UCM Series IP PBX Firmware Release Notes

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FIRMWARE VERSION 1.0.5.4

PRODUCT NAME

UCM6301, UCM6302, UCM6304, UCM6308

DATE

Beta: 02/10/2021

Official: 03/08/2021

FIRMWARE FILE INFORMATION

- UCM6301/6302 firmware file name: ucm6302fw.bin

  MD5: eecc87ff5744069e15df6c08818a349d

- UCM6304/6308 firmware file name: ucm6308fw.bin

  MD5: f59788ad241b9bd70cab9d19c8be495e

IMPORTANT UPGRADING NOTE

- **ALWAYS create a backup of your configuration and data before a firmware upgrade.**
- **After upgrading to 1.0.2.18 or higher, you will no longer be able to downgrade to 1.0.2.17 or lower.**
- **If the device is on a firmware version lower than 1.0.2.15, please upgrade to 1.0.2.15/16/17 first and then to the latest version.**
  - **UCM6301/2:** http://firmware.grandstream.com/Release_UCM6301_6302_1.0.2.17.zip
  - **UCM6304/8:** http://firmware.grandstream.com/Release_UCM6304_6308_1.0.2.17.zip
- **It is recommended to upgrade UCM to latest firmware for product lifespan and security improvements.**

CHANGES SINCE FIRMWARE VERSION 1.0.3.10

ENHANCEMENTS

- **[CDR]** NAT option has been added to the Export File Data filtering option.
- **[Conference]** Improved the organization of the audio and video conference pages. A list of the meetings that have not started and a meeting history list have been added.

- **[Conference]** Added post-meeting reports with participant statistics. This report will be emailed to the conference moderator after the end of a meeting. Note: There is currently no option to disable this.
- **[Conference]** Improved CEI subscription functionality with Wave.
- **[Fax]** Restored Fax support (i.e., T.38 fax related options, Fax Sending, Email2Fax).
- **[IAX]** Restored IAX support.
- **[GDMS]** UCM can now sync system event alerts to GDMS.
- **[Maintenance]** Added the *SRTP Debugging* option to the *Ethernet Capture* page. This option is for acquiring additional files needed for troubleshooting calls encrypted with SRTP. [SRTP TROUBLESHOOTING]
- **[Maintenance]** Added ability to restore backups from the GDMS interface.
- **[Maintenance]** Updated the *System Cleanup/Reset* page interface.
- **[RemoteConnect]** Added support for GDMS call quality monitoring. [REMOTECONNECT CALL QUALITY MONITORING]
- **[SIP Settings]** Added STIR/SHAKEN support. [STIR/SHAKEN SUPPORT]
- **[System Settings]** Added ability to restrict calls and features based on CPU usage and data partition usage. [THRESHOLD-BASED CALL CONTROL & DATA WRITE CONTROL]
- **[Wave Desktop]** Wave desktop client is now supported. [WAVE DESKTOP SUPPORT]
- **[Wave Mobile]** Improved push notification support.
- **[Wave Web]** Added Instant Messaging functionality to Wave Web. [WAVE WEB INSTANT MESSAGING (BETA)]
- **[Wave Web]** LDAP phonebook information will now be synced when viewing the Contacts page.
- **[Web]** Added ability to customize certain parts of the UCM web interface. [WEB PORTAL LOGO CUSTOMIZATION]
- **[ZeroConfig]** Added *Layer 3 QoS for SIP* and *Layer 3 QoS for RTP* options to global policy and relevant templates.
- **[ZeroConfig]** Improved device list import support and added the ability to export devices in the ZC Device List page. [ZERO CONFIG DEVICE LIST EXPORT SUPPORT AND IMPROVED IMPORT SUPPORT]

BUG FIXES

- **[System]**
  - Fixed several system stability issues.
- **[Conference]**
  - Fixed an issue with audio conference status always being displayed as "Not Started" if the conference subject contains an apostrophe (').
  - Fixed an issue with LDAP extensions not ringing when receiving the 2nd conference reminder call.
- **[CDR]**
  - Fixed several CDR display issues.
  - Fixed an issue with External Calls being counted incorrectly.

- Fixed an issue with filtering by Announcement action type.
- Fixed an issue where the play button for call recordings doesn't appear.
- Fixed an issue with the NAT values not appearing properly in the CDR CSV report.

- **[Event List]**
  - Fixed an issue with extension status not being displayed properly in certain cases.
  - Fixed an issue with presence status of imported extensions not being synced to event lists properly.
  - Fixed several display issues.
- **[Extensions]**
  - Fixed an issue with saving and applying changes after changing presence status to "Custom".
  - Fixed an issue with updating existing FXS extensions via CSV file import.
  - Fixed an issue where FXS/IAX extensions could not display DND status.
  - Fixed an issue where a rejected call will be forwarded to the extension's Ring Simultaneously number and Follow Me number.
  - Fixed an issue with the DND whitelist not working properly after adding an extension.
- **[Extension Group]**
  - Fixed an issue with being unable to add FXS extensions to an extension group.
- **[Feature Code]**
  - Fixed an issue with seamless transfer failure if the used outbound route has the Strip field configured.
- **[FXO/FXS]**
  - Fixed an issue with PSTN Auto Detect not working properly.
  - Fixed an issue with calls not being released properly after FXO ports lost power.
- **[GDMS]**
  - Fixed an issue where the extension numbers of devices would be incorrect if registered via TCP/TLS.
  - Fixed an issue with incorrectly counting different types of calls.
- **[HTTPS API]**
  - Fixed an issue with responses to the updateIAXTrunk command.
- **[IAX]**
  - Fixed an issue with batch adding extensions if the CID Number field had "e" as the value.
  - [IAX] Fixed an issue with hearing a second dial tone after a ringback tone when dialing to an IVR via an IAX trunk.
- **[IVR]**
  - Fixed an issue with missing voice prompts in the IVR.
- **[LCD]**
  - Fixed an issue where the LAN gateway IP would not be displayed properly when using Route mode.
  - [6304/08 only] Fixed an issue with entering menus on the LCD.
- **[LDAP]**

- Fixed an issue where LDAP syncing between servers with IPv6 addresses would fail if the addresses are not compressed with 0 bits.

- **[Login Settings]**
  - Fixed an issue where banned users would not have their usernames displayed.
- **[Maintenance]**
  - Fixed an issue with the wrong number of records being kept when the Keep Last X Records option is used.
  - Fixed an issue with displaying operation log details if LDAP numbers are involved.
  - Fixed an issue with SIP Peer Trunk Status alert not working properly.
  - Fixed an issue where the operation log would not correctly show the name of an uploaded custom prompt file.
  - Fixed an issue with system event alert email sending frequency.
  - Fixed an issue with SSH connection timing out after about 30 minutes.
  - Fixed an issue with network captures not being saved properly.
  - Fixed an issue with generating two system alerts after abnormal call termination.
- **[Paging/Intercom]**
  - Fixed an issue with being unable to saving and applying changes when the custom prompt file name is too long.
- **[Queue]**
  - Fixed an issue where calls would not be handed over properly to the next agent if the ringing queue member's extension ring time is set to specific values.
- **[Recording]**
  - Fixed an issue with using the recording feature code when receiving a call from an analog trunk.
  - When a DISA call is routed to an extension configured with Ring Simultaneously to an external number, the call recording will be abnormal if the external number answers the call.
  - Fixed an issue with duplicate recording files after doing a seamless transfer to an extension with Ring Simultaneously configured.
- **[RemoteConnect]**
  - Fixed an issue where the Media NAT Traversal Service would be disabled after deleting and adding a plan.
  - Fixed an issue with remote SSH connection not being available after a reboot or network reconnection.
- **[Ring Group]**
  - Fixed an issue with calling the next LDAP number if the first LDAP call fails.
- **[Routing]**
  - Fixed an issue with being unable to call external numbers that contain hyphen (-).
  - Fixed an issue with custom ringtones not applying when calling ring group and queues.
- **[SCA]**
  - Fixed an issue with ringback tone not being played correctly for incoming calls to SCA extensions.
- **[SIP Settings]**

- Fixed an issue where call feature headers would not be in the SIP INVITE, causing abnormal behavior with GS endpoint softkeys and feature codes.
  - Fixed an issue with calling parties that authenticate incoming INVITE messages.
  - Fixed an issue with passing along division header when *Keep Original CID* is enabled.
  - Fixed an issue with processing VoIP trunk calls when *Keep Original CID* is enabled and an incoming call's CID contains asterisk (*).
  - Fixed an issue with no min-expires header in 423 message responses.
  - Fixed several issues with sending the wrong diversion header when a call is forwarded out from a UCM feature such as Ring Groups and IVR.
  - Fixed an issue where UCM would forward re-INVITE messages with sendrecv instead of passing along the original attribute of the re-INVITE messages.
  - Fixed an issue with calls not working after uploading TLS certificates.

- **[User Management]**
  - Fixed several issues with custom privilege users not being able to save and apply changes on certain pages.
- **[Voice Prompts]**
  - Fixed several issues with prompts being played incorrectly.
- **[VoIP Trunks]**
  - Fixed an issue with processing original CIDs that contain asterisk (*).
  - Fixed an issue with PPI header value when *Keep Original CID* is enabled.
  - Fixed an issue with being unable to create Register SIP trunks with passwords that contain special characters.
  - Fixed an issue where trunks cannot connect after changing the trunk's transport mode to TLS.
- **[Wave Mobile]**
  - Fixed an issue with no audio when both the caller and callee have Direct Media enabled.
- **[Wave Web]**
  - Fixed an issue with the CEI Notify message that is sent after a scheduled meeting is finished.
  - Fixed an issue with seeing the wrong page when joining a recurring scheduled meeting.
  - Fixed an issue where meeting search functionality was case-sensitive.
  - Fixed several display issues.
  - Fixed an issue with abnormal call termination when transferring while sharing screen.
  - Fixed an issue where video phones registered to the UCM's public address cannot see screen shares.
  - Fixed an issue with a busy prompt being played after timing out of inviting more participants mid-meeting.
  - Fixed an issue with creating n-way conferences if a call was established via DISA.
  - Fixed an issue with FXS extensions being detected as talking in a video conference even if it is on hold.
  - Fixed an issue where Wave Web would not work properly in IPv6-only network environments.
- **[Web]**

    - Fixed several issues with the search functionality.
    - Fixed several issues with "No Permission" and "Sensitive Characters" error messages. Fixed several issues with a "sensitive characters" error message.
- **[ZeroConfig]**
    - Fixed an issue with Auto Discover not working properly.

NEW LIMITATIONS

- **[Conference]** Maximum number of video conference room for all UCM6300 models has been increased:
    - 6301: 2 → 4
    - 6302: 3 → 6
    - 6304: 4 → 8
    - 6308: 8 → 10
- **[Conference]** Maximum number of video conference members for all UCM6300 models has been increased (includes audio-only and video participants):
    - 6301: 12 → 20
    - 6302: 20 → 30
    - 6304: 40 →60
    - 6308: 60 → 80
- **[FXO/FXS] [UCM6301 only]** Auto Detect Destination channel option has been removed due to constraints of the UCM6301 hardware.
- **[Voicemail]** Voicemail password limitation is now 4-32 characters.
- **[Voice Prompts]** Open parenthesis (is no longer supported for voice prompt file names.
- **[VoIP Trunks]** The character limits for the PAI header number and name fields have been increased to 256.
- **[Wave Web]** Call recording for an ongoing call will now stop if it is changed to an N-way conference.

KNOWN ISSUES

- **[Conference]**
    - In an audio conference, participants invited via calls will see their display name as the meeting name after joining the meeting.
- **[Email Settings]**
    - Text format emails are currently not supported.
- **[Extensions]**
    - If attempting to apply changes to an extension page where a user’s email address has more than one period (.) in it, an error message “failed to update data” will appear. To fix this, please file a Grandstream Support ticket and provide SSH access.
- **[FXO/FXS]**
    - When using NTT mode for FXO, CID name cannot be displayed.
- **[IAX]**
    - RemoteConnect currently does not support IAX trunks.

- **[LDAP]**
  - UCM RemoteConnect addresses currently cannot be used for LDAP sync.
  - UCM6300 can download from UCM6200/6510, but not vice-versa. A new firmware for UCM6200/6510 will be provided at a later time to address this.
- **[Wave Desktop]**
  - Wave desktop client can only work with UCM6300 versions 1.0.5.x and above.
  - If a PC goes to sleep during a meeting and wakes up, the meeting will not be able to resume properly. In this scenario, participants must leave and rejoin the meeting.
  - If a participant is disconnected from a meeting due to network instability and rejoins, they will not be able to see the meeting chat and participant list. Additionally, the user will not be able to receive any calls after leaving the meeting and must log out and log back in.
- **[Wave Mobile]**
  - If the UCM's session timer is disabled, abnormalities will occur if the network connection changes during a call (e.g., changing from Wi-Fi to LTE connection).
- **[Wave Web]**
  - When using Firefox versions 76-79 and when dialing into a video conference room with audio only, an extra black video feed will appear.
  - If a PC goes to sleep during a meeting and wakes up, the meeting will not be able to resume properly. In this scenario, participants must leave and rejoin the meeting.
  - If a participant is disconnected from a meeting due to network instability and rejoins, they will not be able to see the meeting chat and participant list. Additionally, the user will not be able to receive any calls after leaving the meeting and must log out and log back in.
  - When using Firefox, packet loss may occur while screen sharing.
  - Audio and video cannot be transmitted properly under a NAT64 network environment.

NEW FEATURES OVERVIEW

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

WAVE WEB INSTANT MESSAGING (BETA)

Wave Web now supports instant messaging between WebRTC extensions. The **Chat** option is available in the Wave Web side bar. From here, users can create chats with contacts. Users can create group chats with multiple contacts.

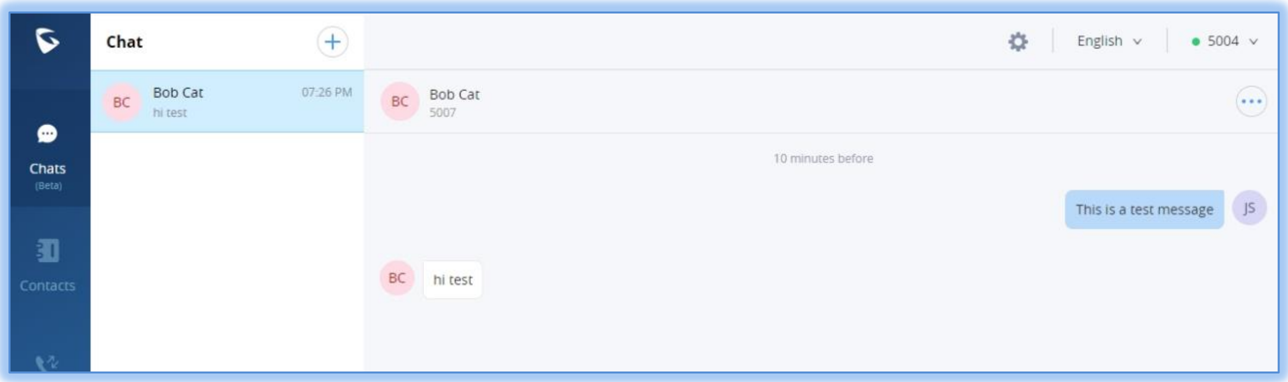

Figure 1 - Wave Web IM

Users can also quickly send a message to specific contacts by finding them in the **Contacts** page and clicking on chat icon.

Instant messaging (IM) data can be cleared from the UCM's **Maintenance→System Cleanup/Reset→Cleaner** page.

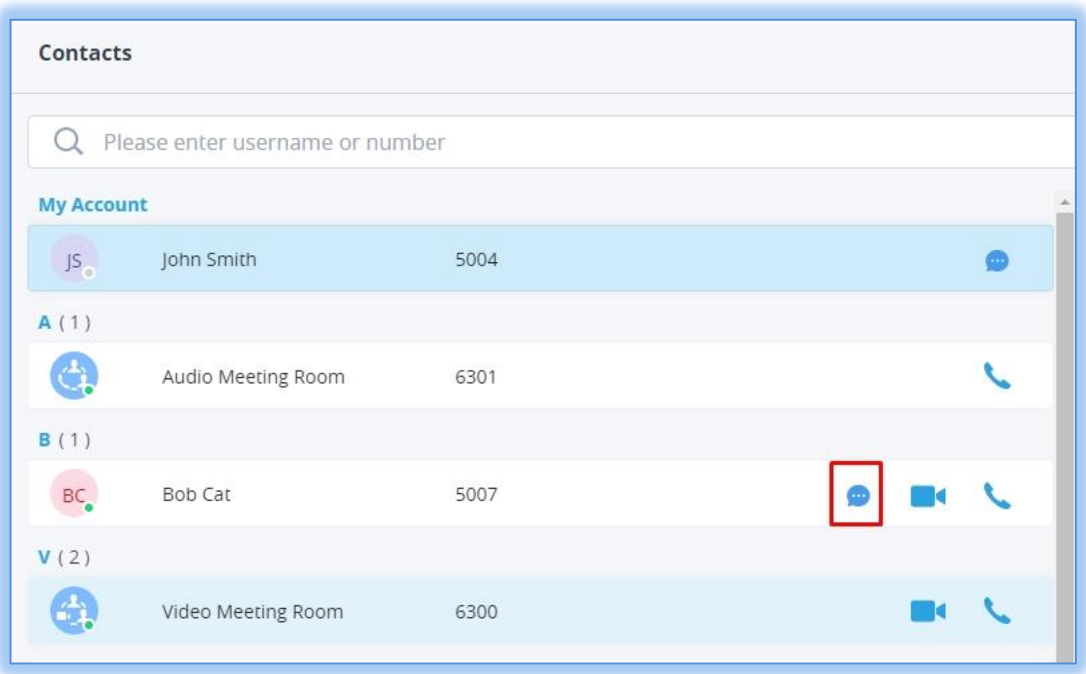

Figure 2 - Wave Web Contacts IM

WAVE DESKTOP SUPPORT

Wave Desktop is now available and supported with UCM version 1.0.5.4. Users can download the client from https://fw.gdms.cloud/wave/download/ or by clicking on the **Download Client** button located on the Wave Web interface.

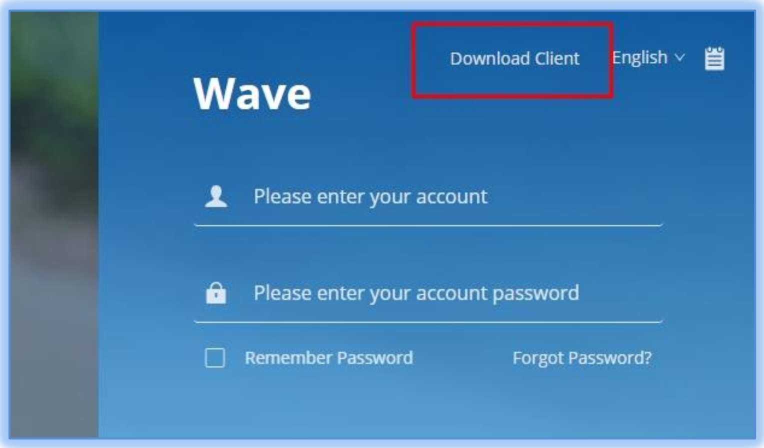

Figure 3 - Wave Desktop Download

STIR/SHAKEN SUPPORT

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

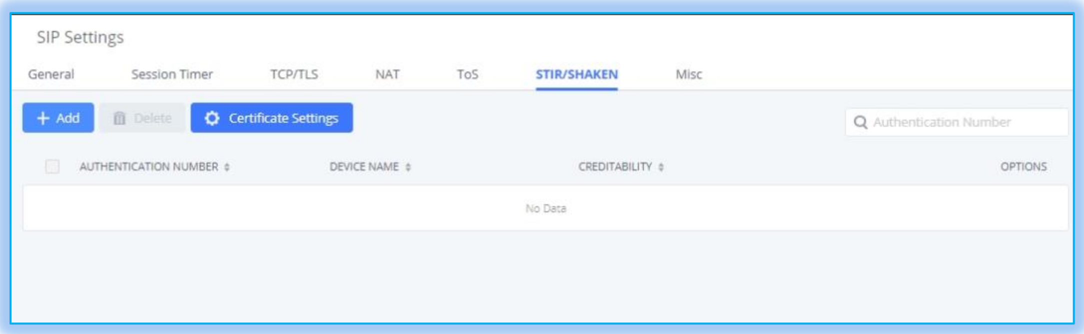

Figure 4 - STIR/SHAKE Main Page

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

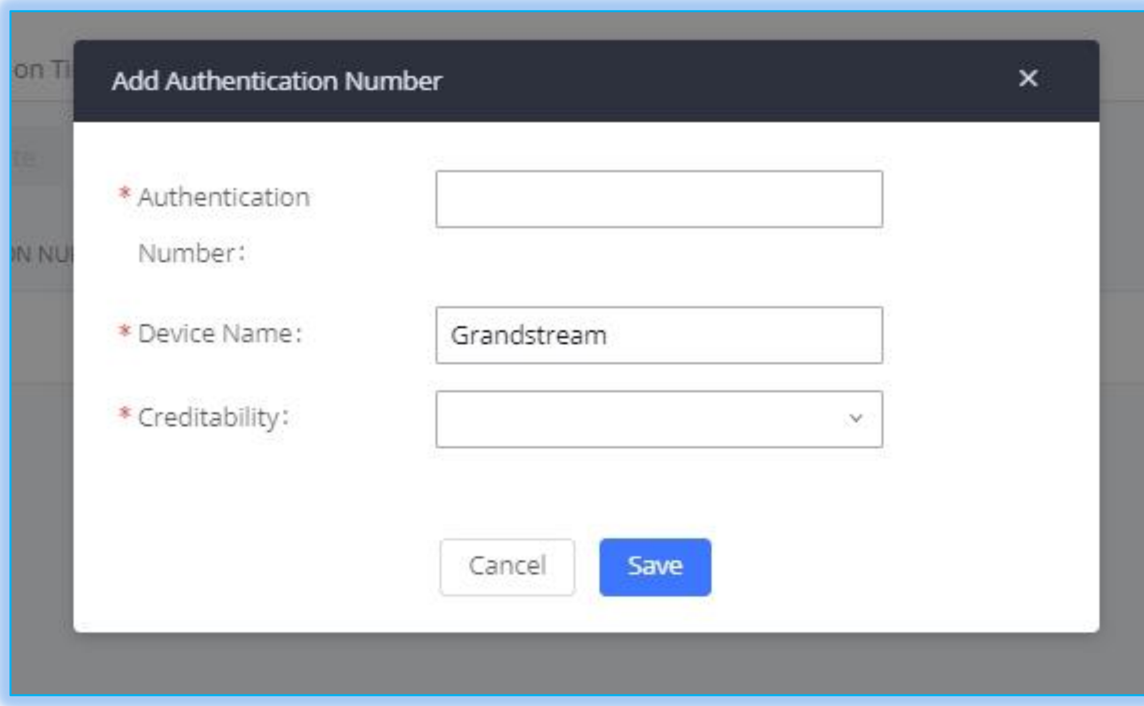

Figure 5 - STIR/SHAKEN Adding Authorized CIDs

- **Authentication Number** (or Authorized CID) - The authorized caller ID
- **Device Name** - The name of the device
- **Creditability** (or Attestation) **-** the level of confidence of the carrier that the CID has not been spoofed. The following options are available:
  - A (Full attestation) - The carrier is associated with the caller and the number. There is high confidence that the CID has not been spoofed.
  - B (Partial attestation) - The carrier is associated with the caller but not the number. There is uncertainty about whether the CID has been spoofed or not.
  - C (Gateway attestation) - The carrier is not associated with the caller and has no confidence at all about the number. Generally used for traceback.

Clicking on the *Certificate Settings* button will bring up the following window:

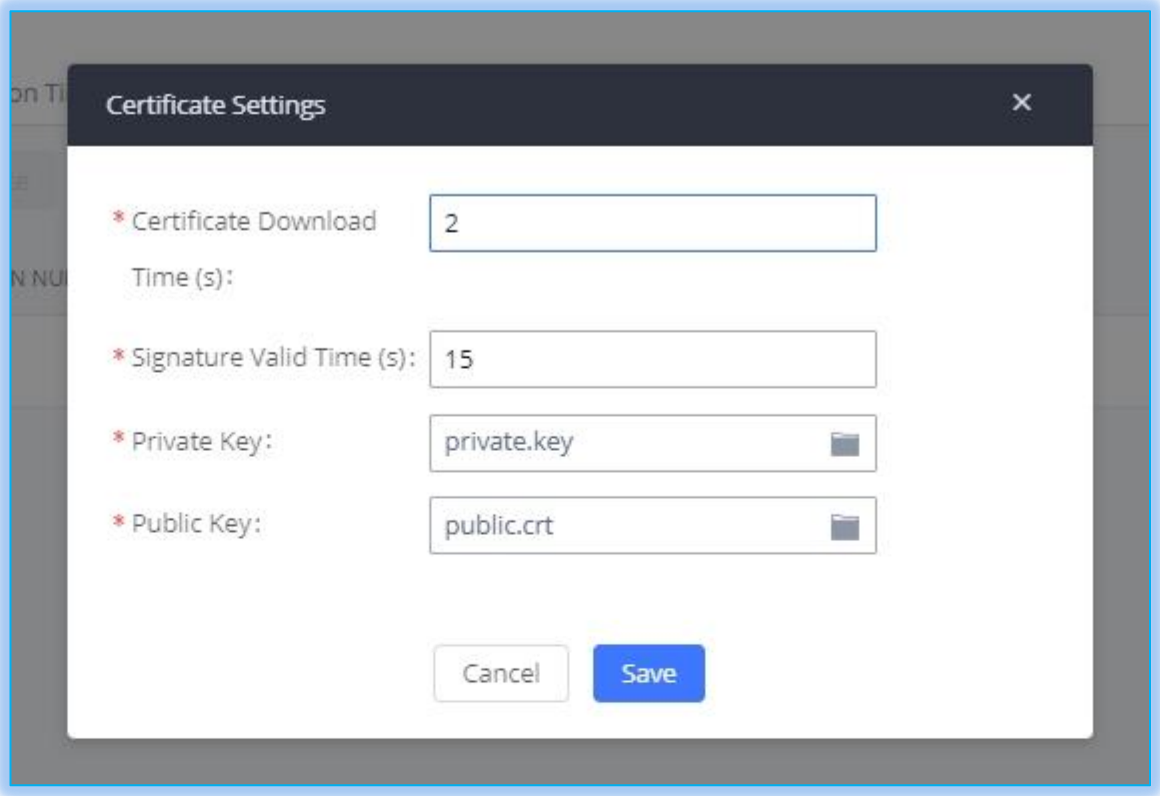

Figure 6 - Certificate Settings

- Certificate Download Time (s) - Public key download timeout period. Default is 2 seconds.
- Signature Valid Time (s) - The amount of seconds the digital signature will be valid for. Default is 15 seconds.
- Private/Public Key - Uploaded files must be less than 2MB in file size, in .crt format and be ECC type.

THRESHOLD-BASED CALL CONTROL & DATA WRITE CONTROL

System administrators can prevent the UCM from making calls and/or writing to the data partition (e.g., CDR, recordings, etc.) once the system reaches a specified threshold of CPU usage and storage usage respectively. These options are located in the *System Settings→General Settings* page.

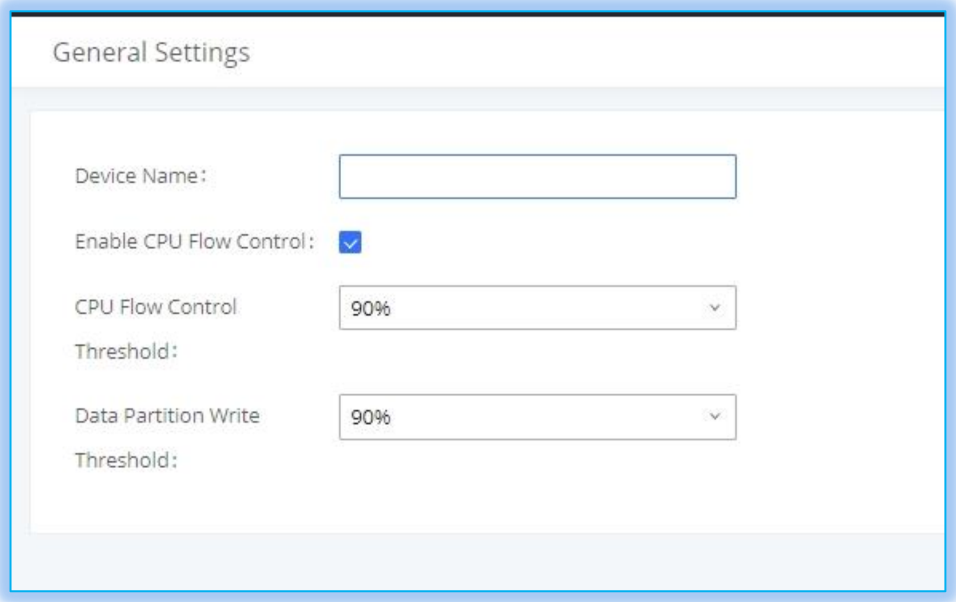

Figure 7 - Threshold Settings

A new system event alert has been added for this feature. Alert notifications will be sent when thresholds have been exceeded or returned to below the threshold.

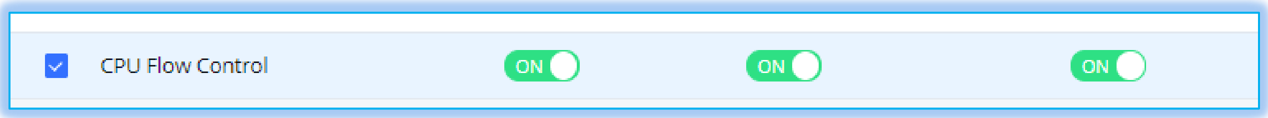

Figure 8 - Threshold System Alert

ZERO CONFIG DEVICE LIST EXPORT SUPPORT AND IMPROVED IMPORT SUPPORT

Users can now export their devices to a CSV file. This file can be imported to another UCM to quickly set it up with the original UCM's devices.

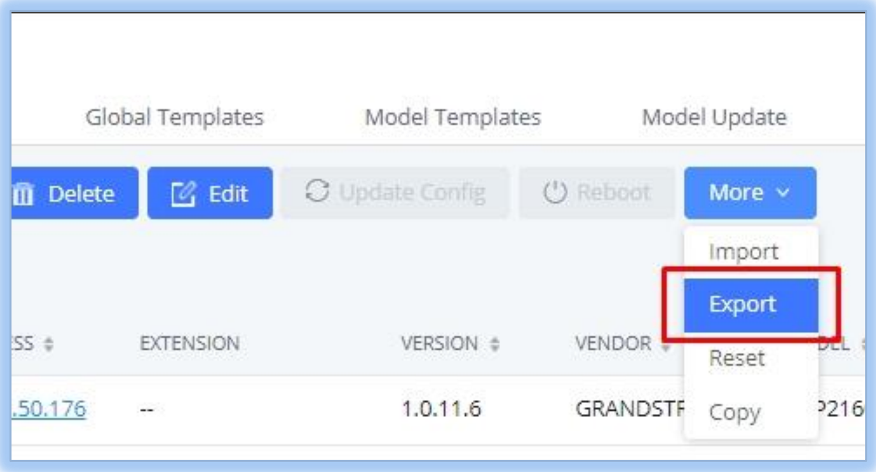

Figure 9 - Zero Config Export

The supported format of imports has been modified.

	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
1	======== Device Start ========														
2	config_na	vendor	state	ip	account_s	file_url	url_parameter	last_acces	mac	version	ad_state	model	hot_deski	port	
3		Grandstre	7	192.168.50.176		https://192.168.50.161:8089		########	000B8260(	1.0.11.6	0	GXP2160	no	5080	
4															
5	******** Advanced Settings ********														
6	value	mac	field_nam	entity_na	element_number										
7	2	000B8260(	UpgradeS	Choice	1										
8		000B8260(	UpgradeS	Path	1										
9															
10	======== Device Start ========														
11	config_na	vendor	state	ip	account_s	file_url	url_parameter	last_acces	mac	version	ad_state	model	hot_deski	port	
12		Grandstre	1	192.168.50.106		https://192.168.50.161:8089		########	000B8279:	1.0.3.224	0	GXV3240	no	5060	
13															
14	======== Device Start ========														
15	config_na	vendor	state	ip	account_s	file_url	url_parameter	last_acces	mac	version	ad_state	model	hot_deski	port	
16		Grandstre	1	192.168.50.33		https://192.168.50.161:8089		########	000B8289	1.0.11.10	0	GXP2170	no	5080	
17															
18	======== Device Start ========														
19	config_na	vendor	state	ip	account_s	file_url	url_parameter	last_acces	mac	version	ad_state	model	hot_deski	port	
20		Grandstre	1	192.168.50.210		https://192.168.50.161:8089		########	000B8294:	1.0.19.11	0	--	no	5060	
21															
22	======== Device Start ========														
23	config_na	vendor	state	ip	account_s	file_url	url_parameter	last_acces	mac	version	ad_state	model	hot_deski	port	
24		Grandstre	1	192.168.50.20		https://192.168.50.161:8089		########	000B829A	1.0.4.152	0	GXP1630	no	5080	
25															
26	======== Device Start ========														
27	config_na	vendor	state	ip	account_s	file_url	url_parameter	last_acces	mac	version	ad_state	model	hot_deski	port	
28		Grandstre	1	192.168.50.237		https://192.168.50.161:8089		########	000B82AC	1.0.19.11	0	--	no	5060	
29															
30	======== Device Start ========														
31	config_na	vendor	state	ip	account_s	file_url	url_parameter	last_acces	mac	version	ad_state	model	hot_deski	port	
32		Grandstre	7	192.168.50.191		https://192.168.50.161:8089		########	000B82FC:	1.0.5.15	0	GRP2614	no	5080	
33															
34	######## Basic Settings ########														
35	value	mac	entity_na	element_	element										
36	3000	000B82FC:	AccountCl	3	Account										
37		000B82FC:	Descriptic	1	SoftKey										
38	20	000B82FC:	Mode	1	SoftKey										

Figure 10 - CSV Export

SRTP TROUBLESHOOTING

The **Enable SRTP Debugging** option has been added to the **Maintenance→Network Troubleshooting→Ethernet Capture** page.

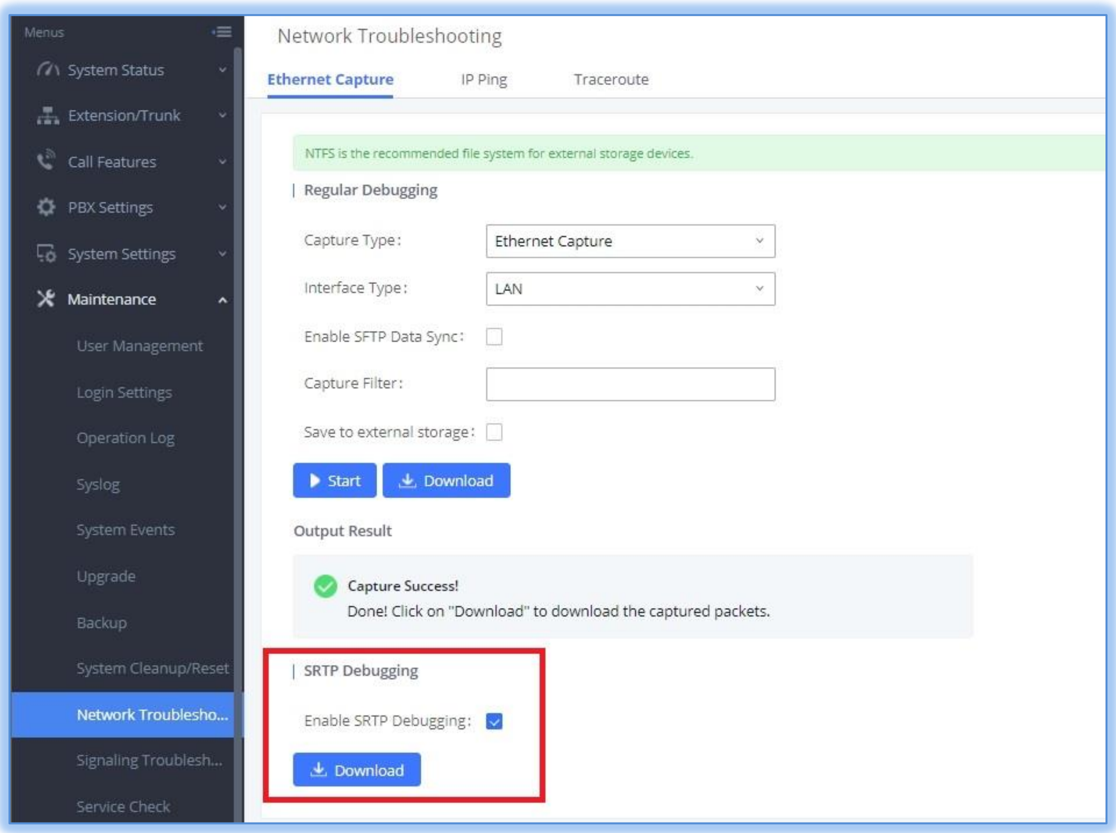

Figure 11 - SRTP Debugging

This option will allow Grandstream Support to more easily troubleshoot calls encrypted with SRTP. After ending a packet capture, clicking on the **Download** button under the **SRTP Debugging** section will generate a file that can then be provided to Grandstream Support alongside the regular network capture file.

REMOTECONNECT CALL QUALITY MONITORING

RemoteConnect users can now monitor UCM call quality from the GDMS portal. The page is located under **UCMRC→Call Quality**.

Note: Only the following devices and firmware are supported at this time:

- GRP260x fw 1.0.1.3+
- DP75x fw 1.0.13.11+
- GXV338 fw 1.0.3.21+

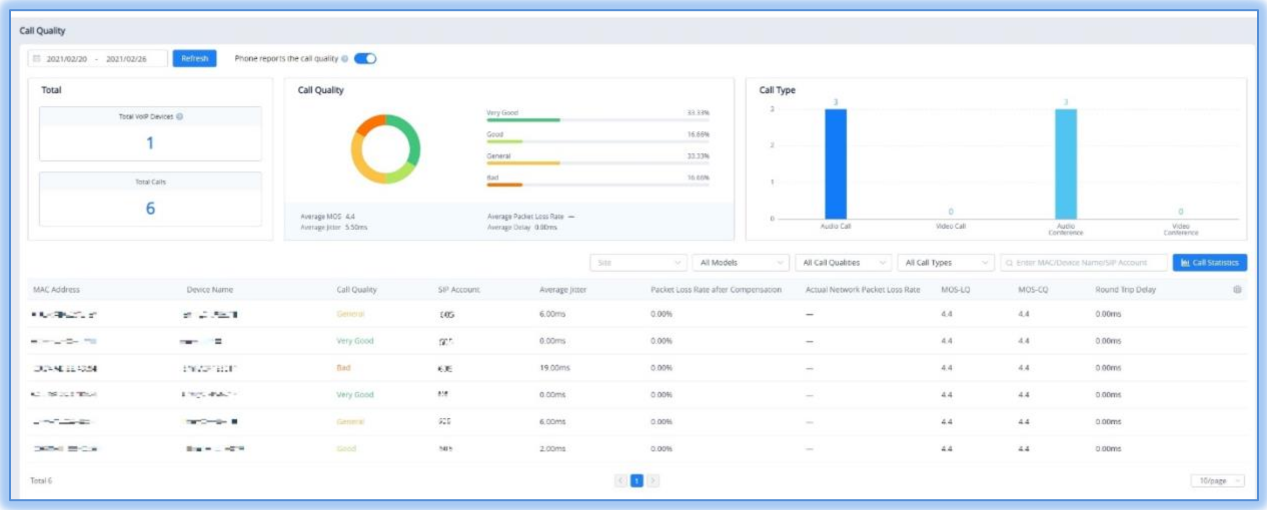

Figure 12 - RemoteConnect Call Quality Monitoring

The following statistics are measured:

- **Call Quality** – overall call performance indicator based on average jitter, average packet loss rate, network latency, and average MoS score.
- **Average Jitter** – Delay between packet arrivals. High jitter will cause delays in audio and video.
- **Packet Loss Rate** – Percentage of network packets lost per call. High packet loss rate will result in choppy low quality audio and video.
- **MOS** – Stands for Mean Opinion Score. It assigns a number to quantify end user satisfaction of sound quality. There are two ways testing:
  - MOS-CQ (conversational quality) – Overall quality of the call, which includes the listening quality, speech and noise levels for both audio streams, and echoes.
  - MOS-LQ (listening quality) – Value based on audio quality/distortion and predicts how people would rate the audio of the call.

WEB PORTAL LOGO CUSTOMIZATION

Users can now customize certain parts of UCM web portal with custom images. This option is located in **Value-added Features→UCM RemoteConnect→Custom Logo.**

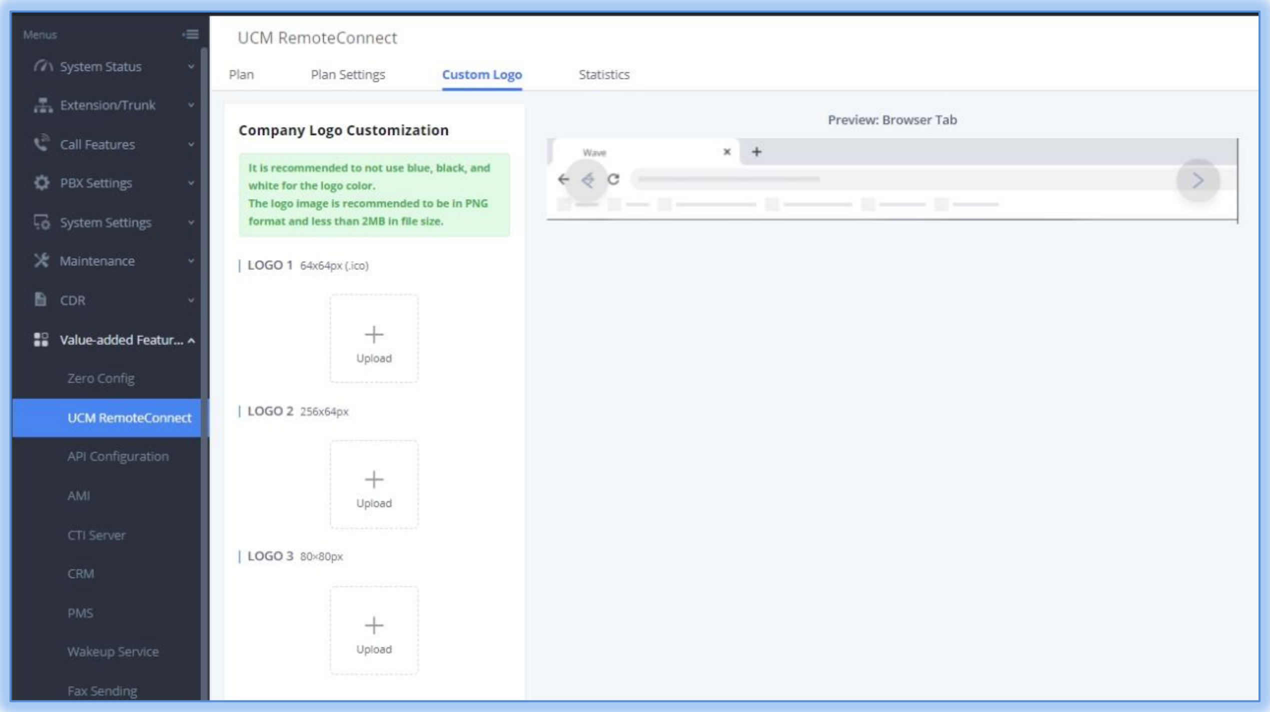

Figure 13 - Web Logo Customization

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