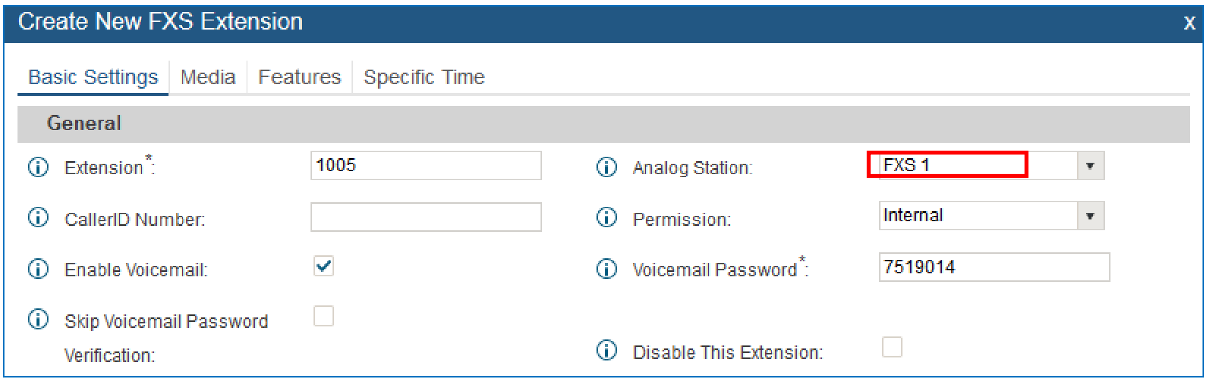


Figure 135: Configure Extension for Fax Machine: FXS Extension

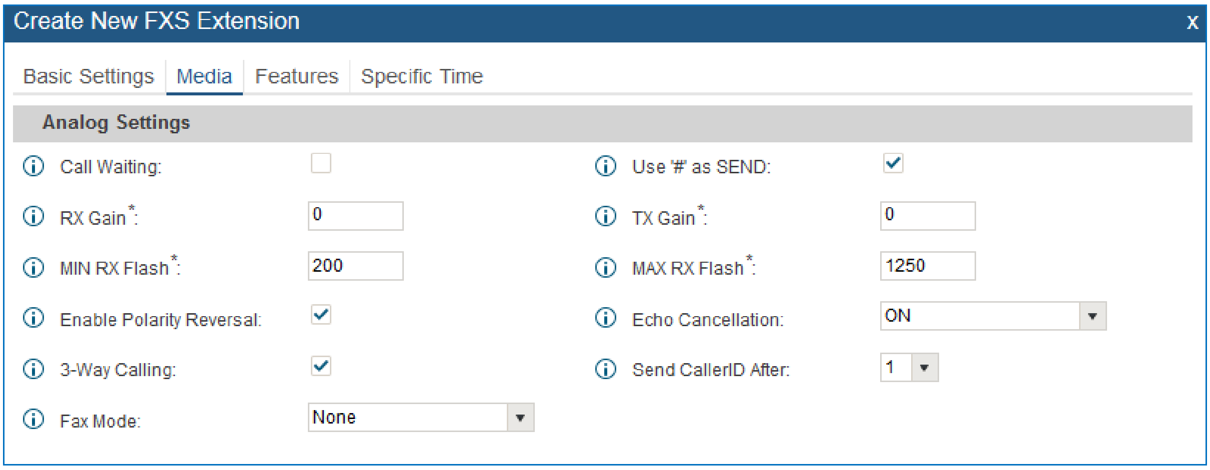


Figure 136: Configure Extension for Fax Machine: Analog Settings

7. Go to web GUI->**PBX**->**Basic/Call Routes**->**Inbound Routes** page.
8. Create an inbound route to use the Fax analog trunk. Select the created extension for Fax machine in step 4 as the default destination.

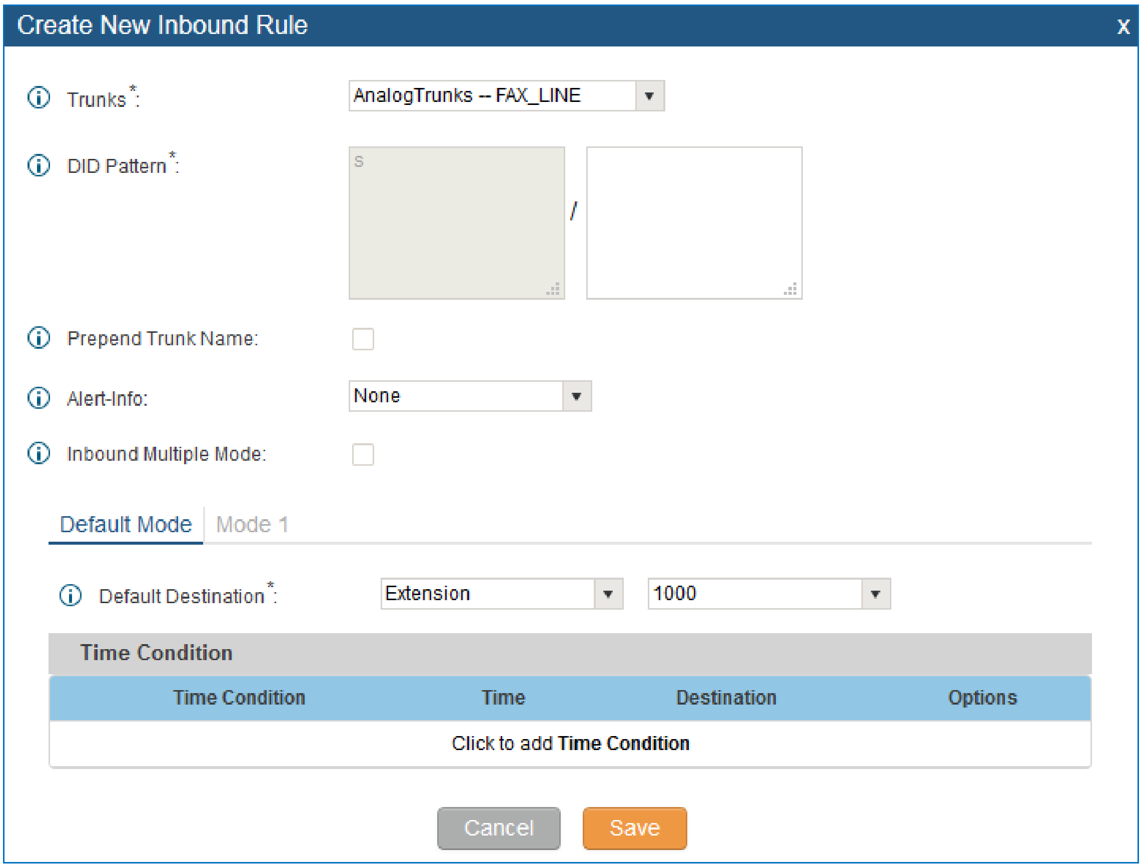


Figure 137: Configure Inbound Rule for Fax

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

SAMPLE CONFIGURATION FOR FAX-TO-EMAIL

The following instructions describe a sample configuration on how to use Fax-to-Email feature on the UCM6200.

1. Connect PSTN line to the UCM6200 FXO port.
2. Go to UCM6200 web GUI->**Internal Options**->**Fax/T.38** page. Create a new Fax extension.

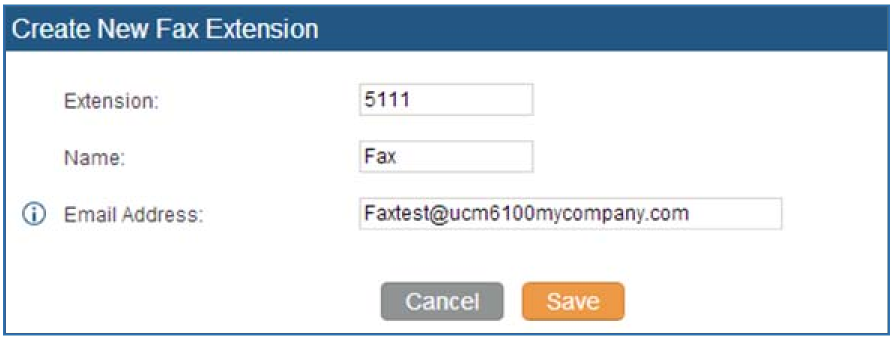


Figure 138: Create Fax Extension

3. Go to UCM6200 web GUI->**Basic/Call Routes**->**Analog Trunks** page. Create a new analog trunk. Please make sure "Fax Detection" is set to "No".
4. Go to UCM6200 web GUI->**Basic/Call Routes**->**Inbound Routes** page. Create a new inbound route and set the default destination to the Fax extension.

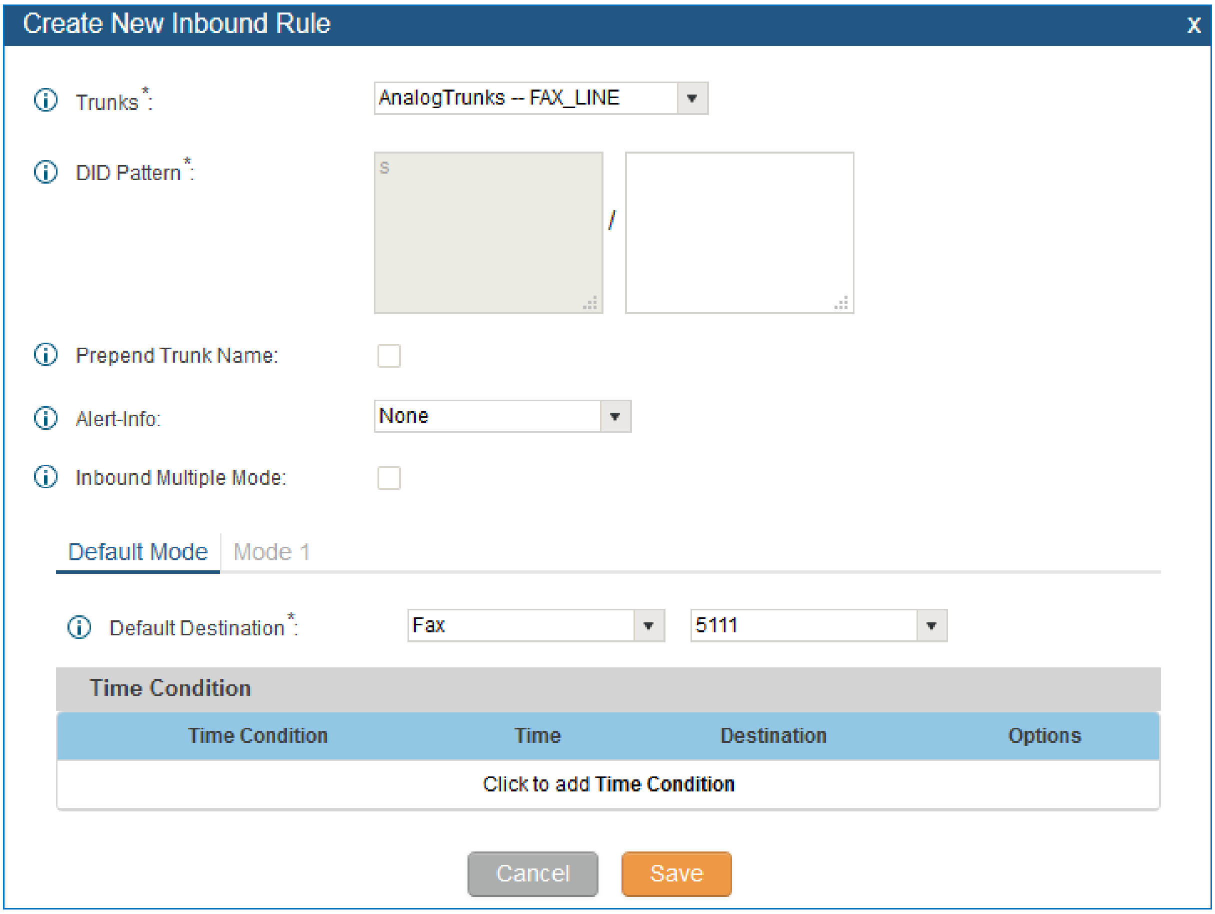


Figure 139: Inbound Route to Fax Extension

5. Once successfully configured, the incoming Fax from external Fax machine to the PSTN line number will be converted to PDF file and sent to the Email address **Faxtest@ucm6200mycompany.com** as attachment.

ASTERISK MANAGER INTERFACE (RESTRICTED ACCESS)

The UCM6200 supports Asterisk Manager Interface (AMI) with restricted access. AMI allows a client program to connect to an Asterisk instance commands or read events over a TCP/IP stream. It's particularly useful when the system admin tries to track the state of a telephony client inside Asterisk.

User could configure AMI parameters on UCM6200 web GUI->**PBX**->**Internal Options**->**AMI**. For details on how to use AMI on UCM6200, please refer to the following AMI guide:

http://www.grandstream.com/sites/default/files/Resources/ucm6200_AMI_guide.pdf

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⚠ **Warning:**

Please do not enable AMI on the UCM6200 if it is placed on a public or untrusted network unless you have taken steps to protect the device from unauthorized access. It is crucial to understand that AMI access can allow AMI user to originate calls and the data exchanged via AMI is often very sensitive and private for your UCM6200 system. Please be cautious when enabling AMI access on the UCM6200 and restrict the permission granted to the AMI user. By using AMI on UCM6200 you agree you understand and acknowledge the risks associated with this.

BUSY CAMP-ON

The UCM6200 supports busy camp-on/call completion feature that allows the PBX to camp on a called party and inform the caller as soon as the called party becomes available given the previous attempted call has failed.

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

http://www.grandstream.com/sites/default/files/Resources/ucm6200_busy_camp_on_guide.pdf

FOLLOW ME

Follow Me is a feature on the UCM6200 that allows users to direct calls to other phone numbers and have them ring all at once or one after the other. Calls can be directed to users' home phone, office phone, mobile and etc. The calls will get to the user no matter where they are. Follow Me option can be found under web **GUI-> PBX-> Call Features->Follow Me.**

To configure follow me:

- Click on "Create New Follow Me" and then select an extension to be configured with Follow Me.

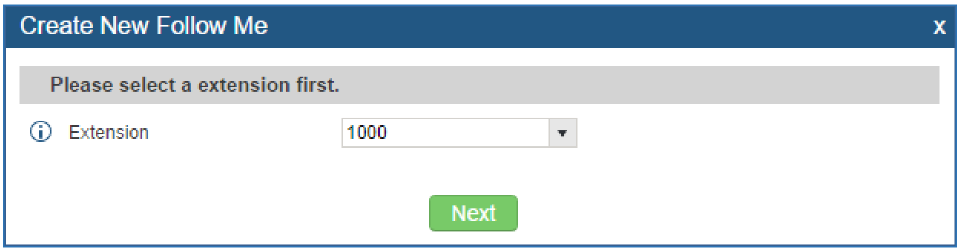


Figure 140: Create Follow Me

- Click on "Next" to continue editing Follow Me configuration.

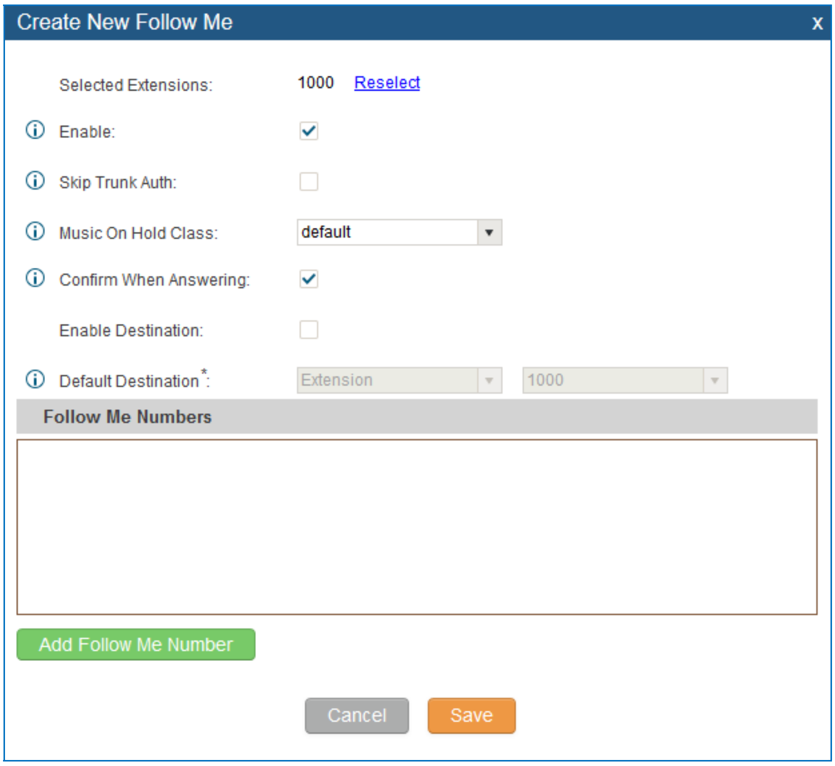


Figure 141: Edit Follow Me

- Click on "Add Follow Me Number" to add local extensions or external numbers to be called after ringing the extension selected in the first step.
- Once created, it will be displayed on the follow me web page list. Click on to edit the Follow Me configuration. Click on to delete the Follow Me.

The following table shows the Follow Me configuration parameters.

Table 70: Follow Me Settings

Enable	Configure to enable or disable Follow Me for this user.
Skip Trunk Auth	If external number is added in the Follow Me, please make sure this option is enabled or the "Skip Trunk Auth" option of the extension is enabled, otherwise the external Follow Me number cannot be reached.
Music On Hold Class	Configure the Music On Hold class that the caller would hear while tracking the user.
Confirm When Answering	By default it is enabled and user will be asked to press 1 to accept the call or to press 2 to reject the call after answering a Follow Me call. If it is disabled, the Follow Me call will be established once after the user answers it.
Enable Destination	When enabled, the call will be routed to the default destination if no one in the Follow Me extensions answers the call.
Default Destination	Configure the destination if no one in the Follow Me extensions answers the call. The available options are: • Extension • Voicemail • Queues • Ring Group • Voicemail Group • IVR • External Number
Follow Me Numbers	The added numbers are listed here. Click on to arrange the order. Click on to delete the number. Click on to add new numbers.
New Follow Me Number	Add a new Follow Me number which could be a 'Local Extension' or 'External Number'. The selected dial plan should have permissions to dial the defined external number.
Dialing Order	Select the order in which the Follow Me destinations will be dialed to reach the user: ring all at once or ring one after the other.

- Click on "Follow Me Options" to enable or disable the options listed in the following table.

Table 71: Follow Me Options

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

ONE-KEY DIAL

The UCM6200 supports One-Key Dial that allows users to call a certain destination by pressing one digit 0 to 9 on the keypad. This creates a system-wide speed dial access for all the extensions on the UCM6200.

To enable One-Key Dial, on the UCM6200 web GUI, go to page **PBX->Call Features->One-Key Dial**.

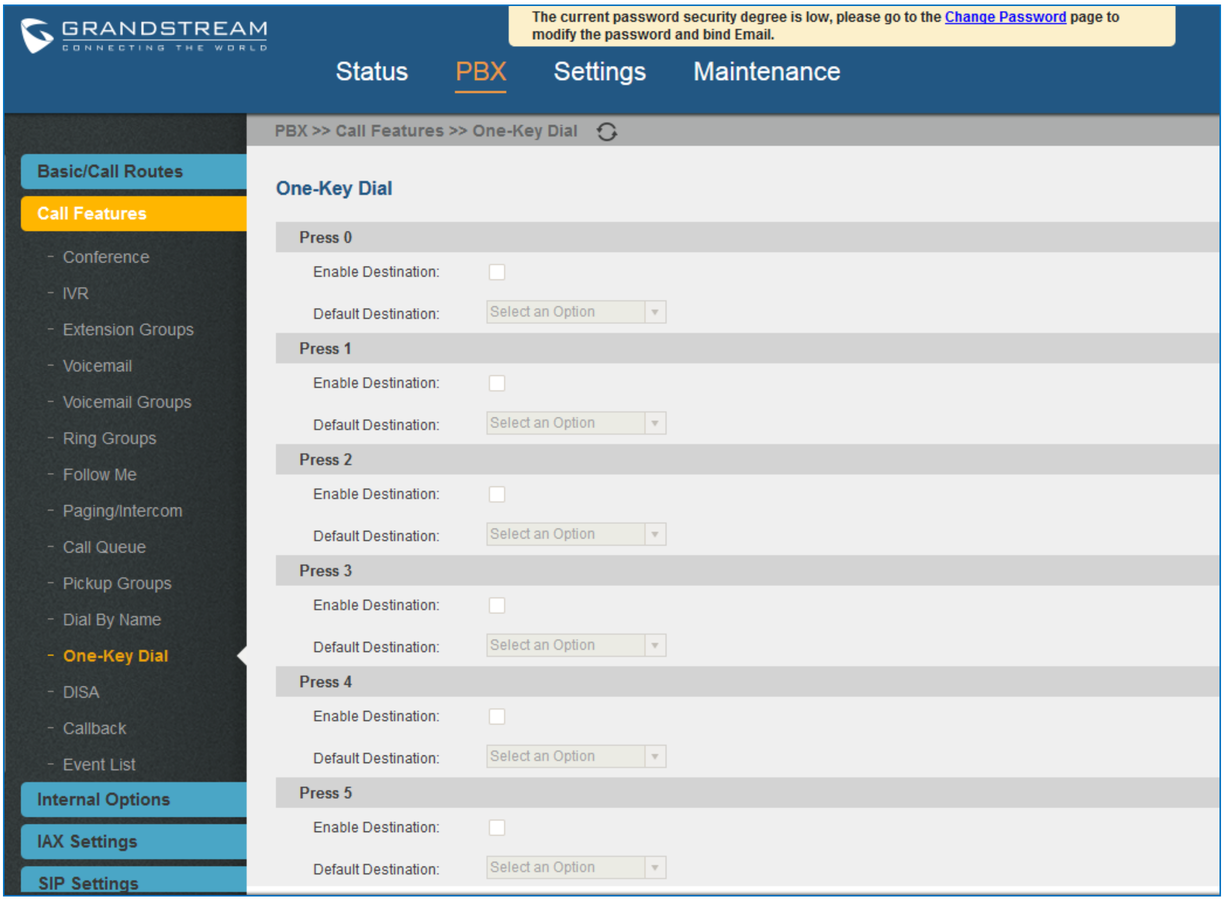


Figure 142: Configure One-Key Dial

User should first decide a digit used for One-Key Dial and check the option “Enable Destination” for the digit. Then select a dial destination from “Default Destination”. The supported destinations include extension, voicemail, conference room, voicemail group, IVR, ring group, call queue, page group, fax, DISA, Dial by Name and external number.

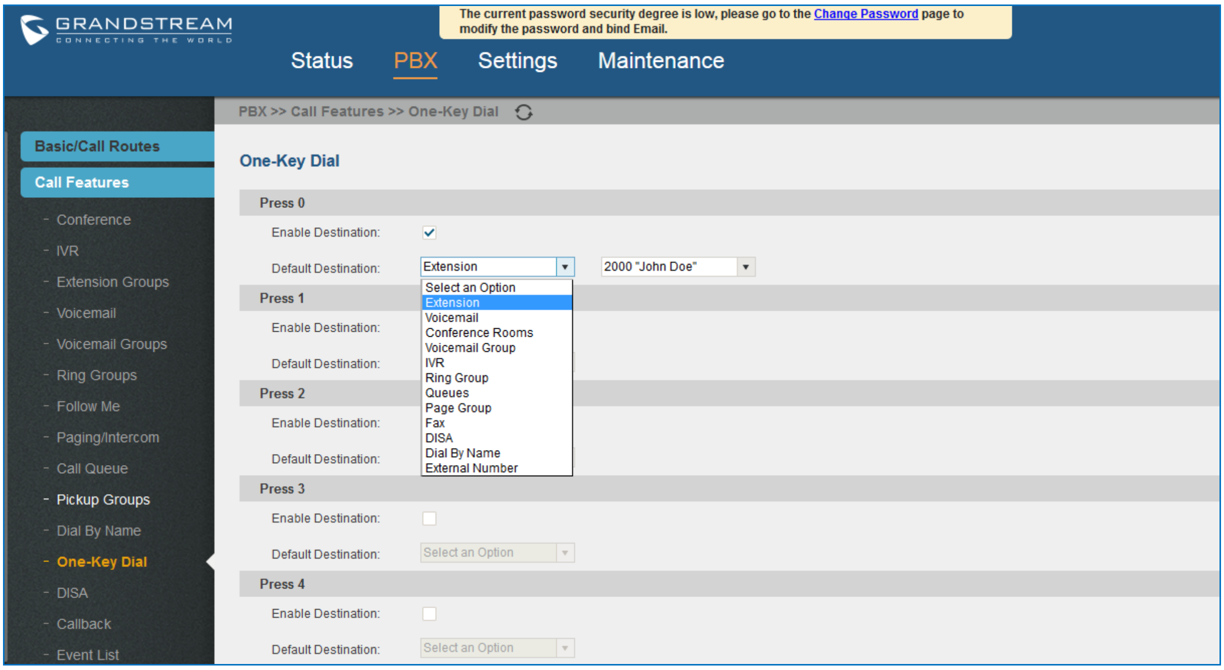


Figure 143: One-Key Dial Destinations

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's using his cell phone, pay phone, regular PSTN, etc. After calling into UCM6200, the user can then dial out via the SIP trunk or PSTN trunk connected to UCM6200 as it is an internal extension.

The UCM6200 supports DISA to be used in IVR or inbound route. Before using it, create new DISA under web GUI->**Call Features**->**DISA**.

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

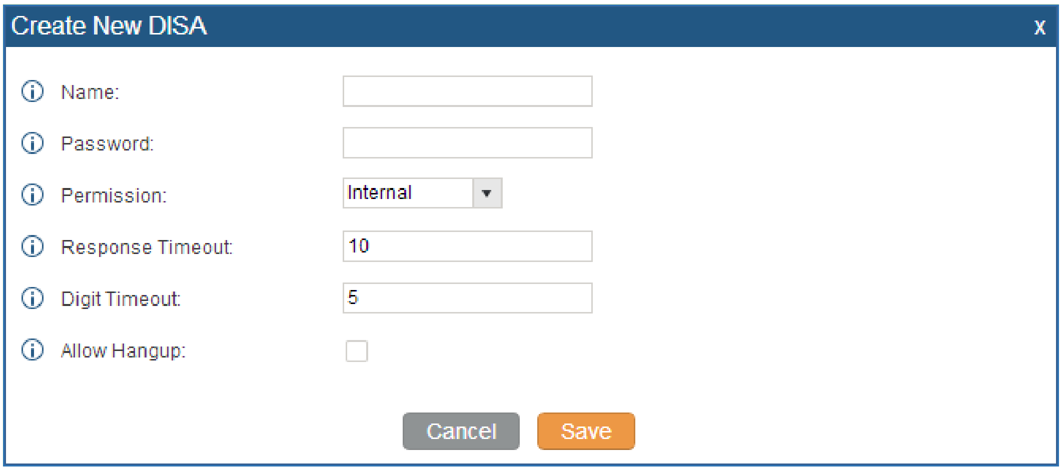


Figure 144: Create New DISA

Table 72: DISA Settings

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 has to 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 UCM6200 will compared 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 UCM6200 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 UCM6200 hangup feature code (by default it's *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".

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 FEATURE

Callback is mainly 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 UCM6200.
2. On the UCM6200, configure destination of the inbound route for analog trunk to callback.
3. Save and apply the settings.
4. The user calls the PSTN number of the UCM6200 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 UCM6200 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 UCM6200 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 UCM6200, go to web GUI->**PBX**->**Call Features**->**Callback** page and click on Create New Callback. Configuration parameters are listed in the following table.

Table 73: Callback Configuration Parameters

Name	Configure a name to identify the Callback.
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: • IVR • DISA The caller can then enter the desired number to dial out via UCM6200 trunk.

BLF AND EVENT LIST

BLF

The UCM6200 supports BLF monitoring for extensions, ring group, call queue, conference room and parking lot. For example, on the user's phone, configure the parking lot number 701 as the BLF monitored number. When there is a parked call on 701, the LED for this BLF key will light up in red, meaning a call is parked against this parking lot. Pressing this BLF key can pick up the call from this parking lot.

⚠ **Note:**

On the Grandstream GXP series phones, the MPK supports "Call Park" mode, which can be used to park the call by configuring the MPK number as call park feature code (e.g., 700). MPK "Call Park" mode can also be used to monitor and pickup parked call if the MPK number is configured as parking lot (e.g., 701).

EVENT LIST

Besides BLF, users can also configure the phones to monitor event list. In this way, both local extensions on the same UCM6200 and remote extensions on the VOIP trunk can be monitored. The event list setting is under web GUI->**Call Features**->**Event List**.

- Click on "Create New Event List" to add a new event list.
- Click on  to edit the event list configuration.
- Click on  to delete the event list.

Table 74: Event List Settings

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 UCM6200. The valid characters are letters, digits, _ and -.
Local Extensions	Select the available extensions/Extension Groups listed on the local UCM6200 to be monitored in the event list.
Remote Extensions	If LDAP sync is enabled between the UCM6200 and the peer UCM6200, 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

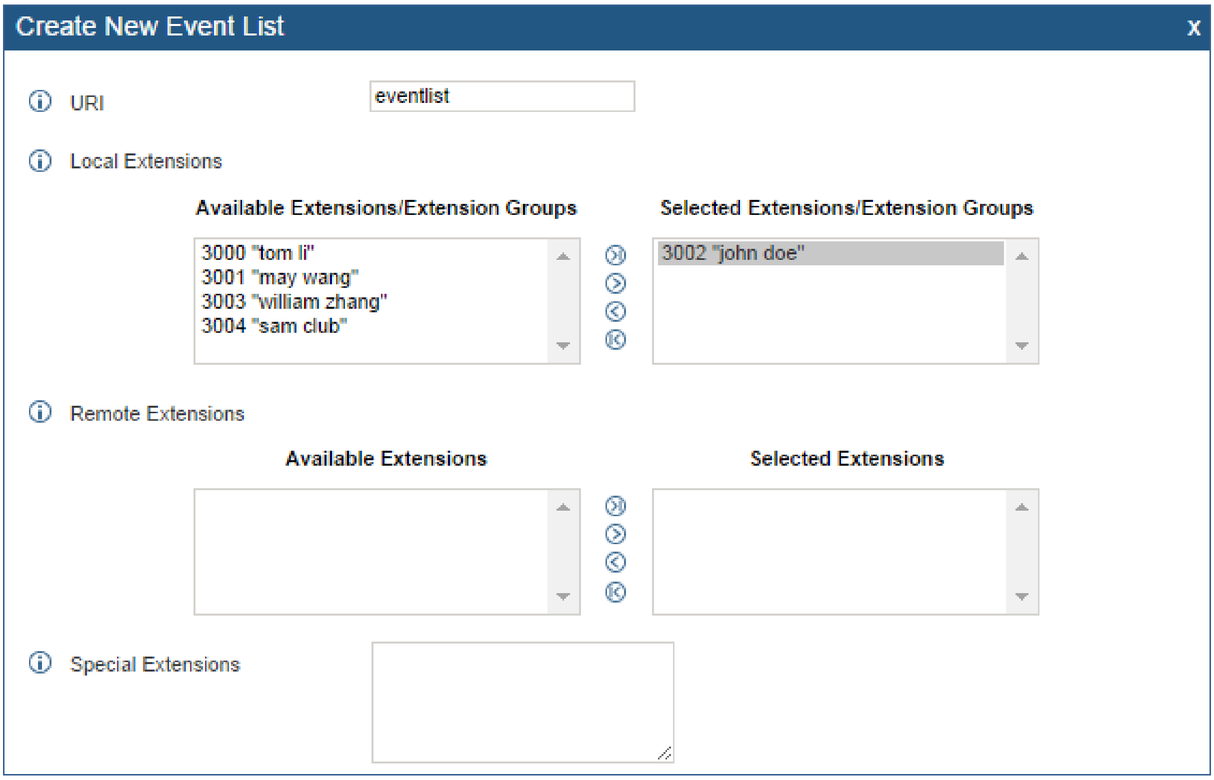


Figure 145: Create New Event List

Remote extension monitoring works on the UCM6200 via event list BLF, among Peer SIP trunks or Register SIP trunks (register to each other). Therefore, please properly configure SIP trunks on the UCM6200 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 UCM6200 and remote extensions are added to the list, the UCM6200 will send out SIP SUBSCIRBE to the remote UCM6200 to obtain the remote extension status. When the SIP end points registers and subscribes to the local UCM6200 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.

⚠ **Note:**

- To configure LDAP sync, please go to UCM6200 web GUI->**PBX**->**Basic/Call Routes**->**VoIP Trunk**. You will see "Sync LDAP Enable" option. Once enabled, please configure password information for the remote peer UCM6200 to connect to the local UCM6200. Additional information such as port number, LDAP outbound rule, LDAP Dialed Prefix will also be required. Both the local UCM6200 and remote UCM6200 need enable LDAP sync option with the same password for successful connection and synchronization.
- Currently LDAP sync feature only works between two UCM6200s.
- (Theoretically) Remote BLF monitoring will work when the remote PBX being monitored is non-UCM6200 PBX. However, it might not work the other way around depending on whether the non-UCM6200 PBX supports event list BLF or remote monitoring feature.

DIAL BY NAME

Dial By Name is a feature on the PBX that allows caller to search a person by first or last name via his/her phone's keypad. The administrator can define the Dial By Name directory including the desired extensions in the directory and the searching type by "first name" or "last name". After dialing in, the PBX IVR/Auto Attendant will guide the caller to spell the digits to find the person in the Dial By Name directory. This feature allows customers/clients to use the guided automatic system to get in touch with the enterprise employees without having to know the extension number, which brings convenience and improves business image for the enterprise.

DIAL BY NAME CONFIGURATION

The administrators can create the dial by name group under web GUI->**PBX**->**Call Features**->**Dial By Name**.

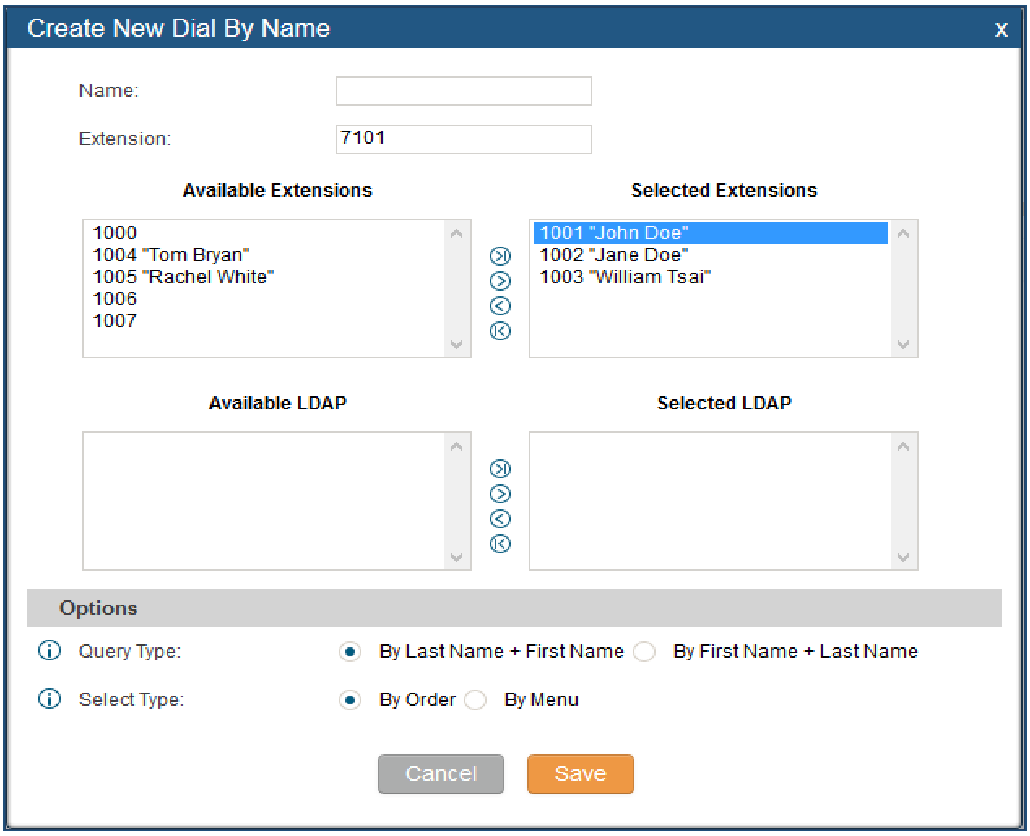


Figure 146: Create Dial By Name Group

1. **Group Name**

Enter the Group Name. This is to identify the Dial By Name group. 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.

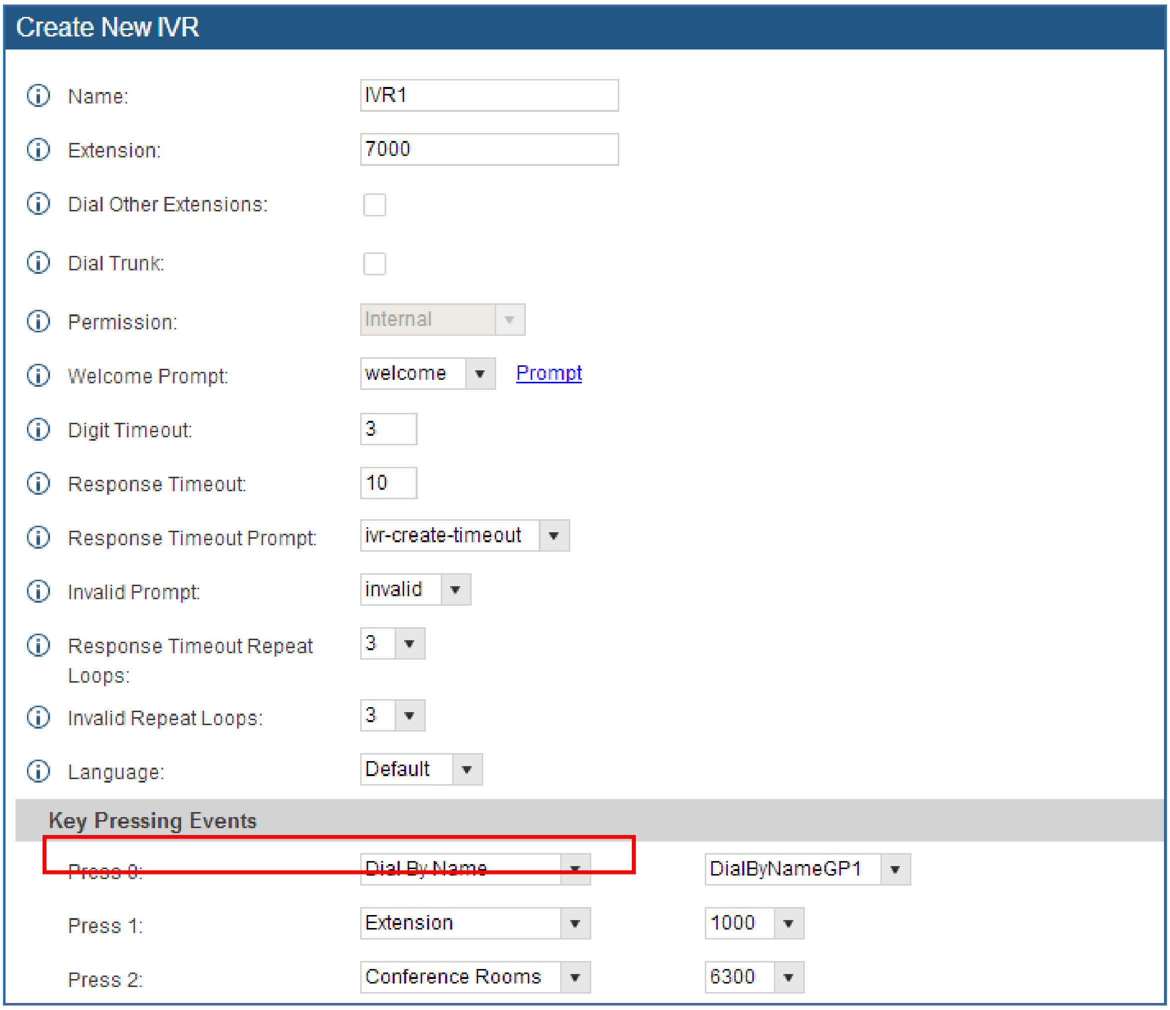


Figure 147: Dial By Name Group In IVR Key Pressing Events

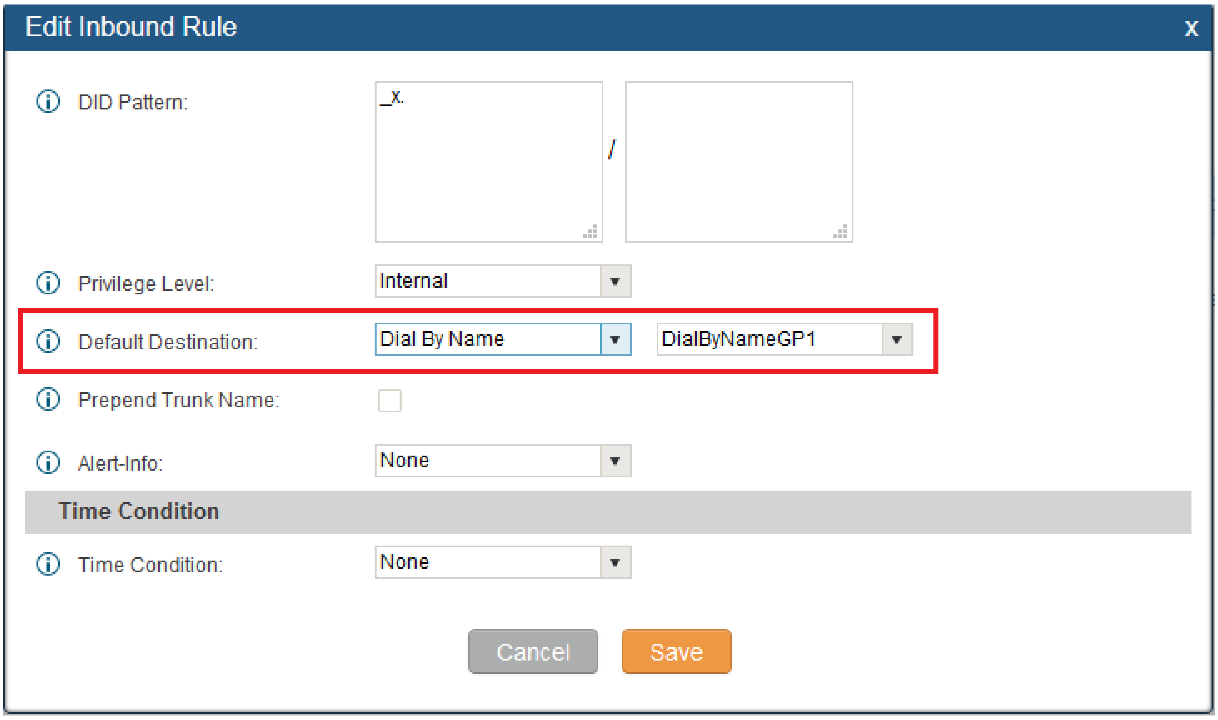


Figure 148: Dial By Name Group In Inbound Rule

2. **Extension**

   Configure the direct dial extension for the Dial By Name group.

3. **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->**PBX**->**Basic/Call Routes**->**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 doesn't want to receive them directly.

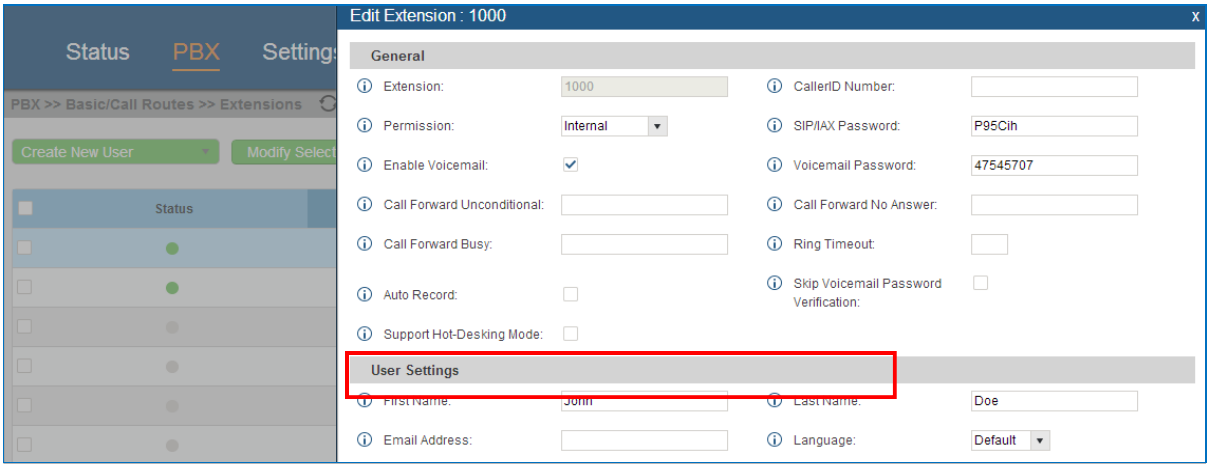


Figure 149: Configure Extension First Name and Last Name

4. **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.
   By Full Name: enter the first 3 digits of the first name or last name to search the directory.

5. **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's the destination party, or press * to listen to the next matching result if it's 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.

ACTIVE CALLS AND MONITOR

The active calls on the UCM6200 are displayed in web UI->**Status**->**Active Calls** page. Users can monitor the status, hang up the call as well as barge in the active calls in real time manner.

ACTIVE CALLS STATUS

To view the status of active calls, navigate to web GUI->**Status**->**Active Calls**. The following figure shows extension 1000 is calling 1001. 1001 is ringing.

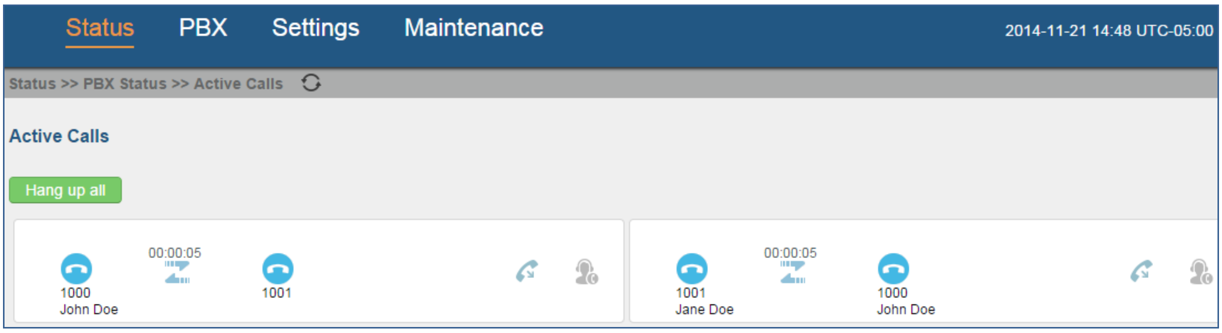


Figure 150: Status->PBX Status->Active Calls - Ringing

The following figure shows the call between 1000 and 1001 is established.

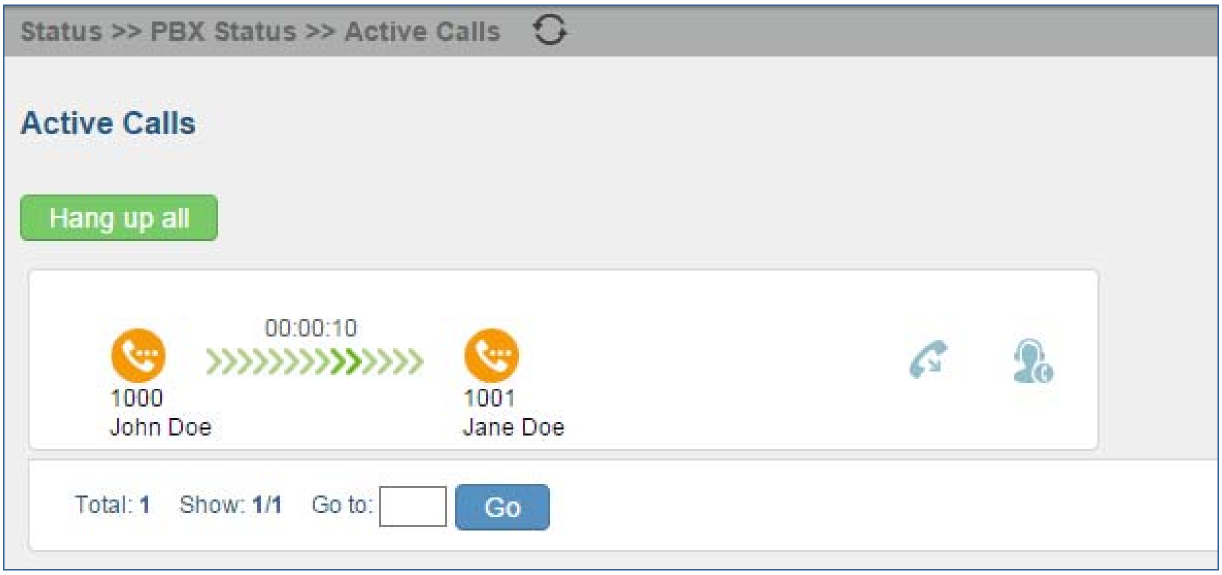


Figure 151: Status->PBX Status->Active Calls – Call Established

In active call web page, click on to refresh the active call status.

HANG UP ACTIVE CALLS

To hang up an active call, click on icon in the active call dialog. Users can also click on Hang up all to hang up all active calls.

CALL MONITOR

During an active call, click on icon and the monitor dialog will pop up.

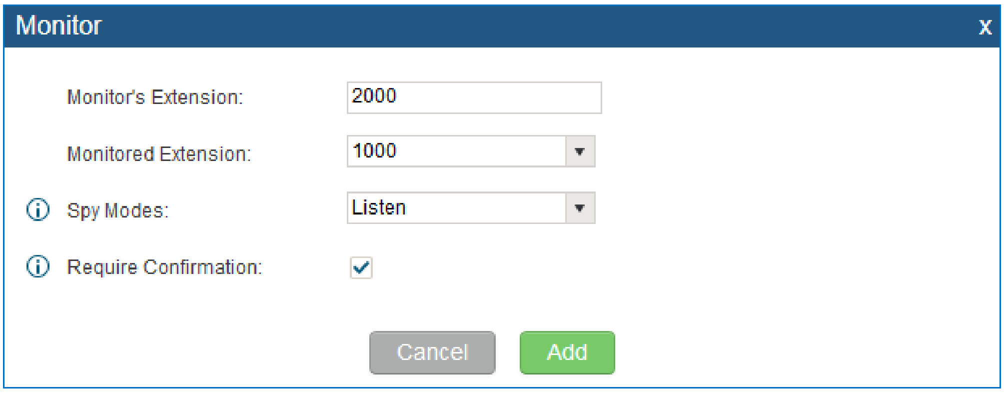


Figure 152: Configure to Monitor an Active Call

In the “Monitor” dialog, configure the following to monitor an active call:

1. Enter an available extension for “Monitor’s Extension” which will be used to monitor the active call.
2. “Monitored Extension” must be one of the parties in the active call to be monitored.
3. Select spy mode. There are three options in “Spy Mode”.
    - Listen
      In “Listen” mode, the extension monitoring the call can hear both parties in the active call but the audio of the user on this extension will not be heard by either party in the monitored active call.
    - Whisper
      In “Whisper” mode, the extension monitoring the call can hear both parties in the active call. The user on this extension can only talk to the selected monitored extension and he/she will not be heard by the other party in the active call. This can be usually used to supervise calls.
    - Barge
      In “Barge” mode, the extension monitoring the call can talk to both parties in the active call. The call will be established similar to three-way conference.

4. Enable or disable “Require Confirmation” option. If enabled, the confirmation of the invited monitor’s extension is required before the active call can be monitored. This option can be used to avoid adding participant who has auto-answer configured or call forwarded to voicemail.
5. Click on “Add”. An INVITE will be sent to the monitor’s extension. The monitor can answer the call and start monitoring. If “Require Confirmation” is enabled, the user will be asked to confirm to monitor the call.

Another way to monitor active calls is to dial the corresponding feature codes from an extension. Please refer to ***[Table 75: UCM6200 Feature Codes]*** and ***[ENABLE SPY]*** section for instructions.

CALL FEATURES

The UCM6200 supports call recording, transfer, call forward, call park and other call features via feature code. This section lists all the feature codes in the UCM6200 and describes how to use the call features.

FEATURE CODES

Table 75: UCM6200 Feature Codes

Feature Maps	
Blind Transfer	• 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 **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	• 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. • Options **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.
Disconnect	• Default code: *0. • Enter the code during active call. It will disconnect the call. • Options **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	• Default code: #72. • Enter the code during active call to park the call. • Options **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.
Audio Mix Record	• Default code: *3. • Enter the code followed by # or SEND to start recording the audio call and the UCM6200 will mix the streams natively on the fly as the call is in progress. • Options **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.
DND/Call Forward	
Do Not Disturb (DND) Activate	• Default code: *77.
Do Not Disturb (DND) Deactivate	• Default code: *78.
Call Forward Busy Activate	• Default Code: *90. • Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward Busy Deactivate	• Default Code: *91.
Call Forward No Answer Activate	• Default Code: *92. • Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward No Answer Deactivate	• Default Code: *93.
Call Forward Unconditional Activate	• Default Code: *72. • Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward Unconditional Deactivate	• Default Code: *73.
Feature Misc	

Feature Code Digits Timeout	• Default Setting: 1000. • Configure the maximum interval (in milliseconds) between the digits input to activate the feature code.
Call Park	• Default Extension: 700. • During an active call, initiate blind transfer and then enter this code to park the call.
Parked Lots	• 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	• 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)	• Default setting: 300. • 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.
Feature Codes	
Voicemail Access Code	• 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	• Default Code: *97. • Press *97 to access the voicemail box.
Agent Pause	• Default Code: *83. • Pause the agent in all call queues.
Agent Unpause	• Default Code: *84. • Unpause the agent in all call queues.
Paging Prefix	• Default Code: *81. • To page an extension, enter the code followed by the extension number.
Intercom Prefix	• Default Code: *80. • To intercom an extension, enter the code followed by the extension number.

Blacklist Add	• Default Code: *40. • To add a number to blacklist for inbound route, dial *40 and follow the voice prompt to enter the number.
Blacklist Remove	• Default Code: *41. • To remove a number from current blacklist for inbound route, dial *41 and follow the voice prompt to remove the number.
Call Pickup on Ringing	• Default Code: **. • To pick up a call for any extension xxxx, enter the code followed by the extension number xxxx.
Pickup Extension	• Default Code: *8. • This code is for the pickup group which can be assigned for each extension on the extension configuration page. • If there is an incoming call to an extension, the other extensions within the same pickup group can dial *8 directly to pick up the call.
Direct Dial Voicemail Prefix	• Default Code: * • This code is for the user to directly dial or transfer to an extension's voicemail. • For example, directly dial *5000 will have to call go into the extension 5000's voicemail. If the user would like to transfer the call to the extension 5000's voicemail, enter *5000 as the transfer target number.
Call Completion Request	• Default Code: *11 • This code is for the user who wants to use Call Completion to complete a call.
Call Completion Cancel	• Default Code: *12 • This code is for the user who wants to cancel Call Completion request.
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. Monitor’s line will be muted, and neither party will hear from the monitor’s extension. The default setting is *54.
Whisper Spy	This is the feature code to speak to one side of the call (for example, whisper to employees to help them handle a call). Only one side will be able to hear from the monitor’s extension. The default setting is *55.

Barge Spy	This is the feature code to join in on the call to assist both parties. The default setting is *56.
Enable Inbound Multiple Mode	If enabled, user can switch between different inbound route modes with feature code. By default, this option is disabled.
Inbound Default Mode	This feature code is used to switch inbound route mode to default mode. The default setting is *61.
Inbound Mode 1	This feature code is used to switch inbound route mode to mode 1. The default setting is *62.

CALL RECORDING

The UCM6200 allows users to record audio during the call. If "Auto Record" is turned on for an extension, ring group, call queue or trunk, the call will be automatically recorded when there is established call with it. Otherwise, please follow the instructions below to manually record the call.

1. Make sure the feature code for "Audio Mix Record" is configured and enabled.

2. After establishing the call, enter the "Audio Mix Record" feature code (by default it's *3) followed by # or SEND to start recording.

3. To stop the recording, enter the "Audio Mix Record" feature code (by default it's *3) followed by # or SEND again. Or the recording will be stopped once the call hangs up.

4. The recording file can be retrieved under Web GUI->**Status**->**CDR**. Click on to play the recording or click on to download the recording file.

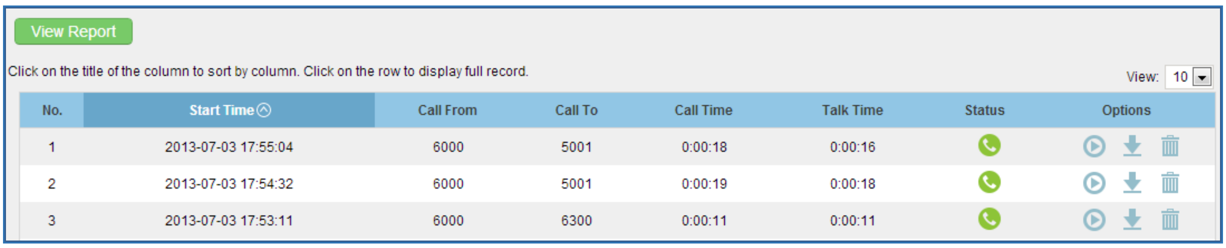


Figure 153: Download Recording File from CDR Page

The above recorded call's recording files are also listed under the UCM6200 web GUI->**CDR**->**Recording Files**.

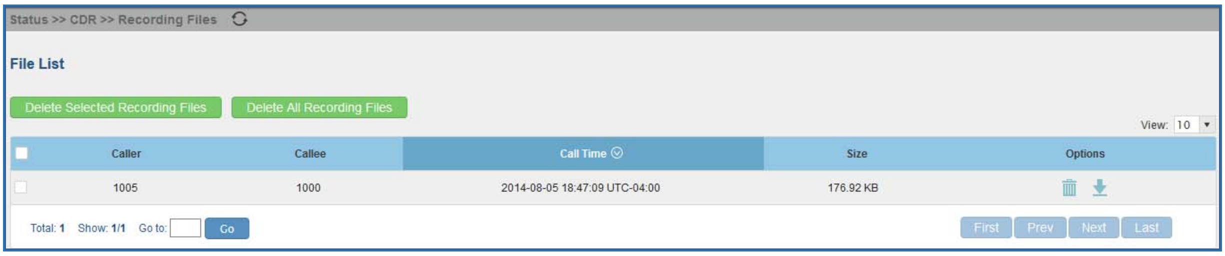


Figure 154: Download Recording File from Recording Files Page

CALL PARK

The UCM6200 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 THE 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.

ENABLE SPY

If "Enable Spy" option is enabled, feature codes for Listen Spy, Whisper Spy and Barge Spy are available for users to dial from any extension to perform the corresponding actions.

Assume a call is on-going between extension A and extension B, user could dial the feature code from extension C to listen on their call (*54 by default), whisper to one side (*55 by default), or barge into the call (*56 by default). Then the user will be asked to enter the number to call, which should be either side of the active call, extension A or B in this example.

⚠ **Caution:**

"Enable Spy" allows any user to listen to any call by feature codes. This may result in the leakage of user privacy.

INTERNAL OPTIONS

This section describes internal options that haven't been mentioned in previous sections yet. The settings in this section can be applied globally to the UCM6200, including general configurations, jitter buffer, RTP settings, ports config and STUN monitor. The options can be accessed via Web GUI->**PBX**->**Internal Options-> General**.

INTERNAL OPTIONS/GENERAL

Table 76: Internal Options/General

General Preferences	
Global OutBound CID	Configure the global CallerID used for all outbound calls when no other CallerID is defined with higher priority. If no CallerID is defined for extension or trunk, the global outbound CID will be used as CallerID.
Global OutBound CID Name	Configure the global CallerID Name used for all outbound calls. If configured, all outbound calls will have the CallerID Name set to this name. If not, the extension's CallerID Name will be used.
Operator Extension	Specify the operator extension, which will be dialed when users press 0 to exit voicemail application. The operator extension can also be used in IVR option.
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	Configure the maximum duration of call-blocking.
Record Prompt	If enabled, users will hear voice prompt before recording is started or stopped. For example, before recording, the UCM6200 will play voice prompt "The call will be recorded". The default setting is "No".
Extension Preferences	
Enforce Strong Passwords	If enabled, strong password will be enforced for the password created on the UCM6200. The default setting is enabled. Strong Password Rules: 1. Password for voicemail, voicemail group, outbound route, DISA, call queue and conference requires non-repetitive and non-sequential

	digits, with a minimum length of 4 digits. Repetitive digits pattern (such as 0000, 1111, 1234, 2345, and etc), or common digits pattern (such as 111222, 321321 and etc) are not allowed to be configured as password. 2. Password for extension registration, web GUI admin login, LDAP and LDAP sync requires alphanumeric characters containing at least two categories of the following, with a minimum length of 4 characters. • Numeric digits • Lowercase alphabet characters • Uppercase alphabet characters • Special characters
Enable Random Password	If enabled, random password will be generated when the extension is created. The default setting is "Yes". It is recommended to enable it for security purpose.
Enable Auto Email To User	If enabled, UCM6200 will send Email notification to user automatically after editing extension settings or adding a new extension.
Disable Extension Range	If set to "Yes", users could disable the extension range pre-configured/configured on the UCM6200. The default setting is "No". **Note:** It is recommended to keep the system assignment to avoid inappropriate usage and unnecessary issues.
Extension Ranges	The default extension range assignment is: • User Extensions: 1000-6299 User Extensions is referring to the extensions created under web UI->**PBX**->**Basic/Call Routes**->**Extensions** page. • Pick Extensions: 4000-4999 This refers to the extensions that can be manually picked from end device when being provisioned by the UCM6200. There are two related options in zero config page->Auto Provision Settings, "Pick Extension Segment" and "Enable Pick Extension". If "Enable Pick Extension" under zero config settings is selected, the extension list defined in "Pick Extension Segment" will be sent out to the device after receiving the device's request. This "Pick Extension Segment" should be a subset of the "Pick Extensions" range here. This feature is for the GXP series phones that support selecting extension to be provisioned via phone's LCD.

	• Auto Provision Extensions: 5000-6299 This sets the range for "Zero Config Extension Segment" which is the extensions can be assigned on the UCM6200 to provision the end device. • Conference Extensions: 6300-6399 • Ring Group Extensions: 6400-6499 • Queue Extensions: 6500-6599 • Voicemail Group Extensions: 6600-6699 • IVR Extensions: 7000-7100 • Dial By Name Extensions: 7101-7199 • Fax Extensions: 7200-8200

INTERNAL OPTIONS/JITTER BUFFER

Table 77: Internal Options/Jitter Buffer

SIP Jitter Buffer	
Enable Jitter Buffer	Select to enable jitter buffer on the sending side of the SIP channel. The default setting is "No".
Jitter Buffer Size	Configure the time (in ms) to buffer. This is the jitter buffer size used in "Fixed" jitter buffer, or used as the initial time for "adaptive" jitter buffer. The default setting is 100.
Max Jitter Buffer	Configure the maximum time (in ms) to buffer for "Adaptive" jitter buffer implementation, or used as the jitter buffer size for "Fixed" jitter buffer implementation. The default setting is 200.
Implementation	Configure the jitter buffer implementation on the sending side of a SIP channel. The default setting is "Fixed". • **Fixed** The size is always equal to the value of "Max Jitter Buffer". • **Adaptive** The size is adjusted automatically and the maximum value equals to the value of "Max Jitter Buffer".

INTERNAL OPTIONS/RTP SETTINGS

Table 78: Internal Options/RTP Settings

RTP Start	Configure the RTP port starting number. The default setting is 10000.
RTP End	Configure the RTP port ending address. The default setting is 20000.
Strict RTP	Configure to enable or disable strict RTP protection. If enabled, RTP packets that do not come from the source of the RTP stream will be dropped. The default setting is "Disable".
RTP Checksums	Configure to enable or disable RTP Checksums on RTP traffic. The default setting is "Disable".
ICE Support	Configure whether to support ICE. The default setting is enabled. ICE is the integrated use of STUN and TURN structure to provide reliable VoIP or video calls and media transmission, via a SIP request/ response model or multiple candidate endpoints exchanging IP addresses and ports, such as private addresses and TURN server address.
STUN Server	Configure STUN server address. STUN protocol is a Client/Server and also a Request/Response protocol. It’s used to check the connectivity between the two terminals, such as maintaining a NAT binding entries keep-alive agreement. The default STUN Server is stun.ipvideotalk.com. Valid format: [(hostname | IP-address) [':' port] The default port number is 3478 if not specified.

INTERNAL OPTIONS/PAYLOAD

The UCM6200 payload type for audio codecs and video codes can be configured here.

Table 79: Internal Options/Payload

AAL2-G.726	Configure payload type for ADPCM (G.726, 32kbps, AAL2 codeword packing). The default setting is 112.
DTMF	Configured payload type for DTMF. The default setting is 101.

G.721 Compatible	Configure to enable/disable G.721 compatible. The default setting is Yes.
G.726	Configure the payload type for G.726 if "G.721 Compatible" is disabled. The default setting is 111.
iLBC	Configure the payload type for iLBC. The default setting is 97.
H.264	Configure the payload type for H.264. The default setting is 99.
H.263P	Configure the payload type for H.263+. The default setting is 100 103.
VP8	Configure the payload type for VP8. The default settings is 108.

IAX SETTINGS

The UCM6200 IAX global settings can be accessed via Web GUI->**PBX**->**IAX Settings**.

IAX SETTINGS/GENERAL

Table 80: IAX Settings/General

Bind Port	Configure the port number that the IAX2 will be allowed to listen to. The default setting is 4569.
Bind Address	Configure the address that the IAX2 will be forced to bind to. The default setting is 0.0.0.0, which means all addresses.
IAX1 Compatibility	Select to configure IAX1 compatibility. The default setting is "No".
No Checksums	If selected, UDP checksums will be disabled and no checksums will be calculated/checked on systems supporting this features. The default setting is "No".
Delay Reject	If enabled, the IAX2 will delay the rejection of calls to avoid DOS. The default setting is "No".
ADSI	Select to enable ADSI phone compatibility. The default setting is "No".
Music On Hold Interpret	Specify which Music On Hold class this channel would like to listen to when being put on hold. This music class is only effective if this channel has no music class configured and the bridged channel putting the call on hold has no "Music On Hold Suggest" setting.
Music On Hold Suggest	Specify which Music On Hold class to suggest to the bridged channel when putting the call on hold.
Bandwidth	Configure the bandwidth for IAX settings. The default setting is "Low".

IAX SETTINGS/REGISTRATION

Table 81: IAX Settings/Registration

IAX Registration Options	
Min Reg Expire	Configure the minimum period (in seconds) of registration. The default setting is 60.
Max Reg Expire	Configure the maximum period (in seconds) of registration. The default setting is 3600.
IAX Thread Count	Configure the number of IAX helper threads. The default setting is 10.
IAX Max Thread Count	Configure the maximum number of IAX threads allowed. The default setting is 100.

Auto Kill	If set to "yes", the connection will be terminated if ACK for the NEW message is not received within 2000ms. Users could also specify number (in milliseconds) in addition to "yes" and "no". The default setting is "yes".
Authentication Debugging	If enabled, authentication traffic in debugging will not show. The default setting is "No".
Codec Priority	Configure codec negotiation priority. The default setting is "Reqonly". • Caller Consider the callers preferred order ahead of the host's. • Host Consider the host's preferred order ahead of the caller's. • Disabled Disable the consideration of codec preference all together. • Reqonly This is almost the same as "Disabled", except when the requested format is not available. The call will only be accepted if the requested format is available.
Type of Service	Configure ToS bit for preferred IP routing.
IAX Trunk Options	
Trunk Frequency	Configure the frequency of trunk frames (in milliseconds). The default setting is 20.
Trunk Time Stamps	If enabled, time stamps will be attached to trunk frames. The default setting is "No".

IAX SETTINGS/STATIC DEFENSE

Table 82: IAX Settings/Static Defense

Call Token Optional	Enter a single IP address (e.g., 11.11.11.11) or a range of IP addresses (11.11.11.11/22.22.22.22) for which call token validation is not required.
Max Call Numbers	Configure the maximum number of calls allowed for a single IP address.
Max Unvalidated Call Numbers	Configure the maximum number of unvalidated calls for all IP addresses.
Call Number Limits	Configure to limit the number of calls for a give IP address of IP range.
IP or IP Range	Enter the IP address (11.11.11.11) or a range of IP addresses (11.11.11.11/22.22.22.22) to be considered for call number limits.

SIP SETTINGS

The UCM6200 SIP global settings can be accessed via Web GUI->**PBX**->**SIP Settings**.

SIP SETTINGS/GENERAL

Table 83: SIP Settings/General

Realm For Digest Authentication	Configure the host name or domain name for the UCM6200. Realms MUST be globally unique according to RFC3261. The default setting is Grandstream.
Bind UDP Port	Configure the UDP port used for SIP. The default setting is 5060.
Bind IP Address	Configure the IP address to bind to. The default setting is 0.0.0.0, which means binding to all addresses.
Allow Guest Calls	If enabled, the UCM6200 allows unauthorized INVITE coming into the PBX and the call can be made. The default setting is "No". **Warning:** Please be aware of the potential security risk when enabling "Allow Guest Calls" as this will allow any user with the UCM6200 address to dial into the UCM6200.
Allow Transfer	If set to "No", all transfers initiated by the endpoint in the UCM6200 will be disabled (unless enabled in peers or users). The default setting is "Yes".
MWI From	When sending MWI NOTIFY requests, this value will be used in the "From:" header as the "name" field. If no "From User" is configured, the "user" field of the URI in the "From:" header will be filled with this value.

SIP SETTINGS/MISC

Table 84: SIP Settings/Misc

Outbound SIP Registrations	
Register Timeout	Configure the register retry timeout (in seconds). The default setting is 20.
Register Attempts	Configure the number of registration attempts before the UCM6200 gives up. The default setting is 0, which means the UCM6200 will keep trying until the server side accepts the registration request.
Video	
Max Bit Rate (kb/s)	Configure the maximum bit rate (in kb/s) for video calls. The default setting is 384.

Support SIP Video	Select to enable video support in SIP calls. The default setting is "Yes".
Reject Non-Matching INVITE	If enabled, when rejecting an incoming INVITE or REGISTER request, the UCM6200 will always reject with "401 Unauthorized" instead of notifying the requester whether there is a matching user or peer for the request. This reduces the ability of an attacker to scan for valid SIP usernames. The default setting is "No".

SIP SETTINGS/SESSION TIMER

Table 85: SIP Settings/Session Timer

Session Timers	Select the session timer mode. The default setting is "Accept". The options are: • Originate Always request and run session timer. • Accept Run session timer only when requested by other UA. • Refuse Do not run session timer.
Session Expire	Configure the maximum session refresh interval (in seconds). The default setting is 1800.
Min SE	Configure the minimum session refresh interval (in seconds). The default setting is 90.
Session Refresher	Select the session refresher to be UAC or UAS. The default setting is UAC.

SIP SETTINGS/TCP AND TLS

Table 86: SIP Settings/TCP and TLS

TCP Enable	Configure to allow incoming TCP connections with the UCM6200. The default setting is "No".
TCP Bind Address	Configure the IP address for TCP server to bind to. 0.0.0.0 means binding to all interfaces. The port number is optional. If not specified, 5060 will be used.
TLS Enable	Configure to allow incoming TLS connections with the UCM6200. The default setting is "No".
TLS Bind Address	Configure the IP address for TLS server to bind to. 0.0.0.0 means binding to all interfaces. The port number is optional. If not specified, 5061 will be used.

	Note: The IP address must match the common name (hostname) in the certificate. Please do not bind a TLS socket to multiple IP addresses. For details on how to construct a certificate for SIP, please refer to the following document: http://tools.ietf.org/html/draft-ietf-sip-domain-certs
TLS Client Protocol	Select the TLS protocol for outbound client connections. The default setting is TLSv1.
TLS Do Not Verify	If enabled, the TLS server's certificate won't be verified when acting as a client. The default setting is "Yes".
TLS Self-Signed CA	This is the CA certificate if the TLS server being connected to requires self-signed certificate, including server's public key. This file will be renames as "TLS.ca" automatically. **Note:** The size of the uploaded ca file must be under 2MB.
TLS Cert	This is the Certificate file (*.pem format only) used for TLS connections. It contains private key for client and signed certificate for the server. This file will be renamed as "TLS.pem" automatically. **Note:** The size of the uploaded certificate file must be under 2MB.
TLS CA Cert	This file must be named with the CA subject name hash value. It contains CA's (Certificate Authority) public key, which is used to verify the accessed servers. **Note:** The size of the uploaded CA certificate file must be under 2MB.
TLS CA List	Display a list of files under the CA Cert directory.

SIP SETTINGS/NAT

Table 87: SIP Settings/NAT

External Host	Configure a static IP address and port (optional) used in outbound SIP messages if the UCM6200 is behind NAT. If it is a host name, it will only be looked up once.
Use IP address in SDP	If enabled, the SDP connection will use the IP address resolved from the external host.

External TCP Port	Configure the externally mapped TCP port when the UCM6200 is behind a static NAT or PAT.
External TLS Port	Configures the externally mapped TLS port when UCM6200 is behind a static NAT or PAT.
Local Network Address	Specify a list of network addresses that are considered inside of the NAT network. Multiple entries are allowed. If not configured, the external IP address will not be set correctly. A sample configuration could be as follows: 192.168.0.0/16

SIP SETTINGS/TOS

Table 88: SIP Settings/ToS

ToS For SIP	Configure the Type of Service for SIP packets. The default setting is None.
ToS For RTP Audio	Configure the Type of Service for RTP audio packets. The default setting is None.
ToS For RTP Video	Configure the Type of Service for RTP video packets. The default setting is None.
Default Incoming/Outgoing Registration Time	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 UCM6200. The default setting is 3600.
Min Registration/Subscription Time	Configure the minimum duration (in seconds) of incoming registration and subscription allowed by the UCM6200. 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 RFC2833. If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit codec PCMU and PCMA are required. When "Auto" is selected, "RFC2833" will be used if offered, otherwise "Inband" will be used. The default setting is "RFC2833".
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 doesn't 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.
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	Configure whether the UCM6200 should generate inband ringing or not. The default setting is "Never". • Yes: The UCM6200 will send 180 Ringing followed by 183 Session Progress and in-band audio. • No: The UCM6200 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 UCM6200 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 UCM6200.
Send Compact SIP Headers	If enabled, compact SIP headers will be sent. The default setting is "No".
100rel	Configure the 100rel setting on UCM6200. The default setting is “Yes”.

PORTS CONFIG

The analog hardware (FXS port and FXO port) on the UCM6200 will be listed in this page. Click on to edit signaling preference for FXS port or configure ACIM settings for FXO port.

Select "Loop Start" or "Kewl Start" for each FXS port. And then click on "Update" to save the change.

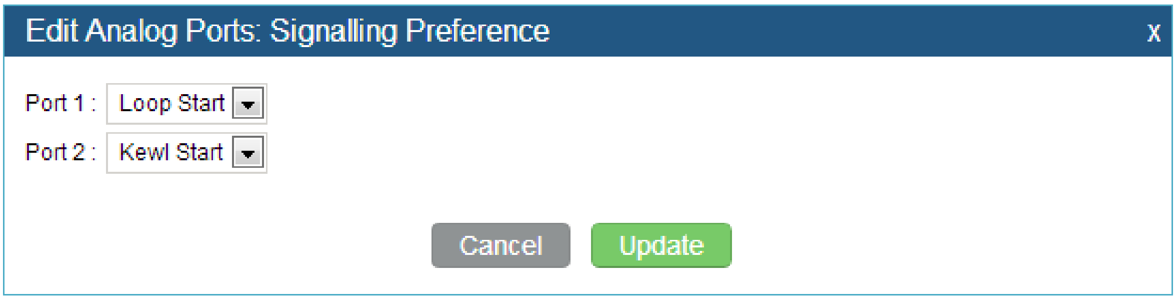


Figure 155: FXS Ports Signaling Preference

For FXO port, users could manually enter the ACIM settings by selecting the value from dropdown list for each port. Or users could click on "Detect" for the UCM6200 to automatically detect the ACIM value. The detecting value will be automatically filled into the settings.

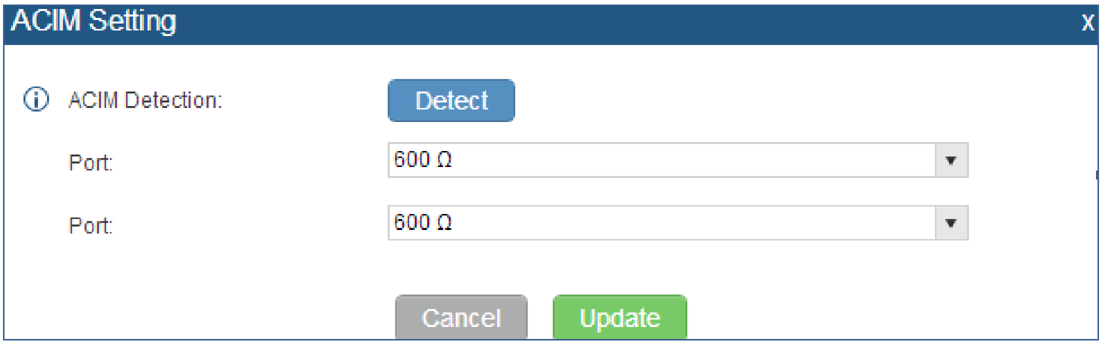


Figure 156: FXO Ports ACIM Settings

Table 89: Internal Options/Ports Config

Tone Region	Select country to set the default tones for dial tone, busy tone, ring tone and etc to be sent from the FXS port. The default setting is "United States of America (USA)".
Advanced Settings	
FXO Opermode	Select country to set the On Hook Speed, Ringer Impedance, Ringer

	Threshold, Current Limiting, TIP/RING voltage adjustment, Minimum Operational Loop Current, and AC Impedance as predefined for your country's analog line characteristics. The default setting is "United States of America (USA)".
FXS Opermode	Select country to set the On Hook Speed, Ringer Impedance, Ringer Threshold, Current Limiting, TIP/RING voltage adjustment, Minimum Operational Loop Current, and AC Impedance as predefined for your country's analog line characteristics. The default setting is "United States of America (USA)".
FXS TISS Override	Configure to enable or disable override Two-Wire Impedance Synthesis (TISS). The default setting is No. If enabled, users can select the impedance value for Two-Wire Impedance Synthesis (TISS) override. The default setting is 600Ω.
PCMA Override	Select the codec to be used for analog lines. North American users should choose PCMU. All other countries, unless already known, should be assumed to be PCMA. The default setting is PCMU. **Note:** This option requires system reboot to take effect.
Boost Ringer	Configure whether normal ringing voltage (40V) or maximum ringing voltage (89V) for analog phones attached to the FXS port is required. The default setting is "Normal".
Fast Ringer	Configure to increase the ringing speed to 25HZ. This option can be used with "Low Power" option. The default setting is "Normal".
Low Power	Configure the peak voltage up to 50V during "Fast Ringer" operation. This option is used with "Fast Ringer". The default setting is "Normal".
Ring Detect	If set to "Full Wave", false ring detection will be prevented for lines where Caller ID is sent before the first ring and proceeded by a polarity reversal, as in UK. The default setting is "Standard".
FXS MWI Mode	Configure the type of Message Waiting Indicator on FXS lines. The default setting is "FSK". • FSK: Frequency Shift Key Indicator • NEON: Light Neon Bulb Indicator.

VALUE-ADDED FEATURES

FAX SENDING

The UCM6200 supports sending Fax via web UI access. This feature can be found on web UI->**PBX**->**Value-added Features**->**Fax Sending** page. In order to send fax, pre-setup for analog trunk and outbound route is required. Please refer to ***[ANALOG TRUNKS], [VOIP TRUNKS]*** and ***[OUTBOUND ROUTES]*** sections for configuring analog trunk and outbound route.

After making sure analog trunk or VoIP Trunk is setup properly and UCM6200 can reach out to PSTN numbers via the trunk, on Fax Sending page, enter the fax number and upload the file to be faxed. Then click on “Send” to start. The progress of sending fax will be displayed in web UI. Users can also view the sending history is in the same web page.

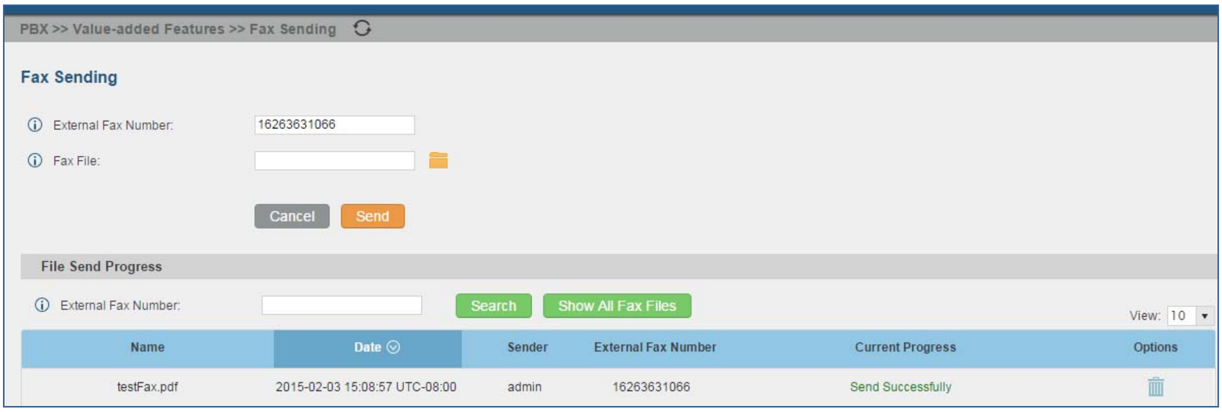


Figure 157: Fax Sending in Web UI

ANNOUNCEMENTS CENTER

The UCM6200 supports Announcements Center feature which allows users to pre-record and store voice message into UCM6200 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.

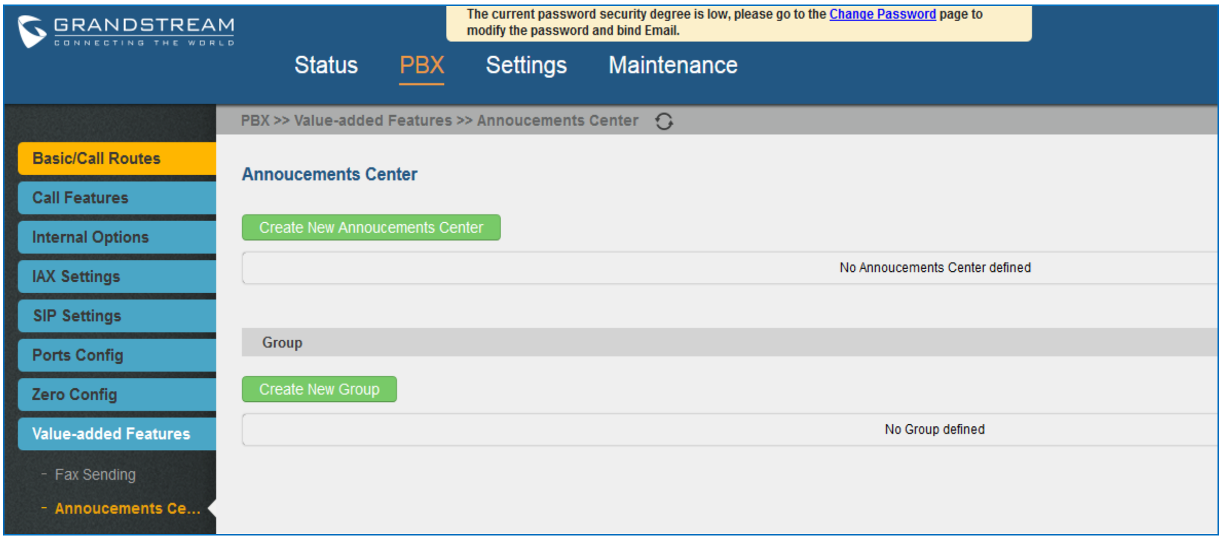


Figure 158: Announcements Center

ANNOUNCEMENTS CENTER SETTINGS

Table 90: Announcements Center Settings

Name	Configure a name for the newly created Announcements 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.
Ring Timeout	Configure the ring timeout for the group members. The default value is 30 seconds.

GROUP SETTINGS

Table 91: Group Settings

Name	Configure a name for the newly created group to identify the group.

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.

Announcements Center feature can be found under **web UI->PBX->Value-added Features-> Announcements Center**. The following example demonstrates the usage of this feature.

1. Click Create New Group to create new group.
2. Give a name to the newly created group.
3. Create a group number which is used with code to send voice message.
4. Select the extensions to be included in the group, who will receive the voice message.

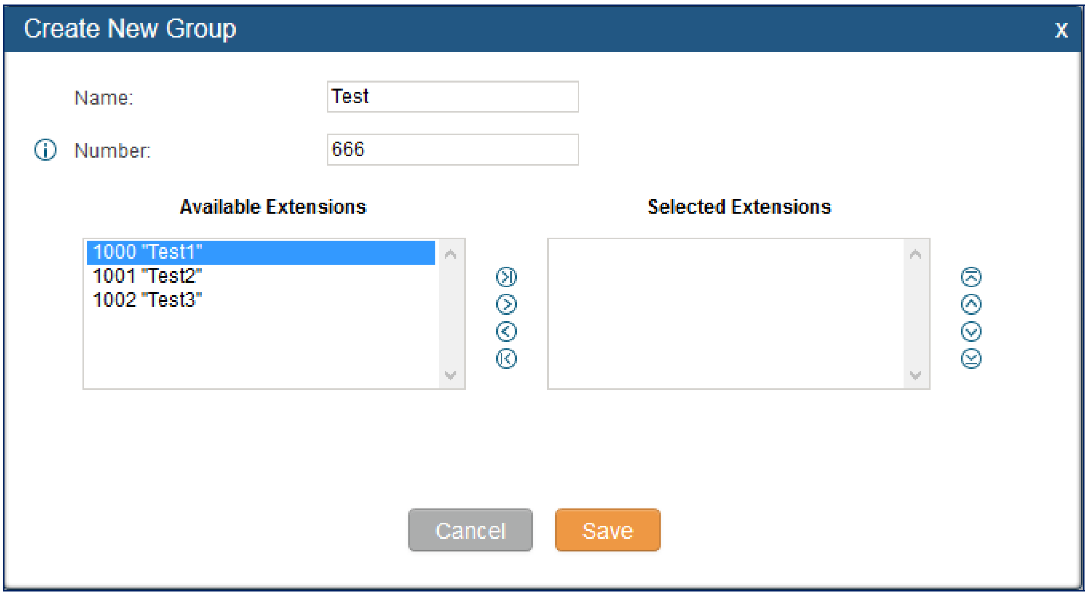


Figure 159: Announcements Center Group Configuration

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

5. Click Create New Annoucements 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.

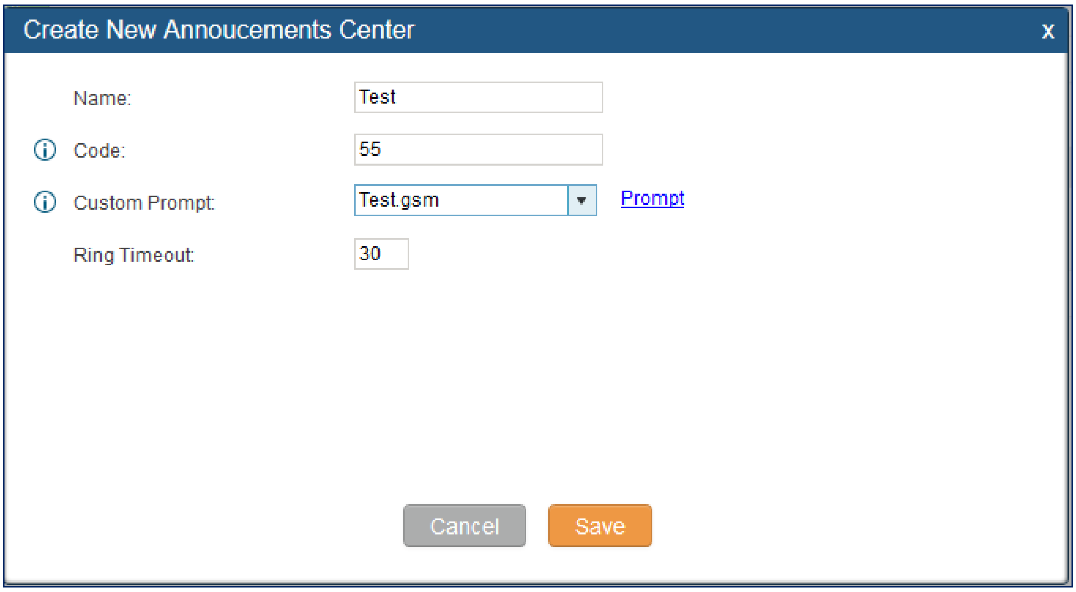


Figure 160: Announcements Center Code Configuration

Code and Group number are used together to direct specified message to the target group. All extensions in the group will receive the message. For example, we can send code 55 to group 666 by dialing 55666 from any extension registered to the UCM6200. All the members in group 666 which are extension 1000, 1001 and 1002 will receive this voice message after they pick up the call.

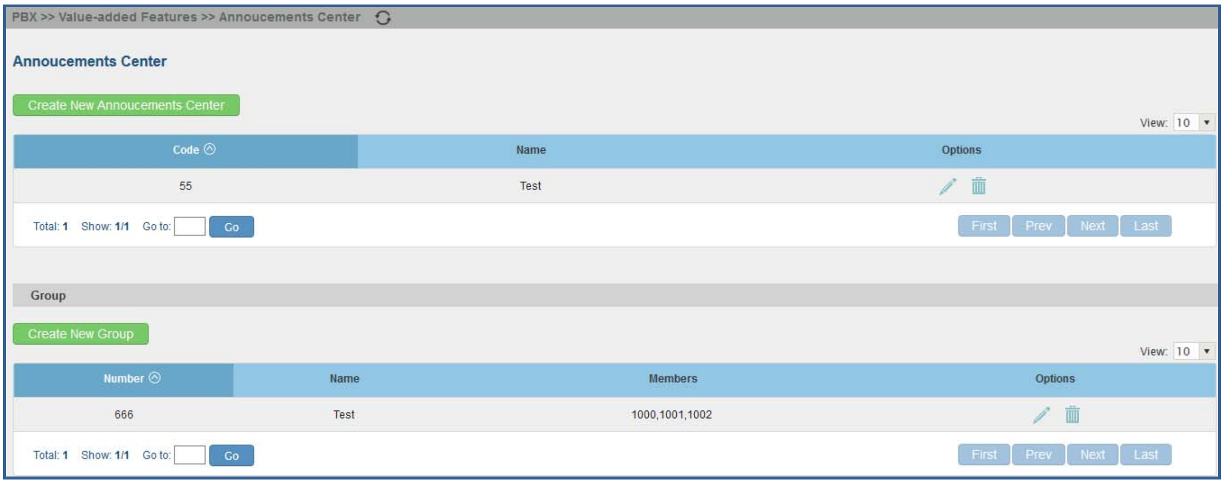


Figure 161: Announcements Center Example

STATUS AND REPORTING

PBX STATUS

The UCM6200 monitors the status for Trunks, Extensions, Queues, Conference Rooms, Interfaces and Parking lot. It presents administrators the real time status in different sections under web GUI->**Status**->**PBX Status**.

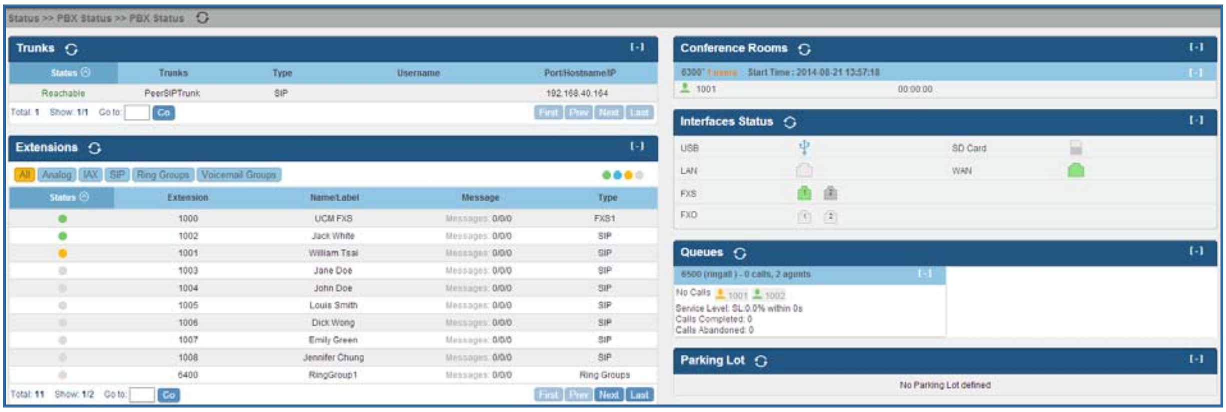


Figure 162: Status->PBX Status

TRUNKS

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

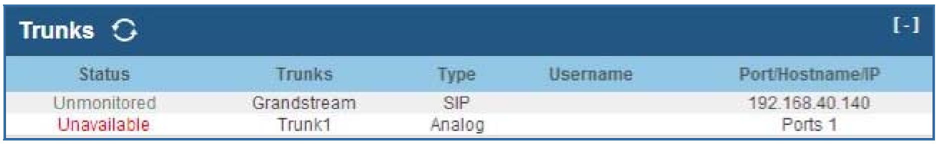


Figure 163: Trunk Status

Table 92: Trunk Status

Status	Display trunk status. • Analog trunk status: **Available** **Busy** **Unavailable** **Unknown Error** • SIP Peer trunk status: **Unreachable**: The hostname cannot be reached.

	Unmonitored: QUALIFY feature is not turned on to be monitored. **Reachable**: The hostname can be reached. • SIP Register trunk status: **Registered** **Unrecognized Trunk**
Trunks	Display trunk name
Type	Display trunk Type: • Analog • SIP • IAX
Username	Display username for this trunk.
Port/Hostname/IP	Display Port for analog trunk, or Hostname/IP for VoIP (SIP/IAX) trunk.

Other operations are also available in trunk status section:

- Click on "Trunks", the web page will redirect to trunk configuration page which can also be accessed via web GUI->**PBX**->**Basic/Call Routes**->**Analog Trunks**.
- Click on to refresh the trunk status.
- Click on [+] to expand the status detail table.
- Click on [-] to hide the status detail table.

EXTENSIONS

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

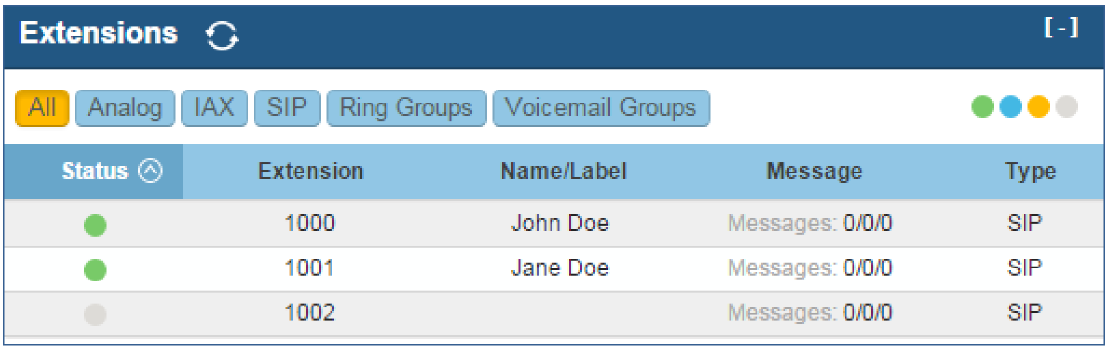


Figure 164: Extension Status

Table 93: Extension Status

Status	Display extension number (including feature code). The color indicator has the following definitions. • Green: Free • Blue: Ringing • Yellow: In Use • Grey: Unavailable
Extension	Display the extension number.
Name/Label	First name and last name of the extension.
Message	Display message status for the extension. Example: 2/4/1 Description: There are 2 urgent messages, 4 messages in total and 1 message that has been already read.
Type	Displays extension type. • SIP User • IAX User • Analog User • Ring Groups • Voicemail Groups

Other operations are also available in extension status section:

- Click on "Extensions", the web page will redirect to extension configuration page which can also be accessed via web GUI->**PBX**->**Basic/Call Routes**->**Extensions**.
- Click on to refresh the extension status.
- Click on one of the tabs All Analog IAX SIP Ring Groups Voicemail Groups to display the corresponding extensions accordingly.
- Click on [+] to expand the status detail table.
- Click on [-] to hide the status detail table.

QUEUES

Users could see all the configured call queue status in this section. The following figure shows the call queue 6500 being in used.

Figure 165: Queue Status

The current call status (caller ID, duration), agent status, service level, calls summary (completed/abandoned) are shown for the call queue. The agent status is defined as below.

Table 94: Agent Status

	The agent is available/idle.
	The agent is ringing.
	The agent is talking/busy.
	The agent has been logged out.

On the UCM6200, **Service Level** is defined as the percentage of high-quality calls over all calls in the call queue, where high-quality call means calls answered within 10 seconds.

Other operations are also available in queue status section:

- Click on "Queues", the web page will redirect to call queue configuration page which can also be accessed via web GUI->**PBX**->**Call Features**->**Call Queue**.
- Click on to refresh the call queue status.
- Click on [+] to expand the call queue detail.
- Click on [-] to hide the call queue detail.

CONFERENCE ROOMS

Users could see all the conference room status in this section. It shows all the configured conference rooms, current users, call duration for each user and conference call.

Figure 166: Conference Room Status

Other operations are also available in conference room status section:

- Click on "Conference Rooms", the web page will redirect to conference room configuration page which can also be accessed via web GUI->**PBX**->**Call Features**->**Conference**.
- Click on  to refresh the conference room status.
- Click on [+] to expand the conference room details.
- Click on [-] to hide the conference room details.

INTERFACES STATUS

This section displays interface/port connection status on the UCM6200. The following example shows the interface status for UCM6204 with USB, WAN port, FXS1, FXS2 and FXO1 connected.

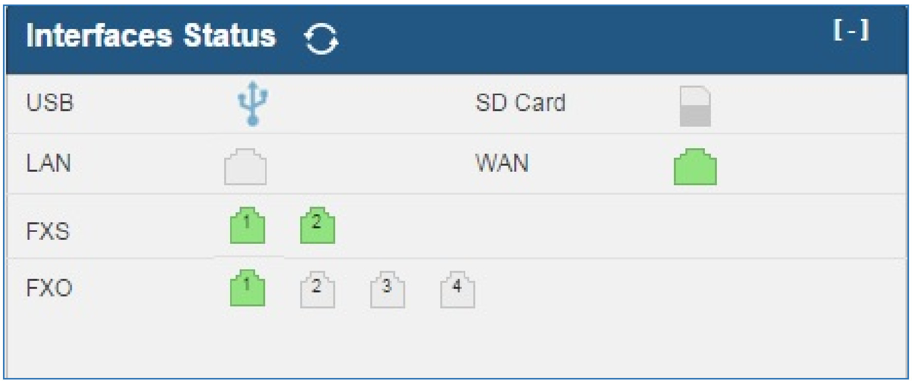


Figure 167: UCM6204 Interfaces Status

Table 95: Interface Status Indicators

	USB connected.
	USB disconnected.
	SD Card connected.
	SD Card disconnected.

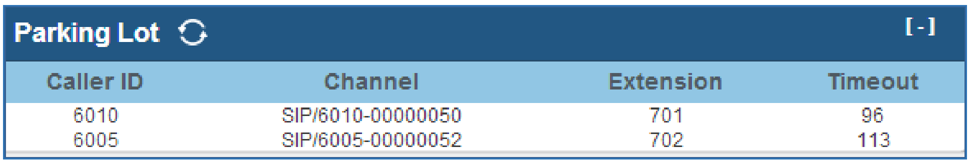
	LAN/WAN connected.
	LAN/WAN not configured.
	LAN/WAN disconnected.
	FXS/FXO connected.
	FXS/FXO waiting.
	FXS/FXO busy.
	FXS/FXO not configured.
	FXS/FXO disconnected.

Other operations are also available in interface status section:

- Click on "Interfaces Status", the web page will redirect to ports configuration page which can also be accessed via web GUI->**PBX**->**Internal Options**->**Ports Config**.
- Click on to refresh the interface status.
- Click on [+] to expand the interface details.
- Click on [-] to hide the interface details.

PARKING LOT

The UCM6200 supports call park using feature code. When there is call being parked, this section will display the parking lot status.



Parking Lot [-]

Caller ID	Channel	Extension	Timeout
6010	SIP/6010-00000050	701	96
6005	SIP/6005-00000052	702	113

Figure 168: Parking Lot Status

Table 96: Parking Lot Status

Caller ID	Display the caller ID who parks the call.
Channel	Display channel for the call park.
Extension	Display the parking lot number where the call is parked/retrieved.
Timeout	Display timeout (in seconds) for the parked call. The status page will dynamically update this timer from 120 seconds (default) to 0. When the timer

	reaches 0, the caller who parks the call will be called back.

Other operations are also available in parking lot status section:

- Click on "Parking Lot", the web page will redirect to feature codes page which can also be accessed via web GUI->**PBX**->**Internal Options**->**Feature Codes**.
- Click on to refresh the parking lot status.
- Click on [+] to expand the parking lot details.
- Click on [-] to hide the parking details.

SYSTEM STATUS

The UCM6200 system status can be accessed via Web GUI->**Status**->**System Status**, which displays the following system information.

- **General**
- **Network**
- **Storage Usage**
- **Resource Usage**

GENERAL

Under Web GUI->**Status**->**System Status**->**General**, users could check the hardware and software information for the UCM6200. Please see details in the following table.

Table 97: System Status->General

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

Recovery	Recovery version.

NETWORK

Under Web GUI->**Status**->**System Status**->**Network**, users could check the network information for the UCM6200. Please see details in the following table.

Table 98: System Status->Network

Status -> System Status -> Network	
MAC Address	Global unique ID of device, in HEX format. The MAC address can be found on the label coming with original box and on the label located on the bottom of the device.
IP Address	IP address.
Gateway	Default gateway address.
Subnet Mask	Subnet mask address.
DNS Server	DNS Server address.

STORAGE USAGE

Users could access the storage usage information from web UI->**Status**->**System Status**->**Storage Usage**. It shows the available and used space for the following partitions.

- Configuration partition
  This partition contains PBX system configuration files and service configuration files.
- Data partition
  Voicemail, recording files, IVR file, Music on Hold files and etc.
- USB disk
  USB disk will display if connected.
- SD Card
  SD Card will display if connected.

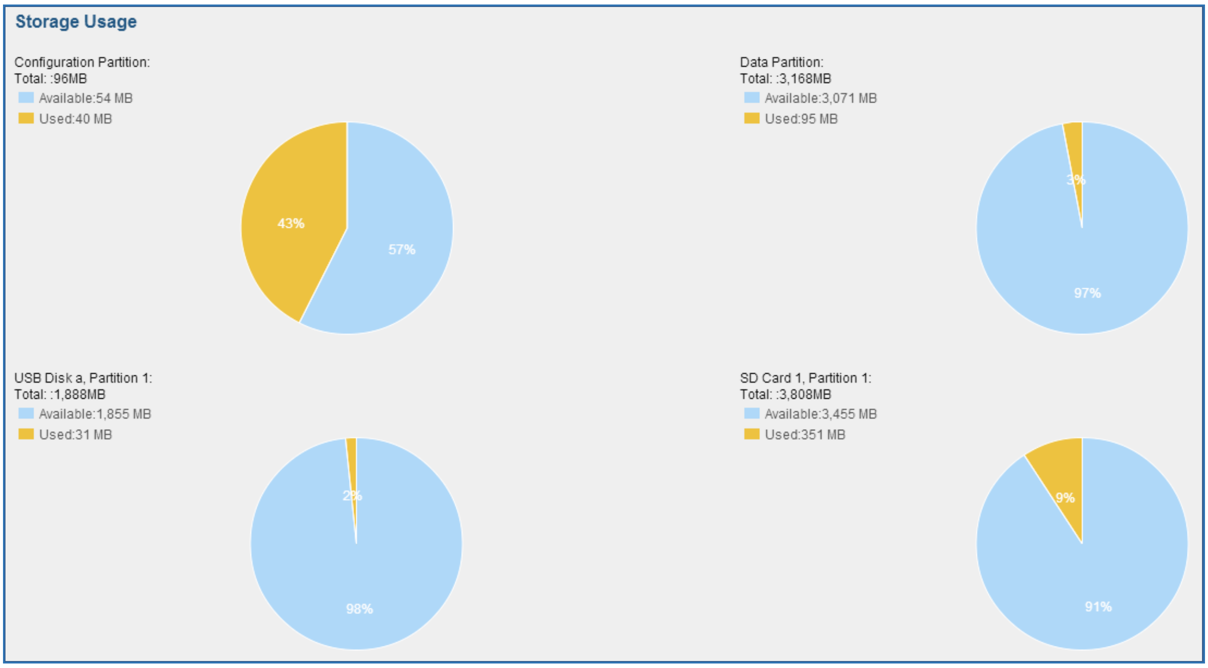


Figure 169: System Status->Storage Usage

RESOURCE USAGE

When configuring and managing the UCM6200, users could access resource usage information to estimate the current usage and allocate the resources accordingly. Under web UI->**Status**->**System Status**->**Resource Usage**, the current CPU usage and Memory usage are shown in the pie chart.

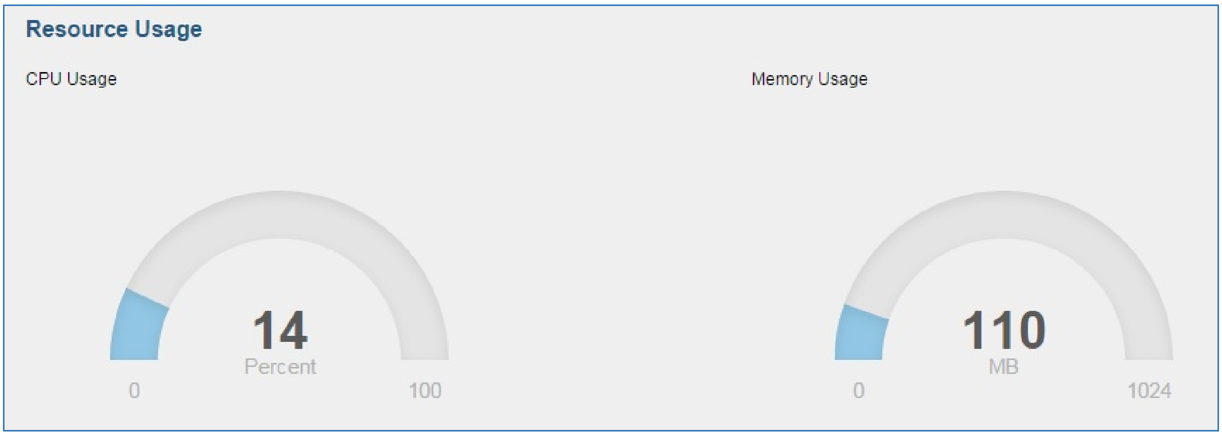


Figure 170: System Status->Resource Usage

SYSTEM EVENTS

The UCM6200 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->**Status**->**System Events**->**Alert Events List**. The following event are currently supported on the UCM6200 which will have alert and/or Email generated if occurred:

Disk Usage
Modify Admin Password
Memory Usage
System Reboot
System Update
System Crash
Register SIP Failed
Register SIP Trunk Failed
Restore Config
User Login Success
User Login Failed
SIP Internal Call Failure
SIP Outgoing Call through Trunk Failure
Fail2ban Blocking
SIP Lost Registration
SIP Peer Trunk Status

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

1. Disk Usage.

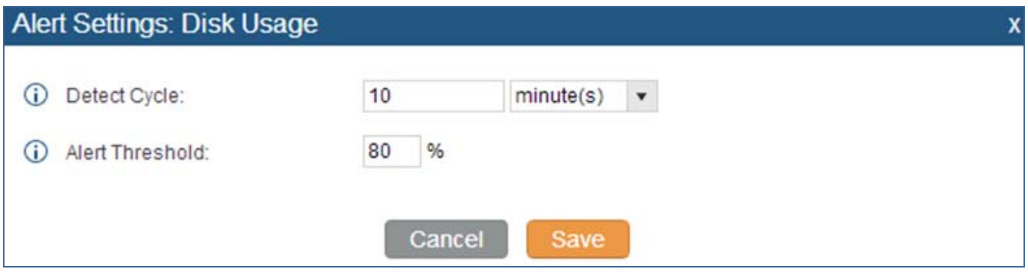


Figure 171: System Events->Alert Events Lists: Disk Usage

- **Detect Cycle**: The UCM6200 will perform the internal disk usage detection based on this cycle. Users can enter the number and then select second(s)/minute(s)/hour(s)/day(s) to configure the cycle.
- **Alert Threshold**: If the detected value exceeds the threshold (in percentage), the UCM6200 system will send the alert.

2. Memory Usage

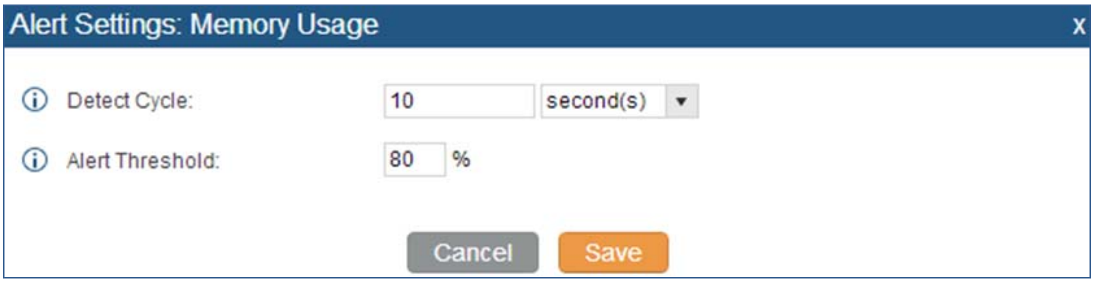


Figure 172: System Events->Alert Events Lists: Memory Usage

- **Detect Cycle**: The UCM6200 will perform the memory usage detection based on this cycle. Users can enter the number and then select second(s)/minute(s)/hour(s)/day(s) to configure the cycle.
- **Alert Threshold**: If the detected value exceeds the threshold (in percentage), the UCM6200 system will send the alert.

3. System Reboot

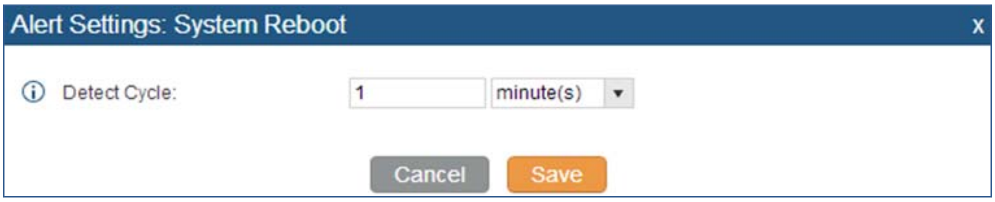


Figure 173: System Events->Alert Events Lists: System Reboot

- **Detect Cycle**: The UCM6200 will check the system reboot based on this cycle. Users can enter the number and then select second(s)/minute(s)/hour(s)/day(s) to configure the cycle.

4. System Crash

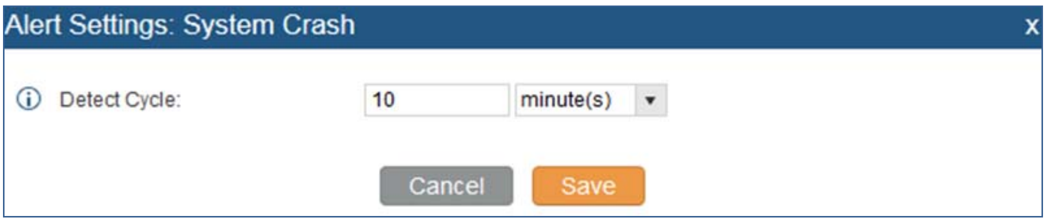


Figure 174: System Events->Alert Events Lists: System Crash

- **Detect Cycle**: The UCM will detect the event at each cycle based on the specified time. Users can enter the number and then select second(s)/minute(s)/hour(s)/day(s) to configure the cycle.

Click on the switch 

to turn on/off the alert and Email notification for the event. Users could also select the checkbox for each event and then click on button "Alert On", "Alert Off", "Email Notification On", "Email Notification Off" to control the alert and Email notification configuration.

ALERT LOG

Under Web GUI->**Status**->**System Events**->**Alert Log**, system messages from triggered system events are listed as alert logs. The following screenshot shows system crash alert logs.

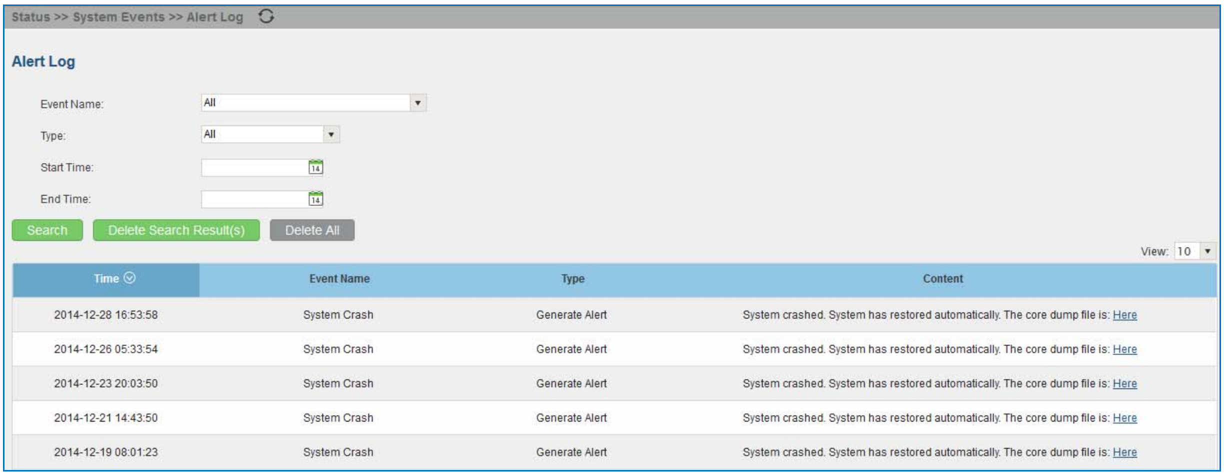


Figure 175: System Events->Alert Log

User could also filter alert logs by selecting a certain event category, type of alert log, and/or specifying a certain time period. The matching results will be displayed after clicking on 

. Alert logs are classified into two types by the system:

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

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

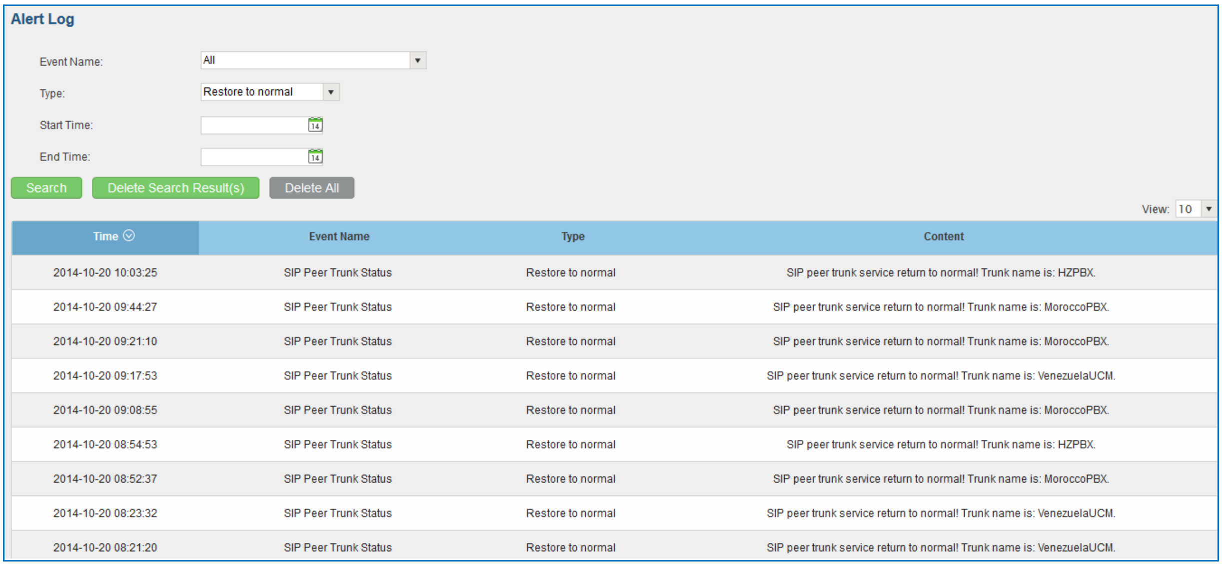


Figure 176: Filter for Alert Log

ALERT CONTACT

Users could add administrator's Email address under Web GUI->**Status**->**System Events**->**Alert Contact** to send the alert notification to. Up to 10 Email addresses can be added.

CDR

A Call Detail Record (CDR) is a data record produced by telephone exchange activities or other telecommunications equipment documenting the details of a phone call that passed through the PBX. The CDR is composed of the following data fields on the UCM6200.

- **Start Time**. Format: 2016-02-27 16:47:03.
- **Call From**. Format: "John Doe"<6012>.
- **Call To**. Format: 6005.
- **Answered By**. Format: 6005.
- **Call Time**. Format: 0:00:10.
- **Talk Time**. Format: 0:00:10
- **Status**. Format: NO ANSWER, BUSY, ANSWERED, or FAILED.
- **Options**. Voice record playing/downloading/deleting.

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. Then click on "View Report" button to display the generated report.

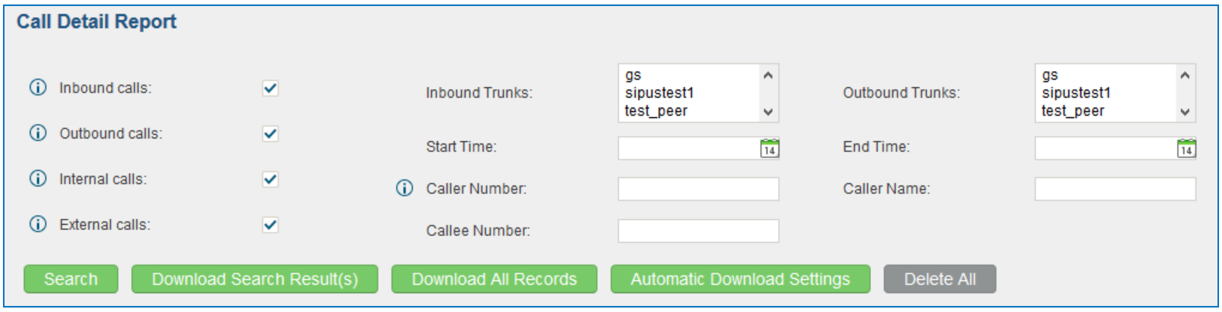


Figure 177: CDR Filter

Table 99: CDR Filter Criteria

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.
Inbound Trunks	Select certain inbound trunk(s) and the CDR of calls going inbound through the trunk(s) will be filtered out.
Outbound Trunks	Select certain outbound trunk(s) and the CDR of calls going outbound through the trunk(s) will be filtered out.
Start Time	Specify the start time to filter the CDR report. Click on the calendar icon on the right and the calendar will show for users to select the exact date and time.
End Time	Specify the end time to filter the CDR report. Click on the calendar icon on the right and the calendar will show for users to select the exact date and time.
Caller Number	Enter the caller number to filter the CDR report. CDR with the matching caller number will be filtered out. 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. For example: 3XXX: It will filter out CDR that having caller number with leading digit 3 and of 4 digits length. 3.: It will filter out CDR that having caller number with leading digit 3 and of any length.
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.

The call report will display as the following figure shows.

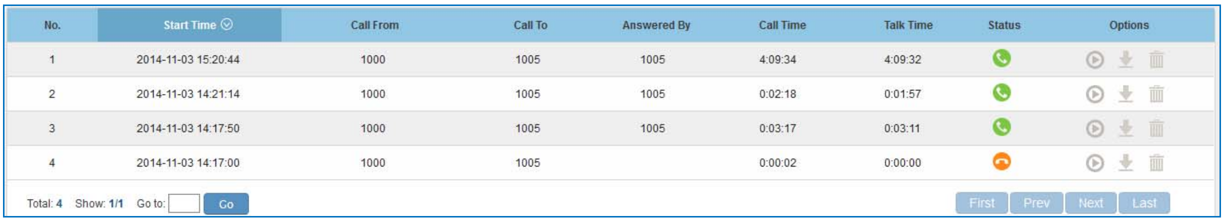


Figure 178: Call Report

Users could perform the following operations on the call report.

- **Sort**
  Click on the header of the column to sort by this category. For example, clicking on "Start Time" will sort the report according to 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**
  On the bottom of the page, click on "Delete All" button to remove all the call report information.

- **Play/Download/Delete Recording File (per entry)**
  If the entry has audio recording file for the call, the three icons on the most right column will be activated for users to select. In the following picture, the second entry has audio recording file for the call.

  Click on to play the recording file; click on to download the recording file in .wav format; click on to delete the recording file (the call record entry will not be deleted).

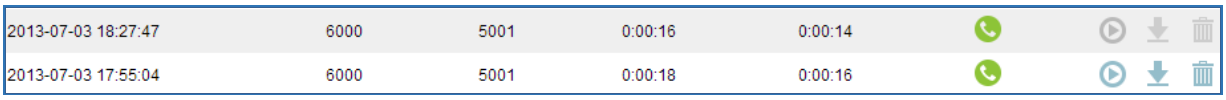


Figure 179: Call Report Entry with Audio Recording File

- **Automatic Download CDR Records**

  User could configure the UCM6200 to automatically download the CDR records and send the records to an Email address. Click on “Automatic Download Settings”, and configure the parameters in the dialog below.

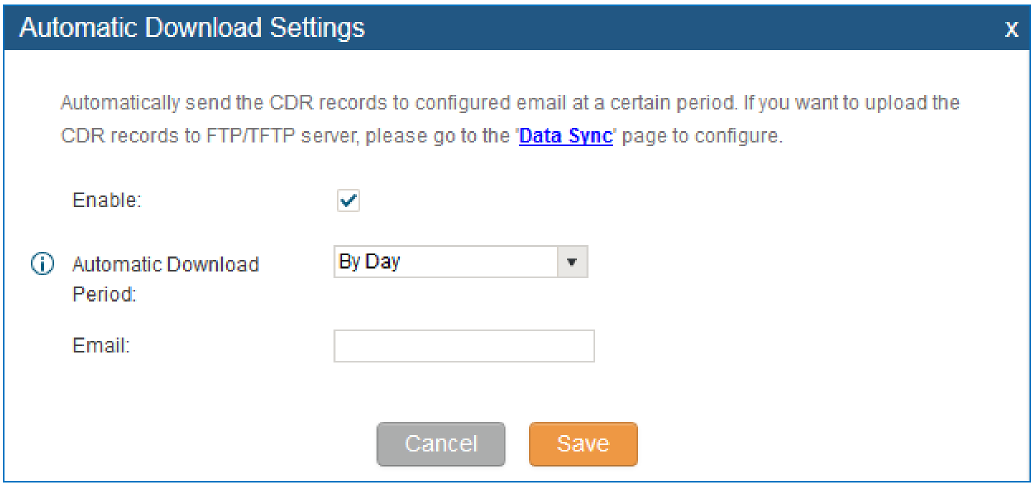


Figure 180: Automatic Download Settings

To receive CDR record automatically from Email, check “Enable” and select a time period “By Day” “By Week” or “By Month” for the automatic download period. Make sure you have entered an Email address to receive the CDR records.

CDR IMPROVEMENT

Starting from UCM6200 firmware 1.0.10.x, transferred call will no longer be displayed as a separate call entry in CDR. It will display within call record in the same entry. CDR new features can be found under **web UI-> Status->CDR->CDR**. The user can click on the option icon for a specific call log entry to view details about this entry, such as premier caller and transferred call information.

Click on the title of the column to sort by column. Click on the row to display full record.

View: 10

No.	Start Time	Call Type	Call From	Call To	Call Time	Talk Time	Status	Recording File Options	Options
1	2015-08-28 17:52:16	DIAL	1000	1001	0:01:07	0:01:04		No Recording Files	
2	2015-08-28 17:51:56	DIAL	1000	1001	0:00:15	0:00:12		No Recording Files	
3	2015-08-28 17:50:33	DIAL	1000	1001	0:00:14	0:00:11		No Recording Files	
4	2015-08-28 17:50:17	DIAL	1000	1002	0:00:05	0:00:02		No Recording Files	
5	2015-08-28 17:49:59	DIAL	1000	1001	0:00:29	0:00:25		No Recording Files	

Figure 181: CDR Report

Call Detail

Start Time	Premier Caller	Call Type	Call From	Call To	Call Time	Talk Time	Status	Recording File Options
2015-08-28 17:52:16	1000	DIAL	1000	1001	0:01:04	0:01:01		No Recording Files
2015-08-28 17:53:21	1000	TRANSFER	1001	1002	0:00:03	0:00:03		No Recording Files

Figure 182: Detailed CDR Information

DOWNLOADED CDR FILE

The downloaded CDR (.csv file) has different format from the web UI CDR. Here are some descriptions.

- **Call From, Call To**

"Call From": the caller ID.
"Call To": the callee ID.

If "Call From" shows empty, "Call To" shows "s" (see highlight part in the picture below) and the "Source Channel" contains "DAHDI", this means the call is from FXO/PSTN line. For FXO/PSTN line, we only know there is an incoming request when there is incoming call but we don't know the number being called. So we are using "s" to match it where "s" means "start".

call from	call to	context	start time	answer time	end time	call time	talk time	source channel	dest channel	status
610	19097622990	from-internal	1/29/2014 14:28	1/29/2014 14:28	1/29/2014 14:31	153	150	SIP/610-00000074	DAHDI/1-1	ANSWERED
	s	default	1/29/2014 14:33		1/29/2014 14:33	8	0	DAHDI/pseudo-149089967		NO ANSWER
	s	default	1/29/2014 14:33		1/29/2014 14:33	9	0	DAHDI/pseudo-1067045536		NO ANSWER
601	688	from-internal	1/29/2014 14:33	1/29/2014 14:33	1/29/2014 14:33	9	9	SIP/601-00000077		ANSWERED
	s	default	1/29/2014 14:34		1/29/2014 14:34	22	0	DAHDI/pseudo-1124093033		NO ANSWER
	s	default	1/29/2014 14:34		1/29/2014 14:34	22	0	DAHDI/pseudo-1719498666		NO ANSWER

Figure 183: Downloaded CDR File Sample - Call To Shows "s"

- **Context**

There are different context values that might show up in the downloaded CDR file. The actual value can vary case by case. Here are some sample values and their descriptions.

from-internal: internal extension makes outbound calls.
ext-did-XXXXX: inbound calls. It starts with "ext-did", and "XXXXX" content varies case by case, which also relate to the order when the trunk is created.
ext-local: internal calls between local extensions.

- **Source Channel, Dest Channel**

Sample 1:

call from	call to	context	start time	answer time	end time	call time	talk time	source channel	dest channel	status
3122731439	s	ext-did-1	1/30/2014 14:27	1/30/2014 14:27	1/30/2014 14:27	37	35	DAHDI/1-1		ANSWERED

Figure 184: Downloaded CDR File Sample - Source Channel and Dest Channel 1

DAHDI means it is an analog call, FXO or FXS.
For UCM6202, DAHDI/(1-2) are FXO ports, and DAHDI(3-4) are FXS ports.
For UCM6204, DAHDI/(1-4) are FXO ports, and DAHDI(5-6) are FXS ports.
For UCM6208, DAHDI/(1-8) are FXO ports, and DAHDI(9-10) are FXS ports.

Sample 2:

call from	call to	context	start time	answer time	end time	call time	talk time	source channel	dest channel	status
609	619	from-internal	1/30/2014 14:31	1/30/2014 14:32	1/30/2014 14:32	9	3	SIP/609-00000150	SIP/619-00000151	ANSWERED

Figure 185: Downloaded CDR File Sample - Source Channel and Dest Channel 2

"SIP" means it's a SIP call. There are three possible format:

(a) **SIP/NUM-XXXXXX,** where NUM is the local SIP extension number. The last XXXXX is a random string and can be ignored.

(c) **SIP/trunk_X/NUM,** where trunk_X is the internal trunk name, and NUM is the number to dial out through the trunk.

(c) **SIP/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.

Sample 3:

call from	call to	context	start time	answer time	end time	call time	talk time	source channel	dest channel	status
	s	default	1/30/2014 14:30		1/30/2014 14:37	386	0	DAHDI/pseudo-1665832080		NO ANSWER
	s	default	1/30/2014 14:30		1/30/2014 14:37	390	0	DAHDI/pseudo-1946772436		NO ANSWER

Figure 186: Downloaded CDR File Sample - Source Channel and Dest Channel 3

This is a very special channel name. If it shows up, most likely it means a conference call.

There are some other possible values, but these values are almost the application name which are used by the dialplan.

IAX2/NUM-XXXXXXX: it means this is an IAX call.
Local/@from-internal-XXXXX: it is used internally to do some special feature procedure. We can simply ignore it.
Hangup: the call is hung up from the dialplan. This indicates there are some errors or it has run into abnormal cases.
Playback: play some prompts to you, such as 183 response or run into an IVR.
ReadExten: collect numbers from user. It may occur when you input PIN codes or run into DISA

STATISTICS

CDR Statistics is an additional feature on the UCM6200 which provides users a visual overview of the call report across the time frame. Users can filter with different criteria to generate the statistics chart.

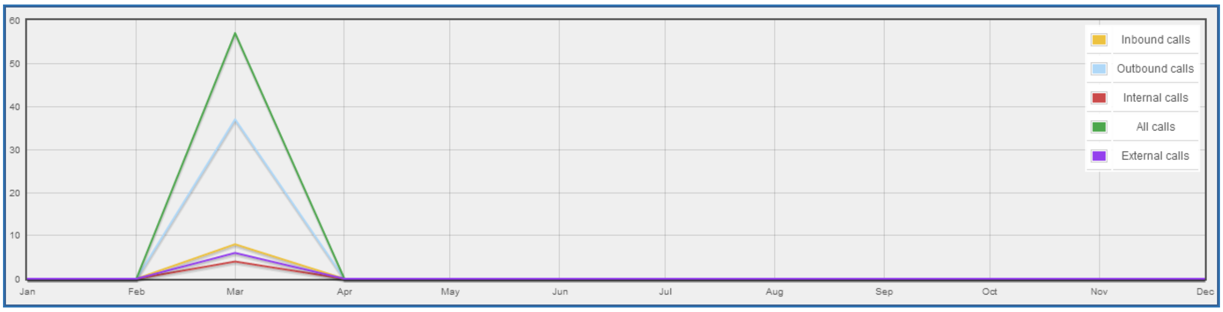


Figure 187: CDR Statistics

Table 100: CDR Statistics Filter Criteria

Trunk Type	Select one of the following trunk type. • All • SIP Calls • PSTN Calls
Call Type	Select one or more in the following checkboxes. • Inbound calls • Outbound calls • Internal calls • External calls • All calls
Time Range	• By month (of the selected year). • By week (of the selected year). • By day (of the specified month for the year). • By hour (of the specified date). • By range. For example, 2016-01 To 2016-03.

RECORDING FILES

This page lists all the recording files recorded by "Auto Record" per extension/ring group/call queue/trunk, or via feature code "Audio Mix Record". If external storage device is plugged in, for example, SD card or USB drive, the files are stored on the external storage. Otherwise, internal storage will be used on the UCM6200.

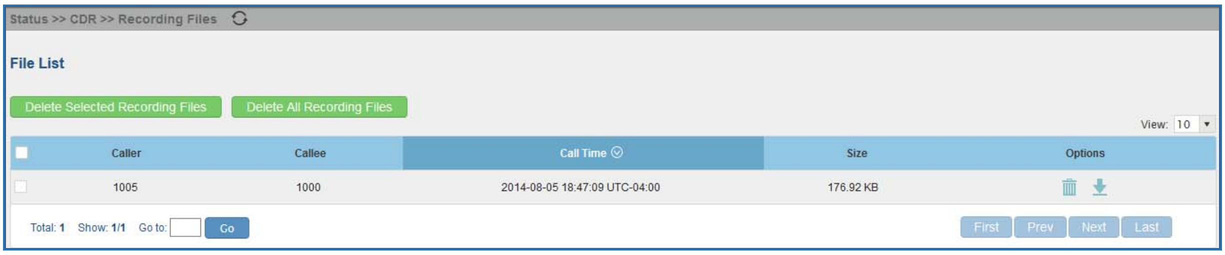


Figure 188: CDR->Recording Files

- Click on “Delete Selected Recording Files” to delete the recording files.
- Click on “Delete All Recording Files” to delete all recording files.
- Click on  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.

API CONFIGURATION

The UCM6200 supports third party billing interface API for external billing software to access CDR and call recordings on the PBX. The API uses HTTPS to request the CDR data and call recording data matching given parameters as configured on the third party application.

Before accessing the API, the administrators need enable API and configure the access/authentication information on the UCM6200 first. The API configuration parameters are listed in the table below.

Table 101: API Configuration Files

Enable	Enable/Disable API. The default setting is disabled.
TLS Bind Address	Configure the IP address for TLS server to bind to. "0.0.0.0" means binding to all interfaces. The port number is optional and the default port number is 8443. The IP address must match the common name (host name) in the certificate so that the TLS socket won't bind to multiple IP addresses. The default setting is 0.0.0.0:8443.
TLS Private Key	Upload TLS private key. The size of the key file must be under 2MB. This file will be renamed as 'private.pem' automatically.
TLS Cert	Upload TLS cert. The size of the certificate must be under 2MB. This is the certificate file (*.pem format only) for TLS connection. This file will be renamed as "certificate.pem" automatically. It contains private key for the client and signed certificate for the server.

Username	Configure the Username for API Authentication.
Password	Configure the Password for API Authentication.
Permitted	Specify a list of IP addresses permitted by API. This creates an AIP-specific access control list. Multiple entries are allowed. For example, "192.168.40.3/255.255.255.255" denies access from all IP addresses except 192.168.40.3. The default setting is blank, meaning all IPs will be denied. Users must set permitted IP address before connecting to the API.

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:
http://www.grandstream.com/sites/default/files/Resources/ucm6200_cdr_rec_api_guide.pdf

UPGRADING AND MAINTENANCE

UPGRADING

The UCM6200 can be upgraded to a new firmware version remotely or locally. This section describes how to upgrade your UCM6200 via network or local upload.

UPGRADING VIA NETWORK

The UCM6200 can be upgraded via TFTP/HTTP/HTTPS by configuring the URL/IP Address for the TFTP/HTTP/HTTPS server and selecting a download method. Configure a valid URL for TFTP, HTTP or HTTPS; the server name can be FQDN or IP address.

Examples of valid URLs:
firmware.grandstream.com/BETA

The upgrading configuration can be accessed via **Web GUI**->**Maintenance**->**Upgrade**.

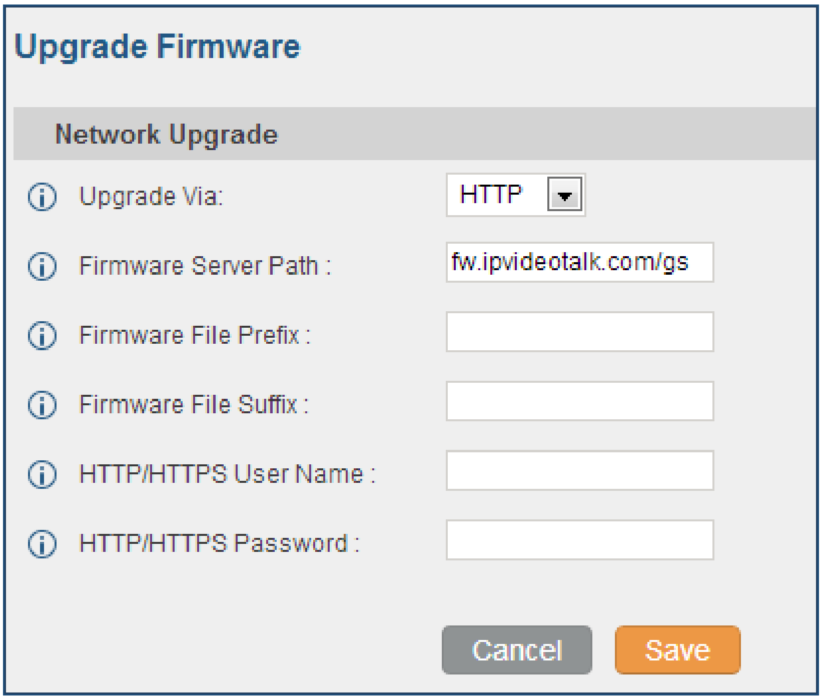


Figure 189: Network Upgrade

Table 102: Network Upgrade Configuration

Upgrade Via	Allow users to choose the firmware upgrade method: TFTP, HTTP or HTTPS.
Firmware Server Path	Define the server path for the firmware server.
Firmware File Prefix	If configured, only the firmware with the matching encrypted prefix will be downloaded and flashed into the UCM6200.
Firmware File Suffix	If configured, only the firmware with the matching encrypted postfix will be downloaded and flashed into the UCM6200.
HTTP/HTTPS User Name	The user name for the HTTP/HTTPS server.
HTTP/HTTPS Password	The password for the HTTP/HTTPS server.

Please follow the steps below to upgrade the firmware remotely.

- Enter the firmware server path under **web UI**->**Maintenance**->**Upgrade**.
- Click on "Save". Then reboot the device to start the upgrading process.
- Please be patient during the upgrading process. Once done, a reboot message will be displayed in the LCD.
- Manually reboot the UCM6200 when it's appropriate to avoid immediate service interruption. After it boots up, log in the web GUI to check the firmware version.

UPGRADING VIA LOCAL UPLOAD

If there is no HTTP/TFTP server, users could also upload the firmware to the UCM6200 directly via Web GUI. Please follow the steps below to upload firmware locally.

- Download the latest UCM6200 firmware file from the following link and save it in your PC. http://www.grandstream.com/support/firmware

- Log in the Web GUI as administrator in the PC.
- Go to Web GUI->**Maintenance**->**Upgrade**, upload the firmware file by clicking on  and select the firmware file from your PC. The default firmware file name is ucm6200fw.bin

Figure 190: Local Upgrade

- Click on  to start upgrading.

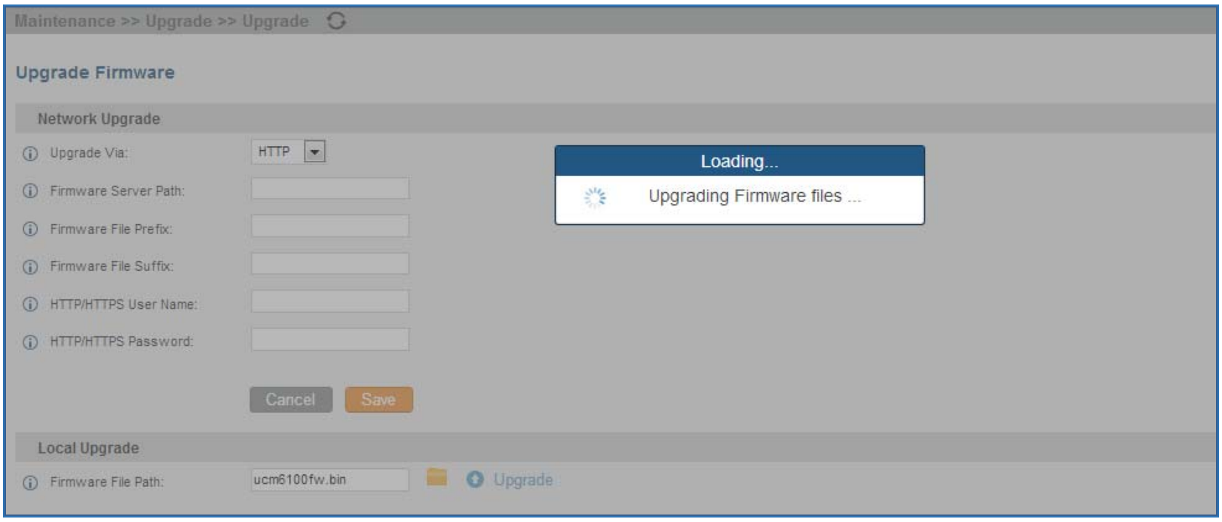


Figure 191: Upgrading Firmware Files

- Wait until the upgrading process is successful and a window will be popped up in the Web GUI.

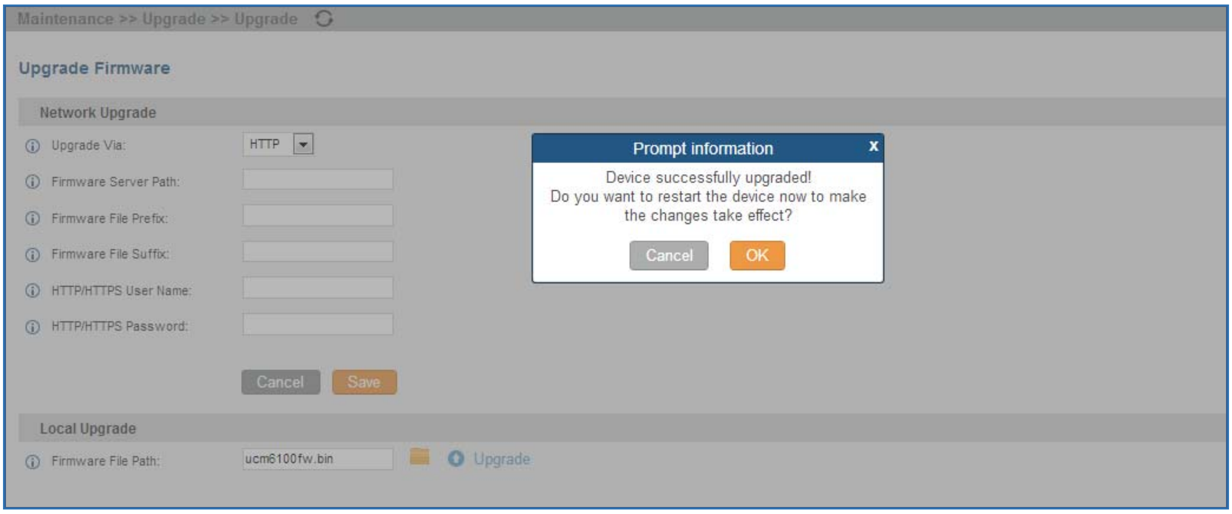


Figure 192: Reboot UCM6200

- Click on "OK" to reboot the UCM6200 and check the firmware version after it boots up.

Note:

Please do not interrupt or power cycle the UCM6200 during upgrading process.

NO LOCAL FIRMWARE SERVERS

Service providers should maintain their own firmware upgrade servers. For users who do not have TFTP/HTTP/HTTPS server, some free windows version TFTP servers are available for download from
http://www.solarwinds.com/products/freetools/free_tftp_server.aspx
http://tftpd32.jounin.net

Please check our website at http://www.grandstream.com/support/firmware for latest firmware.

Instructions for local firmware upgrade via TFTP:

1. Unzip the firmware files and put all of them in the root directory of the TFTP server;
2. Connect the PC running the TFTP server and the UCM6200 to the same LAN segment;
3. Launch the TFTP server and go to the File menu->Configure->Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade;
4. Start the TFTP server and configure the TFTP server in the UCM6200 web configuration interface;
5. Configure the Firmware Server Path to the IP address of the PC;
6. Update the changes and reboot the UCM6200.

End users can also choose to download a free HTTP server from http://httpd.apache.org/ or use Microsoft IIS web server.

BACKUP

The UCM6200 configuration can be backed up locally or via network. The backup file will be used to restore the configuration on UCM6200 when necessary.

BACKUP/RESTORE

Users could backup the UCM6200 configurations for restore purpose under Web GUI->**Maintenance**->**Backup**->**Local Backup**.

Click on 

to create a new backup file. Then the following dialog will show.

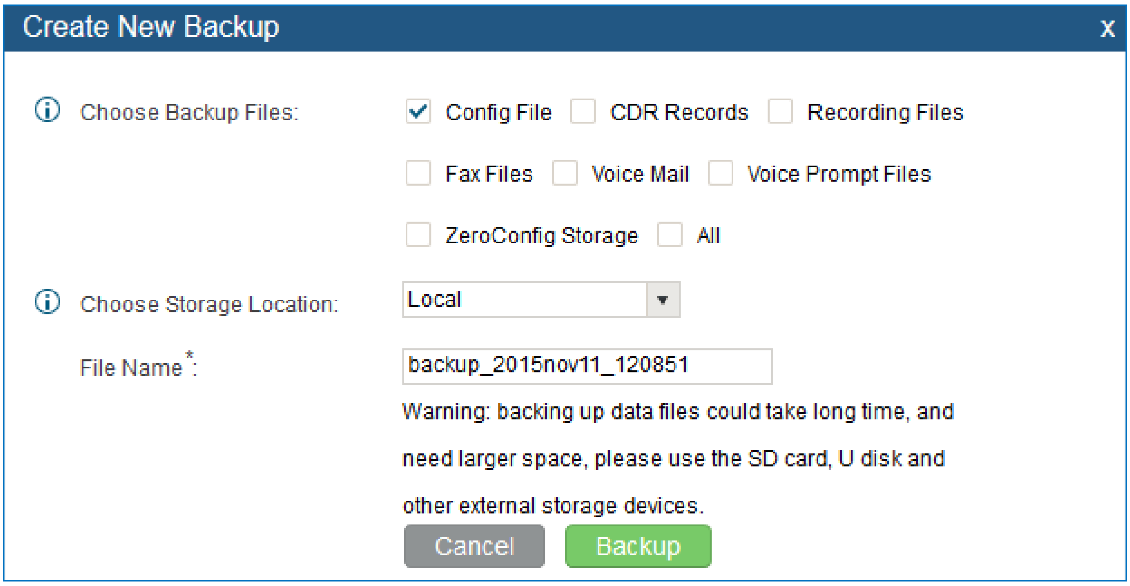


Figure 193: Create New Backup

1. Choose the type(s) of files to be included in the backup.
2. Choose where to store the backup file: USB Disk, SD Card or Local.
3. Name the backup file.
4. Click on "Backup" to start backup.

Once the backup is done, the list of the backups will be displayed with date and time in the web page. Users can download, restore, or delete it from the UCM6200 internal storage or the external device.

Click on  to upload backup file from the local device to UCM6200. The uploaded backup file will also be displayed in the web page and can be used to restore the UCM6200.

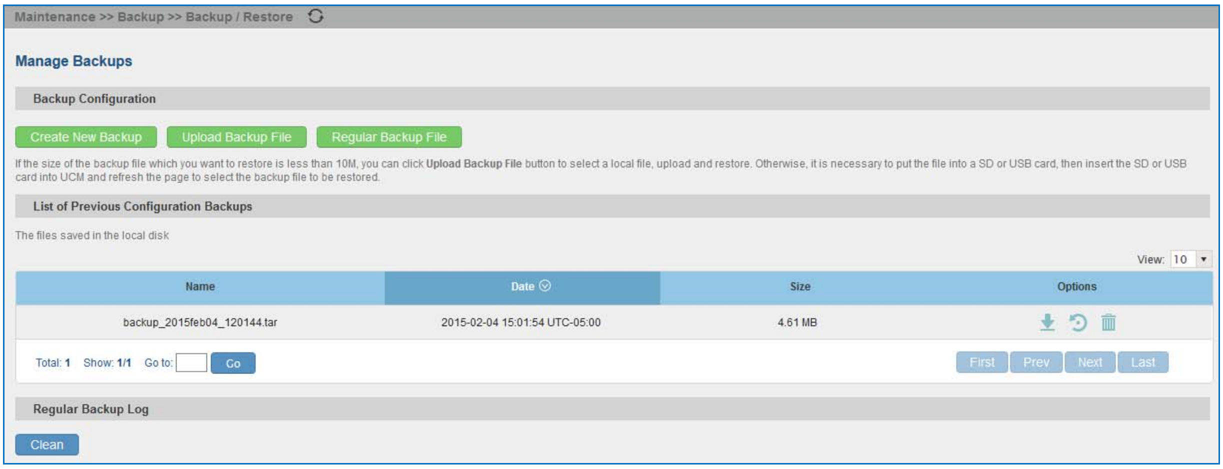


Figure 194: Backup / Restore

Regular Backup File option allows UCM to perform automatically backup on the user specified time. Regular backup file can only be stored in USB / SD card / SFTP server. User is allowed to set backup time from 0-23 and how frequent the backup will be performed.

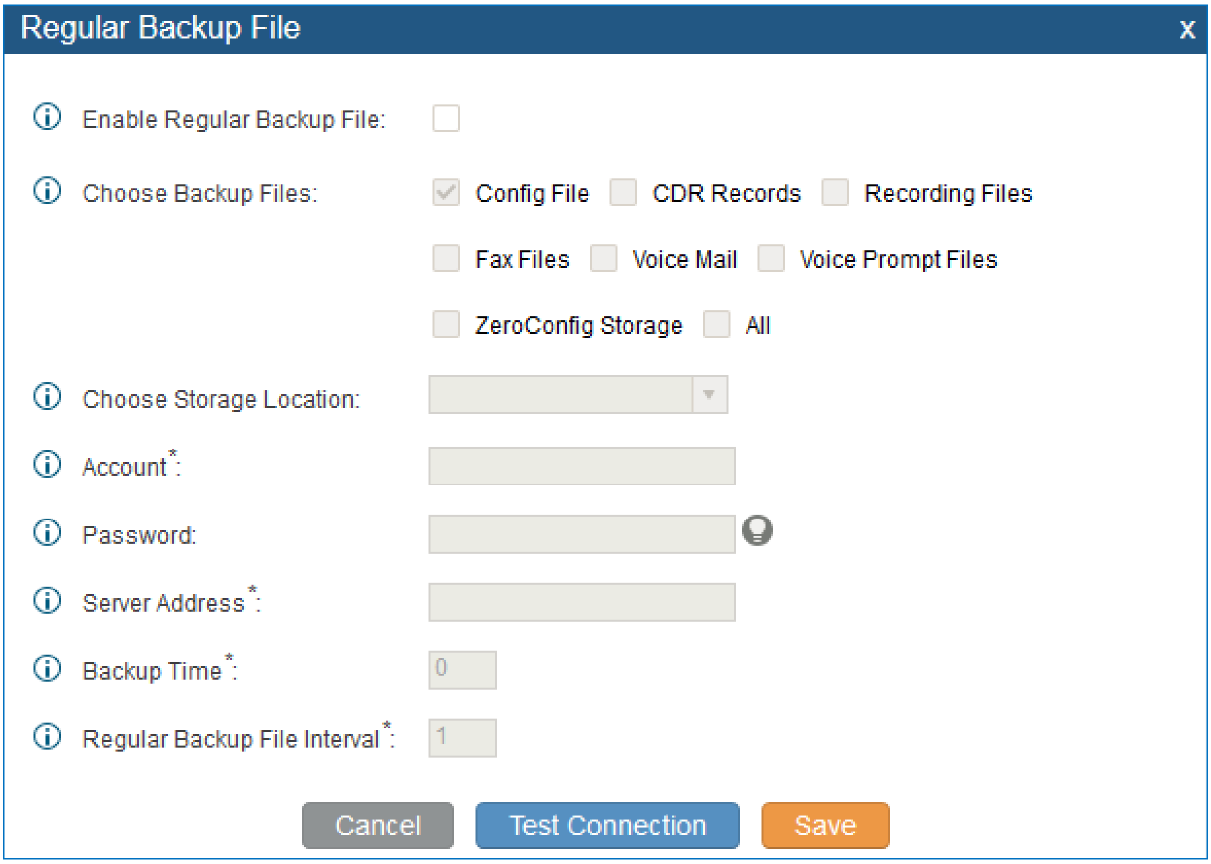


Figure 195: Local Backup

DATA SYNC

Besides local backup, users could backup the voice records/voice mails/CDR/FAX in a daily basis to a remote server via SFTP protocol automatically under Web GUI->**Maintenance**->**Backup**->**Data Sync**.

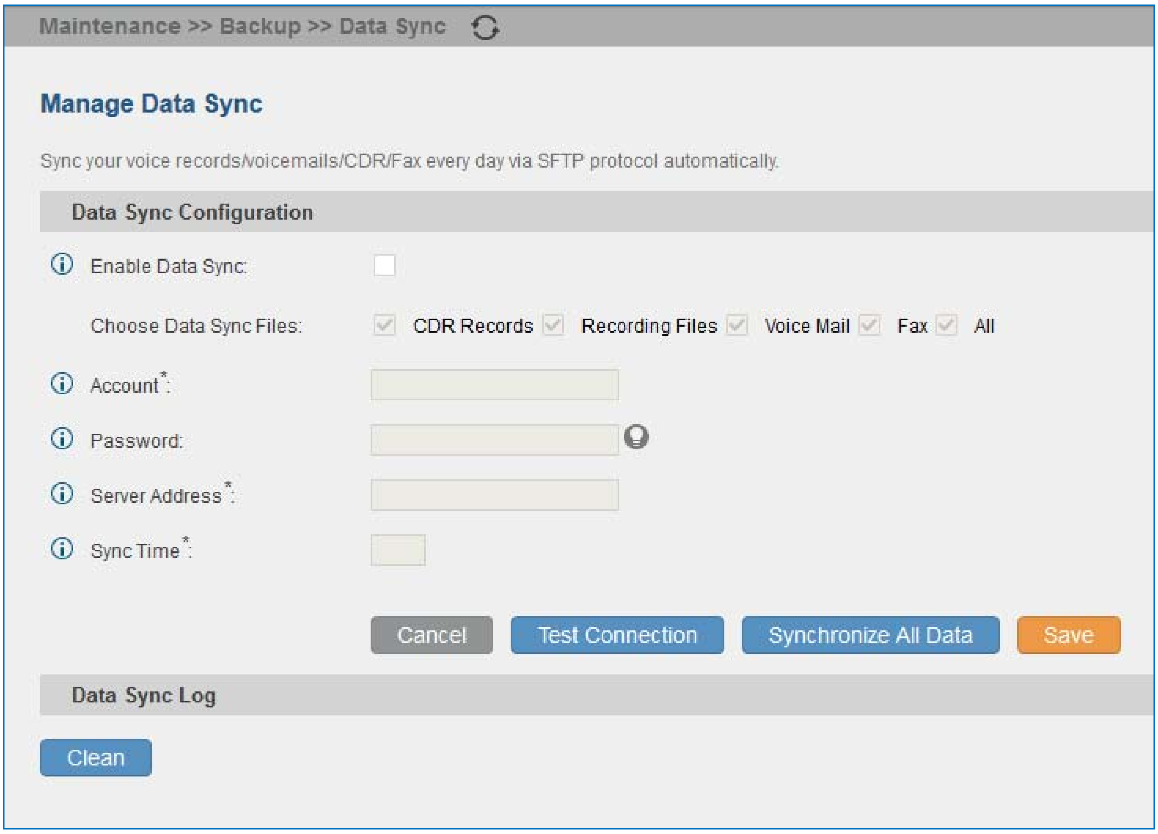


Figure 196: Data Sync

Table 103: Data Sync Configuration

Enable Data Sync	Enable the auto data sync function. The default setting is "No".
Account	Enter the Account name on the SFTP backup server.
Password	Enter the Password associate with the Account on the SFTP backup server.
Server Address	Enter the SFTP server address.
Sync Time	Enter 0-23 to specify the backup hour of the day.

Before saving the configuration, users could click on "Test Connection". The UCM6200 will then try connecting the server to make sure the server is up and accessible for the UCM6200. Save the changes and all the backup logs will be listed on the web page. After data sync is configured, users could also manually synchronize all data by clicking on Synchronize All Data instead of waiting for the backup time interval to come.

RESTORE CONFIGURATION FROM BACKUP FILE

To restore the configuration on the UCM6200 from a backup file, users could go to Web GUI->**Maintenance**->**Backup**->**Local Backup**.

- A list of previous configuration backups is displayed on the web page. Users could click on of the desired backup file and it will be restored to the UCM6200.
- If users have other backup files on PC to restore on the UCM6200, click on "Upload Backup File" first and select it from local PC to upload on the UCM6200. Once the uploading is done, this backup file will be displayed in the list of previous configuration backups for restore purpose. Click on to restore from the backup file.

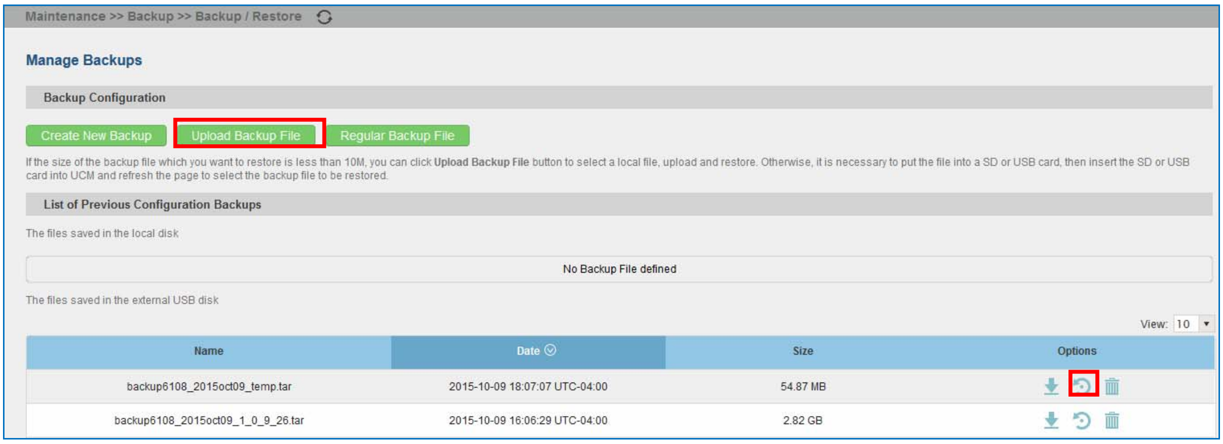


Figure 197: Restore UCM6200 from Backup File

Note:

- The uploaded backup file must be a tar file with no special characters like *,!,#,@,&,$,%,^,(,),/,\,space in the file name.
- The uploaded back file size must be under 10MB.

CLEANER

Users could configure to clean the Call Detail Report/Voice Records/Voice Mails/FAX automatically under Web GUI->**Maintenance**->**Cleaner**.

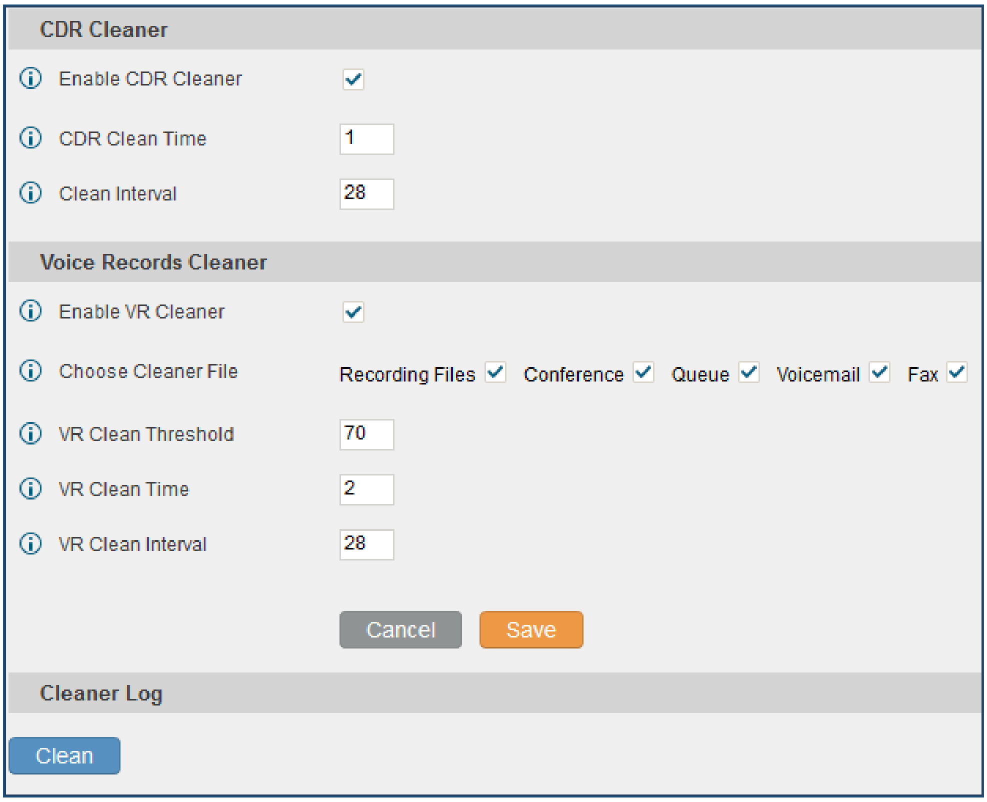


Figure 198: Cleaner

Table 104: Cleaner Configuration

Enable CDR Cleaner	Enable the CDR Cleaner function.
CDR Clean Time	Enter 0-23 to specify the hour of the day to clean up CDR.
Clean Interval	Enter 1-30 to specify the day of the month to clean up CDR.
Enable VR Cleaner	Enter the Voice Records Cleaner function.
Choose Cleaner File	Select the files for system automatic clean. • Recording Files • Conference • Queue • Voicemail • Fax
VR Clean Threshold	Specify the Voice Records threshold from 0 to 99 by using local storage status in percentage.

VR Clean Time	Enter 0-23 to specify the hour of the day to clean up Voice Records.
Clean Interval	Enter 1-30 to specify the day of the month to clean up Voice Records.

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

⚠ **Note:**

Cleaner will delete data based on Recording Storage selection. If **USB Disk** is selected, Cleaner will only clean data in USB and local data will leave untouched. If **Enable auto change** is selected and USB disk is connected, Cleaner will only delete data in USB drive. Recordings Storage function can be found under web **UI-> Settings-> Recordings Storage-> Recordings Storage**.

RESET AND REBOOT

Users could perform reset and reboot under Web GUI->**Maintenance**->**Reset and Reboot**.

To factory reset the device, select the mode type first. There are two different types for reset.

- User Data: All the data including voicemail, recordings, IVR Prompt, Music on Hold, CDR and backup files will be cleared.
- All: All the configurations and data will be reset to factory default.

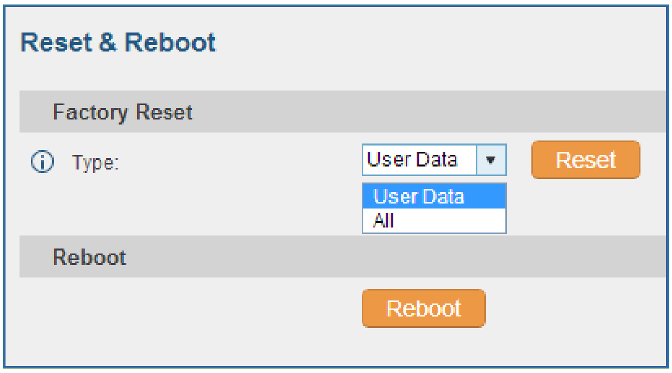


Figure 199: Reset and Reboot

SYSLOG

On the UCM6200, 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.

The default syslog level for all modules is "error", which is recommended in your UCM6200 settings because it can be helpful to locate the issues when errors happen.

Some typical modules for UCM6200 functions are as follows and users can turn on "notic" and "verb" levels besides "error" level.

pbx: This module is related to general PBX functions.
chan_sip: This module is related to SIP calls.
chan_dahdi: This module is related to analog calls (FXO/FXS).
app_meetme: This module is related to conference bridge.

⚠ **Note:**

Syslog is usually for debugging and troubleshooting purpose. Turning on all levels for all syslog modules is not recommended for daily usage. Too many syslog print might cause traffic and affect system performance.

TROUBLESHOOTING

On the UCM6200, users could capture traces, ping remote host and traceroute remote host for troubleshooting purpose under Web GUI->**Maintenance**->**Troubleshooting**.

ETHERNET CAPTURE

The captured trace can be downloaded for analysis. Also the instructions or result will be displayed in the web GUI output result.

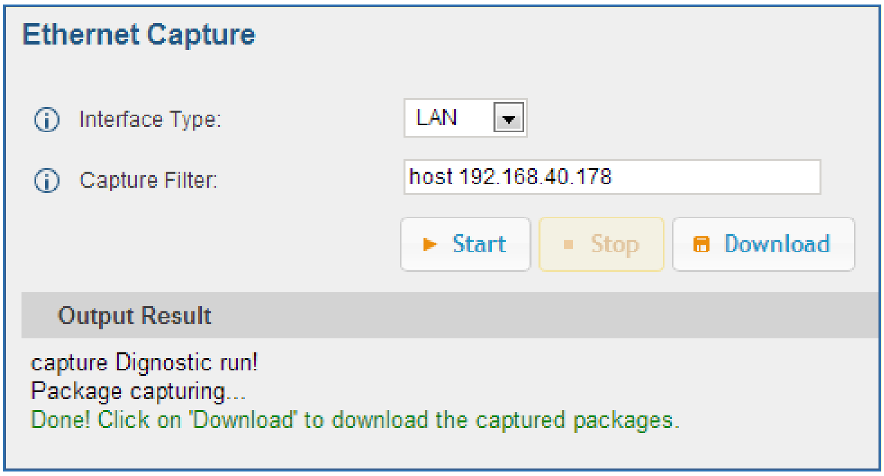


Figure 200: Ethernet Capture

The output result is in .pcap format. Therefore, users could specify the capture filter as used in general network traffic capture tool (host, src, dst, net, protocol, port, port range) before starting capturing the trace.

IP PING

Enter the target host in host name or IP address. Then press "Start" button. The output result will dynamically display in the window below.

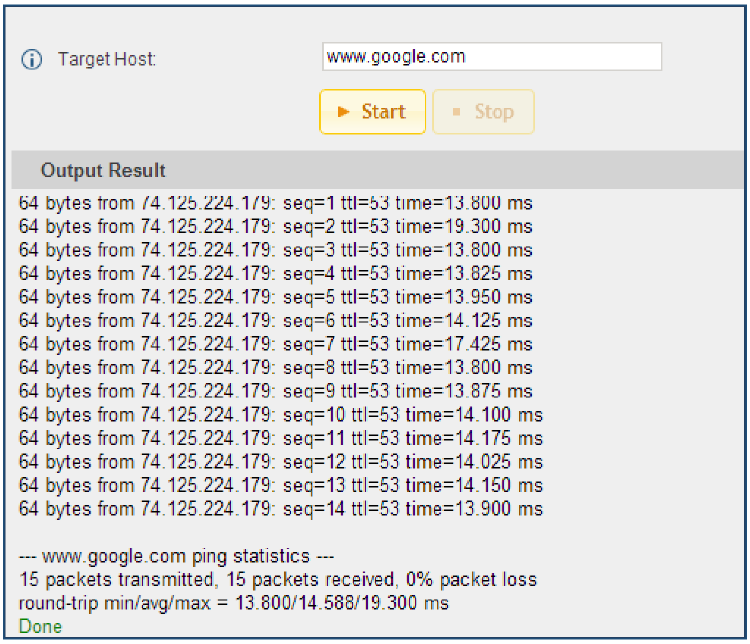


Figure 201: PING

TRACEROUTE

Enter the target host in host name or IP address. Then press "Start" button. The output result will dynamically display in the window below.

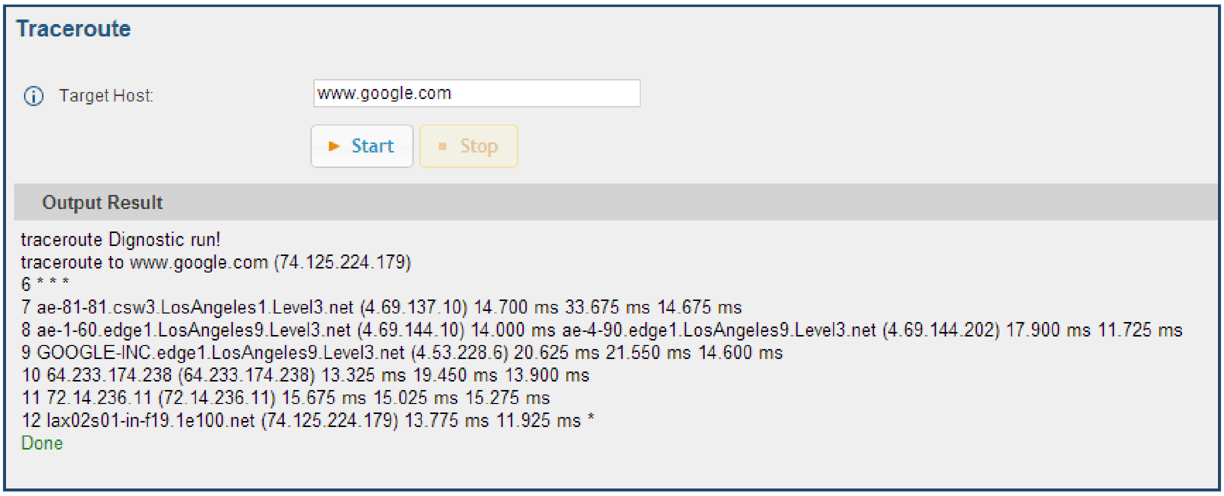


Figure 202: Traceroute

ANALOG RECORD TRACE

Analog record trace can be used to troubleshoot analog trunk issue, for example, the UCM6200 user has caller ID issue for incoming call from Analog trunk. Users can access analog record trance under web GUI->**Maintenance**->**Troubleshooting** ->**Analog Record Trace**.

Here is the step to capture trace:

1. Select FXO or FXS for "Record Ports". If the issue happens on FXO 1, select FXO port 1 to record the trace.
2. Select "Record Direction".
3. Select "Record File Mode" to separate the record per direction or mix.
4. Click on "Start".
5. Make a call via the analog port that has the issue.
6. Once done, click on "Stop".
7. Click on "Download" to download the analog record trace.

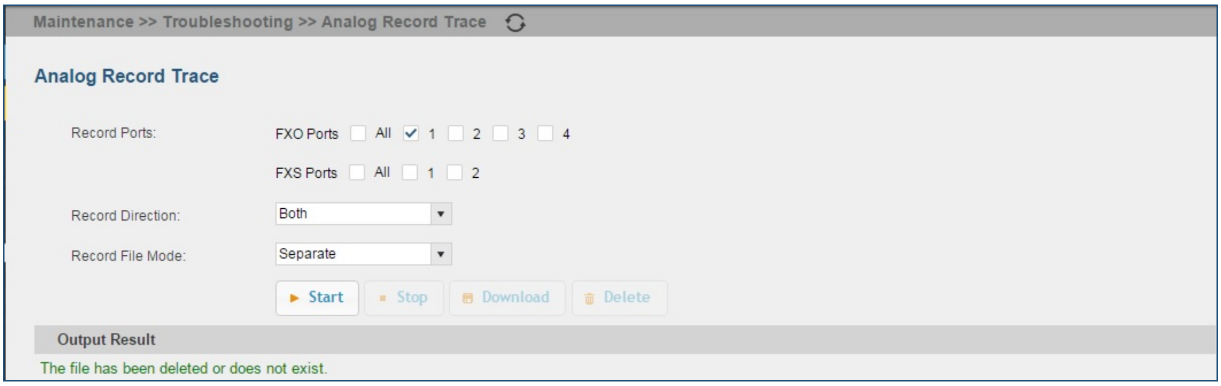


Figure 203: Troubleshooting Analog Trunks

After capturing the trace, users can download it for basic analysis. Or you can contact Grandstream Technical support in the following link for further assistance if the issue is not resolved.
http://www.grandstream.com/index.php/support

SERVICE CHECK

Enable Service Check to periodically check UCM6200. 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 UCM6200. The default setting is 3. If there is no response from UCM6200 after 3 attempts (default) to check, current status will be stored and the internal service in UCM6200 will be restarted.

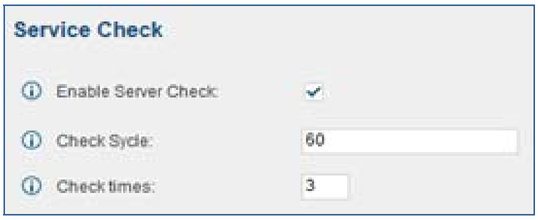


Figure 204: Service Check

NETWORK STATUS

In UCM6200 web UI->Maintenance->Troubleshooting->Network Status, the users can view active Internet connections. This information can be used to troubleshoot connection issue between UCM6200 and other services.

Maintenance >> Troubleshooting >> Network Status

Network Status

Active Internet Connections (Servers And Established)

Proto	Recv-Q	Send-Q	Local-Address	Foreign-Address	State
tcp	0	0	0.0.0.0:7777	0.0.0.0:*	LISTEN
tcp	0	0	0.0.0.0:5060	0.0.0.0:*	LISTEN
tcp	0	0	0.0.0.0:5061	0.0.0.0:*	LISTEN
tcp	0	0	0.0.0.0:389	0.0.0.0:*	LISTEN
tcp	0	0	0.0.0.0:80	0.0.0.0:*	LISTEN
tcp	0	0	0.0.0.0:8089	0.0.0.0:*	LISTEN
tcp	0	0	192.168.40.151:8089	192.168.40.191:2852	ESTABLISHED
tcp	0	1074	192.168.40.151:8089	192.168.40.191:2849	ESTABLISHED
tcp	0	0	192.168.40.151:8089	192.168.40.191:2848	ESTABLISHED
tcp	0	0	192.168.40.151:8089	192.168.40.191:2851	ESTABLISHED
tcp	0	0	127.0.0.1:52450	127.0.0.1:7777	ESTABLISHED
tcp	0	0	192.168.40.151:8089	192.168.40.191:2850	ESTABLISHED
tcp	0	0	192.168.40.151:8089	192.168.40.191:2845	TIME_WAIT
tcp	0	0	127.0.0.1:7777	127.0.0.1:52450	ESTABLISHED
tcp	0	0	:::389	:::*	LISTEN

Figure 205: Network Status

REMOTE ACCESS

SSH ACCESS

SSH switch now is available via web UI and LCD. User can enable or disable SSH access directly from web UI or LCD screen. For web SSH access, please log in UCM6200 web interface and go to **Maintenance->Remote Access->SSH Access.** By default, SSH access is disabled for security concerns. It is highly recommended to only enable SSH access for debugging purpose.

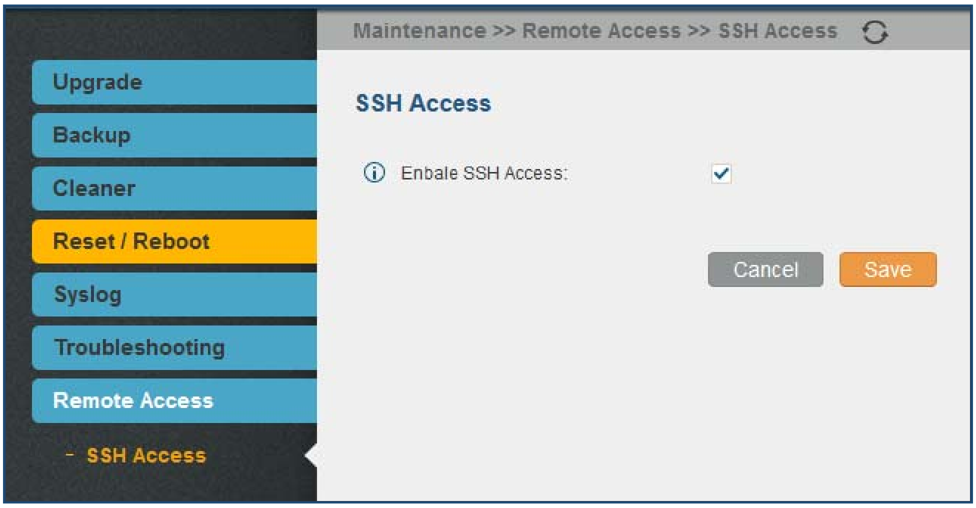


Figure 206: SSH Access

EXPERIENCING THE UCM6200 SERIES IP PBX

Please visit our website: http://www.grandstream.com to receive the most up- to-date updates on firmware releases, additional features, FAQs, documentation and news on new products.

We encourage you to browse our product related documentation, FAQs and User and Developer Forum for answers to your general questions. If you have purchased our products through a Grandstream Certified Partner or Reseller, please contact them directly for immediate support.

Our technical support staff is trained and ready to answer all of your questions. Contact a technical support member or submit a trouble ticket online to receive in-depth support.

Thank you again for purchasing Grandstream UCM6200 series IP PBX appliance, it will be sure to bring convenience and color to both your business and personal life.

FCC Caution:

Any Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.

- Increase the separation between the equipment and receiver.

- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

- Consult the dealer or an experienced radio/TV technician for help.

Regulatory Information

U.S. FCC Part 68 Statement

This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. The unit bears a label on the back which contains among other information a product identifier in the format US: GNIIS01BUCM6208. If requested, this number must be provided to the telephone company.

This equipment uses the following standard jack types for network connection: RJ11C.

This equipment contains an FCC compliant modular jack. It is designed to be connected to the telephone network or premises wiring using compatible modular plugs and cabling which comply with the requirements of FCC Part 68 rules.

The Ringer Equivalence Number, or REN, is used to determine the number of devices which may be connected to the telephone line. An excessive REN may cause the equipment to not ring in response to an incoming call. In most areas, the sum of the RENs of all equipment on a line should not exceed five (5.0).

In the unlikely event that this equipment causes harm to the telephone network, the telephone company can temporarily disconnect your service. The telephone company will try to warn you in advance of any such disconnection, but if advance notice isn't practical, it may disconnect the service first and notify you as soon as possible afterwards. In the event such a disconnection is deemed necessary, you will be advised of your right to file a complaint with the FCC.

From time to time, the telephone company may make changes in its facilities, equipment, or operations which could affect the operation of this equipment. If this occurs, the telephone company is required to provide you with advance notice so you can make the modifications necessary to obtain uninterrupted service.

There are no user serviceable components within this equipment. See Warranty flyer for repair or warranty information.

It shall be unlawful for any person within the United States to use a computer or other electronic device to send any message via a telephone facsimile unless such message clearly contains, in a margin at the top or bottom of each transmitted page or on the first page of the transmission, the date and time it is sent and an identification of the business, other entity, or individual sending the message and the telephone number of the sending machine or of such business, other entity, or individual. The telephone number provided may not be a 900 number or any other number for which charges exceed local or long distance transmission charges. Telephone facsimile machines manufactured on and after December 20, 1992, must clearly mark such identifying information on each transmitted message. Facsimile modem boards manufactured on and after December 13, 1995, must comply with the requirements of this section.

This equipment cannot be used on public coin phone service provided by the telephone company.

Connection to Party Line Service is subject to state tariffs. Contact your state public utility commission, public service commission, or corporation commission for more information.

If trouble is experienced with this equipment, please contact (Agent in the US):

Company Name: Grandstream Networks, Inc.
Address: 126 Brookline Ave, 3rd Floor Boston, MA 02215, USA
Tel: +1 617-566-9300
Fax: 844-350-7572