

UCM Series IP PBX Firmware Release Notes

Table of Content

FIRMWARE VERSION 1.0.20.17	2
PRODUCT NAME.....	2
DATE	2
FIRMWARE FILE INFORMATION	2
IMPORTANT UPGRADING NOTE.....	2
PREVIOUS FIRMWARE VERSIONS.....	2
CHANGES SINCE FIRMWARE VERSION 1.0.19.27	3
<i>MAJOR ENHANCEMENTS</i>	3
<i>OTHER ENHANCEMENTS</i>	4
<i>BUG FIXES</i>	7
<i>NEW LIMITATIONS</i>	12
<i>KNOWN ISSUES</i>	13
NEW FEATURES OVERVIEW	15
<i>ANNOUNCEMENT</i>	15
<i>CONFERENCE CONTACT GROUPS</i>	15
<i>CONFERENCE REPORTS</i>	16
<i>CALL FORWARDING REMOTE MANAGEMENT</i>	17
<i>GDMS SIP ACCOUNT SYNCING</i>	18
<i>NEW HTTP API</i>	21
<i>NEW CLEANING CONDITIONS</i>	22
<i>VIRTUAL QUEUE KEY CUSTOMIZATION</i>	23
<i>QUEUE STATISTICS IMPROVEMENTS</i>	24
<i>QUEUOMETRICS SUPPORT</i>	27
<i>PRIVATE INTERCOM (GSC3510 ONLY)</i>	27
<i>PMS VOICMAIL BACKUP ON CHECK-OUT</i>	28
<i>CALLING INTO IPVT MEETINGS</i>	29
<i>ASSIGNING DOD TO FAX SENDING</i>	31

FIRMWARE VERSION 1.0.20.17

PRODUCT NAME

UCM6202, UCM6204, UCM6208, and UCM6510

DATE

2/3/2020

FIRMWARE FILE INFORMATION

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





MD5: e3a785127a866a50741c887675b246bd
- UCM6510 firmware file name: ucm6500fw.bin










































MD5: 73c349f691c2df97fedc504a4498828c

IMPORTANT UPGRADING NOTE

- **ALWAYS create a backup of your configuration and data before a firmware upgrade.**
- **If downgrading from 1.0.20.x to 1.0.19.x, please restore the 1.0.19.x backup after the downgrade to avoid issues with saving settings and backing up.**
- **For UCM6xxx on 1.0.19.x, please upgrade to 1.0.20.x directly.**
- **For UCM6xxx on firmware earlier than 1.0.20.x, please upgrade to 1.0.19.x first, then upgrade to 1.0.20.x. To upgrade to 1.0.19.x, please check 1.0.19.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:**
 - **UCM62xx:** http://firmware.grandstream.com/Release_UCM62xx_1.0.20.17.zip
 - **UCM6510:** http://firmware.grandstream.com/Release_UCM6510_1.0.20.17.zip

PREVIOUS FIRMWARE VERSIONS

FW Version	UCM6100 Series	UCM6200 Series	UCM6510
1.0.19.29		 	 

1.0.18.17	 		
1.0.18.13	 	 	 
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 FIRMWARE VERSION 1.0.19.27

MAJOR ENHANCEMENTS

- **[Announcement]** Added Announcement feature. [ANNOUNCEMENT]
- **[Backup/Restore]** Added NAS as a backup/restore location.
- **[Conference]** Added conference contact groups. [CONFERENCE CONTACT GROUPS]
- **[Conference]** Added conference call statistics and reports. [CONFERENCE REPORTS]
- **[CRM]** Added support for Zoho CRM v2 API.
- **[Extensions]** DND Whitelist now supports patterns and no longer has the 10-number restriction.
- **[Feature Codes]** Added feature codes for remote management of extension call forwarding. [CALL FORWARDING REMOTE MANAGEMENT]
- **[GDMS]** Added GDMS SIP account syncing. [GDMS SIP ACCOUNT SYNCING]
- **[HTTPS API]** Add a new HTTPS API. [NEW HTTP API]
- **[Maintenance]** Added several more conditions for cleaning files and CDR. [NEW CLEANING CONDITIONS]
- **[Maintenance]** Added the ability to download files on external storage from the *USB Disk/SD Card File Management* page.
- **[Queue]** Added the ability to customize the keys used for virtual queue. [VIRTUAL QUEUE KEY CUSTOMIZATION]
- **[Queue]** Improved queue statistics page [QUEUE STATISTICS IMPROVEMENTS]

- **[Queue]** Added a *Welcome Prompt* option to the *Edit Queue* page. Users can now upload an audio file to play when callers call into the queue.
- **[Queue]** Added QueueMetrics support. [QUEUOMETRICS SUPPORT]
- **[Paging]** Added *Private Intercom* paging type (GSC3510 only). [PRIVATE INTERCOM (GSC3510 ONLY)]
- **[Routing]** Users can now add up to 10 inbound modes from the *Set Global Inbound Mode* page.
- **[Security]** Added *TLS Security* tab to allow users to manage the highest and lowest TLS versions the UCM will support.
- **[SIP Settings]** Replaced RFC2833 DTMF option with RFC4733.

OTHER ENHANCEMENTS

- **[Active Calls]**
 - Parked calls will now be displayed in the *Active Calls* page.
- **[Basic Calls]**
 - Added *International Call Prefix* field to *PBX Settings* → *General Settings* page.
 - After an attended transfer is completed, endpoints will now be updated with the CID of the other party instead of keeping the transferer's CID.
 - When UCM passes along SDP with no media attribute, it will use *sendrecv* as a media attribute by default instead of *inactive*.
- **[Backup/Restore]**
 - Added event logs and email notifications for scheduled SFTP backup and data sync results.
- **[CDR]**
 - Added *Extension Groups* as a CDR filter.
- **[Conference]**
 - Added a field to configure the Kick Prompt Interval.
 - Conference participant names will be announced when joining/leaving the conference even when the conference is on hold.
- **[CRM]**
 - *Add Unknown Number* option has been added for ACT! CRM
 - Added <https://www.zohoapis.eu> to the Zoho CRM Server Address dropdown list.
- **[Dial by Name]**
 - Added the ability to upload a custom Dial by Name prompt.
- **[Digital Trunks]**
 - *Direct Callback* option will now be hidden if E&M Immediate or E&M Wink is selected for the *Edit T1 Port* → *Signaling* field due to not being applicable.
- **[Email Settings]**
 - Added new variable `#{CONFR_MEMBERS}`, which shows conference participant details.
 - Added the following email type filters to the *Email Send Log* page: Send Fax, Call Queue Statistics, Conference Report.

- Updated default email template. To use the new template, users must click on the *Restore Default Template* option in the *Edit Template* page.
- **[Emergency Calls]**
 - Users can now strip the same amount of numbers as the emergency number length itself.
- **[Eventlist]**
 - Added a search bar for eventlists
- **[Extensions]**
 - *Enable CC* option added as batch edit extension option.
 - The “Local Subnet Only” and “Specified IP Address” options for the *Strategy* field (now called *ACL Policy*) have merged into the “Local Network Address” option.
 - Subnet mask can now be specified when configuring subnets that can register to the extension.
 - Updated extension DND response.
 - Voicemail count information has been moved to the “More” pop-up menu that appears when clicking on the extension number in the Extensions List.
- **[Grandstream Wave Web]**
 - A conference participant's CID number will now be displayed instead of extension number if configured.
 - Improved extension searching when transferring.
 - An error message will now appear when the maximum number of allowed registrations has been reached.
 - Hosts can now generate a link that can be used to invite others to the conference.
- **[IVR]**
 - Users can now use up to 5 IVR welcome prompts.
- **[LDAP]**
 - *LDAP Client CA Cert* field no longer supports .ca file uploads. The following file types are supported: .crt .der .pem.
 - LDAP now supports downloading phonebooks from domains.
 - Added space support to the *LDAP Client Username* and *LDAP Server RootDN* fields.
- **[Maintenance]**
 - Added Conference Call Statistics Report Cleaner
- **[Miscellaneous]**
 - GXP21xx phones can now unlock doors controlled by GDS.
- **[OpenVPN]**
 - Added option *Allow Weak SSL Ciphers*.
- **[Paging]**
 - Added an option to allow scheduled paging to play during holidays.
- **[Parking]**
 - Ring All Callback on Timeout will not be available if *Forward to Destination on Timeout* is enabled.
- **[PMS]**

- The *User Name* and *Guest Category Code* columns in the Room Status page can now be sorted.
- **[Queue]**
 - Agents can now view recordings of their calls in their user portal.
 - A queue can now have 3 chairmen to manage it.
- **[Ring Group]**
 - Added the option to skip busy members when receiving an incoming call.
- **[SIP Settings]**
 - The *TLS Self-signed CA* field no longer supports .ca file uploads. The following file types are supported: .ca .crt .der .pem.
- **[Recording]**
 - MoH will now be recorded when a party is on hold.
- **[Routing]**
 - Users can now add patterns to the blacklist to restrict calls from several numbers.
 - Inbound route imports/exports now have the following columns: Fax Detection, Fax Intelligent Routing, and Fax Destination
 - Users can now export PIN Groups.
 - The *PIN Groups with Privilege Level* option has been added to allow PIN groups to work alongside *Privilege Level* and *Filter on Source Caller ID* settings.
- **[System Information]**
 - Added duplex type and network port connection speed information to *System Information* → *Network* page.
- **[Time Settings]**
 - Users can now configure holidays for the next 4 years.
 - Added time condition *Office Time and Out of Holiday*.
 - Default NTP server is now pool.ntp.org.
 - Israel now has its own time zone to account for daylight savings.
 - São Paulo / Brazil time zone is now GMT-3:00.
- **[User Management]**
 - Added the following custom privileges:
 - CDR Records
 - CDR Statistics
 - CDR Recordings
 - Voice Prompt
 - Inbound Routes
 - Outbound Routes
 - Added the option to change Super Administrator username in the *Change Information* page. Warning: This username cannot be changed back to “admin”.
 - Regular users will now be logged out automatically if an admin resets their extension.
- **[Voicemail]**
 - The voicemail system will now mention if a voicemail is from a ring group.

- **[VoIP Trunks]**
 - When editing a register trunk or trunk group and TLS is selected in the Transport field, users can now select between SIP and SIPS URI scheme.
 - Domain names can now be sent in the request URL by configuring the From Domain field.
- **[Wakeup]**
 - The wakeup service prompt now has an option to set a wakeup for the next day.

BUG FIXES

- **[Security]**
 - Fixed a security vulnerability that allowed unauthorized outbound calls to be made.
- **[System]**
 - Fixed an issue with crashing when users with custom privileges perform certain actions.
 - Fixed an issue with crashing when receiving 120 simultaneous calls using TLS.
 - Fixed an issue where the UCM would be unable to register to trunks with domains set in the Host Name field after rebooting with no Ethernet connection.
 - Fixed an issue with potential crashing when using Zero Config to update or reboot endpoints.
 - Fixed an issue with crashing in HA environments.
- **[Active Calls]**
 - Fixed duplicate entries display when analog trunk has auto record enabled, and a transfer made from that trunk is completed.
 - Fixed an issue with displaying GS Wave Web calls.
 - Fixed an issue where if a blind transfer destination is in a peer trunk, the destination CID name would be incorrect.
 - Fixed an issue with speed dial destinations being displayed incorrectly.
 - Fixed an issue where if a call participant has a CID number configured, the call would not be monitored.
- **[API Configuration]**
 - Fixed an issue where recordings located on NAS could not be downloaded via API.
 - Fixed an issue with being unable to retrieve fax files via CDR API.
- **[Backup/Restore]**
 - Fixed an issue where *Data Sync* → *Test Connection* would not create files in the configured file path.
- **[Basic Calls]**
 - Fixed an issue with *Use Callee DOD on FWD or RS* option not working properly.
 - Fixed an issue where if OPUS was selected as top priority, and if auto record was enabled, calls would drop immediately or after pickup.
 - Fixed an issue where if *Direct Media* is enabled, and a call is rejected, the caller will not hear the call failure tone.
 - Fixed an issue where UCM does not reply to 407 Proxy Authentication challenge message.
 - Fixed an issue where Call Forward Busy would not work if call waiting was disabled.

- Fixed an issue with one-way audio when the UCM passes along SDP with no media attribute.
- **[BLF Eventlist]**
 - Fixed an issue where subscriber information disappears after rebooting or reconnecting to the network.
- **[Call Completion]**
 - Fixed an issue where the call completion success prompt would play when it has actually failed.
- **[Conference]**
 - Fixed an issue with incorrect conference participant information when participant is blind transferred to another extension.
 - Fixed an issue where the conference management buttons would still be grayed out for an empty locked conference.
 - Fixed an issue with displaying the CID Name of external numbers.
 - Fixed an issue where conference room participant limit information would not be sent to endpoints that support CEI notifications.
 - Fixed an issue where if a user places the conference on hold, the user's talking indicator would be constantly displayed.
 - Fixed an issue with the mute button not responding after clicking on it several times.
 - Fixed an issue with batch operation buttons being inconsistently inoperable when several members are selected across different conferences.
- **[CDR]**
 - Fixed an issue with repeated unanswered calls and faxes not being displayed properly when the used inbound route has time conditions to exclude holidays.
 - Fixed an issue with being unable to play recordings created from Ring Simultaneously calls.
 - Fixed an issue where inviting external numbers to conferences would not generate CDR entries.
 - Fixed an issue with CDR displaying incorrectly when there are multiple call legs are involved.
 - Fixed several issues with downloading filtered CDR entries.
 - Fixed an issue with displaying inbound calls to fax destinations.
 - Fixed an issue with statistics not matching CDR information for previous months.
 - Fixed an issue where the *Delete All* button on the CDR page would not delete recordings located on NAS.
- **[CRM]**
 - Fixed an issue where if an incoming call's CID is modified by the inbound route configuration, the call would not be reported to the CRM.
 - Fixed an issue with missing description information in the CTI pop-up for incoming calls when using VitgerCRM.
- **[Email Settings]**

- Fixed an issue with deleting HTML message content.
- Fixed an issue with embedded images not being displayed correctly in emails sent using Microsoft Outlook.
- **[Fax]**
 - Fixed an issue where sorting by name or email address in the Fax/T.38 page did not work properly.
 - Fixed an issue with dadhi channels not being cleared properly after sending a T.38 fax via FXS.
 - Fixed an issue with not being able to upload PDFs with Arabic characters in the file name when using the Fax Sending feature.
- **[Feature Codes]**
 - Fixed an issue where the *Disconnect* feature code (*0 by default) does not work while transferring a call originating from a trunk.
- **[Grandstream Wave Web]**
 - Fixed several video display issues
 - Fixed an issue with displaying special characters in GS Wave-related emails.
 - Fixed an issue with valid scheduling time slots being considered conflicting.
 - Fixed some Firefox compatibility issues with Firefox.
 - Fixed an issue with being unable to display conference participant information when they are using GXP21xx devices.
 - Fixed an issue with line breaks not working properly in video conference chat.
 - Fixed an issue with log names having the incorrect creation time.
 - Fixed an issue with *Login Timeout* setting in the web portal affecting GS Wave Web.
 - Fixed an issue with being unable to use the microphone and camera after dialing into a disabled video conference.
 - Fixed an issue with not being able to receive chat messages in certain cases.
 - Fixed an issue with where video conference extensions could not be configured as default destination in the *Inbound Route* pages for analog trunks and register SIP trunks.
 - Fixed an issue where the conference host would not have 1080p video if using Google Chrome.
 - Fixed an issue with delays with sending emails regarding video conferences.
 - Fixed an issue where calls will drop when starting a video call in the middle of an audio call with Mozilla Firefox browser.
- **[HTTP Server]**
 - Fixed an issue where the HTTP server port could not be set to 65535.
- **[IAX]**
 - *Max Call Numbers* and *Max Invalidated Call Numbers* fields are now required fields with 0 as default value (this was already the value in previous firmware).
- **[Jitter Buffer]**
 - Fixed an issue where jitter buffer settings did not apply to SCA calls.
- **[LDAP]**

- Fixed an issue with loading loops when the LDAP port is set to certain values.
- Fixed an issue with LDAP syncing with peer trunks when the peer trunk name contains a space.
- Fixed an issue with syncing when both peered UCMs are behind NATs.
- **[Music on Hold]**
 - Fixed an issue where uploaded archive files are not uploaded to the correct playlist.
- **[NAS]**
 - Fixed an issue with reestablishing NAS connection after failover.
- **[Paging]**
 - Fixed an issue with prompt file names not being case-sensitive, causing conflicts between uppercase and lowercase versions of the same name.
 - Fixed an issue where the recording prompt would play during multicast pages under certain circumstances.
 - Fixed an issue with ports being blocked when a paging call originator makes an attended transfer.
- **[Parking]**
 - Fixed an issue where if a caller dials into an empty parking lot extension, and if there is a park timeout, the call will drop on timeout instead of calling back the caller.
- **[PMS]**
 - Fixed an issue with being able change to rooms that are in repair or are being cleaned.
- **[Queue]**
 - Fixed an issue where the queue chairman could not terminate calls in virtual queue.
 - Fixed an issue with seamless transfers using the wrong outbound route and failing.
 - Fixed an issue with incorrect available agent count when agent extensions are not registered.
 - Fixed an issue where if two calls come into the queue simultaneously and to the same agent, one of the calls will be logged as a missed call.
- **[Recording]**
 - Fixed an issue with recording files not being created for DISA calls through trunks and call queues.
 - Fixed an issue with VoIP trunk auto call recordings not being displayed in the *CDR* page.
 - Fixed an issue with creating different recordings having the same audio if an attended transfer is completed before the transfer destination answers.
- **[Routing]**
 - Fixed an issue with outbound route's *Call Duration Limit* value not applying to seamless transfers.
 - Fixed an issue with inbound mode BLF status light not updating when using feature codes to change the inbound route mode and ending the call before the voice prompt finishes playing.
- **[SCA]**
 - Fixed an issue with incorrect eventlist status when SCA private number is ringing.

- **[Setup Wizard]**
 - Fixed an issue with a time zone not displaying properly in the list.
 - Fixed an issue with repeating reboot prompts after saving.
- **[SIP Settings]**
 - Fixed an issue where the *Enable Diversion Header* option was not working.
- **[Speed Dial]**
 - Fixed an issue where speed dials using 1000 as the rigger will fail.
- **[System Events]**
 - Fixed an issue generating duplicate alert log entries after call failure 488 errors.
 - Fixed an issue with usernames with special characters being displaying incorrectly in alert logs.
 - Fixed an issue with sending out notifications via HTTPS.
- **[Time Settings]**
 - Fixed an issue with 12:00 – 1:00 PM displaying as AM instead when using 12-hour time display format.
 - Fixed an issue where daylight savings was not applied to Adelaide, Australia, and New Zealand time zones.
- **[Trunk Groups]**
 - Fixed an issue where entering "--" into the *CallerID Name* field would result in the trunk group entry being deleted.
- **[Voicemail]**
 - Fixed an issue where the date/time format in emails would not be correct.
 - Fixed an issue where voicemail for voicemail groups and ring groups would not be deleted even when *Keep Voicemail after Emailing* is disabled.
 - Fixed an issue where batch resetting extensions would not reset the voicemail password.
- **[Voice Prompts]**
 - Fixed an issue where a NOTIFY won't be sent immediately after changing voicemail type.
 - Fixed an issue with some prompts playing in English instead of Hebrew when the Hebrew voice prompt package is selected.
- **[VoIP Trunks]**
 - Fixed an issue where the *From Domain* setting does not work properly until the trunk is disabled and re-enabled.
- **[Web]**
 - Corrected and modified several tooltips.
 - Fixed an issue where Internet Explorer could not access the UCM web portal.
 - Fixed an issue where users would be logged out while changing password even if the old password was wrong.
 - Fixed an issue where IAX and fax extensions could not see calls on their user portal switchboard.
 - Fixed an issue with being unable to reboot devices from the user portal without the Zero Config permission.

- Fixed an issue where after adding and saving a server address in the *STUN Server Address* field, the server address would be between two brackets [].
- Fixed an issue where after uploading a custom prompt, it would not appear in the prompt dropdown lists.
- Fixed an issue with sorting not working properly on the LDAP phonebook page.
- Fixed an issue with user portal not displaying the correct amount of voicemail.
- Fixed an issue where the *Recordings* page could not refresh the file list after plugging and unplugging an external device several times.
- **[ZeroConfig]**
 - Fixed an issue with custom field descriptions not showing in previews.
 - Fixed an issue with being not being able to provision admin password to GDS3710
 - Fixed an issue with HT813 templates not having account setting fields.
 - Fixed an issue with configured p-values getting deleted if description contains certain words.
 - Fixed an issue with being unable to upload files to a directory with Chinese characters.
- **[UCM6510] [HA]**
 - Fixed an issue with template package updates not syncing properly.
 - Fixed an issue with syncing deleted PMS voicemail.

NEW LIMITATIONS

- [AMI] AMI username can no longer exceed 64 characters.
- [Announcement Center] Announcement Center group name and numbers now have a character limit of 64.
- [Call Completion] The number of CC requests a SIP extension can make is now limited by its concurrent registration setting.
- [Conference] Conference extension number can no longer be used as the conference password if *Strong Password* is enabled.
- [Conference] Conferences can now be scheduled to last 8 hours.
- [DOD] DOD names and numbers now have a character limit of 32
- [Extensions] *Extension* and *Auth ID* fields now support plus signs (+).
- [LDAP] Plus signs (+) are now supported when creating contacts and receiving calls with "+" in the CID.
- [LDAP] Character limit changes for the following fields:
 - LDAP Server→Root Password: 32
 - LDAP Server→Root DN: 64

- Phonebook Download Configurations→User Name: 64
- [LDAP] The *User Name* field will no longer require "cn=" to precede username.
- [Paging] "=" is no longer supported in the Alert-Info Header field.
- [Paging] Failover Destination field now has a character limit of 32.
- [Queue] Callers can no longer use feature codes in established callbacks.
- [Ring Group] The character limit of the email address fields for ring group voicemail boxes has been changed from 256 to 128.
- [Routing] Outbound routes is now limited to 50 routes.
- [Routing] Inbound routes is now limited to 50 routes per trunk.
- [Voicemail] Voicemail group member limit increased from 16 to 27.
- [Voice Prompt] Voice prompt package upload file size limit increased to 50MB.
- [Voice Prompt] Custom voice prompt file name character limits for the Voice Prompt and Announcement Center pages have been changed to 100.
- [VoIP Trunks] PAI headers now have a character limit of 64 and can only contain alphanumeric characters and/or special characters #*-_+.

KNOWN ISSUES

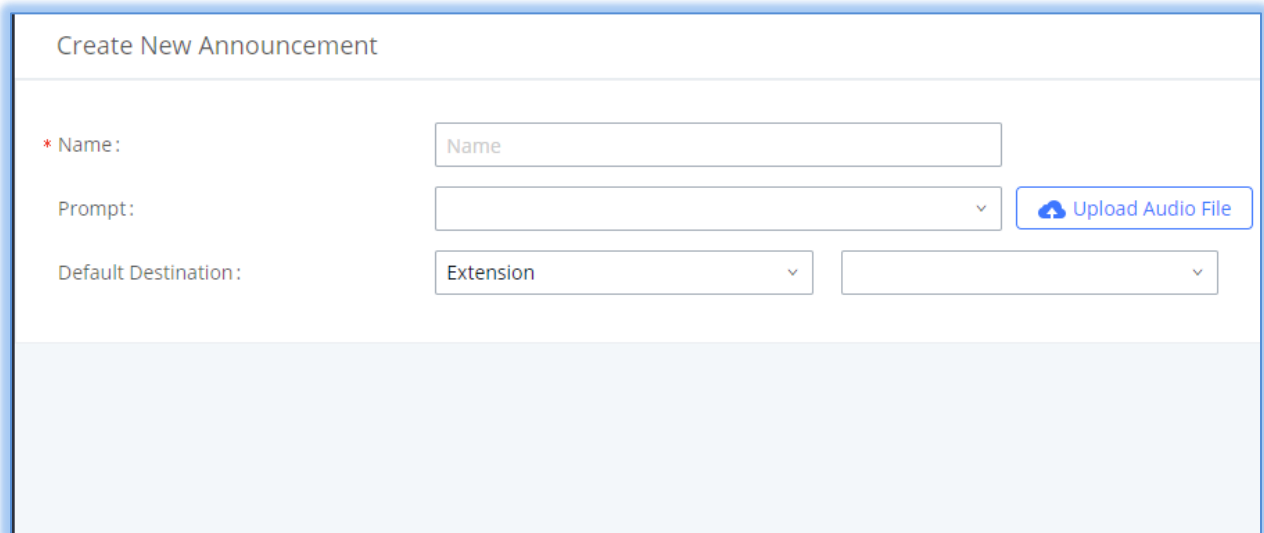
- [GDMS] The GDMS password field is unable to accept many special characters.
- [Grandstream Wave Web] If BFCP TCP is used for video conferencing, crashing may occur when toggling video and screen sharing.
- [Routing] If an exported inbound route has the Alert-Info field configured with a value other than "None" or "Custom", it cannot be imported.
- [System] Crashing may occur if *PBX Settings*→*SIP Settings*→*Misc*→*Enable Use of Final SDP* is toggled on.
- [System] Slight chance of crashing when loading or reloading PJSIP extensions (e.g., applying changes to extensions).
- [ZeroConfig] If a template has custom parameters with names containing the character "P", the values of these parameters may be wiped after upgrading from 1.0.19.29 to 1.0.20.17 and saving the template page.

NEW FEATURES OVERVIEW

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

ANNOUNCEMENT

The Announcement feature (not to be confused with Announcement Paging and Announcement Center) is a feature that allows users to set an unskippable audio file to play to callers before routing them to a configured destination. Announcements can be configured as a destination in the *Inbound Routes* page.



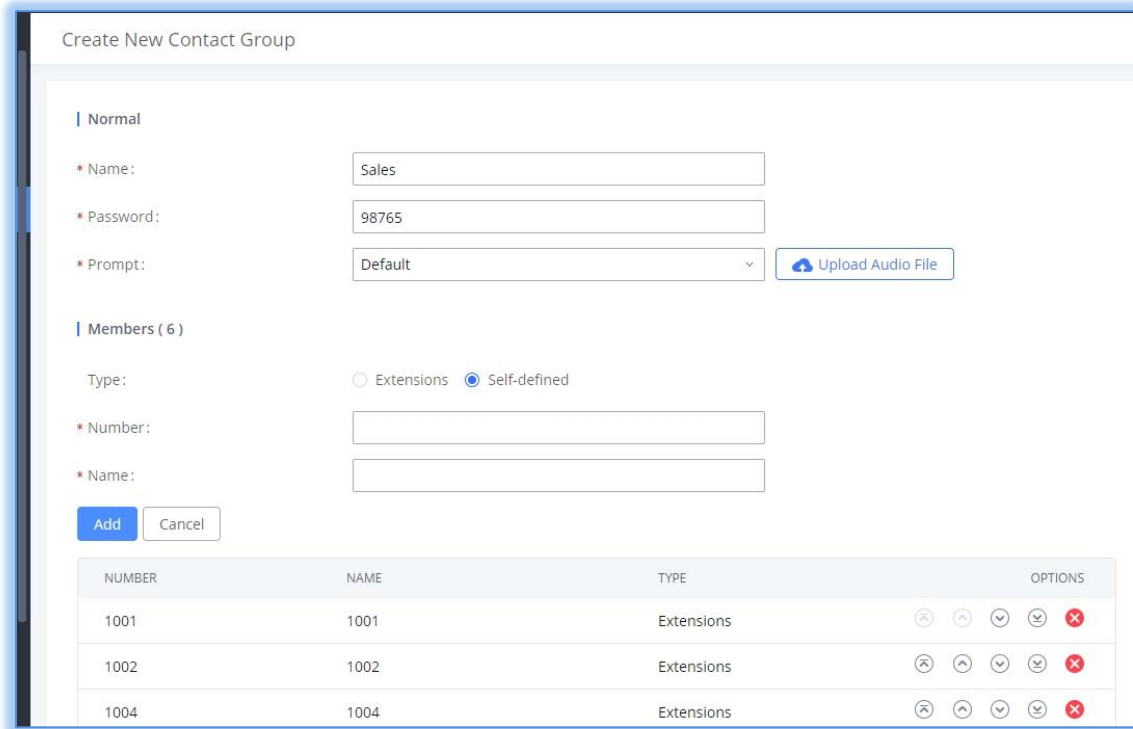
The screenshot shows a web form titled "Create New Announcement". It contains the following fields and controls:

- Name:** A required text input field with the placeholder "Name".
- Prompt:** A dropdown menu with a downward arrow.
- Default Destination:** A dropdown menu with the placeholder "Extension" and a downward arrow.
- Upload Audio File:** A button with a cloud upload icon and the text "Upload Audio File".

Figure 1 - Announcement page

CONFERENCE CONTACT GROUPS

Users can now quickly invite multiple participants at once to a conference via conference contact groups. Up to 5 contact groups can be created. The maximum allowed number of contacts per group is based on the UCM model's conference participant limit: 25 for 6202/6204, 32 for 6208, 64 for 6510.



Create New Contact Group

Normal

* Name:

* Password:

* Prompt:

Members (6)

Type: Extensions Self-defined

* Number:

* Name:

NUMBER	NAME	TYPE	OPTIONS
1001	1001	Extensions	<input type="button" value="Up"/> <input type="button" value="Down"/> <input type="button" value="Refresh"/> <input type="button" value="Delete"/>
1002	1002	Extensions	<input type="button" value="Up"/> <input type="button" value="Down"/> <input type="button" value="Refresh"/> <input type="button" value="Delete"/>
1004	1004	Extensions	<input type="button" value="Up"/> <input type="button" value="Down"/> <input type="button" value="Refresh"/> <input type="button" value="Delete"/>

Figure 2 - Contact Group

Each contact group must have a password configured, which will be required when inviting the specified contact group to a conference. Additionally, an audio file can be uploaded to each group to be used to announce the contact group name such as “Sales” or “Marketing”. The default announcement for each group is “Conference Contact Group 1”, “Conference Contact Group 2”, etc.

CONFERENCE REPORTS

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

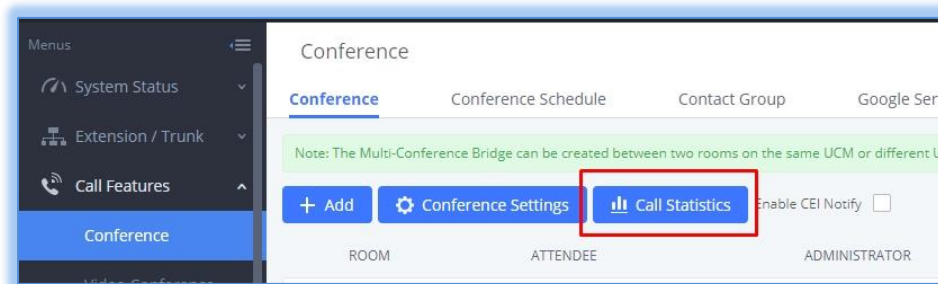


Figure 3 - Call Statistics

< Call Statistics

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

CONFERENCE	START TIME	TIME
6301	2019-11-07 16:12:16	00:01:16
1001 Name	1001 Number	Received Admission Mode Answered State
Sales Name	-- Number	Invitation Admission Mode 0 / 4 State
Conference invitation Name	1002 Number	Invitation Admission Mode Failed State
Conference invitation Name	1005 Number	Invitation Admission Mode Failed State

Figure 4 - Conference Report on Web

	A	B	C	D	E	F	G
1	Room	Start Time	Duration Time				
2	6301	11/7/2019 16:12	0:01:16				
3	Contact Number	Name	Way	Status	Contact Group Name		
4	1002	Conference invitatio	INVITE	FAILURE	Sales		
5	1005	Conference invitatio	INVITE	FAILURE	Sales		
6	1004	Conference invitatio	INVITE	FAILURE	Sales		
7	1003	Conference invitatio	INVITE	FAILURE	Sales		
8	1001	1001	CALLIN	ANSWER			
9							

Figure 5 - Conference Report on CSV

CALL FORWARDING REMOTE MANAGEMENT

Users can now remotely change the call forwarding settings of another extension via feature codes. Additionally, a whitelist can be configured to restrict this ability to specific users.

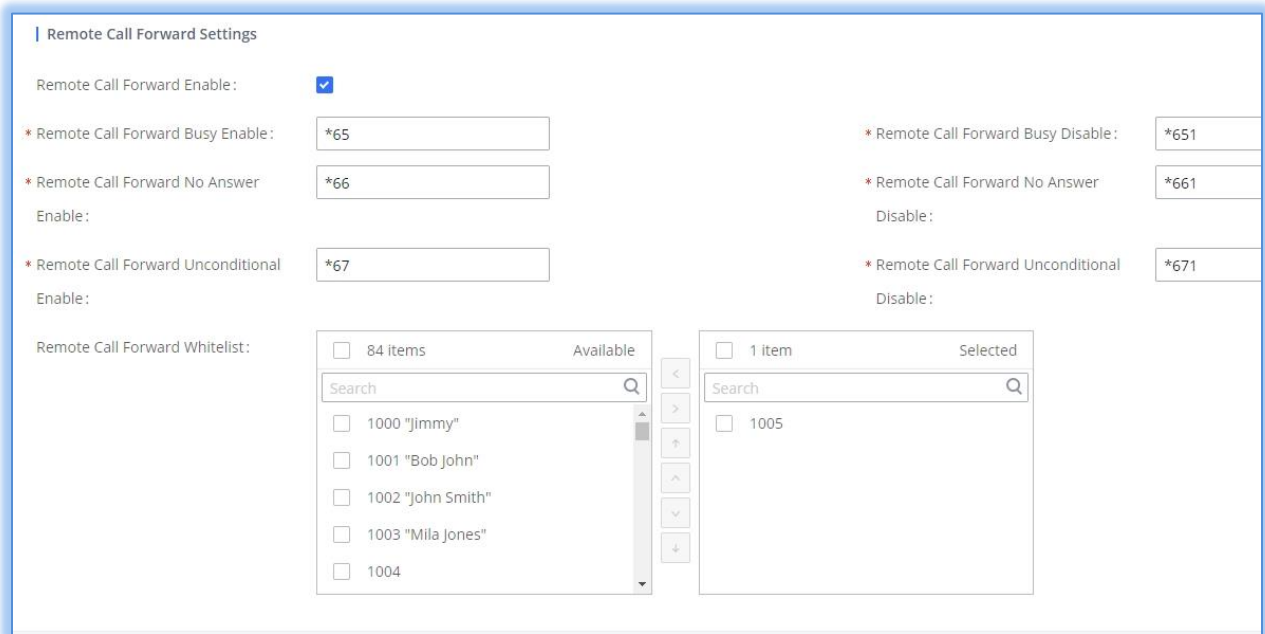


Figure 6 - Remote Call Forwarding Feature Codes

GDMS SIP ACCOUNT SYNCING

UCM can now sync its SIP accounts to GDMS cloud management system. To do get started, log into your GDMS account and navigate to the *System* → *API Developer* page. Click on the *Enable API Developer Mode* button if it has not been enabled yet.

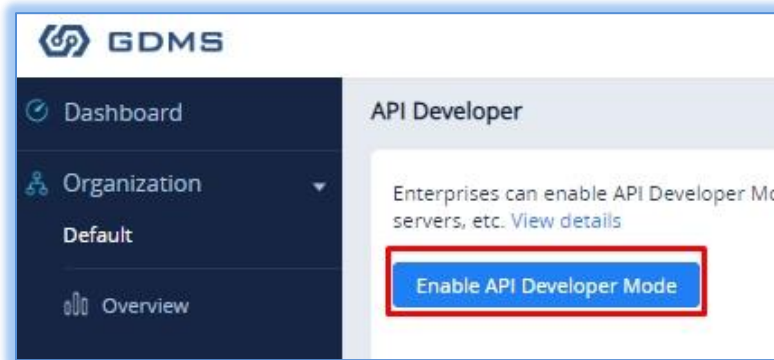


Figure 7 - GDMS Developer Mode Button

After clicking on it, the API ID and Secret Key will be displayed. Note down these credentials.

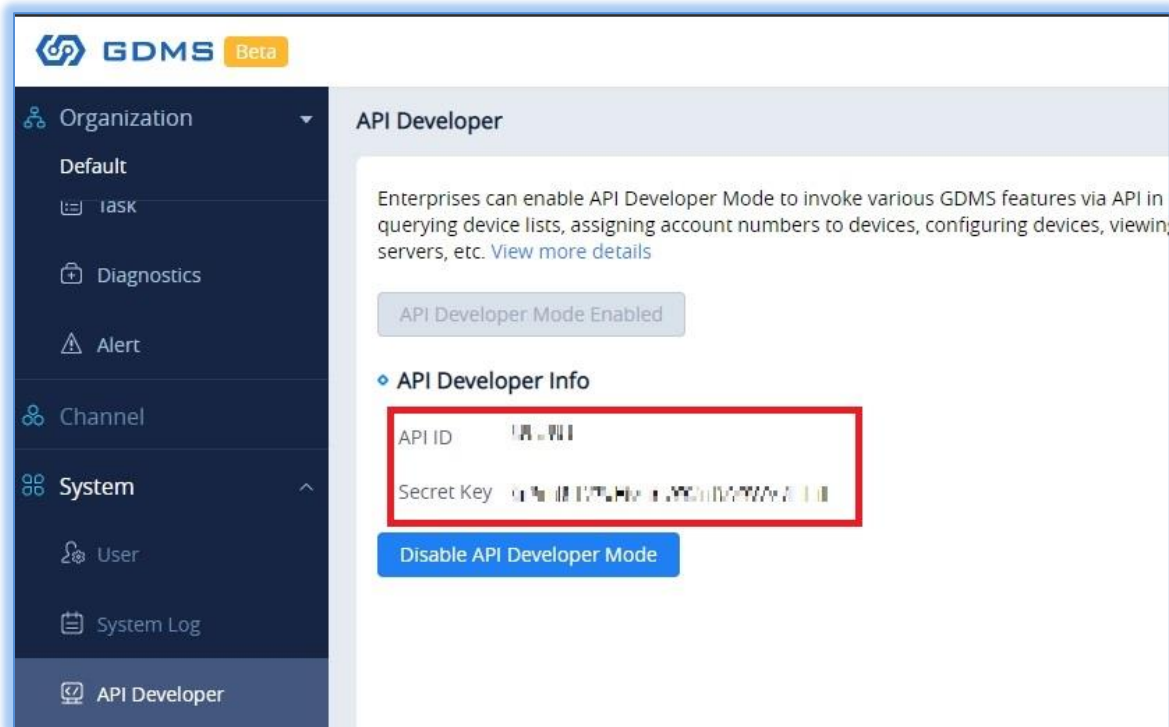


Figure 8- GDMS API Credentials

On the UCM, navigate to *System Settings* → *GDMS Settings*. Check the *Enable* option if it has not been toggled on.

GDMS Settings

UCM supports syncing all SIP extensions to GDMS Cloud Platform. After authentication, select the organization to sync extensions to. This function

Enable:

* Server Region:

* Account Email/Username:

* Password:

* API ID:

* Secret Key:

Connection Status: No authentication

Organization:

Authenticate

Figure 9 - GDMS Settings

Enter your GDMS account credentials and the API developer credentials. For the *Account* field, enter either your account username or email address. Once all the information has been entered, click on the *Authenticate* button to connect to GDMS. If successful, you will now be able to select an organization on GDMS to sync all UCM SIP accounts to. After selecting, saving and applying changes, the UCM will then start the syncing process. Once the initial sync is complete, you will now be able to see the UCM and its extensions on GDMS, which will have an orange label with the words “UCM”. Any future creation, deletion, or modification of SIP accounts will automatically be synced to GDMS.

SIP Server

	Server Name		Server Address
<input type="checkbox"/>	192.168.42.63 UCM		192.168.42.63

Total 1

Figure 10 - UCM on GDMS

SIP Account

Delete

<input type="checkbox"/>	User ID ↕	Account Name ↕	Display Name ↕	SIP Server ↕
<input type="checkbox"/>	1000 UCM	1000	1000	192.168.42.63
<input type="checkbox"/>	1001 UCM	1001	1001	192.168.42.63
<input type="checkbox"/>	1002 UCM	1002	1002	192.168.42.63
<input type="checkbox"/>	1003 UCM	1003	1003	192.168.42.63

Figure 11 - UCM SIP Extensions on GDMS

Any future creation, modification, and deletion of the UCM's SIP extensions will automatically be synced to GDMS.

Note: UCM extensions' names currently cannot be synced.

NEW HTTP API

A new HTTP API has been added and is slated to replace the previous HTTP API. The previous HTTP API will still be available in 1.0.20.x firmware but will no longer be available in the near future. Please see the HTTP API Configuration Guide on our website for more details on usage.

API Configuration

< **HTTPS API Settings(New)** HTTPS API Settings(Old) CDR Real-time

The UCM provides a second API interface to query, edit PBX settings and implement multiple call f other server. Additionally, legacy CDR API, REC API and PMS API are supported. Please see the use

Enable:

* Username:

* Password:

Call Control:

Figure 12 - HTTPS API

NEW CLEANING CONDITIONS

The CDR and file cleaner now offers more conditions to allow more control over when cleaning takes place. The following options have been added to the cleaners for CDR, queue statistics reports, and conference call statistics reports:

By Schedule – This was the only available condition in previous firmware. CDR and reports are deleted every X days at the configured *Clean Time*.

Keep Last X Records – Keep the last X entries and delete everything else at the configured *Clean Time*.

Keep Last X Days – Keep the last X days of entries and delete everything older than X days at the configured *Clean Time*.

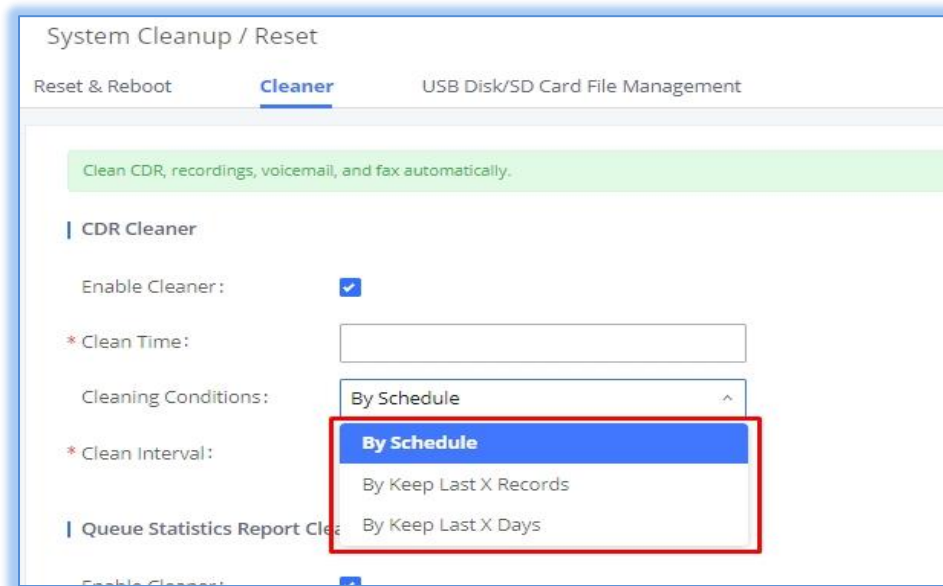


Figure 13 - CDR and Statistics Report Cleaning Options

The following options have been added to the file cleaner:

By Threshold – Delete files once internal storage usage has reached the percentage configured in the *File Clean Threshold* field. If this option is selected, the *Keep Last X Days* option will also be available to allow users to delete only files older than the configured number of days. If unconfigured, all files will be deleted at the configured *Clean Time* upon reaching the threshold.

Keep Last X Days – Delete all files older than the number of days configured in the *Keep Last X Days* field at the configured *Clean Time*.

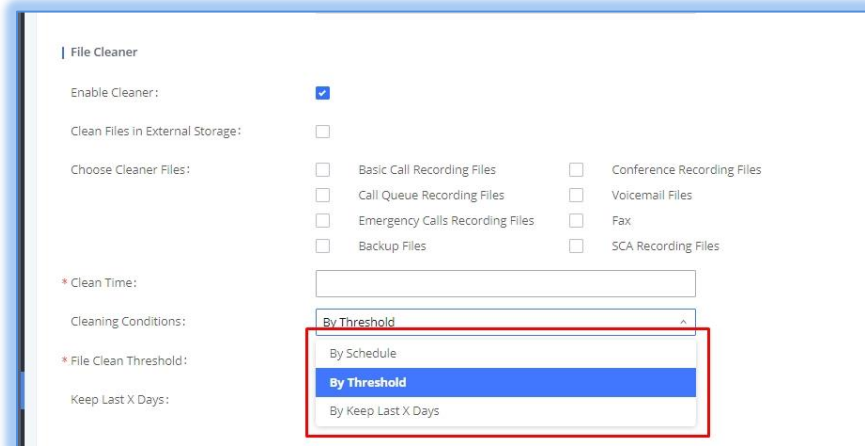


Figure 14 - File Cleaning Options

VIRTUAL QUEUE KEY CUSTOMIZATION

Users can now customize the keys for virtual queue options. These options are found in the new *Global Queue Settings* page alongside dynamic agent login settings.

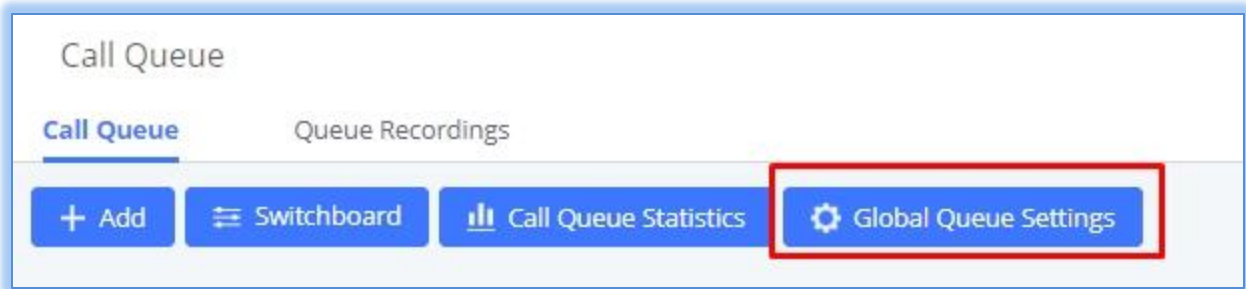


Figure 15 - Global Queue Settings

Global Queue Settings

Dynamic Agent Login Settings

Agent Login Code Suffix:

Agent Logout Code Suffix:

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

Virtual Queue Callback Key Settings

* Call Back Current Number:

* Custom Callback Number:

* Continue Waiting:

Figure 16 - Virtual Queue Key Customization

QUEUE STATISTICS IMPROVEMENTS

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

< Call Queue Statistics

[Download](#)
[Reset Statistics](#)
[Automatic Download](#)

[Overview](#)
[Agent Details](#)
[Login Record](#)
[Pause Log](#)

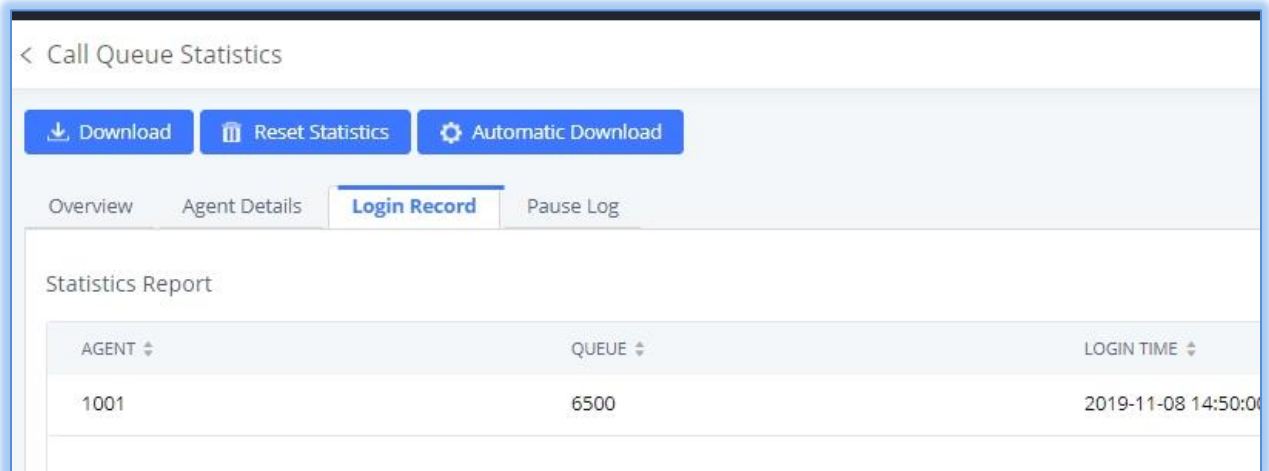
Statistics Report

TIME ↕	AGENT ↕	QUEUE ↕	CALLER ID NU
2019-11-08 14:40:27	1001	6500	1002
2019-11-08 14:41:07	1001	6500	1002
2019-11-08 14:41:37	1001	6500	1002
2019-11-08 14:41:46	1001	6500	1002
2019-11-08 14:45:25	1001	6500	1002
2019-11-08 14:51:38	1001	6500	1002

Figure 17 - Agent Details

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

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



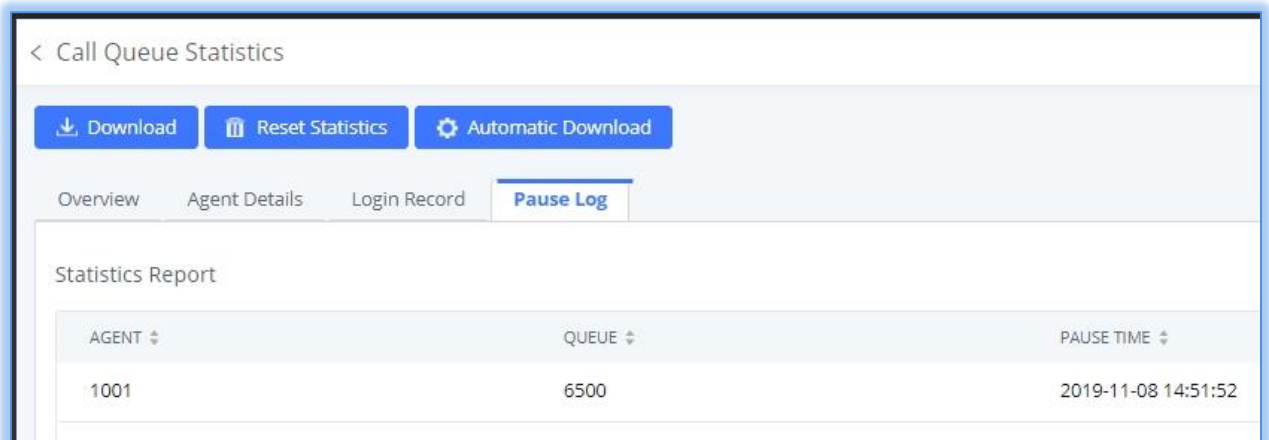
The screenshot shows the 'Call Queue Statistics' interface. At the top, there are three buttons: 'Download', 'Reset Statistics', and 'Automatic Download'. Below these are four tabs: 'Overview', 'Agent Details', 'Login Record' (which is selected), and 'Pause Log'. The main content area is titled 'Statistics Report' and contains a table with the following data:

AGENT ↕	QUEUE ↕	LOGIN TIME ↕
1001	6500	2019-11-08 14:50:00

Figure 18 - Login Record

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

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



The screenshot shows the 'Call Queue Statistics' interface. At the top, there are three buttons: 'Download', 'Reset Statistics', and 'Automatic Download'. Below these are four tabs: 'Overview', 'Agent Details', 'Login Record', and 'Pause Log' (which is selected). The main content area is titled 'Statistics Report' and contains a table with the following data:

AGENT ↕	QUEUE ↕	PAUSE TIME ↕
1001	6500	2019-11-08 14:51:52

Figure 19 - Pause Log

Pause Log is a report that shows the times of agent pauses and unpauses and calculates the amount of time that agents are paused. If an agent is part of several queues, an entry will be created for each queue.

An entry will only be created after an agent unpauses. The following information is available:

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

QUEUOMETRICS SUPPORT

UCM now supports QueueMetrics. To get started on using this, navigate to the **Value-added Features**→**QueueMetrics** page and enter your QueueMetrics server address and credentials. Please see the QueueMetrics Configuration documentation on our website for more details on usage.

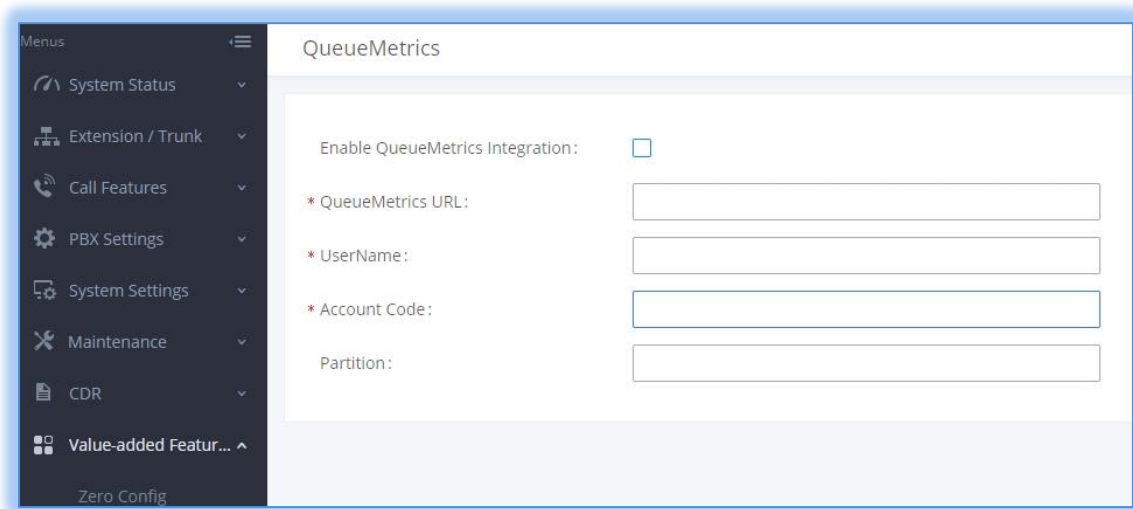


Figure 20- QueueMetrics Page

PRIVATE INTERCOM (GSC3510 ONLY)

Private Intercom is a new paging type that is meant to be used with Grandstream GSC3510. In a private intercom:

- The initiator can be heard by all parties
- The initiator can hear only one of the intercom members, which is determined by whose audio is initially detected. Audio from other members cannot be heard until the first responder is done talking.
- Intercom members can hear only the initiator's audio and not other intercom members'.

Create New Paging/Intercom Groups

* Name:

* Type: ^

* Extension:

Replace Display Name:

* Maximum Call Duration.:

Custom Prompt:

Members: 84 items Available 0 item

Figure 21 - Private Intercom option

PMS VOICMAIL BACKUP ON CHECK-OUT

Users now have the option to enable automatic backup of voicemail recordings from room extensions upon guest check-out. Voicemail can be backed up to SD card, USB disk, NAS, or an SFTP server. Additionally, users can configure an email address to send voicemail recordings to upon backup.

PMS

Basic Settings Room Status Wakeup Service Mini Bar Maid

PMS Module: v

Wakeup Prompt: v

* Username:

* Password:

Back Up Voicemail Recordings:

SD Card USB Disk NAS SFTP Server

Email Address:

Figure 22 - PMS Voicemail Backup

CALLING INTO IPVT MEETINGS

UCM extensions can now dial into IPVT meetings by creating a peer trunk to an IPVT server. However, users must make sure that the IPVT server they are peering to also has a peer trunk to their UCM configured. This setting can be found in *Admin Center* → *SIP Trunk Configuration*.

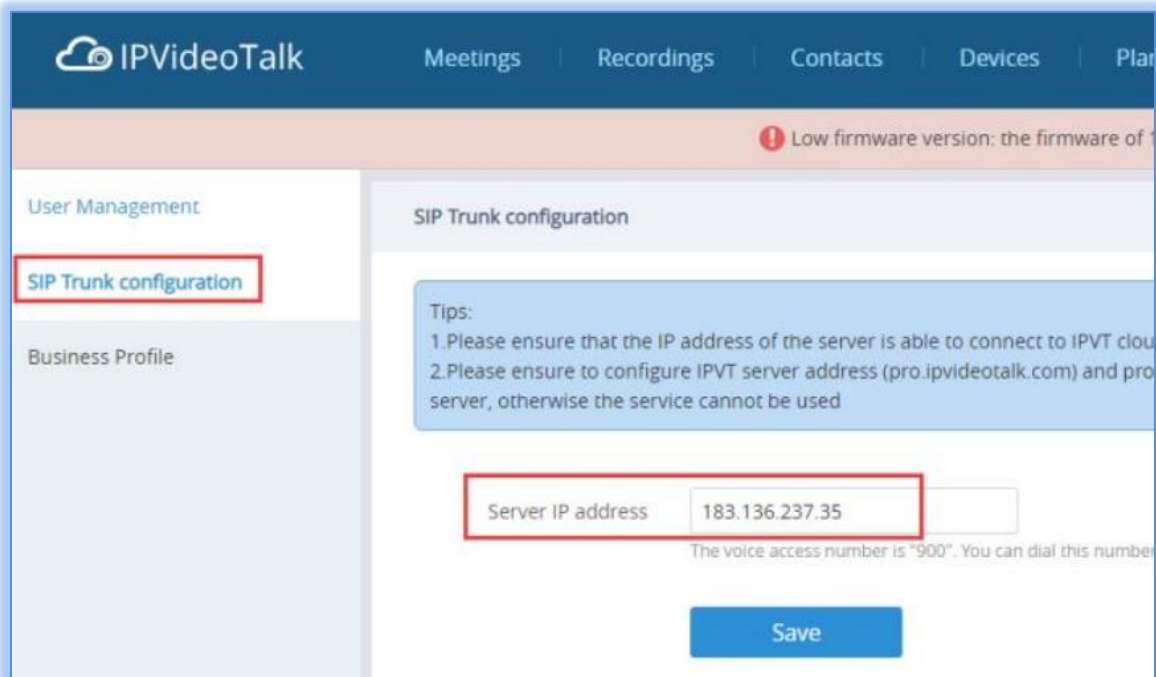


Figure 23 - IPVT SIP Trunk page

Next, users must create a peer trunk on the UCM to the IPVT server. Enter one of the following addresses based on the desired connection protocol:

- TCP: pro.ipvideotalk.com:20000
- TLS: pro.ipvideotalk.com:20001

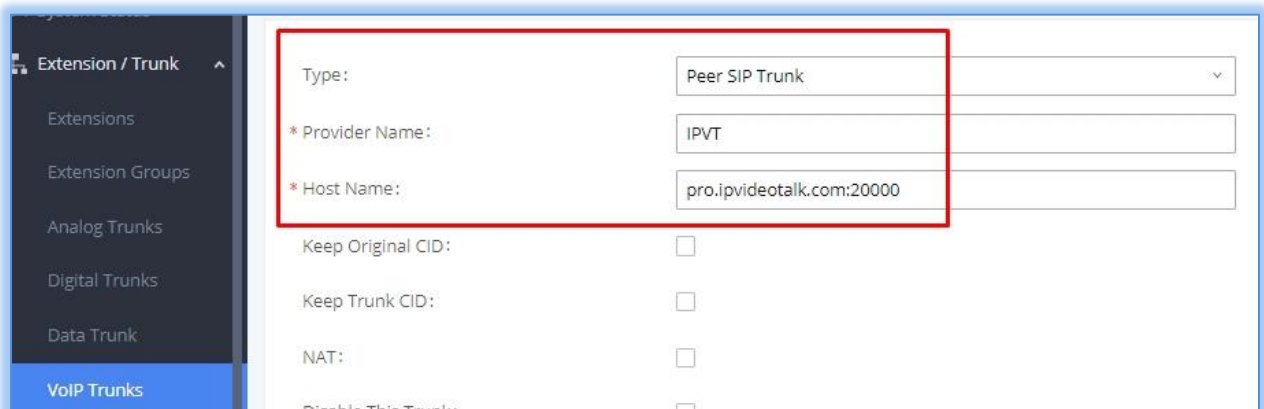


Figure 24 - Peer Trunk to IPVT

Make sure that the *Transport* field is either “TCP” or “TLS”. Save and apply changes to create the trunk.

Next, edit the newly created trunk and click on the *Advanced Settings* tab. Make sure the *IPVT Mode* option is checked. Otherwise, you may experience audio issues when dialing into IPVT.

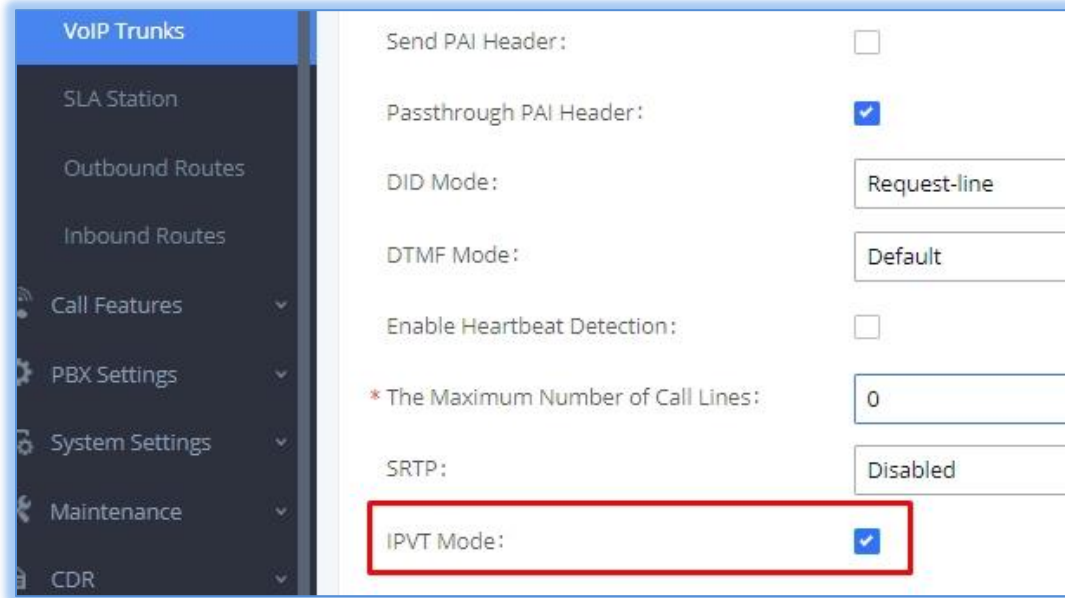


Figure 25 - IPVT Mode

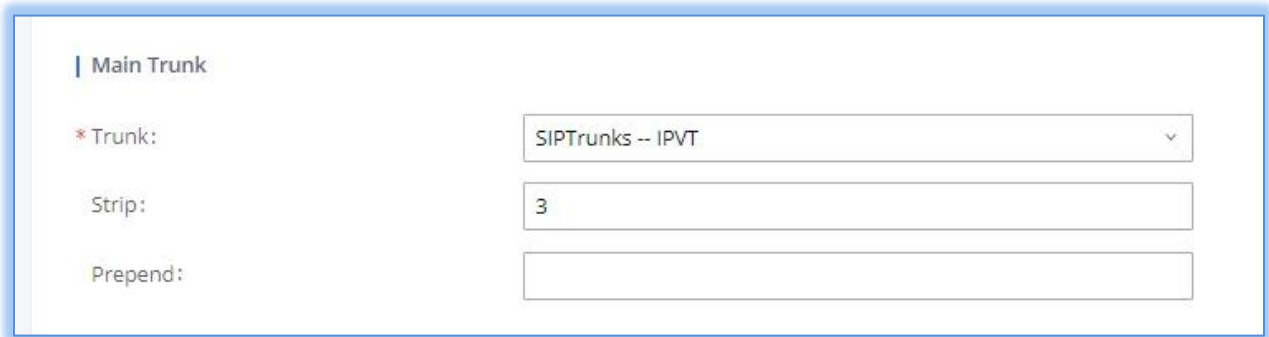
Finally, create an outbound route for this trunk. This route will be used to dial IPVT meeting IDs. Due to IPVT meeting IDs having a random assortment of numbers, it is recommended to use a unique code to precede the meeting ID so that UCM can direct calls to the IPVT trunk without fail (e.g., *99). In the below image, “x.” would be the meeting ID.



Figure 26 - IPVT Outbound Pattern

However, if a unique code is used, users must also configure the *Strip* field to remove the unique code from

the meeting ID before the call is sent to IPVT.



| Main Trunk

* Trunk: SIPTrunks -- IPVT

Strip: 3

Prepend:

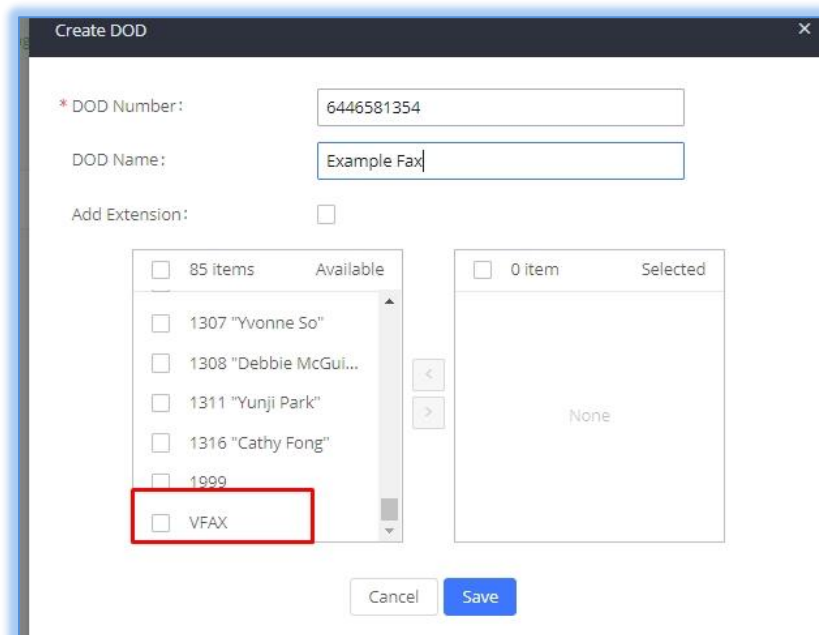
Figure 27 - IPVT Outbound Strip

In this example, the *Strip* field has “3” configured to remove the example unique code *99 from the dialed number before the call is routed out to the IPVT server. Once this outbound route has been created, users can now use a UCM extension to dial IPVT meeting rooms.

Note: An IPVT account can have only 1 SIP trunk peered to it.

ASSIGNING DOD TO FAX SENDING

DOD can now be assigned to the UCM's **Fax Sending** feature. To do this, select “VFAX” from the extension list when creating or editing a DOD number.



Create DOD

* DOD Number: 6446581354

DOD Name: Example Fax

Add Extension:

Available	Selected
<input type="checkbox"/> 85 items <input type="checkbox"/> 1307 "Yvonne So" <input type="checkbox"/> 1308 "Debbie McGul..." <input type="checkbox"/> 1311 "Yunji Park" <input type="checkbox"/> 1316 "Cathy Fong" <input type="checkbox"/> 1999 <input type="checkbox"/> VFAX	<input type="checkbox"/> 0 item None

Cancel Save

Figure 28 - Fax Sending DOD