UCM Series IP PBX Firmware Release Notes

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FIRMWARE VERSION 1.0.7.12

PRODUCT NAME

UCM6301, UCM6302, UCM6304, UCM6308, UCM6300A, UCM6302A, UCM6304A, UCM6308A

DATE

6/7/2021

FIRMWARE FILE INFORMATION

• UCM6301/6302 firmware file name: ucm6301_ucm6302fw.bin

MD5: 486020bb270499ea02a9584561171664

• UCM6304/6308 firmware file name: ucm6304_ucm6308fw.bin

MD5: 72ab35d5374c90da5d2f33caabc61ac4

• UCM6300A/6302A firmware file name: ucm6300A_ucm6302Afw.bin

MD5: b9068764e316213ad5b1104656172fe9

• UCM6304A firmware file name: ucm6304Afw.bin

MD5: d47ed67159c8b9b1eaad81858d939384

• UCM6304A firmware file name: ucm6308Afw.bin

MD5: 94f17a615d5a9627a3184c7e006d8db7

IMPORTANT UPGRADING NOTE

- Firmware 1.0.7.12 has added security improvement. Upgrading to 1.0.7.12 is highly recommended.
- 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. Here is the link to download firmware 1.0.2.17:
 - o UCM6301/2: http://firmware.grandstream.com/Release_UCM6301_6302_1.0.2.17.zip
 - o 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 1.0.7.9

ENHANCEMENTS

- [Security] Improved internal security processes
- [IM] Improved extension status syncing process to the IM server.
- [Meetings] After editing the time of a single recurrence of a scheduled meeting, a cancelation email will now be sent out followed by a meeting update email.

BUG FIXES

• [System] Fixed an API crashing issue

FIRMWARE VERSION 1.0.7.9

PRODUCT NAME

UCM6301, UCM6302, UCM6304, UCM6308

DATE

5/21/2021

FIRMWARE FILE INFORMATION

• UCM6301/6302 firmware file name: ucm6302fw.bin

MD5: ef63eb65b3c6c2a560cac5b05f1aecf9

• UCM6304/6308 firmware file name: ucm6308fw.bin

MD5: e31735068711f62c59dc4a88d1e9ca25

IMPORTANT UPGRADING NOTE

- ALWAYS create a backup of your configuration and data before a firmware upgrade.
- <u>After upgrading to 1.0.2.18 or higher, you will no longer be able to downgrade to 1.0.2.17 or</u>
 <u>lower.</u>
- 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.
 - o UCM6301/2: http://firmware.grandstream.com/Release_UCM6301_6302_1.0.2.17.zip
 - o 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 1.0.5.4

ENHANCEMENTS

[CDR] Added Concurrent Call Statistics page to monitor the number of concurrent calls for specific trunks throughout an entire year or entire month. [

- TRUNK CONCURRENT CALL THRESHOLD]
- **[CloudIM]** Users can now configure a cloud IM server for storage of instant messaging data used for Wave. Only applicable if a UCM Cloud IM plan or Grandstream IM server was purchased. [CLOUD IM CONFIGURATION]
- **[Conferencing]** All instances of "Conference" and "Moderator" in the webUI and voice prompts have been replaced with "Meeting" and "Host" respectively. All updates related to conferencing will now be in the [Meeting] category in these release notes.
- **[Email Settings]** Added a *Download Wave* link to the *Extension* email template to download the Wave Desktop application.
- [Email Settings] Restored Text format for email templates.
- [LDAP] LDAP syncing now works with SIP trunks to RemoteConnect addresses.
- [Maintenance] The report cleaner can now automatically clear scheduled meeting history.
- [Meeting] Users can now schedule video meetings in randomly generated temporary meeting rooms without needing to manually create and configure meeting rooms. [TEMPORARY MEETING ROOMS]
- [Meeting] Users can now enter meeting rooms up to 10 minutes in advance by default.
- [Meeting] Updated the look of meeting reports
- **[Meetings]** Added email reminder option to Audio Meetings configuration page.
- [Pickup Groups] Added the ability to import/export pickup groups
- [RemoteConnect] Added Click2Call Integrated Customer Service SDK [CLICK2CALL INTEGRATED CUSTOMER SERVICE SDK]

[Ring Group] Users can now customize the various voicemail prompts for ring group voicemail. [CLICK2CALL INTEGRATED CUSTOMER SERVICE SDK

The UCM now provides a method for users to embed a simple click2call link into their websites. To get started, navigate to Value-added Features→UCM RemoteConnect→Integrated Customer Service SDK and configure the following:

Enable Click2Call :		
Call Queue :	testQueue v	
	⊥ Download	

Figure 6 - Integrated Customer Service SDK Options

Toggle on **Enable Click2Call** and select the desired call queue to take calls made via Click2Call.Click on the **Download** button to download the SDK. Once the SDK is downloaded, extract the files.

	1 34-5	llasse ll
webRTC	File folder	
😹 debug.js	JavaScript File	55 KB
😹 gsRTC.min.js	JavaScript File	382 KB
😹 jquery-3.6.0.min.js	JavaScript File	88 KB
😹 jsSipIndex.js	JavaScript File	10 KB
😹 jsSipStack.js	JavaScript File	204 KB
📄 jsSipStack.wasm	WASM File	330 KB
😹 jsSipStackWorker.js	JavaScript File	9 KB
READEME.md	MD File	1 KB
💽 webRTC.html	Microsoft Edge H	7 KB
😹 webRTC.js	JavaScript File	5 KB
webrtc_settings.json	JSON File	1 KB

Figure 7 - SDK Files

Once these files are available, open the **webrtc_settings.json** file with the desired application and edit the value of the "**websocket_proxy_url**" parameter with your UCM's RemoteConnect server address. Save the file.



Figure 8 - Editing webrtc_settings.json

Embed the HTML file into the desired web page. Once embedded, the following icon should now appear on your website (assuming no other items in SDK package were modified):



Clicking on this icon in the website should then send an audio call to the configured UCM and to the selected call queue.

- RING GROUP VOICEMAIL PROMPT CUSTOMIZATION]
- [Ring Group] Ring Group Voicemail can now be toggled on and off.

[SIP Trunks] Added the ability to configure and monitor concurrent call thresholds for each trunk. [CLICK2CALL INTEGRATED CUSTOMER SERVICE SDK

The UCM now provides a method for users to embed a simple click2call link into their websites. To get started, navigate to Value-added Features→UCM RemoteConnect→Integrated Customer Service SDK and configure the following:

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	1 26-2	
📊 webRTC	File folder	
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😹 jsSipIndex.js	JavaScript File	10 KB
😹 jsSipStack.js	JavaScript File	204 KB
📄 jsSipStack.wasm	WASM File	330 KB
😹 jsSipStackWorker.js	JavaScript File	9 KB
READEME.md	MD File	1 KB
🔁 webRTC.html	Microsoft Edge H	7 KB
😹 webRTC.js	JavaScript File	5 KB
📄 webrtc_settings.json	JSON File	1 KB

Figure 7 - SDK Files

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Figure 8 - Editing webrtc_settings.json

Embed the HTML file into the desired web page. Once embedded, the following icon should now appear on your website (assuming no other items in SDK package were modified):



Clicking on this icon in the website should then send an audio call to the configured UCM and to the selected call queue.

RING GROUP VOICEMAIL PROMPT CUSTOMIZATION

Users can now customize various parts of the ring group's voicemail prompt. The following parts can be modified:

ng Groups	Default Destination :	Extensions			
	Voicemail :				
	Voicemail Password :				
	Email Address :			•	
			Add Er	nail Address	
		hen user enters voicemail. Priority: Tempora 16 bits at 8000Hz mono with mp3/wav forma	ry Prompt > Busy Prompt/U		
			ry Prompt > Busy Prompt/U		
	Sound file must be PCM encoded, 1	16 bits at 8000Hz mono with mp3/wav forma	ry Prompt > Busy Prompt/U it, or raw ulaw/alaw/gsm fil	e with .mp3/.wav/.ulaw	//.alaw/.gsm suffix
	Sound file must be PCM encoded, 1 Busy Prompt :	16 bits at 8000Hz mono with mp3/wav forma	y Prompt > Busy Prompt/U It, or raw ulaw/alaw/gsm fil	e with .mp3/.wav/.ulaw	//.alaw/.gsm suffix

Figure 10 - Ring Group Voicemail Prompts

- TRUNK CONCURRENT CALL THRESHOLD]
- [SIP Trunks] Added the STIR/SHAKEN toggle to the trunk group configuration page.
- [SIP Trunks] Added ability to manipulate DOD based on the extension dialing out. [DOD MANIPULATION]
- [System Events] Added the TLS Cert Expiration warning system alert.
- [System Events] Added the *Trunk Concurrent Calls* system alert.
- **[System Settings]** CPU usage call control will now apply to enabling video and screen sharing in endpoint-to-endpoint Wave calls and video meetings.
- [Time Settings] Added the ability to import/export Office Time and Holiday configurations.
- [Time Settings] Added an All option to the Year dropdown list to include all years.
- **[User Portal]** Removed the video conferencing page in the regular user portal. Meetings are now managed from the Wave interface.
- [Wave General] Added support for multiple call lines.
- **[Wave General]** *Meeting List* page will now display the user's meeting history along with the participant list and meeting duration of each meeting.
- **[Wave General]** Added popular chat features to Wave's instant messaging service, such as quoting and replying to specific messages, forwarding them to another chat, deleting them entirely and even managing entire chat groups by pinning, hiding, and/or muting them.
- **[Wave General]** Added a Meeting Assistant chat bot that will notify and remind users of upcoming meetings.
- **[Wave Desktop/Web]** Wave endpoint-to-endpoint calls can now use features previously available only to video meetings such as screen sharing and in-call chat. This is only applicable if both endpoints are using Wave Desktop/Web.
- **[Wave Mobile]** Added ability to edit a single instance of a recurring meeting.

BUG FIXES

- [System]
 - Fixed several system stability issues.
 - Fixed a memory leak issue that occurs when faxing is frequently done.
 - Reduced CPU usage for concurrent call recordings.
- [Active Calls]
 - Fixed several display issues
- [Backup/Restore]
 - Fixed an issue with not syncing CDR after date splits.
 - Fixed an issue with abnormal IM data syncing if NAS is enabled.
- [Basic Calls]
 - \circ $\;$ Fixed an issue with audio calls being received as video calls.
 - Fixed an issue with transfer destinations ringing only 1 second even when Blind Transfer Timeout is not configured.
 - o Fixed an issue with calls randomly having no audio after a period of time.
 - Fixed an issue with no audio when falling back from video to audio via transfer.

- [CDR]
 - Fixed several CDR display issues.
 - Fixed an issue with exporting CSV that includes CALL TYPE information.
- [Email Settings]
 - Fixed an issue with email-to-fax functionality when the fax file contains Arabic characters.
 - Fixed an issue with not displaying the email log entry for reboot warning emails when email type is MTA.
- [Extension]
 - Fixed an issue with Follow Me calls not containing the Alert-info header.
 - Fixed an issue with diversion headers not being included in follow-me calls to extensions that also have follow-me enabled.
- [Fax]
 - Fixed an issue with extension-to-extension faxing
 - \circ Fixed an issue with displaying the incorrect Fax Sending status.
 - Fixed an issue with no fax tone when the caller is using opus.
- [Feature Code]
 - Fixed an issue where a transferred call via *2 feature code to FXO external number and parked would be shown as anonymous.
 - Fixed an issue with the wrong division header being included after blind transferring via feature code twice.
 - Fixed an issue with creating wakeup calls via feature code if the extension prefix starts with +
- [FXO/FXS]
 - Fixed an issue with DTMF detection when Caller ID Scheme is set to Auto Detect.
 - o [FXO/FXS] Fixed an issue with CID detection when Caller ID Scheme is NTT.
- [IAX]
 - Fixed an issue with establishing video calls if PCMU is not used.
 - Improved stability of IAX functionality
 - Fixed an issue where an IAX extension's forwarding settings are ignored if transferred to via feature code.
 - Fixed an issue with continuously hearing ringback tone when establishing calls via IAX register trunks.
 - Fixed an issue with callees ringing even after IAX callers cancel the calls.
 - Fixed an issue with calls with no audio.
 - Fixed an issue with no audio or video when doing a seamless transfer to an IAX extension.
- [IVR]
 - Configured voice prompt fields will be reset to default if prompts are deleted.
 - Fixed an issue with calls not being redirected back to the IVR menu.
- [LCD]
 - Fixed an issue with the LCD randomly displaying all white.
- [Maintenance]
 - \circ $\;$ Fixed an issue with the Operation Log not displaying reboot via GDMS events.

- Fixed an issue where the "Restore to normal" system alert would not be generated after "Register SIP trunk failed" issues are resolved
- Fixed an issue with sending HTTP system alert notifications if the notification is too long.

• [Meetings]

- Fixed several participant name display issues.
- Fixed an issue with the email sent to hosts when an audio meeting is scheduled.
- Fixed an issue with usernames being displayed as multi-conference bridge in the email meeting reports.
- Fixed issues with incorrect voice prompts playing
- Fixed an issue with recurring scheduled meetings being incorrectly displayed as ongoing.
- Fixed an issue with not sending three meeting reminder calls to SCA extensions.
- Fixed an issue with poor audio quality of voice prompts after ending a meeting.
- Fixed an issue with sending abnormal CEI information.
- Fixed an issue with system performance when using Opus Fullband.
- Fixed an issue with the email sent to hosts when an audio meeting is scheduled.
- Fixed an issue with scheduling a meeting with a description that contains special characters.
- Fixed an issue with usernames being displayed as multi-conference bridge in the email meeting reports.
- [NAS]
 - Fixed an issue with the NAS Username changing after restoring backup.
 - [Network Settings]
 - Fixed an issue with connecting to VPN via PPP protocol when the UCM is in route mode.
- [Paging]
 - Fixed an issue with displaying video answer when sending an announcement page to a Wave user.
 - Fixed an issue with call disconnection after holding a paging call.
 - Fixed an issue where paging members do not hear the recording prompt.
- [Parking]
 - Fixed an issue where the Parking Lot Timeout Alert-Info would use the Alert-Info configuration in Inbound Routes or IVR instead.
- [PMS]
 - Fixed an issue with being unable to create a wakeup service call via the feature code.
 - Fixed an issue with backing up voicemail to SD card.
- [Queue]
 - Fixed an issue with Call Statistics mislabeling inbound calls to DISA that are then redirected to conference.
 - Fixed an issue with average wait time and average talk time on switchboard being different from queue statistics.
 - [Queue] Fixed an issue with logging into queue via GXP/GRP softkey "UCM-CCenter" after the UCM reboots.
- [Recording]

- Fixed an issue with auto record not working when calling a party with Follow Me configured and then blind transferred.
- Fixed an issue with recording file names not including the correct callee number.
- Fixed an issue with no recording prompt when calling in via analog trunk.
- Fixed an issue where the hold music would not be included in recordings of ring group member calls.
- Fixed several issues with no audio in call recordings.
- Fixed an issue with recording after a seamless transfer.
- Fixed several issues with recording prompts not playing.

• [RemoteConnect]

- Fixed an issue with backing up recordings to GDMS.
- Fixed an issue with SIP server address on GDMS being reset to default after upgrading the RemoteConnect plan.
- Fixed an issue where GDMS cloud storage was displayed as KB instead of MB.
- Fixed an issue with incorrect error messages when configuring HTTP whitelist during a RemoteConnect session.
- Fixed an issue with displaying the actual IP address of RemoteConnect users in various pages of the management portal.
- Fixed an issue where UCM syslog capture cannot be toggled on in GDMS.
- [Ring Group]
 - Fixed an issue where the permission level of the ring group member would not be used when transferring to an external number.
 - Fixed an issue with delayed calls to ring group members.
 - Fixed an issue with ringing subsequent LDAP members if the first LDAP member rejects the call.
 - Fixed an issue with dialing out of failover trunks when a ring group failover destination is set to an external number.

• [Routing]

- Fixed an issue with importing inbound routes that contain time conditions.
- Fixed an issue with importing inbound routes if Prepend Trunk Name and Set CallerID fields are configured with Yes.
- [SCA]
 - Fixed an issue with displaying remote party information in a speed dial call.
- [SIP Settings]
 - Fixed an issue with missing Contact URI User field in the SIP 200 OK.
 - Fixed an issue with sending the wrong server name for the Grandstream endpoint UCM Detect feature.
- [SIP Trunks]
 - Fixed an issue with video not displaying properly when IPVT invites a UCM extension to conference.
- [Wave General]
 - Fixed an issue with terminating SCA calls when Wave initiates an N-way conference.

• Fixed an issue where held Wave calls through RC peer trunks cannot be unheld after repeatedly holding and unholding.

• [Wave Desktop/Web]

- Fixed an issue with being unable to unlock GDS if the account has registrations on other endpoints.
- Fixed an issue with inviting external numbers for n-way conferences if the used outbound route has a password configured.
- Fixed an issue with wakeup calls being received as video calls.
- Fixed an issue with poor screen share quality.
- Fixed an issue with error message displaying when typing in certain characters into the search bar.
- Fixed an issue with Wave Desktop being displayed as Chrome in the email meeting report.
- Fixed an issue with black screen during screen sharing on Firefox.
- Fixed an issue with screen sharing after closing video during a meeting.
- Fixed an issue with call drops after closing screen sharing.

• [Wave Mobile]

- Fixed an issue where the Wave mobile app will receive a new call after disconnecting from and reconnecting to a network.
- [Web]
 - Fixed an issue with saving an email address that has two periods after the @.
 - Fixed an issue with saving configuration when the SIP trunk password contains periods or hyphens.
 - Fixed an issue where password would be reset to default after a certificate verification and reboot.
 - Fixed an issue with the Dashboard showing the incorrect number of Remote Users.

NEW LIMITATIONS

- **[LDAP]** Username is now a required field in the LDAP Server->LDAP Phonebook->Phonebook Download Configuration page.
- [Meeting] Audio meetings and video meetings now share the same meeting participant limit pool.
- [Meetings] Special character forward slash (/) is now supported in the Meeting Subject field.
- **[SIP Settings]** Increased the character limit of the STIR/SHAKEN Authorized CID field from 18 to 64.
- **[SIP Trunks]** Modified the register trunk group password limitations to match register trunk password limitations.

NEW FEATURES OVERVIEW

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

CLOUD IM CONFIGURATION

If users have bought a Cloud IM plan or Grandstream IM server, they will be provided credentials to enter and configure on the UCM. The following page can be found under **System Settings**->Cloud IM:

Cloud IM		
Enable Cloud IM :		
* Cloud IM Server Address:	Humble:	
* Service ID :	1000-0	
* Key:		
* Department Name:		
Prepend:		



Warning: Changing the IM server will make the data and message history from the previous storage location inaccessible. If there are existing chat groups in the UCM's local storage, users will not be able access them after configuring a cloud IM server unless they toggle off *Enable Cloud IM*. Likewise, users will not be able to access their data and chat history stored in the configured cloud server if *Enable Cloud IM* is not enabled.

TEMPORARY MEETING ROOMS

Management portal users and Wave users can schedule meetings without needing to create and/or select a meeting room. Upon scheduling, a randomly generated meeting room will be used for the meeting, and this meeting room will be deleted once the meeting has completed. Participants can be selected via the UCM phonebook, existing chat group members, local phone contacts, and email address.

	Meet Now	×
* Subject :	Weekly Monday Meeting	
Invitees:	5804 "Test Ken" (Add Email) x	
	johns.smith@gmail.com ×	
	Add	
	Cancel Meet Now	

Figure 2 - Meet Now (Wave Web)

As mentioned, Wave Mobile users can also schedule meetings in these temporary meeting rooms.

<	Invite Participants	
En		ľ
h	Company Contacts >	
2	Phone Contacts	
	Email Invitation >	
2	Group Member >	
Selec	cted: 2 ^	

Figure 3 - Wave Mobile Participant Invite

Once participants have been selected, users can then start the meeting and invite the selected participants.



Figure 4 - Temporary Meeting Room

There are no limits to the number of temporary meeting rooms. **This is currently not applicable to Audio Meetings.**

If users elect to not use temporary meeting rooms, they can still create fixed meetings rooms and select them when scheduling a meeting. These fixed meeting rooms are now categorized as **Public Rooms** in Wave.

Meeting Room :	✓ Use Existing Meeting Room	
	6300	~

Figure 5 - Use Existing Meeting Rooms

CLICK2CALL INTEGRATED CUSTOMER SERVICE SDK

The UCM now provides a method for users to embed a simple click2call link into their websites. To get started, navigate to Value-added Features->UCM RemoteConnect->Integrated Customer Service SDK and configure the following:

	Enable Click2Call :		
	Call Queue :	testQueue	~
L		🛃 Download	

Figure 6 - Integrated Customer Service SDK Options

Toggle on **Enable Click2Call** and select the desired call queue to take calls made via Click2Call.Click on the **Download** button to download the SDK. Once the SDK is downloaded, extract the files.

0.000	1 200	Il and a second
webRTC	File folder	
😹 debug.js	JavaScript File	55 KB
😹 gsRTC.min.js	JavaScript File	382 KB
😹 jquery-3.6.0.min.js	JavaScript File	88 KB
😹 jsSipIndex.js	JavaScript File	10 KB
📓 jsSipStack.js	JavaScript File	204 KB
📄 jsSipStack.wasm	WASM File	330 KB
😹 jsSipStackWorker.js	JavaScript File	9 KB
READEME.md	MD File	1 KB
🔁 webRTC.html	Microsoft Edge H	7 KB
😹 webRTC.js	JavaScript File	5 KB
📄 webrtc_settings.json	JSON File	1 KB



Once these files are available, open the **webrtc_settings.json** file with the desired application and edit the value of the "**websocket_proxy_url**" parameter with your UCM's RemoteConnect server address. Save the file.



Figure 8 - Editing webrtc_settings.json

Embed the HTML file into the desired web page. Once embedded, the following icon should now appear on your website (assuming no other items in SDK package were modified):



Figure 9 - Click2Call Icon

Clicking on this icon in the website should then send an audio call to the configured UCM and to the selected call queue.

RING GROUP VOICEMAIL PROMPT CUSTOMIZATION

Users can now customize various parts of the ring group's voicemail prompt. The following parts can be modified:

ng Groups	Default Destination :	Extensions		
	Voicemail :	V		
	Voicemail Password :			
	Email Address :	-		•
			Add Email Address	0
		nen user enters voicemail. Priority: Temporary Prompt > 16 bits at 8000Hz mono with mp3/wav format, or raw ul	Busy Prompt/Unavailable P	rompt > Greet Prompt
		nen user enters voicemail. Priority: Temporary Prompt > 16 bits at 8000Hz mono with mp3/wav format, or raw ul Choose File to Upload	Busy Prompt/Unavailable P	rompt > Greet Prompt .wav/.ulaw/.alaw/.gsm suffix. T
	Sound file must be PCM encoded, 1	16 bits at 8000Hz mono with mp3/wav format, or raw ul	Busy Prompt/Unavailable F aw/alaw/gsm file with .mp3/	rompt > Greet Prompt .wav/.ulaw/.alaw/.gsm suffix. T vnload
	Sound file must be PCM encoded, 1 Busy Prompt: Greet Prompt:	16 bits at 8000Hz mono with mp3/wav format, or raw ul Choose File to Upload Choose File to Upload	Busy Prompt/Unavailable F aw/alaw/gsm file with .mp3/	rompt > Greet Prompt .wav/.ulaw/.alaw/.gsm suffix. T mload în Delete vnload în Delete
	Sound file must be PCM encoded, 1 Busy Prompt :	16 bits at 8000Hz mono with mp3/wav format, or raw ul	Busy Prompt/Unavailable P aw/alaw/gsm file with .mp3/	rompt > Greet Prompt .wav/.ulaw/.gsm suffix. T vnload îî Delete vnload îî Delete

Figure 10 - Ring Group Voicemail Prompts

TRUNK CONCURRENT CALL THRESHOLD

Users can now configure concurrent call thresholds for specific trunks. This can be toggled and configured from the *Edit SIP Trunk*-*Basic Settings* page.

VolP Trunks	Disable This Trunk :	
SLA Station	TEL URI :	Disabled v
Outbound Routes	CallerID Number:	
Inbound Routes	CallerID Name:	
🗳 Call Features 🛛 🗸	From Domain :	
🔅 PBX Settings 🗸 🗸	Jitter Buffer:	NetEQ ~
😳 System Settings 🗸 🗸	Direct Callback :	
🗶 Maintenance 🗸 🗸	RemoteConnect Mode :	
🖺 CDR 🗸 🗸	Enable Concurrent Call Alert :	
📲 Value-added Featur 🗸	* Concurrent Call Threshold :	40
	* Outgoing Concurrent Call Threshold :	20
	* Incoming Concurrent Call Threshold:	10

Figure 11 - SIP Trunk Concurrent Call Threshold

- Concurrent Call Threshold the maximum allowed number of incoming and outgoing calls
- Outgoing Concurrent Call Threshold the maximum allowed number of concurrent outgoing calls
- Incoming Concurrent Call Threshold the maximum allowed number of concurrent incoming calls

Once these options are configured, navigate to the *Maintenance→System Events→Alert Events List→Edit Trunk Concurrent Calls* page and toggle on the desired alerts.

-J-10		
System Events	Register SIP failed	OFF
Upgrade	SIP Lost Registration	OFF
Backup	SIP Internal Call Failure	OFF
System Cleanup/Reset	Remote Concurrent Calls 🕢	OFF
Network Troublesho	Trunk Concurrent Calls	

Figure 12 - Trunk Concurrent Calls Alert

Settings: Trunk Concur	rent Calls	
t Cycle:	10	minute(s) v
]

Figure 13 - Trunk Concurrent Calls Alert Options

Users can then configure how often to check whether the thresholds have been exceeded.

Finally, users can view the *CDR→Statistics→Concurrent Call Statistics* page to view the number of concurrent calls throughout a year or a month.

Menus	-	Statistics				
	*	CDR Statistics	Concurrent Call Statistics			
	*					
	e:	Trunks :	EPT anto - unclabered PT			
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	re *					
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Figure 14 - Trunk Concurrent Calls Statistics

DOD MANIPULATION

Users can now manipulate the DOD number based on the extension that is used to dial out of the trunk. The following options have been added to the DOD Add/Edit page:

- Add Extension If enabled, the user dialing out will have his extension added to the end of the DOD number.
- **Strip Extension** If enabled, the specified number of digits configured in the following **Strip** field will be removed from the beginning of the extension number.
- **Strip** The number of digits to remove from the beginning of the dialing extension number.

ipa	Create DC	D					×	
	* DOD Number : DOD Name : Add Extension : Strip Extension :		12					
	* Strip:	2 items Search VFAX Extension Grou	2 Available Q Ip test	× >	1 item Search 3000	Selected		
			Cance	1	Save			

Figure 15 - DOD Manipulation

Example: 12 is configured as the DOD number. The dialing extension is 3000, and the strip value is 2. Two digits will be removed from the beginning of 3000, making it 00. 00 will then be added to the end of the DOD number. The resulting final DOD is 1200.

MEETING ASSISTANT

Wave's instant messaging service now includes a chat bot that will remind and notify users of upcoming meetings. However, this is only applicable if the user's extension was used for the invitation.



Figure 16 - Meeting Assistant