UCM Series IP PBX Firmware Release Notes

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FIRMWARE VERSION 1.0.18.12

PRODUCT NAME

UCM6102, UCM6104, UCM6108, UCM6116, UCM6202, UCM6204, UCM6208, and UCM6510

EOL NOTICE: 1.0.18.12 will be the last supported firmware for the UCM6100 series. UCM6100 devices will no longer receive new features, but will continue to receive critical security updates and major bug fixes.

DATE

09/28/2018

FIRMWARE FILE INFORMATION

- UCM6102/6104/6108/6116 firmware file name: ucm6100fw.bin

  MD5: 7bca157eaec8e32c586b1accf3c369d1

- UCM6202/6204/6208 firmware file name: ucm6200fw.bin

  MD5: 4f082bc65994adc4b29b412d415bae44

- UCM6510 firmware file name: ucm6500fw.bin

  MD5: 4fba6739dfa8b0c42db047976e33704c

IMPORTANT UPGRADING NOTE

- **ALWAYS create a backup of your configuration and data before a firmware upgrade.**
- **After upgrading to 1.0.18.12, if you downgrade to 1.0.17.x and issues are encountered after downgrading, factory reset the UCM and restore the backup.**
- **For UCM6xxx on 1.0.17.x, please upgrade to 1.0.18.x directly.**
- **For UCM6xxx on firmware earlier than 1.0.17.16, please upgrade to 1.0.17.16 first, then upgrade to 1.0.18.x. To upgrade to 1.0.17.16, please check 1.0.17.x release note "Important Upgrading Note" section and follow the instructions.**
- **It is recommended to upgrade UCM to latest firmware for product lifespan and security improvements.**
- **Firmware download links below:**
  - **UCM61xx:** http://firmware.grandstream.com/Release_UCM61xx_1.0.18.12.zip
  - **UCM62xx:** http://firmware.grandstream.com/Release_UCM62xx_1.0.18.12.zip
  - **UCM6510:** http://firmware.grandstream.com/Release_UCM6510_1.0.18.12.zip

PREVIOUS FIRMWARE VERSIONS

FW Version	UCM6100 Series	UCM6200 Series	UCM6510
1.0.17.16	⬇ 📄	⬇ 📄	⬇ 📄
1.0.16.20	⬇ 📄	⬇ 📄	⬇ 📄
1.0.15.16	⬇ 📄	⬇ 📄	⬇ 📄
1.0.14.24	⬇	⬇	⬇
1.0.13.14	⬇	⬇	⬇
1.0.12.19	⬇	⬇	⬇
1.0.11.27	⬇		⬇
1.0.10.44	⬇		⬇
1.0.9.97	⬇		
1.0.2.97			⬇

CHANGES SINCE BETA FIRMWARE VERSION 1.0.18.9

ENHANCEMENTS

- o [CDR API] CDR API configuration page moved from *CDR* to *Value-Added Features*.
- o [CDR API] Added ability to upload voice prompt files to via API.
- o [Grandstream Wave Web] Added ability to transfer to custom numbers and not just extensions.

BUG FIXES

- o [CDR] Fixed an issue where an error would occur when attempting to download a filtered CDR.
- o [Fax] Fixed an issue where if both a fax extension is set as an inbound fax destination and fax detection is enabled on the inbound trunk, the fax would fail.
- o [Grandstream Wave Web] Fixed an issue where audio conference invites could not be accepted via the web page.
- o [Grandstream Wave Web] Fixed an issue where there would be no audio when answering the first voice call after logging in.
- o [Grandstream Wave Web] Fixed several crashing issues.
- o [Paging/Intercom] Fixed an issue where there would be no audio in multicast pages when the page is initiated by the wakeup service and forwarded to the paging group.
- o [ZeroConfig] Fixed an issue where the VMPK section for GXP2130/40/60 templates would not show the correct number of supported VMPKs. **Note: If the templates have existing VMPK**

configurations prior to upgrading, users will need to refresh the VMPK section by clicking on *Options->Phone* **to see all available configurable VPMK fields. If the VMPK section is still not visible, clear browser cache or open the web portal page in incognito mode.**

o [ZeroConfig] Fixed an issue where batch editing selected devices would wipe existing account information on those devices.

CHANGES SINCE FIRMWARE VERSION 1.0.17.16

ENHANCEMENTS

- **Major Enhancements**
  o [WebRTC] Added Grandstream Wave WebRTC video calling and video conferencing functionality for UCM62XX/6510. [[UCM62XX/6510] GRANDSTREAM WAVE WEBRTC VIDEO CALLING & CONFERENCING]
  o [AMI] Added new commands: *GSAgents, PJSIPShowEndpoints,* and *PJSIPShowEndpoint* [NEW AMI COMMANDS]
  o [CDR] Added ability to customize the data columns included in exported CDR reports. [CDR EXPORT COLUMN CUSTOMIZATION]
  o [Extensions] Added ability to view a specified extension's membership in Call Queues/Ring Groups and other details. [EXTENSION DETAILS]
  o [Fax] Added Email-to-Fax functionality. [EMAIL-TO-FAX]
  o [Music on Hold] Added ability to disable audio files. [DISABLING MUSIC ON HOLD FILES]
  o [Parking] Added *Timeout Destination* and *Ring-All Callback on Timeout* options to *Parking Lot* page. [NEW PARKING OPTIONS: TIMEOUT DESTINATION & RING-ALL CALLBACK]
  o [Routing] Added inbound rule importing/exporting functionality. [INBOUND ROUTE IMPORTING & EXPORTING]
  o [ZeroConfig] Added GXV3370 model template. Please note GXV3370 firmware 1.0.1.21 does not support UCM ZeroConfig. Please use firmware 1.0.1.12 or 1.0.1.18. To provision GXV3370 using UCM ZeroConfig, please download GXV3370 model template from UCM webUI->Value-added Features->Zero Config->Model Update first.

- **Other Enhancements**
  o [Active Calls] CID number and name will replace DOD number and name if applicable.
  o [Extensions] CID numbers now have a minimum character limit of 2.
  o [Extensions] Custom statuses can no longer contain periods (.).
  o [Paging/Intercom] Added *Maximum Call Duration* option to paging/intercom group configuration page.
  o [Paging/Intercom] Forwarded paging calls will contain call-info, alert-info, and diversion headers.
  o [Paging/Intercom] Paging/Intercom can now be scheduled weekly.
  o [Recordings] The *Recording Files* page will how the recordings of the current month by default if no filters are applied.
  o [Ring Groups] Ring Group extensions can no longer have letters.

- o [SIP Settings] UCM will now forward FROM headers and add privacy headers to anonymous calls if applicable.
- o [SIP Settings] Added *Passthrough PAI Header* option to allow or block PAI headers as they go through the UCM.
- o [System] If *Clean Files in External Storage* is enabled, files stored in NAS will also be cleaned.
- o [System] All voice prompts will be reset to default prompts after a factory reset.
- o [Voicemail] Added ability to change voicemail audio format to WAV.
- o [Voicemail] Resetting an extension will change its *Voicemail* setting to "Enable Local Voicemail".
- o [VoIP Trunks] Added *DOD Name* column to DOD list page.
- o [VoIP Trunks] Added Supervoice to SIP Trunk provider list.
- o [WebUI] Password character limit changed to 30.
- o [WebUI] Added "All" options for selecting days and months.

BUG FIXES

- **[Active Calls]**
  - o Fixed several issues causing duplicate Active Call entries.
  - o Fixed several issues with incorrect entries when SCA extensions and private numbers are part of ring groups and paging/intercom groups.
  - o Fixed an issue with incorrect active call display when dialing emergency numbers via trunk group.
  - o Fixed an issue with incorrect active call display when monitoring an external extension connected via VoIP trunk.
  - o Fixed an issue with incorrect active call entry if a ring group is dialed, and the ring group member has an LDAP number configured.
  - o Fixed an issue with calls transferred through the ring group would not be included in the page.
- **[AMI]**
  - o Fixed an issue with inserting IP addresses ending in ".255" in *AMI->Permitted IP*.
- **[Basic Calls]**
  - o Fixed an issue where calls routed to external numbers would get the highest permission level during any time condition even if the original caller had lower permissions.
  - o Fixed an issue where if *Can Direct Media* is enabled, holding/unholding a call and transferring will cause a loss of audio on both ends.
- **[Backup/Restore]**
  - o Fixed an issue with Data Sync not backing up recordings placed in root directories.
  - o Fixed an issue with scheduled backups would not back up all CDR when the scheduled time is the same as Data Sync's backup time.
- **[CDR]**
  - o Fixed several issues where CDR would not show the correct call details in certain scenarios.
  - o Fixed an issue where CDR entries would be deleted if both *CDR->Automatic Download* and *Call Queue->Automatic Download* are enabled and the email address configured for both are the same.
  - o Fixed an issue where CDR *Call Times* and *Talk Times* had the same value.

- o Fixed an issue where the CDR would not show the current month's CDR immediately after changing system time.
- o Fixed an issue with missing call originator and caller information when emergency call notifies all included members simultaneously.
- o Fixed an issue with blank CDR statistics pages after sorting by time from newest to oldest.
- **[Conference]**
  - o Fixed an issue with call failure messages being played twice when attempting to invite participants.
  - o Fixed an issue with sending the wrong notification to endpoints after getting locked and unlocked.
- **[Email Settings]**
  - o Fixed an issue with email send log where filtering a second time will show all logs rather than the filtered results.
- **[Extension Groups]**
  - o Fixed an issue where an extension group could not be edited if a member was previously assigned a DOD number.
- **[Follow Me]**
  - o Fixed an issue with calls not reaching voicemail if *Confirm when Answering* was disabled.
- **[HTTP Server]**
  - o Fixed an issue where certificate requests keep getting server unavailable errors.
- **[LDAP]**
  - o Fixed an issue where blank values are overwritten with "None" when importing XML phonebooks.
- **[Maintenance]**
  - o Fixed an issue where queue logs generated after a scheduled cleaning time would still be cleaned.
- **[Network Settings]**
  - o Fixed an issue with the DHCP client list showing no entries.
- **[Operation Log]**
  - o Fixed an issue where analog record signal tracing events would be shown as SS7 signal tracing.
  - o Fixed an issue with Operation Log time format being inconsistent with system time format.
- **[Paging/Intercom]**
  - o Fixed an issue where transfers to paging/intercom groups would fail despite the caller being in the paging/intercom whitelist.
- **[Routing]**
  - o Fixed an issue with users being able to import duplicate outbound blacklist rules and rules with more than 32 characters.
  - o Fixed an issue with capital letters in imported outbound blacklist rules being converted to lowercase after the importing.
  - o Fixed an issue with adding back deleted outbound blacklist rules.
  - o Fixed an issue where users were able to import empty blacklist rules.
  - o Fixed an issue that caused some outbound calls to fail despite matching time conditions.
- **[SCA]**
  - o Fixed an issue where if an Announcement Center member is an SCA extension and rejects a call, it would be forwarded to voicemail.

- o Fixed an issue where an SCA extension and its private numbers could all accept the same conference invite.
  - o Fixed an issue with incorrect SCA status after disabling and enabling it during a call.
  - o Fixed an issue with trunk auto record not working when routing to an SCA extension.
  - o Fixed an issue where SCA extensions keep hearing Music on Hold even after being transferred.
- **[SIP Settings]**
  - o Fixed an issue with Remote Party ID information not being forwarded properly on call transfers.
- **[System]**
  - o Fixed an issue where the UCM would keep crashing if mp3 files were present in the Music on Hold playlists and *Voice Prompts* page.
- **[System Events]**
  - o Fixed an issue where 2 alarm logs would be generated when *Sip Internal Call Failure* alerts are enabled.
- **[Troubleshooting]**
  - o Fixed an issue with external numbers being cleared after refreshing the page during tracing.
- **[Trunk Groups]**
  - o Fixed an issue where duplicate users could be created.
- **[User Portal]**
  - o Fixed an issue with call transfer information not being displayed when extension DND is enabled.
  - o Fixed an issue with queue chairman user portal's login time displaying abnormally.
  - o Fixed an issue with SCA users not being able to see recordings.
  - o Fixed an issue where both SCA and Follow Me features could be enabled at the same time.
- **[Voicemail]**
  - o Fixed an issue with long voicemail passwords not being reflected properly in the web portal when created through the voice portal.
  - o Fixed an issue with calls entering into the wrong voicemail after extensive call rerouting.
- **[Voice Prompt]**
  - o Fixed an issue with automated recording prompts being in the wrong language when a configured extension has call forwarding set up.
- **[WebUI]**
  - o Fixed an issue with *RTP Settings* page modifications not being reflected on the web page.
  - o Fixed an issue where the *Voicemail Group* list page would change sort order after refreshing the page.
  - o Fixed an issue where changing FXS extension presence status via feature code would not reflect on the web UI.
- **[Zero Config]**
  - o Fixed an issue where model template configurations disappear after upgrading from 1.0.16.20 to 1.0.17.16.
  - o Fixed an issue where GXV devices could not retrieve CUST files uploaded to the UCM.
- **[UCM6510] [HA]**

o   Fixed an issue where if the recording file storage path is set to NAS, and failover occurs, the storage path would change to local storage.

NEW FEATURES OVERVIEW

This section describes the major new features/changes introduced in the update and provides instructions for usage.

[UCM62XX/6510] GRANDSTREAM WAVE WEBRTC VIDEO CALLING & CONFERENCING

Web audio and video calls and conferencing can now be achieved through the UCM's new WebRTC page. To ensure the best experience, please use Google Chrome (v67 or higher) or Mozilla Firefox (v60). Please note this feature is supported on UCM62xx and UCM6510 only.

Warning: Video conferencing can be resource-intensive and may cause performance issues with the UCM when used. More details found in the *Specifications* section below.

To get started with this new feature:

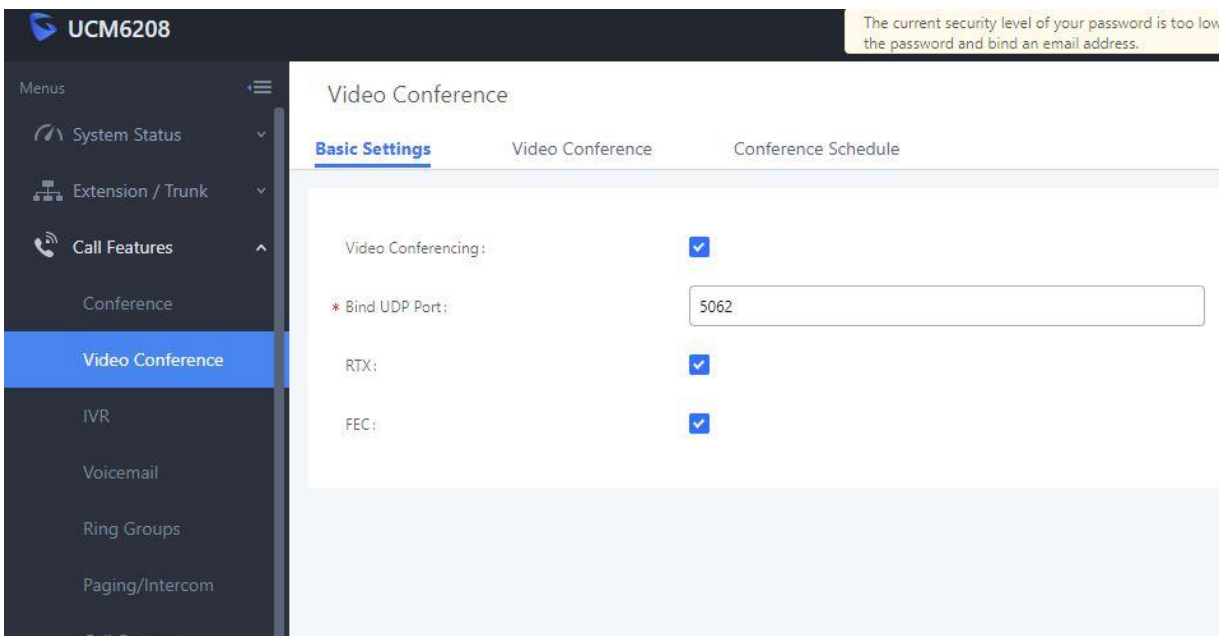1) Navigate to *Value-Added Features->WebRTC* and enable WebRTC support.
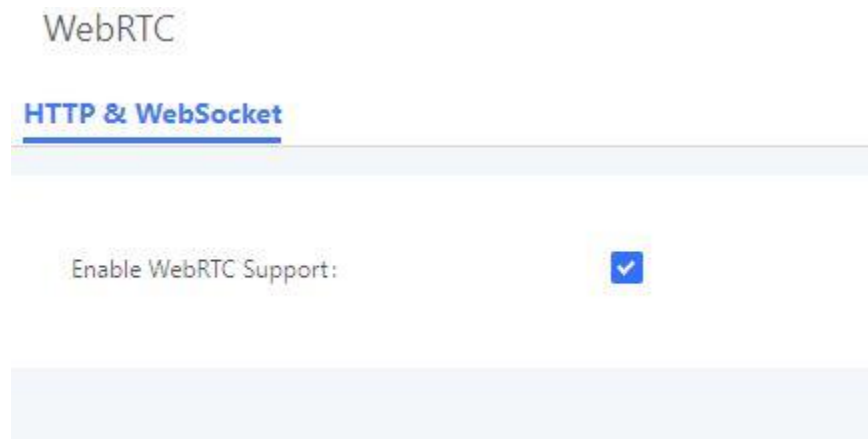
2) Select the extensions that would use WebRTC and enable WebRTC support on them.

Other Settings

Ring Timeout :	
* Skip Trunk Auth :	No
Support Hot-desking Mode :	☐
Enable WebRTC Support :	☑
Call Duration Limit :	☐
Call Waiting :	☑

3) Navigate to *Call Features->Video Conference* and check the *Video Conferencing* option.

4) Click on the *Video Conference* tab and create a new video conference room.

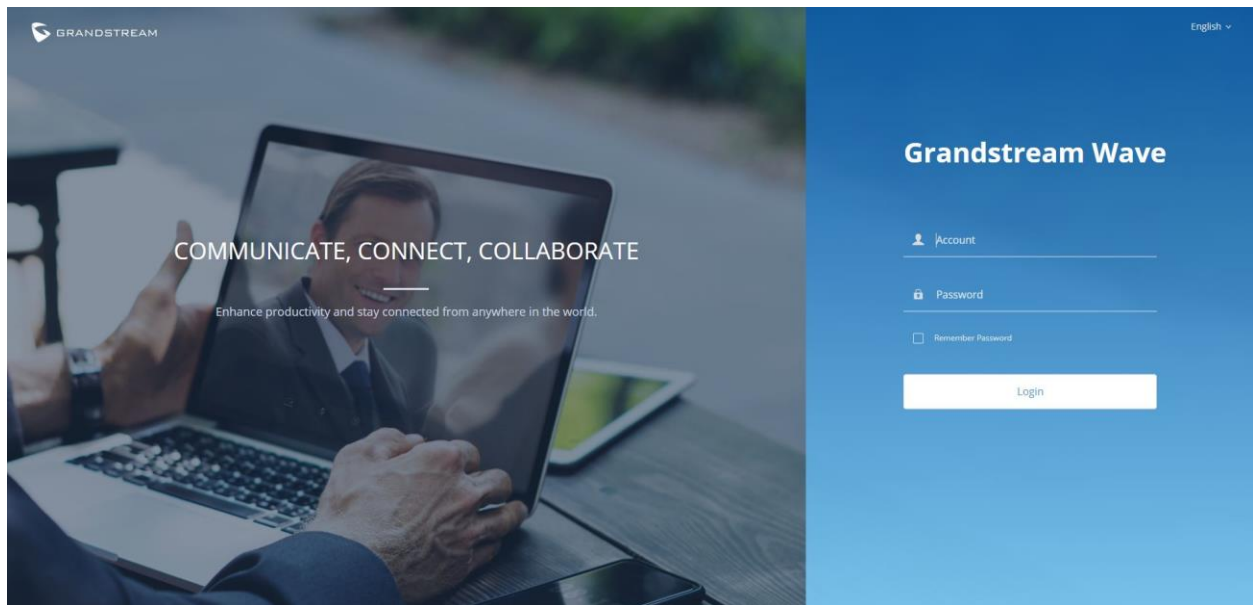Video Conference

Basic Settings   **Video Conference**   Conference Schedule

Please ensure that ICE Support is enabled and that STUN/TURN server is configured with NAT.
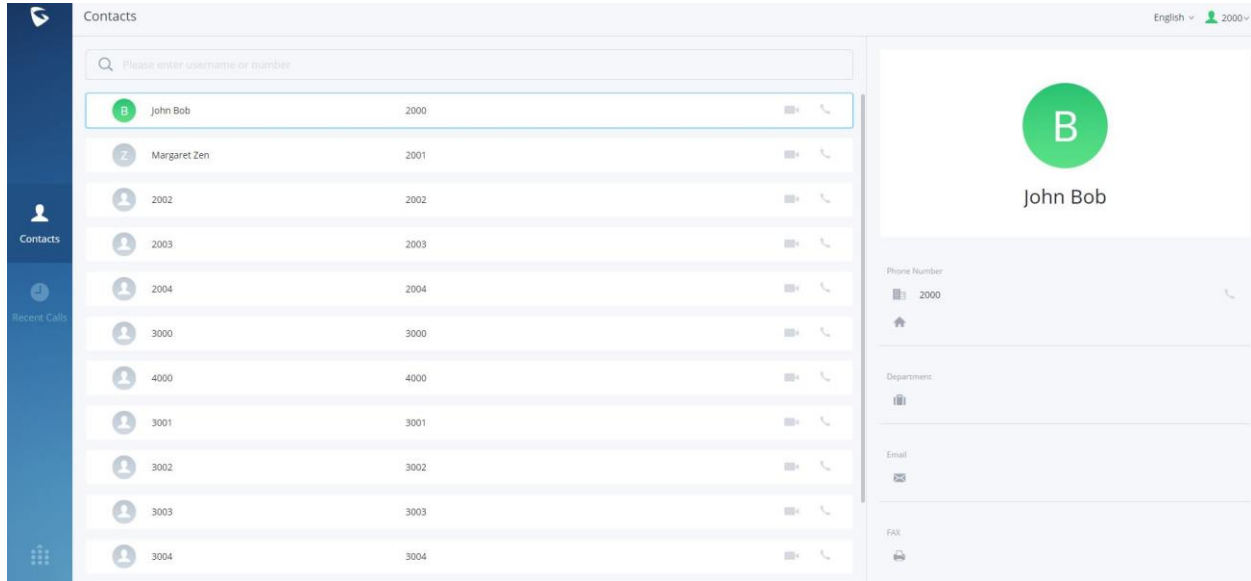
+ Create New Conference Room

Room
6302

5) Access the page by adding "/gswave" after the UCM's server address and port. For example, https://my.ucm.com:8089/gswave


6) Enter an extension number and that extension's SIP password. Once logged in successfully, the following page will appear:

Note: The same IP address cannot log into both the admin/user portal and the Grandsream Wave WebRTC portal at the same time.

7) From here on, users can initiate an audio or video call to individual extensions and conference rooms. After entering a video conference room, users can invite other members to join. If starting a call for the first time on a browser, users may be asked to allow the web page to use the devices's mic and/or webcam. Users may be required to drop and re-establish the call after allowing webcam/microphone access.
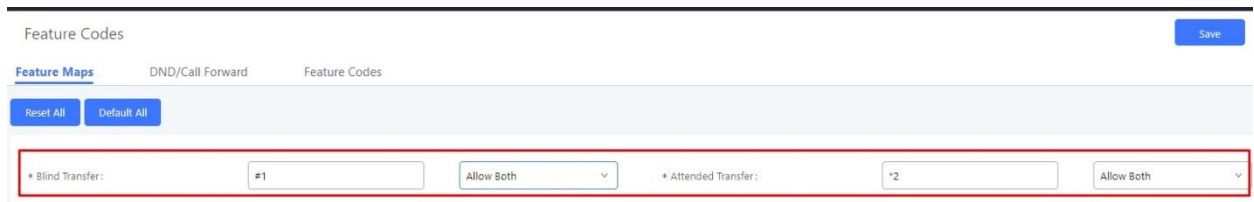

Note 1: To transfer calls, the UCM's transfer feature codes must be enabled.

**Note 2:** To complete an attended transfer, one must hang up the call.

Note 3: While users can dial into regular audio conference rooms via Grandstream Wave Web, extensions must not include Opus as a supported codec. If users find themselves unable to dial into audio conference rooms, please check the extension codec settings.
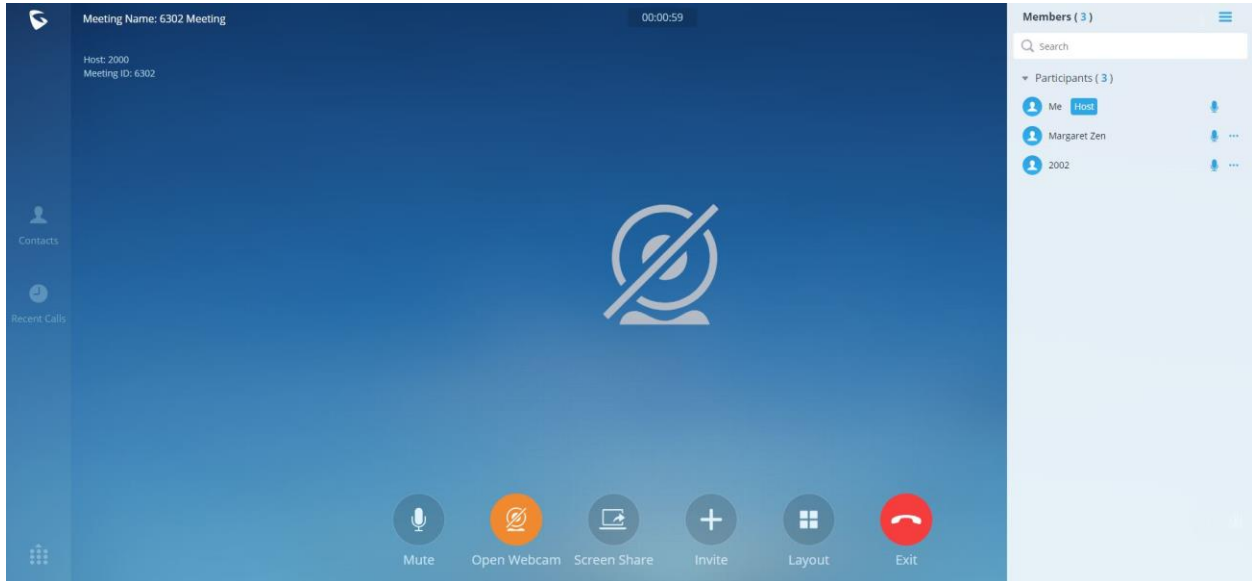
8) Once a video conference call is established, users can mute themselves, enable and disable their video feed, screen share, invite other people to the conference, and change the video layout. Currently, only Equal (grid) and Focus video layouts are available.

9) Screen sharing allows users to share their whole screen, an application, or a specific browser tab to all participants.
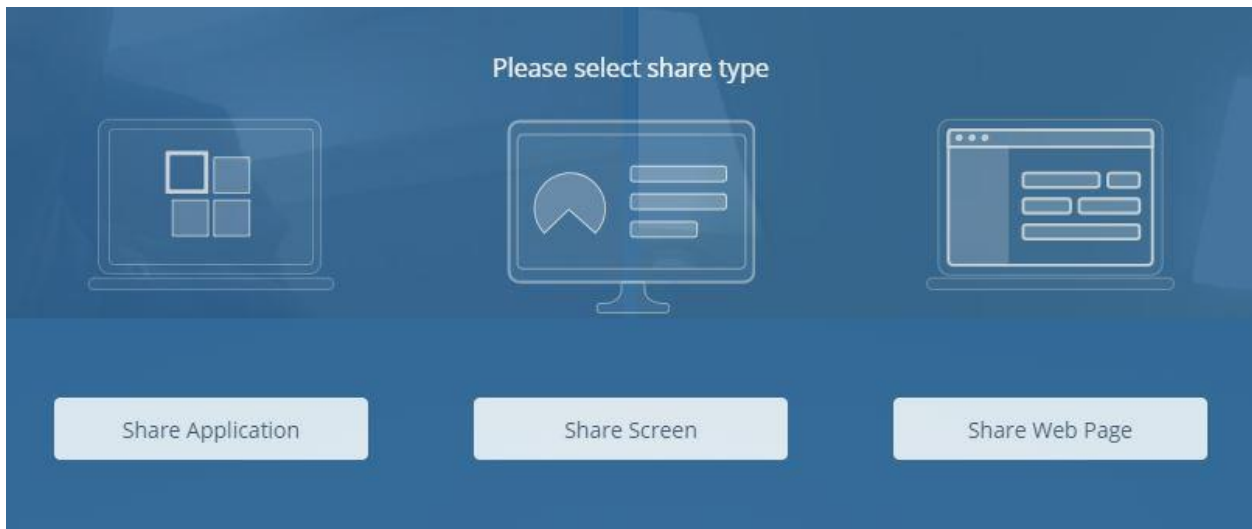
Note 1: Screen sharing requires the *IPVT Screen Capturing* browser extension/add-on to work properly. If the browser does not have it, a prompt will appear asking to install or allow it. Users may need to disable their ad-blockers to have the prompts appear.
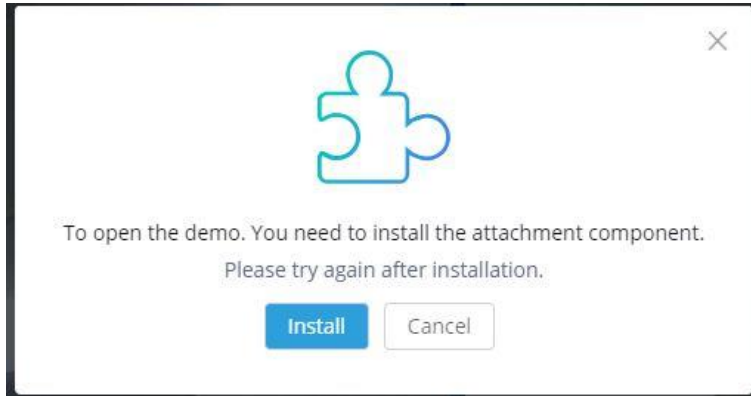
Note 2: GXV and GVC devices will need to enable presentation to have the screen share appear.
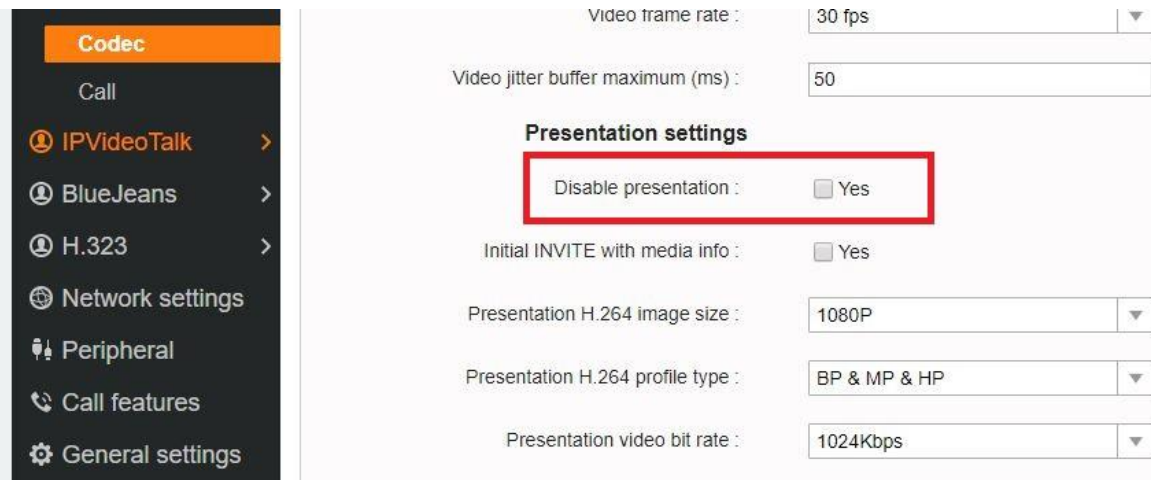
10) The host of a conference has the following privileges:

    a.   Lock and unlock conferences to control access to it.

    b.   Mute and unmute conference participants

    c.   Transfer host privileges to one of the participants.

11) To schedule a video conference, navigate back to the *Video Conference* page on the UCM admin portal and click on the *Conference Schedule* tab.

Schedule New Conference

* Conference Subject :		* Conference Room : 6302
Conference password :		* Kick Time (m) : 10
* Schedule Time : Select date		* Duration (m) : 15
Period : No Repeat		Host : None
Members :		
Special Extension : FirstName Telephone Email ⊕		
Shareable Link :		
Description :		

    a.   The following fields are available to configure:

        i.   **Conference Subject** – Purpose of the conference.

        ii.   **Conference Room** – Extension of the conference room to dial into.

        iii.   **Conference Password** – Password to enter the conference.

        iv.   **Kick Time (m)** – The amount of minutes before the conference start time to play a warning prompt. Any participants still in the conference room 5 minutes after the prompt plays will be automatically kicked out.

        v.   **Schedule Time** – Date and time of the conference. Conferences cannot be scheduled earlier than the current time and must be scheduled at least 10 minutes before the start time.

        vi.   **Duration** – The length of the conference.

        vii.   **Period** – Repeat the conference either daily or weekly.

        viii.   **Host** – Assign an extension as the host of the conference.

ix. Members – Select the extensions to invite to the conference.

x. Special Extension – Add non-extensions to invite to the conference.

xi. Shareable Link – Configure a publicly accessible link with a server address and port that can be clicked on for quick access to the video conference.

xii. Description - Additional details about the conference.

b. Upon scheduling the conference, invites will be sent out to the host and selected participants. These invitations will include the conference details and a link to the conference. Upon clicking the link, participants will be prompted to enter their extension numbers to log in and join the conference.

Hi "Michael Jordan",

You are invited to attend the following conference.

Conference Schedule Details
This is the information of the schedule conference which you will attendee.

Topic:
UCMSFU

Description:

Schedule Time:
2018-07-06 15:53 - 2018-07-06 16:08

Conference Info:
Conference Room: 6300
Conference Password: No Password

LAN Conference Link Address:
Click the link to enter Schedule Conference

WAN Conference Link Address:
No Address

Notes: please open the link address with Chrome or Firefox browser.

12) Specifications

    a. Up to 4 video feeds in a conference call (one 1080p, others being QVGA) and one screen share feed.

    b. Up to 100 WebRTC users can be logged into the Grandstream Wave page at the same time.

    c. Participant limit:

        i. UCM62xx: 8

        ii. UCM6510: 15

        iii. **Warning: A video conference with 4 ongoing video feeds will result in 75-83% CPU usage. UCM62xx and UCM65100 will be able to handle 15 and 30 more concurrent audio calls respectively.**

    d. Audio: PCMU, PCMA, GSM, ILBC, G722.1, G722.1C, G729, G723.1, G726, OPUS

    e. Video: H264, VP8

    f. Framerate: 15 FPS video stream, 5 FPS screen share

    g. FEC (Forward Error Correction)

    h. RTX (Retransmission)

13) Known Issues

    a. Camera/mic hot swapping currently does not work.

    b. GXV3175 and Grandstream Wave app cannot see the host's video feed.

    c. Audio quality issues when:

        i. More than one participant is using G723.1 codec.

        ii. GSM codec is used.

        iii. SRTP key length is not 128-bit

        iv. SRTP is enabled, but not forced.

    d. If two participants open their cameras at the same time, the following may occur:

        i. A participant will be booted from the conference.

ii. The video feeds for other participants will turn black.

e. Audio calls cannot be changed to video calls mid-call.

f. Video conference participants may be kicked out when additional participants join the conference via email link.

g. Packet loss retransmission methods currently do not work with WebRTC video calls.

h. No IPv6 support.

i. Black video feed due to Firefox's native issues with H264 and TURN server support.

j. Firefox v61 issue: Caller cannot hear the callee when Firefox is used to answer the call.

NEW AMI COMMANDS

Three additional AMI commands have been added:

- GSAgents – Lists queue agents and their status. Functionally similar to Asterisk's *Agents* command.
- PJSIPShowEndpoint – Shows the information of an individual SIP endpoint. Functionally similar to Asterisk's *SIPshowpeer* command.
- PJSIPShowEndpoint – Shows the information of all SIP endpoints. Functionally similar to Asteriks's *SIPshowpeer* command.

CDR EXPORT COLUMN CUSTOMIZATION

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

EXTENSION DETAILS

Users can now click on an extension number in the *Extensions* list page and quickly view information about it.

EMAIL-TO-FAX

Email-to-Fax allows users to send an email with a TIFF/PDF attachment to a specified UCM fax extension, which would then be forwarded to the fax extension's configured email address. The detected TIFF/PDF attachments will then be added to the *Fax/T.38* page's fax file list. **Note**: It will take approximately 2-3 minutes for the attachments to be processed and sent out to the designated fax extension.

To set up this feature:

1. Navigate to *System Settings->Email Settings* and configure the following options:

   a. Set *Type* as "Client".

   b. Configure SMTP server settings and email credentials

   c. Tick the *Enable Email-to-Fax* option

   d. Configure POP settings

   e. Configure *Sender* with the same email address set in the SMTP fields.

2. Navigate to *Call Features->Fax/T.38* and create a fax extension.

To use this feature, send an email to the configured email address with the subject "SendFaxMail to XXXX" where XXXX is the fax extension and attach the TIFF/PDF file. After 2-3 minutes, the attachment should appear in the fax file list in the *Call Features->Fax/T.38* page in both TIFF and PDF format.

Restrictions:

1. Only one attachment per email is supported.

2. TIFF/PDF files cannot have pages in landscape orientation.

DISABLING MUSIC ON HOLD FILES

Music on Hold audio files can now be disabled to prevent playback, removing the need to delete audio files just to not have them play.

NEW PARKING OPTIONS: TIMEOUT DESTINATION & RING-ALL CALLBACK

Additional options have been added call park timeouts: *Timeout Destination* and *Ring-All Callbacks.* By default, timed out parked calls will be rerouted to the party that parked the call.

- **Timeout Destination** allows users to control where these timed-out calls will be routed to.

- **Ring-All Callback** only applies to extensions that have concurrent registrations on other endpoint devices. If a parked call times out, the resulting callback will ring all the endpoints that the extension has registrations on.

INBOUND ROUTE IMPORTING & EXPORTING

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