



# Grandstream Networks, Inc.

## UCM6200 Series IP PBX

---

### User Manual



## **COPYRIGHT**

©2019 Grandstream Networks, Inc. <http://www.grandstream.com>

All rights reserved. Information in this document is subject to change without notice. Reproduction or transmittal of the entire or any part, in any form or by any means, electronic or print, for any purpose without the express written permission of Grandstream Networks, Inc. is not permitted.

The latest electronic version of this user manual is available for download here:

<http://www.grandstream.com/support>

Grandstream is a registered trademark and Grandstream logo is trademark of Grandstream Networks, Inc. in the United States, Europe and other countries.

## **CAUTION**

Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

## **WARNING**

Please do not use a different power adaptor with your devices as it may cause damage to the products and void the manufacturer warranty.



## GNU GPL INFORMATION

UCM62xx firmware contains third-party software licensed under the GNU General Public License (GPL). Grandstream uses software under the specific terms of the GPL. Please see the GNU General Public License (GPL) for the exact terms and conditions of the license.

Grandstream GNU GPL related source code can be downloaded from Grandstream web site from:

<http://www.grandstream.com/support/faq/gnu-general-public-license/gnu-gpl-information-download>



## Table of Content

<b>DOCUMENT PURPOSE.....</b>	<b>27</b>
<b>CHANGE LOG.....</b>	<b>28</b>
Firmware Version 1.0.20.23.....	28
Firmware Version 1.0.20.22.....	28
Firmware Version 1.0.20.20.....	28
Firmware Version 1.0.20.17.....	28
Firmware Version 1.0.19.29.....	33
Firmware Version 1.0.19.27.....	33
Firmware Version 1.0.19.21.....	34
Firmware Version 1.0.18.13.....	34
Firmware Version 1.0.18.12.....	34
Firmware Version 1.0.18.9.....	34
Firmware Version 1.0.17.16.....	35
Firmware Version 1.0.16.20.....	36
Firmware Version 1.0.16.18.....	36
Firmware Version 1.0.15.16.....	37
Firmware Version 1.0.14.24.....	37
Firmware Version 1.0.14.23.....	37
Firmware Version 1.0.14.21.....	38
Firmware Version 1.0.13.14.....	38
Firmware Version 1.0.12.19.....	39
Firmware Version 1.0.11.27.....	40
Firmware Version 1.0.0.7.....	40
<b>WELCOME.....</b>	<b>41</b>
<b>PRODUCT OVERVIEW.....</b>	<b>42</b>
Technical Specifications.....	42
<b>INSTALLATION.....</b>	<b>45</b>
Equipment Packaging.....	45
Connect Your UCM6200.....	45
<i>Connect The UCM6202</i> .....	45
<i>Connect The UCM6204</i> .....	46
<i>Connect The UCM6208</i> .....	47
Safety Compliances.....	48





Warranty.....	48
<b>GETTING STARTED.....</b>	<b>49</b>
Use the LCD Menu.....	49
Use the LED Indicators.....	51
Using the Web UI.....	52
<i>Accessing the Web UI.....</i>	52
<i>Setup Wizard.....</i>	53
<i>Main Settings.....</i>	53
<i>Web GUI Languages.....</i>	54
<i>Web GUI Search Bar.....</i>	55
<i>Saving and Applying Changes.....</i>	55
Setting Up an Extension.....	55
<b>SYSTEM SETTINGS.....</b>	<b>57</b>
HTTP Server.....	57
Network Settings.....	58
<i>Basic Settings.....</i>	58
<i>DHCP Client List.....</i>	64
<i>802.1X.....</i>	65
<i>Static Routes.....</i>	67
<i>Port Forwarding.....</i>	69
OpenVPN®.....	71
DDNS Settings.....	73
Security Settings.....	75
<i>Static Defense.....</i>	75
<i>Dynamic Defense.....</i>	78
<i>Fail2ban.....</i>	80
<i>TLS Security.....</i>	81
<i>SSH Access.....</i>	82
LDAP Server.....	82
<i>LDAP Server Configurations.....</i>	83
<i>LDAP Phonebook.....</i>	84
<i>LDAP Client Configurations.....</i>	88
Time settings.....	91
<i>Auto time updating.....</i>	91
<i>Set Time Manually.....</i>	92
<i>NTP Server.....</i>	93
<i>Office Time.....</i>	93
<i>Holiday.....</i>	94
Email Settings.....	96



<i>Email settings</i> .....	96
<i>Email Templates</i> .....	99
<i>Email Send Log</i> .....	100
NTP Server.....	102
Recordings Storage.....	102

## **PROVISIONING.....105**

Overview.....	105
Configuration Architecture for End Point Device.....	105
Auto Provisioning Settings.....	106
Discovery.....	109
Uploading Devices List.....	111
Managing discovered devices.....	112
Global configuration.....	113
<i>Global policy</i> .....	113
<i>Global Templates</i> .....	121
Model configuration.....	123
<i>Model templates</i> .....	123
<i>Model Update</i> .....	125
Device Configuration.....	127
<i>Create New Device</i> .....	127
<i>Manage Devices</i> .....	128
Sample Application.....	134

## **EXTENSIONS..... 139**

Create New User.....	139
<i>Create New SIP Extension</i> .....	139
<i>Create New IAX Extension</i> .....	148
<i>Create New FXS Extension</i> .....	152
Batch Add Extensions.....	157
<i>Batch Add SIP Extensions</i> .....	157
<i>Batch Add IAX Extensions</i> .....	160
Batch Extension Resetting Functionality.....	162
Search and Edit Extension.....	162
Export Extensions.....	163
Import Extensions.....	164
Extension Details.....	171
E-mail Notification.....	172
Multiple Registrations per Extension.....	173
SMS Message Support.....	174

## **EXTENSION GROUPS..... 176**



Configure Extension Groups.....	176
Using Extension Groups.....	177
<b>ANALOG TRUNKS.....</b>	<b>178</b>
Analog Trunk Configuration.....	178
PSTN Detection.....	182
<b>VOIP TRUNKS.....</b>	<b>185</b>
VoIP Trunk Configuration.....	185
Trunk Groups.....	196
Direct Outward Dialing (DOD).....	197
<b>SLA STATION.....</b>	<b>200</b>
Create/Edit SLA Station.....	200
Sample Configuration.....	201
<b>CALL ROUTES.....</b>	<b>203</b>
Outbound Routes.....	203
<i>Configuring Outbound Routes</i> .....	203
<i>Outbound Blacklist</i> .....	206
<i>PIN Groups</i> .....	208
Inbound Routes.....	212
<i>Inbound Rule Configurations</i> .....	213
<i>Inbound Route: Prepend Example</i> .....	217
<i>Inbound Route: Multiple Mode</i> .....	218
<i>Inbound Route: Route-Level Mode</i> .....	220
<i>Inbound Route: Inbound Mode BLF Monitoring</i> .....	221
<i>Inbound Route: Import/Export Inbound Route</i> .....	222
<i>FAX Intelligent Route</i> .....	223
<i>FAX with Two Media</i> .....	223
<i>Blacklist Configurations</i> .....	223
<b>CONFERENCE.....</b>	<b>225</b>
Conference Room Configurations.....	225
Conference Call Operations.....	228
<i>Join a Conference Call</i> .....	228
<i>Invite Other Parties to Join Conference</i> .....	228
<i>During The Conference</i> .....	229
<i>Google Service Settings Support</i> .....	231
Conference Schedule.....	233
<i>Cleaner Options</i> .....	236



<i>Show/Hide Conference Schedule Table</i> .....	236
Contact Group.....	238
<i>Contact Group Configurations</i> .....	238
Conference Recordings.....	239
Conference Call Statistics.....	239
<b>VIDEO CONFERENCE</b> .....	<b>241</b>
Basic Settings.....	241
Video Conference Room Configurations.....	242
Conference Schedule.....	242
Wave WebRTC Video Calling & Conferencing.....	244
<b>IPVIDEOTALK MEETINGS</b> .....	<b>246</b>
<b>IVR</b> .....	<b>249</b>
Configure IVR.....	249
Black/White List in IVR.....	252
Create Custom Prompt.....	253
<b>VOICE PROMPT</b> .....	<b>255</b>
Language Settings.....	255
<i>Download and Install Voice Prompt Package</i> .....	255
<i>Upload Language Package</i> .....	257
Custom Prompt.....	258
<i>Record New Custom Prompt</i> .....	258
<i>Upload Custom Prompt</i> .....	258
<i>Download All Custom Prompt</i> .....	259
Username Prompt Customization.....	259
<i>Upload Username Prompt File from Web GUI</i> .....	259
<i>Record Username via Voicemail Menu</i> .....	260
<b>VOICEMAIL</b> .....	<b>261</b>
Configure Voicemail.....	261
Access Voicemail.....	263
Leaving Voicemail.....	264
Extension Voicemail Count.....	265
Voicemail Email Settings.....	265
Configure Voicemail Group.....	266
<b>RING GROUP</b> .....	<b>268</b>
Configure Ring Group.....	268



Remote Extension in Ring Group.....	271
<b>PAGING AND INTERCOM GROUP.....</b>	<b>275</b>
Configure Paging/Intercom Group.....	275
<i>Configure Multicast Paging.....</i>	<i>275</i>
<i>Configure 2-way Intercom.....</i>	<i>276</i>
<i>Configure 1-way Paging.....</i>	<i>277</i>
<i>Configure Announcement Paging.....</i>	<i>278</i>
<i>Configure Private Intercom.....</i>	<i>279</i>
<i>Paging/Intercom Group Settings.....</i>	<i>281</i>
Configure a Scheduled Paging/Intercom.....	281
<b>CALL QUEUE.....</b>	<b>283</b>
Configure Call Queue.....	283
Call Center Settings and Enhancements.....	288
Queue Statistics.....	290
<i>Agent Details.....</i>	<i>292</i>
<i>Login Record.....</i>	<i>292</i>
<i>Pause Log.....</i>	<i>293</i>
Switchboard.....	293
Global Queue Settings.....	296
<b>PICKUP GROUPS.....</b>	<b>298</b>
Configure Pickup Groups.....	298
Configure Pickup Feature Code.....	298
<b>MUSIC ON HOLD.....</b>	<b>300</b>
<b>FAX SERVER.....</b>	<b>303</b>
Configure Fax/T.38.....	303
Receiving Fax.....	305
<i>Sample Configuration to Receive Fax from PSTN Line.....</i>	<i>305</i>
<i>Sample Configuration for Fax-To-Email.....</i>	<i>307</i>
FAX Sending.....	309
<b>BUSY CAMP-ON.....</b>	<b>310</b>
<b>PRESENCE.....</b>	<b>311</b>
<b>FOLLOW ME.....</b>	<b>314</b>
<b>SPEED DIAL.....</b>	<b>316</b>



<b>DISA.....</b>	<b>317</b>
<b>EMERGENCY.....</b>	<b>319</b>
<b>CALLBACK.....</b>	<b>323</b>
<b>BLF AND EVENT LIST.....</b>	<b>324</b>
BLF.....	324
Event List.....	324
<b>DIAL BY NAME.....</b>	<b>327</b>
Dial by Name Configuration.....	327
Username Prompt Customization.....	330
<i>Upload Username Prompt File from Web GUI.....</i>	<i>330</i>
<i>Record Username via Voicemail Menu.....</i>	<i>331</i>
<b>ACTIVE CALLS AND MONITOR.....</b>	<b>332</b>
Active Calls Status.....	332
Hang Up Active Calls.....	334
Call Monitor.....	334
<b>CALL FEATURES.....</b>	<b>336</b>
Feature Codes.....	336
Parking Lot.....	342
Call Park.....	343
<i>Park a Call.....</i>	<i>343</i>
<i>Retrieve Parked Call.....</i>	<i>344</i>
<i>Monitor Call Park CID Name Information (GXP21xx Phones Only).....</i>	<i>344</i>
Call Recording.....	344
Enable Spy.....	345
Shared Call Appearance (SCA).....	345
Announcement.....	349
<b>PBX SETTINGS.....</b>	<b>352</b>
PBX Settings/General Settings.....	352
PBX Settings/RTP Settings.....	354
<i>RTP Settings.....</i>	<i>354</i>
<i>Payload.....</i>	<i>355</i>
PBX Settings/NAS.....	355
PBX Settings/Voice Prompt Customization.....	356



<i>Record New Custom Prompt</i> .....	356
<i>Upload Custom Prompt</i> .....	357
<i>Download All Custom Prompt</i> .....	357
PBX Settings/ Call Failure Tone Settings.....	358
<i>SIP Trunk Prompt Tone</i> .....	358
<i>General Call Failure Tone</i> .....	359
PBX Settings/Jitter Buffer.....	359
PBX Settings/Recordings Storage.....	360
PBX Settings/NAS.....	362
<b>SIP SETTINGS.....</b>	<b>363</b>
SIP Settings/General.....	363
SIP Settings/MISC.....	364
SIP Settings/Session Timer.....	365
SIP Settings/TCP and TLS.....	365
SIP Settings/NAT.....	366
SIP Settings/TOS.....	366
Transparent Call-Info header.....	368
<b>IAX SETTINGS.....</b>	<b>369</b>
IAX Settings/General.....	369
IAX Settings/Registration.....	369
IAX Settings/Security.....	370
<b>INTERFACE SETTINGS.....</b>	<b>371</b>
<b>GDMS SETTINGS.....</b>	<b>374</b>
<b>API CONFIGURATION.....</b>	<b>377</b>
HTTPS API (New).....	377
HTTPS API (Old).....	378
CDR Real-time Output Settings.....	379
Upload Voice Prompt via API.....	380
<b>CTI SERVER.....</b>	<b>382</b>
<b>ASTERISK MANAGER INTERFACE (RESTRICTED ACCESS).....</b>	<b>383</b>
<b>CRM INTEGRATION.....</b>	<b>384</b>
SugarCRM.....	384
vTigerCRM.....	385



ZohoCRM.....	387
Salesforce CRM.....	389
ACT! CRM.....	390
<b>PMS INTEGRATION.....</b>	<b>392</b>
HMobile PMS Connector.....	392
HSC PMS.....	393
Mitel PMS.....	394
PMS API.....	395
Connecting to PMS.....	396
PMS Features.....	397
<i>Room Status</i> .....	397
<i>Wake Up Service</i> .....	398
<i>Mini Bar</i> .....	399
<b>WAKEUP SERVICE.....</b>	<b>402</b>
Wakeup Service using Admin Login.....	402
Wakeup Service from User Portal.....	403
Wakeup Service using Feature Code.....	404
<b>ANNOUNCEMENTS CENTER.....</b>	<b>405</b>
Announcements Center Settings.....	406
Group Settings.....	406
<b>QUEUOMETRICS INTEGRATION.....</b>	<b>409</b>
API Configuration Parameters.....	409
<b>STATUS AND REPORTING.....</b>	<b>411</b>
PBX Status.....	411
<i>Trunks</i> .....	412
<i>Extensions</i> .....	413
<i>Interfaces Status</i> .....	414
System Status.....	415
<i>General</i> .....	415
<i>Network</i> .....	415
<i>Storage Usage</i> .....	416
<i>Resource Usage</i> .....	417
System Events.....	418
<i>Alert Events List</i> .....	418
<i>Alert Log</i> .....	420
<i>Alert Contact</i> .....	422





CDR.....	423
<i>CDR Improvement</i> .....	427
<i>Downloaded CDR File</i> .....	428
<i>CDR Export Customization</i> .....	429
<i>Statistics</i> .....	430
<i>Recording Files</i> .....	432
<i>API Configuration</i> .....	432
<b>USER PORTAL.....</b>	<b>434</b>
Basic Information.....	436
Personal Data.....	436
Value-added Features.....	436
<b>MAINTENANCE.....</b>	<b>437</b>
User Management.....	437
<i>User Information</i> .....	437
<i>Custom Privilege</i> .....	438
<i>Concurrent Multi-User Login</i> .....	441
<i>Change Password</i> .....	442
<i>Change Username</i> .....	442
<i>Change binding Email</i> .....	443
<i>Login Settings</i> .....	443
Operation Log.....	444
Upgrading.....	446
<i>Upgrading Via Network</i> .....	447
<i>Upgrading Via Local Upload</i> .....	448
<i>No Local Firmware Servers</i> .....	450
Backup.....	450
<i>Backup/Restore</i> .....	451
<i>Data Sync</i> .....	453
<i>Restore Configuration from Backup File</i> .....	455
System Cleanup/Reset.....	455
<i>Reset and Reboot</i> .....	455
<i>Cleaner</i> .....	456
<i>USB/SD Card Files Cleanup</i> .....	460
System Recovery.....	461
Syslog.....	463
Network Troubleshooting.....	463
<i>Ethernet Capture</i> .....	463
<i>IP Ping</i> .....	464
<i>Traceroute</i> .....	465



Signaling Troubleshooting.....	466
<i>Analog Record Trace</i> .....	<b>466</b>
Service Check.....	467
Network Status.....	468
<b>EXPERIENCING THE UCM6200 SERIES IP PBX.....</b>	<b>469</b>



## Table of Tables

Table 1: Technical Specifications.....	42
Table 2: UCM6200 Equipment Packaging.....	45
Table 3: LCD Menu Options.....	50
Table 4: UCM6202/UCM6204 LED Indicators.....	51
Table 5: UCM6208 LED Indicators.....	51
Table 6: HTTP Server Settings.....	57
Table 7: UCM6200 Network Settings→Basic Settings.....	58
Table 8: UCM6200 Network Settings→802.1X.....	67
Table 9: UCM6200 Network Settings→Static Routes.....	67
Table 10: UCM6200 Network Settings→Port Forwarding.....	69
Table 11: UCM6200 System Settings→Network Settings→OpenVPN®.....	72
Table 12: UCM6200 Firewall→Static Defense→Current Service.....	76
Table 13: Typical Firewall Settings.....	76
Table 14: Firewall Rule Settings.....	78
Table 15: UCM6200 Firewall Dynamic Defense.....	79
Table 16: Fail2Ban Settings.....	81
Table 17: Time Auto Updating.....	91
Table 18: Create New Office Time.....	93
Table 19: Create New Holiday.....	95
Table 20: Email Settings.....	96
Table 21: Email Log.....	101
Table 22: Auto Provision Settings.....	108
Table 23: Global Policy Parameters – Localization.....	114
Table 24: Global Policy Parameters – Phone Settings.....	114
Table 25: Global Policy Parameters – Contact List.....	115
Table 26: Global Policy Parameters – Maintenance.....	117
Table 27: Global Policy Parameters – Network Settings.....	119
Table 28: Global Policy Parameters – Customization.....	119
Table 29: Global Policy Parameters – Communication Settings.....	120
Table 30: Create New Template.....	122
Table 31: Create New Model Template.....	123
Table 32: SIP Extension Configuration Parameters – Basic Settings.....	140
Table 33: SIP Extension Configuration Parameters – Media.....	142
Table 34: SIP Extension Configuration Parameters – Features.....	143
Table 35: SIP Extension Configuration Parameters – Specific Time.....	147
Table 36: IAX Extension Configuration Parameters→Basic Settings.....	148
Table 37: IAX Extension Configuration Parameters→Media.....	149
Table 38: IAX Extension Configuration Parameters→Features.....	150



Table 39: IAX Extension Configuration Parameters→Specific Time.....	152
Table 40: FXS Extension Configuration Parameters→Basic Settings.....	152
Table 41: FXS Extension Configuration Parameters→Media.....	153
Table 42: FXS Extension Configuration Parameters→Features.....	154
Table 43: FXS Extension Configuration Parameters→Specific Time.....	156
Table 44: Batch Add SIP Extension Parameters.....	157
Table 45: Batch Add IAX Extension Parameters.....	160
Table 46: SIP extensions Imported File Example.....	165
Table 47: IAX extensions Imported File Example.....	167
Table 48: FXS Extensions Imported File Example.....	169
Table 49: Analog Trunk Configuration Parameters.....	178
Table 50: PSTN Detection for Analog Trunk.....	184
Table 51: Create New SIP Trunk.....	185
Table 52: SIP Register Trunk Configuration Parameters.....	187
Table 53: SIP Peer Trunk Configuration Parameters.....	190
Table 54: Create New IAX Trunk.....	193
Table 55: IAX Register Trunk Configuration Parameters.....	193
Table 56: IAX Peer Trunk Configuration Parameters.....	195
Table 57: SLA Station Configuration Parameters.....	200
Table 58: Outbound Route Configuration Parameters.....	203
Table 59: Outbound Routes/PIN Group.....	208
Table 60: Inbound Rule Configuration Parameters.....	213
Table 61: Conference Room Configuration Parameters.....	225
Table 62: Conference Settings.....	227
Table 63: Conference Caller IVR Menu.....	230
Table 64: Conference Schedule Parameters.....	233
Table 65: Contact Group Parameters.....	239
Table 66: Video Conference Basic Settings.....	241
Table 67: Video Conference room Configuration Parameters.....	242
Table 68: Video Conference Schedule Parameters.....	242
Table 69: IVR Configuration Parameters.....	250
Table 70: Voicemail Settings.....	262
Table 71: Voicemail IVR Menu.....	263
Table 72: Voicemail Email Settings.....	265
Table 73: Voicemail Group Settings.....	267
Table 74: Ring Group Parameters.....	268
Table 75: Multicast Paging Configuration Parameters.....	276
Table 76: 2-way Intercom Configuration Parameters.....	277
Table 77: 1-way Paging Configuration Parameters.....	278
Table 78: Announcement Paging Configuration Parameters.....	279
Table 79: Private Intercom Configuration Parameters.....	280



Table 80: Schedule Paging / Intercom Settings.....	282
Table 81: Call Queue Configuration Parameters.....	283
Table 82: Static Agent Limitation.....	287
Table 83: Call Center Parameters.....	289
Table 84: Switchboard Parameters.....	295
Table 85: Global Queue Settings.....	297
Table 86: FAX/T.38 Settings.....	303
Table 87: SIP Presence Status.....	312
Table 88: Follow Me Settings.....	315
Table 89: Follow Me Options.....	315
Table 90: DISA Settings.....	318
Table 91: Emergency Numbers Parameters.....	321
Table 92: Callback Configuration Parameters.....	323
Table 93: Event List Settings.....	325
Table 94: UCM6200 Feature Codes.....	336
Table 95 : Parking Lot.....	342
Table 96: Add SCA Private Number.....	348
Table 97: Editing the SCA Number.....	348
Table 98: Announcement Parameters.....	350
Table 99: Internal Options/General.....	352
Table 100: Internal Options/RTP Settings.....	354
Table 101: Internal Options/Payload.....	355
Table 102: NAS Settings.....	355
Table 103: Internal Options/Jitter Buffer.....	359
Table 104: NAS Settings.....	362
Table 105: SIP Settings/General.....	363
Table 106: SIP Settings/Misc.....	364
Table 107: SIP Settings/Session Timer.....	365
Table 108: SIP Settings/TCP and TLS.....	365
Table 109: SIP Settings/NAT.....	366
Table 110: SIP Settings/ToS.....	366
Table 111: IAX Settings/General.....	369
Table 112: IAX Settings/Registration.....	369
Table 113: IAX Settings/Static Defense.....	370
Table 114: PBX Interface Settings.....	371
Table 115: API Configuration Parameters.....	377
Table 116: New API Supported Queries.....	377
Table 117: API Configuration Parameters (Old).....	378
Table 118: CDR Real-time Output Settings.....	379
Table 119: SugarCRM Settings.....	384
Table 120: vTigerCRM Settings.....	386



Table 121: ZohoCRM Settings.....	388
Table 122: Salesforce Settings.....	389
Table 123: PMS Supported Features.....	392
Table 124: PMS Basic Settings.....	396
Table 125: PMS Wake up Service.....	398
Table 126: Create New Mini Bar.....	399
Table 127: Create New Maid.....	400
Table 128: Wakeup Service.....	403
Table 129: Max Wakeup Members.....	403
Table 130: Announcements Center Settings.....	406
Table 131: Group Settings.....	406
Table 132: QueueMetrics Configuration Parameters.....	410
Table 133: Trunk Status.....	412
Table 134: Extension Status.....	413
Table 135: Interface Status Indicators.....	414
Table 136: System Status→General.....	415
Table 137: System Status→Network.....	416
Table 138: Alert Events.....	418
Table 139: Alert Contact.....	422
Table 140: CDR Filter Criteria.....	423
Table 141: CDR Statistics Filter Criteria.....	431
Table 142: API Configuration Files.....	433
Table 143: User Management→Create New User.....	438
Table 144: Change Binding Email option.....	443
Table 145: Operation Log Column Header.....	445
Table 146: Network Upgrade Configuration.....	447
Table 147: Data Sync Configuration.....	454
Table 148: Cleaner Configuration.....	458
Table 149: USB/SD Card Files Cleanup.....	461
Table 150: Ethernet Capture.....	464



## Table of Figures

Figure 1: UCM6202 Front View.....	45
Figure 2: UCM6202 Back View.....	46
Figure 3: UCM6204 Front View.....	46
Figure 4: UCM6204 Back View.....	47
Figure 5: UCM6208 Front View.....	48
Figure 6: UCM6208 Back View.....	48
Figure 7: UCM6202 Web GUI Login Page.....	52
Figure 8: UCM6200 Setup Wizard.....	53
Figure 9: UCM6200 Web GUI Language.....	54
Figure 10: Web GUI Search Bar.....	55
Figure 11: UCM6202 Network Interface Method: Route.....	62
Figure 12: UCM6202 Network Interface Method: Switch.....	63
Figure 13: UCM6202 Network Interface Method: Dual.....	64
Figure 14: DHCP Client List.....	64
Figure 15: Add MAC Address Bind.....	65
Figure 16: Batch Add MAC Address Bind.....	65
Figure 17: UCM6200 Using 802.1X as Client.....	66
Figure 18: UCM6200 Using 802.1X EAP-MD5.....	66
Figure 19: UCM6204 Static Route Sample.....	68
Figure 20: UCM6204 Static Route Configuration.....	69
Figure 21: Create New Port Forwarding.....	70
Figure 22: UCM6200 Port Forwarding Configuration.....	71
Figure 23: GXP2160 Web Access using UCM6202 Port Forwarding.....	71
Figure 24: Open VPN® feature on the UCM6200.....	73
Figure 25: Register Domain Name on noip.com.....	74
Figure 26: UCM6200 DDNS Setting.....	74
Figure 27: Using Domain Name to Connect to UCM6200.....	75
Figure 28: Create New Firewall Rule.....	77
Figure 29: Configure Dynamic Defense.....	79
Figure 30: Fail2ban Settings.....	80
Figure 31: TLS Security.....	82
Figure 32: SSH Access.....	82
Figure 33: LDAP Server Configurations.....	83
Figure 34: Default LDAP Phonebook DN.....	84
Figure 35: Default LDAP Phonebook Attributes.....	84
Figure 36: LDAP Server→LDAP Phonebook.....	85
Figure 37: Add LDAP Phonebook.....	85
Figure 38: Edit LDAP Phonebook.....	85



Figure 39: Import Phonebook.....	86
Figure 40: Phonebook CSV File Format.....	86
Figure 41: LDAP Phonebook After Import.....	87
Figure 42: Export Selected LDAP Phonebook.....	87
Figure 43: LDAP Client Configurations.....	88
Figure 44: GXP2170 LDAP Phonebook Configuration.....	90
Figure 45: Set Time Manually.....	92
Figure 46: Create New Office Time.....	93
Figure 47: Settings→Time Settings→Office Time.....	94
Figure 48: Create New Holiday.....	95
Figure 49: Settings→Time Settings→Holiday.....	96
Figure 50: UCM6200 Email Settings.....	98
Figure 51: Email Templates.....	99
Figure 52: Conference Schedule Template.....	100
Figure 53: Email Send log.....	101
Figure 54: Email Logs.....	102
Figure 55: PBX Settings→Recordings Storage.....	103
Figure 56: Recordings Storage Prompt Information.....	103
Figure 57: Recording Storage Category.....	104
Figure 58: Zero Config Configuration Architecture for EndPoint Device.....	106
Figure 59: UCM6200 Zero Config.....	107
Figure 60: Auto Provision Settings.....	108
Figure 61: Auto Discover.....	110
Figure 62: Auto Discover other subnets.....	111
Figure 63: Discovered Devices.....	111
Figure 64: Device list - CSV file sample.....	111
Figure 65: Managing Discovered Devices.....	112
Figure 66: Global Policy Categories.....	114
Figure 67: Edit Global Template.....	122
Figure 68: Edit Model Template.....	125
Figure 69: OEM Models.....	126
Figure 70: Template Management.....	127
Figure 71: Upload Model Template Manually.....	127
Figure 72: Create New Device.....	128
Figure 73: Manage Devices.....	129
Figure 74: Edit Device.....	129
Figure 75: Edit Customize Device Settings.....	131
Figure 76: Modify Selected Devices - Same Model.....	132
Figure 77: Modify Selected Devices - Different Models.....	133
Figure 78: Device List in Zero Config.....	134
Figure 79: Zero Config Sample - Global Policy.....	135





Figure 80: Zero Config Sample - Device Preview 1.....	136
Figure 81: Zero Config Sample - Device Preview 2.....	137
Figure 82: Zero Config Sample - Device Preview 3.....	138
Figure 83: Create New Device.....	139
Figure 84: Manage Extensions.....	162
Figure 85: Export Extensions.....	164
Figure 86: Import Extensions.....	164
Figure 87: Import File.....	165
Figure 88: Import Error.....	171
Figure 89: Extension Details.....	172
Figure 90: E-mail Notification - Prompt Information.....	172
Figure 91: Account Registration Information and QR Code.....	173
Figure 92: LDAP Client Information and QR Code.....	173
Figure 93: Multiple Registrations per Extension.....	174
Figure 94: Extension - Concurrent Registration.....	174
Figure 95: SMS Message Support.....	175
Figure 96: Edit Extension Group.....	176
Figure 97: Select Extension Group in Outbound Route.....	177
Figure 98: UCM6200 FXO Tone Settings.....	182
Figure 99: UCM6200 PSTN Detection.....	182
Figure 100: UCM6200 PSTN Detection: Auto Detect.....	183
Figure 101: UCM6200 PSTN Detection: Semi-Auto Detect.....	183
Figure 102: Trunk Group.....	196
Figure 103: Trunk Group Configuration.....	197
Figure 104: DOD extension selection.....	198
Figure 105: Edit DOD.....	198
Figure 106: Fax Sending DOD.....	199
Figure 107: SLA Station.....	200
Figure 108: Enable SLA Mode for Analog Trunk.....	201
Figure 109: Analog Trunk with SLA Mode Enabled.....	201
Figure 110: SLA Example - SLA Station.....	202
Figure 111: SLA Example - MPK Configuration.....	202
Figure 112: Country Codes.....	207
Figure 113: Blacklist Import/Export.....	208
Figure 114: Create New PIN Group.....	209
Figure 115: PIN Members.....	209
Figure 116: Outbound PIN.....	210
Figure 117: CDR Record.....	210
Figure 118: Importing PIN Groups from CSV files.....	211
Figure 119: Incorrect CSV File.....	211
Figure 120: CSV File Format.....	212



Figure 121: CSV File Successful Upload.....	212
Figure 122: Inbound Route feature: Prepend.....	218
Figure 123: Inbound Route - Multiple Mode.....	219
Figure 124: Inbound Route - Multiple Mode Feature Codes.....	220
Figure 125: Inbound Route - Route-Level Mode.....	220
Figure 126: Global Inbound Mode.....	221
Figure 127: Inbound Mode - Default Mode.....	222
Figure 128: Inbound Mode - Mode 1.....	222
Figure 129: Import/Export Inbound Route.....	222
Figure 130: Blacklist Configuration Parameters.....	224
Figure 131: Blacklist csv File.....	224
Figure 132: Conference.....	228
Figure 133: Conference Invitation from Web GUI.....	229
Figure 134: Google Service Settings→OAuth2.0 Authentication.....	231
Figure 135: Google Service→New Project.....	232
Figure 136: Google Service→Create New Credential.....	232
Figure 137: Google Service→OAuth2.0 Login.....	233
Figure 138: Conference Schedule.....	237
Figure 139: Contact Group Parameters.....	238
Figure 140: Conference Recording.....	239
Figure 141: Conference Call Statistics.....	240
Figure 142: Conference Report on Web.....	240
Figure 143: Conference Report on CSV.....	240
Figure 144: Video Conference Basic settings.....	241
Figure 145: Video Conference Schedule.....	243
Figure 146: Enabling WebRTC Feature.....	244
Figure 147: Enabling WebRTC on Extensions.....	244
Figure 148: Grandstream Wave Interface.....	245
Figure 149: IPVT SIP Trunk page.....	246
Figure 150 - Peer Trunk to IPVT.....	246
Figure 151 - IPVT Mode.....	247
Figure 152 - IPVT Outbound Pattern.....	247
Figure 153 - IPVT Outbound Strip.....	248
Figure 154: Create New IVR.....	249
Figure 155: Key Pressing Events.....	252
Figure 156: Black/White List.....	253
Figure 157: Click on Prompt to Create IVR Prompt.....	254
Figure 158: Language Settings for Voice Prompt.....	255
Figure 159: Voice Prompt Package List.....	256
Figure 160: New Voice Prompt Language Added.....	256
Figure 161: Upload Voice Prompts Package.....	257



Figure 162: Record New IVR Prompt.....	258
Figure 163: Upload IVR Prompt.....	259
Figure 164: Download All Custom Prompt.....	259
Figure 165: Voicemail Settings.....	261
Figure 166: Voicemail Count.....	265
Figure 167: Voicemail Email Settings.....	266
Figure 168: Voicemail Group.....	267
Figure 169: Ring Group.....	268
Figure 170: Ring Group Configuration.....	270
Figure 171: Sync LDAP Server option.....	272
Figure 172: Manually Sync LDAP Server.....	273
Figure 173: Ring Group Remote Extension.....	274
Figure 174: Multicast Paging.....	275
Figure 175: 2-way Intercom.....	276
Figure 176: 1-way Paging.....	277
Figure 177: Announcement Paging.....	278
Figure 178: Private Intercom.....	280
Figure 179: Page/Intercom Group Settings.....	281
Figure 180: Schedule Paging/Intercom page.....	281
Figure 181: Creating a scheduled paging/intercom call.....	282
Figure 182: Call Queue.....	283
Figure 183: Static Agents limit.....	287
Figure 184: Agent Login Settings.....	288
Figure 185: Call Queue Statistics.....	291
Figure 186 : Automatic Download Settings - Queue Statistics.....	291
Figure 187: Agent details.....	292
Figure 188: Login Record.....	293
Figure 189: Pause Log.....	293
Figure 190: Switchboard Summary.....	294
Figure 191: Call Queue Switchboard.....	294
Figure 192: Queue Agent.....	296
Figure 193: Global Queue Settings.....	296
Figure 194: Edit Pickup Group.....	298
Figure 195: Edit Pickup Feature Code.....	299
Figure 196: Music On Hold Default Class.....	300
Figure 197: Play Custom Prompt.....	301
Figure 198: Information Prompt.....	301
Figure 199: Record Custom Prompt.....	302
Figure 200: Fax Settings.....	303
Figure 201: Configure Analog Trunk without Fax Detection.....	305
Figure 202: Configure Extension for Fax Machine: FXS Extension.....	306



Figure 203: Configure Extension for Fax Machine: Analog Settings.....	306
Figure 204: Configure Inbound Rule for Fax.....	307
Figure 205: Create Fax Extension.....	307
Figure 206: Inbound Route to Fax Extension.....	308
Figure 207: List of Fax Files.....	308
Figure 208: Fax Sending in Web GUI.....	309
Figure 209: Fax Send Progress.....	309
Figure 210: SIP Presence Configuration.....	311
Figure 211: SIP Presence Feature Code.....	312
Figure 212: Presence Status CDR.....	313
Figure 213: Edit Follow Me.....	314
Figure 214: Speed Dial Destinations.....	316
Figure 215: List of Speed Dial.....	316
Figure 216: Create New DISA.....	317
Figure 217: Emergency Number Configuration.....	320
Figure 218: 911 Emergency Sample.....	322
Figure 219: Create New Event List.....	325
Figure 220: Create Dial by Name Group.....	327
Figure 221: Configure Extension First Name and Last Name.....	328
Figure 222: Dial By Name Group In IVR Key Pressing Events.....	329
Figure 223: Dial By Name Group In Inbound Rule.....	330
Figure 224: Status→PBX Status→Active Calls - Ringing.....	332
Figure 225: Status→PBX Status→Active Calls – Call Established.....	332
Figure 226: Call Connection less than half hour.....	333
Figure 227: Call Connection between half an hour and one hour.....	333
Figure 228: Call Connection more than one hour.....	334
Figure 229: Configure to Monitor an Active Call.....	334
Figure 230: Enable/Disable Feature codes.....	341
Figure 231: Parking Lot.....	342
Figure 232: New Parking Lot.....	342
Figure 233: Monitored call park CID name.....	344
Figure 234: Download Recording File from CDR Page.....	345
Figure 235: Download Recording File from Recording Files Page.....	345
Figure 236: Enabling SCA option under Extension’s Settings.....	346
Figure 237: SCA Number Configuration.....	346
Figure 238: SCA Private Number Configuration.....	347
Figure 239: SCA Options.....	347
Figure 240: Announcement settings.....	350
Figure 241: Announcement.....	351
Figure 242: Record New Custom Prompt.....	356
Figure 243: Upload Custom Prompt.....	357



Figure 244: Download All Custom Prompt.....	357
Figure 245: SIP Trunk Prompt Tone.....	358
Figure 246: General call Failure Prompts.....	359
Figure 247: Settings→Recordings Storage.....	360
Figure 248: Recordings Storage Prompt Information.....	361
Figure 249: Recording Storage Category.....	361
Figure 250: Transparent Call-Info.....	368
Figure 251: FXS Ports Signaling Preference.....	371
Figure 252: FXO Ports ACIM Settings.....	371
Figure 253: DAHDI Settings.....	373
Figure 254: GDMS Developer Mode Button.....	374
Figure 255: GDMS API Credentials.....	374
Figure 256: GDMS Settings.....	375
Figure 257: UCM on GDMS.....	376
Figure 258: UCM SIP Extensions on GDMS.....	376
Figure 259: Upload Prompt User Configuration.....	380
Figure 260: CTI Server Listening port.....	382
Figure 261: SugarCRM Basic Settings.....	384
Figure 262: CRM User Settings.....	385
Figure 263: vTigerCRM Basic Settings.....	386
Figure 264: CRM User Settings.....	387
Figure 265: ZohoCRM Basic Settings.....	387
Figure 266: CRM User Settings.....	388
Figure 267: Salesforce Basic Settings.....	389
Figure 268: Salesforce User Settings.....	390
Figure 269: Enabling ACT! CRM.....	390
Figure 270: Enabling CRM on the User Portal.....	391
Figure 271: UCM & PMS interaction.....	393
Figure 272: UCM & HSC PMS interaction.....	394
Figure 273: UCM & Mitel PMS interaction.....	394
Figure 274: Enable PMSAPI.....	395
Figure 275: Create New Room.....	397
Figure 276: Room Status.....	397
Figure 277: Add batch rooms.....	398
Figure 278: Create New Wake Up Service.....	398
Figure 279: Wakeup Call executed.....	399
Figure 280: Create New Mini Bar.....	399
Figure 281: Create New Maid.....	400
Figure 282: Create New Consumer Goods.....	400
Figure 283: Mini Bar.....	401
Figure 284: Create New Wakeup Service.....	402



Figure 285: Wakeup Service Feature Code.....	404
Figure 286: Announcements Center.....	405
Figure 287: Announcements Center Group Configuration.....	407
Figure 288: Announcements Center Code Configuration.....	407
Figure 289: Announcements Center Example.....	408
Figure 290: QueueMetrics configuration.....	409
Figure 291: Status→PBX Status.....	411
Figure 292: Trunk Status.....	412
Figure 293: Extension Status.....	413
Figure 294: UCM6204 Interfaces Status.....	414
Figure 295: System Status→Storage Usage.....	417
Figure 296: System Status→Resource Usage.....	417
Figure 297: System Events→Alert Events Lists: Disk Usage.....	419
Figure 298: System Events→Alert Events Lists: External Disk Usage.....	419
Figure 299: System Events→Alert Events Lists: Memory Usage.....	420
Figure 300: System Events→Alert Events Lists: System Crash.....	420
Figure 301: System Events→Alert Log.....	421
Figure 302: Filter for Alert Log.....	421
Figure 303: CDR Filter.....	423
Figure 304: Call Report.....	425
Figure 305: Call Report Entry with Audio Recording File.....	427
Figure 306: Automatic Download Settings.....	427
Figure 307: CDR Report.....	428
Figure 308: Detailed CDR Information.....	428
Figure 309: Downloaded CDR File Sample.....	428
Figure 310: Downloaded CDR File Sample - Source Channel and Dest Channel 1.....	428
Figure 311: Downloaded CDR File Sample - Source Channel and Dest Channel 2.....	429
Figure 312: CDR Export File data.....	430
Figure 313: CDR Statistics.....	431
Figure 314: CDR→Recording Files.....	432
Figure 315: Edit User Information by Super Admin.....	434
Figure 316: User Portal Login.....	435
Figure 317: User Portal Layout.....	435
Figure 318: User Management Page Display.....	437
Figure 319: Create New User.....	437
Figure 320: User Management – New Users.....	438
Figure 321: Assign Backup permission to "Admin" users.....	439
Figure 322: General User.....	440
Figure 323: Create New Custom Privilege.....	441
Figure 324: Multiple User Operation Error Prompt.....	441
Figure 325: Change Password.....	442



Figure 326: Change Username.....	443
Figure 327: Change Binding Email.....	443
Figure 328: Login Timeout Settings.....	444
Figure 329: Operation Logs.....	445
Figure 330: Operation Logs Filter.....	446
Figure 331: Network Upgrade.....	447
Figure 332: Local Upgrade.....	448
Figure 333: Upgrading Firmware Files.....	449
Figure 334: Reboot UCM6200.....	449
Figure 335: Create New Backup.....	451
Figure 336: Backup / Restore.....	452
Figure 337: Local Backup.....	453
Figure 338: Data Sync.....	454
Figure 339: Restore UCM6200 from Backup File.....	455
Figure 340: Reset and Reboot.....	456
Figure 341: Cleaner.....	457
Figure 342: USB/SD Card Files Cleanup.....	461
Figure 343: UCM6202 Recovery Web Page.....	462
Figure 344: Recovery Mode.....	462
Figure 345: Ethernet Capture.....	464
Figure 346: Ping.....	465
Figure 347: Traceroute.....	465
Figure 348: Troubleshooting Analog Trunks.....	466
Figure 349: A Key Dial-up FXO.....	467
Figure 350: Service Check.....	468
Figure 351: Network Status.....	468



## DOCUMENT PURPOSE

The intent of this document is to provide device administrators an overview of the specifications and features of the Grandstream UCM6200 IPPBX system. To learn more about the UCM6200, please visit <http://www.grandstream.com/support> to download additional guides.

This guide covers following main topics:

- [Product overview](#)
- [Installation](#)
- [Getting started](#)
- [System settings](#)
- [Provisioning](#)
- [Extensions](#)
- [Extension groups](#)
- [Analog Trunks](#)
- [VoIP Trunks](#)
- [SLA station](#)
- [Call routes](#)
- [Conference](#)
- [Video Conference](#)
- [IVR](#)
- [Voice prompt](#)
- [Voicemail](#)
- [Ring group](#)
- [Paging and intercom group](#)
- [Call queue](#)
- [Pickup groups](#)
- [Music On Hold](#)
- [Fax Server](#)
- [Busy camp-on](#)
- [Presence](#)
- [Follow me](#)
- [Speed Dial](#)
- [DISA](#)
- [Emergency](#)
- [Callback](#)
- [BLF and event list](#)
- [Dial by name](#)
- [Active calls and monitor](#)
- [Call features](#)
- [Call recording](#)
- [CTI Server](#)
- [Asterisk manager interface \(AMI\)](#)
- [CRM integration](#)
- [PMS integration](#)
- [GDMS](#)
- [API](#)
- [QueueMetrics Integration](#)
- [Wakeup service](#)
- [Announcements center](#)
- [Status and reporting](#)
- [CDR \(Call Details Record\)](#)
- [User Portal](#)
- [Upgrading and maintenance](#)
- [Backup/restore](#)
- [Troubleshooting](#)





## CHANGE LOG

This section documents significant changes from previous versions of the UCM6200 user manuals. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

### Firmware Version 1.0.20.23

- No major changes.

### Firmware Version 1.0.20.22

- No major changes.

### Firmware Version 1.0.20.20

- Increased the maximum limit of IVR entries to 500 and inbound routes to 5000. [IVR] [Inbound Routes]
- Added support for full compliance with Kari's law and Ray Baum's act. [EMERGENCY]
- Increased the maximum number of outbound routes to 500. [Outbound Routes]
- Increased the maximum limit of SIP trunks to 200. [VoIP trunks]

### Firmware Version 1.0.20.17

#### Major Enhancements

- **[Announcement]**
  - Added Announcement feature. [Announcement]
- **[Backup/Restore]**
  - Added NAS as a backup/restore location. [Backup/Restore]
- **[Conference]**
  - Added conference contact groups. [Contact Group]
  - Added conference call statistics and reports. [Conference Call Statistics]
- **[CRM]**
  - Added support for Zoho CRM v2 API. [ZohoCRM]
- **[GDMS]**
  - Added GDMS SIP account syncing. [GDMS SETTINGS]
- **[HTTPS API]**
  - Add a new HTTPS API. [HTTPS API (New)]
- **[Maintenance]**



- Added the ability to download files on external storage from the USB Disk/SD Card File Management page. [USB/SD Card Files Cleanup]
- Added several more conditions for cleaning files and CDR. [Cleaner]
- **[Queue]**
  - Added the ability to customize the keys used for virtual queue. [Virtual Queue Callback Key Settings]
  - Improved queue statistics page. [Queue Statistics]
  - Added a Welcome Prompt option to the Edit Queue page. [Welcome Prompt]
  - Added QueueMetrics support. [QUEUEMETRICS INTEGRATION]
- **[Paging]**
  - Added Private Intercom paging type (GSC3510 only). [Configure Private Intercom]
- **[PMS]**
  - Added the ability to backup voicemail recordings upon guest check-out. [Connecting to PMS]
- **[Call Routing]**
  - Added Outbound Route CID option to the Extensions/Trunk→Outbound Routes page. [Outbound Route CID]
- **[VoIP Trunks]**
  - Users can now dial into IPVT meeting rooms via peer trunks. [IPVIDEOTALKMEETINGS]
  - Added support for DOD to be assigned to UCM's Fax Sending feature. [Direct Outward Dialing (DOD)]
- **[Zero Config]**
  - Auto Discover can now search for devices located on subnets added to the Subnet Whitelist. [Discovery]

### Other Enhancements

- **[Active Calls]**
  - Updated Parked calls which will now be displayed in the Active Calls page. [Active Calls Status]
- **[AMI]**
  - AMI username can no longer exceed 64 characters. [ASTERISK MANAGER INTERFACE (RESTRICTED ACCESS)]
- **[Announcement Center]**
  - Announcement Center group name and numbers now have a character limit of 64. [Group Settings]
- **[Basic Calls]**
  - Added International Call Prefix. [PBX SETTINGS]
  - After an attended transfer is completed, endpoints will now be updated with the CID of the other party instead of keeping the transferor's CID.
  - Added Block collect calls option in SIP settings→General settings. [Block Collect Calls]



- **[Backup/Restore]**
  - Added event logs and email notifications for scheduled SFTP backup and data sync results. [Alert Events List]
- **[Call Completion]**
  - Updated the number of CC agents to be limited by the extensions allowed number of concurrent registrations. [CC Mode] [BUSY CAMP-ON]
- **[CDR]**
  - Added Extension Groups as a CDR filter. [Table 140: CDR Filter Criteria]
- **[Conference]**
  - Added a field to configure the Kick Prompt Interval. [Kick Warning Interval]
  - Added support for Conference participant names to be announced when joining/leaving the conference even when the conference is on hold. [Announce Callers]
  - Extended the Meeting duration to be scheduled up to 8 hours. [Meeting Duration]
  - Conference extension number can no longer be used as the conference password if Strong Password is enabled. [Password]
- **[CRM]**
  - Added “Add Unknown Number” option for ACT! CRM. [ACT! CRM]
  - Added <https://www.zohoapis.eu> to the ZohoCRM Server Address dropdown list. [ZohoCRM]
- **[Dial by Name]**
  - Added the ability to upload a custom Dial by Name prompt. [Custom Prompt]
- **[DOD]**
  - Added DOD names and numbers character limit to 32. [Direct Outward Dialing (DOD)]
- **[Email Settings]**
  - Added new variable `#{CONFR_MEMBERS}`, which shows conference participant details. [Email Templates]
  - Added the following email type filters to the Email Send Log page: Send Fax, Call Queue Statistics, Conference Report. [Email Send Log]
- **[Emergency Calls]**
  - Users can now strip the same amount of numbers as the emergency number length itself. [Strip]
- **[Eventlist]**
  - Added a search bar for eventlists. [Event List]
- **[Extensions]**
  - The Extension and Auth ID fields now support plus signs (+). [Extension] [Auth ID]
  - Enable CC option added as batch edit extension option. [Enable CC]
  - Subnet mask can now be specified when configuring subnets that can register to the extension. [ACL Policy] [Local Network Address]
- **[GS Wave Web]**



- A conference participant's CID number will now be displayed instead of extension number if configured. [Wave WebRTC Video Calling & Conferencing]
- Improved extension searching when transferring. [Wave WebRTC Video Calling & Conferencing]
- An error message will now appear when the maximum number of allowed registrations has been reached. [Wave WebRTC Video Calling & Conferencing]
- Hosts can now generate a link that can be used to invite others to the conference. [Wave WebRTC Video Calling & Conferencing]
- **[IVR]**
  - Added support to upload more than one welcome prompt for IVR. [Welcome Prompt]
- **[LDAP]**
  - Plus signs (+) is now supported when creating contacts and receiving calls with "+" in the CID. [LDAP Phonebook]
  - LDAP Client CA Cert field no longer supports .ca file uploads. The following file types are supported: .crt .der .pem. [LDAP Client CA cert]
  - The Username field will no longer require "cn=" to precede username. [Username]
  - Character limit changed for the following fields:
    - LDAP Server→Root Password: 32 [LDAP Server Configurations]
    - LDAP Server→Root DN: 64 [LDAP Server Configurations]
    - Phonebook Download Configurations→Username: 64 [Username]
  - LDAP now supports downloading phonebooks from domains. [Server Address]
  - Added space support to the LDAP Client Username. [Username]
- **[Maintenance]**
  - Added Conference Call Statistics Report Cleaner. [Conference Call Statistics Report Cleaner]
- **[OpenVPN®]**
  - Added option Allow Weak SSL Ciphers. [Allow Weak SSL Ciphers]
- **[Paging]**
  - Added an option to allow scheduled paging to play during holidays. [Include Holidays]
  - "=" is no longer supported in the Alert-Info Header field. [Paging/Intercom Group Settings]
- **[Parking]**
  - Ring All Callback on Timeout will not be available if Forward to Destination on Timeout is enabled. [Ring All Callback on Timeout]
  - The Failover Destination field now has a character limit of 32. [Failover Destination]
- **[PMS]**
  - The Username and Guest Category Code columns in the Room Status page can now be sorted. [Room Status]
- **[Queue]**
  - Agents can now view recordings of their calls in their user portal. [USER PORTAL]
  - A queue can now have 3 chairmen to manage it. [Queue Chairman]
  - Callers can no longer use feature codes in established callbacks. [Enable Feature Codes]



- **[Ring Group]**
  - The character limit of the email address fields for ring group voicemail boxes has been changed from 256 to 128. [Enable Destination]
  - Added the option to skip busy members when receiving an incoming call. [Skip Busy Agent]
- **[SIP Settings]**
  - The TLS Self-signed CA field no longer supports .ca file uploads. The following file types are supported: .ca .crt .der .pem. [TLS Self-Signed CA]
- **[Recording]**
  - MoH will now be recorded when a party is on hold. [Call Recording]
- **[Routing]**
  - Users can now add patterns to the blacklist to restrict calls from several numbers. [Outbound Blacklist]
  - Inbound route imports/exports now have the following columns: Fax Detection, Fax Intelligent Routing, and Fax Destination. [Inbound Route: Import/Export Inbound Route]
  - Users can now export PIN Groups. [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. [PIN Groups with Privilege Level]
- **[System Information]**
  - Added duplex type and network port connection speed information to System Information→Network page. [Speed] [Duplex Mode]
- **[Time Settings]**
  - Users can now configure holidays for the next 4 years. [Year]
  - Added time condition Office Time and Out of Holiday. [Time Condition]
  - Default NTP server is now pool.ntp.org. [Remote NTP Server]
- **[User Management]**
  - Added the following custom privileges [Custom Privilege]:
    - 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. [Change Username]
  - Regular users will now be logged out automatically if an admin resets their extension. [USER PORTAL]
- **[Voicemail]**
  - Voicemail group member limit increased from 16 to 27. [Member]



- The voicemail system will now mention if a voicemail is from a ring group. [Enable Destination]
- **[Voice Prompt]**
  - Voice prompt package upload file size limit increased to 50MB. [Upload Language Package]
  - Custom voice prompt file name character limits for the Voice Prompt and Announcement Center pages have been changed to 100. [Upload Custom Prompt] [ANNOUNCEMENTS CENTER] [Announcement]
- **[VoIP Trunks]**
  - PAI headers now have a character limit of 64 and can only contain alphanumeric characters and/or special characters #\*-\_+. [PAI Header]
  - 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. [SIP URI Scheme When Using TLS]
  - Domain names can now be sent in the request URL by configuring the From Domain field. [From Domain]
- **[Security]**
  - Added support to specify minimum/maximum TLS version. [TLS Security]

## **Firmware Version 1.0.19.29**

- Updated extension DND response. [General Call Failure Tone]
- Added feature codes for remote management of extension call forwarding. [Remote Call Forward Enable]
- The wakeup service prompt now has an option to set a wakeup for the next day. [Wakeup Service]

## **Firmware Version 1.0.19.27**

- Added new option Email-to-Fax Subject Format to allow email subjects to consist of only the fax number. [Email-to-Fax Subject]
- Added new options Auto Record to automatically record emergency calls and Send Recording File to send these recordings to a configured email address. [Auto Record]
- Increased concurrent registration limit to 300 for UCM62xx's wave web interface. [Wave WebRTC Video Calling & Conferencing]
- Added Emergency Recordings page to display recordings of emergency calls. [EMERGENCY]
- Added a toggle checkbox to enable/disable Announcement Paging. [Enable]
- Queue position will now be announced to the caller upon entering the queue. [Enable Position Announcement]
- Added new option Enable Hold time Announcement to announce estimated wait times to caller. [Enable Hold time Announcement]
- Added a new option "Block the Backward Collect Call" to automatically block collect calls/reverse charge calls. [Block the Backward Collect Call]



- Added Chile time zone. [Auto Time Updating]
- Added IPVT Mode option to VoIP Peer Trunk [IPVT Mode].

### **Firmware Version 1.0.19.21**

- Optimized system performance for UCM6202 and UCM6204 models, which now support up to 50 and 75 concurrent calls, respectively. [Maximum Call Capacity]
- Added ACT! CRM support. [ACT! CRM]
- Added Email-to-Fax Blacklist/Whitelist. [Email-to-Fax Blacklist/Whitelist]
- Added digit prepending and stripping support for Emergency calls. [EMERGENCY]
- Added new functionalities to Grandstream Wave Web. [Wave WebRTC Video Calling & Conferencing]
- Added LDAPS protocol support. [Technical Specifications] [LDAP Server]
- Added support for .conf and .ovpn files for OpenVPN. [OpenVPN@]
- Added Announcement Paging type. [Configure AnnouncementPaging]
- Added PMS API as a PMS options. [PMSAPI]
- Added ability to clear agents call counters. [Reset Agent Call Counter - Enable]
- Added Set CallerID option. [Set Caller ID Info]
- Added ability to monitor and change the inbound mode of individual inbound routes and toggle/monitor them via BLF. [Inbound Mode] [Inbound Route: Route-Level Mode] [Inbound Route: Inbound Mode BLF Monitoring]
- Added search functionality to the web portal to quickly find settings. [Web GUI Search Bar]
- Added HT818 model template to Zero Config Models Templates. [PROVISIONING]
- Added the ability to associate differently named OEM models with their original GS models for provisioning purposes. [OEM device models]

### **Firmware Version 1.0.18.13**

- No major changes.

### **Firmware Version 1.0.18.12**

- CDR API configuration page moved from CDR to Value-Added Features.
- Added ability to upload voice prompt files via API.
- Added ability to transfer to custom numbers, not just extensions using Grandstream Wave Web [Wave WebRTC Video Calling & Conferencing]

### **Firmware Version 1.0.18.9**

- Added new AMI commands. [ASTERISK MANAGER INTERFACE (RESTRICTED ACCESS)]
- Added ability to customize the data columns included in exported CDR reports. [CDR Export Customization]
- Added ability to view a specified extension's membership in Call Queues/Ring Groups and other details. [Extension Details]



- Added ability to disable audio files for Music on Hold. [MUSIC ON HOLD]
- Added Timeout Destination and Ring-All Callback on Timeout options to Parking Lot page. [Parking Lot]
- Added inbound rule importing/exporting functionality. [Inbound Route: Import/Export Inbound Route]
- Added GS Wave WebRTC video calling and video conferencing functionality. [Wave WebRTC Video Calling & Conferencing]

## **Firmware Version 1.0.17.16**

- Added RTX codec support. [Technical Specifications] [Codec Preference]
- Emergency calls will no longer be logged into CRM servers. [CRM INTEGRATION]
- Added batch extension resetting functionality. [Batch Extension Resetting Functionality]
- Added FAX Resend Support. [Enable Fax Resend] [Max Resend Attempts]
- Fail2ban now supports up to 20 whitelist entries. [Fail2Ban Whitelist]
- Added FXO Frequency Tolerance option to Analog Hardware page. [FXO Frequency Tolerance]
- Added NAS support for call recording backup. [NAS]
- Added paging/intercom scheduling functionality. [Configure a Scheduled Paging/Intercom]
- Added Multicast Paging support. [PAGING AND INTERCOM GROUP]
- Added Failover Destination under call parking options. [Failover Destination]
- Added Timeout Callback Ringing All options under call parking options. [Ring All Callback on Timeout]
- Added ability to view CID of parked calls on VPKs/MPKs configured as monitored call park. [Monitor Call Park CID Name Information (GXP21xx Phones Only)]
- Added support for endpoint call forwarding under ring group. [Endpoint Call Forwarding Support]
- Added batch operations and searching functionality to the Inbound/Outbound Blacklist pages. Operations include deleting rules and importing/exporting entire blacklists. [Outbound Blacklist][Blacklist Configurations]
- Added ability to monitor and toggle inbound routing modes via BLF. [Inbound Mode BLF Monitoring]
- Added Shared Call Appearance functionality. [Shared Call Appearance]
- Added Forward HOLD Requests option under Misc page. [Forward HOLD Requests]
- Added support for Trunk Groups. [Trunk Groups]
- Added Forward Voicemail to Peered UCMs option. [Forward Voicemail to Peered UCMs]
- Added a new Voicemail Password field under Voicemail settings. [Voicemail Password]
- Added Catalan language support under voice prompt. [VOICE PROMPT]
- Added DOD Name field under VoIP trunk settings. [Direct Outward Dialing (DOD)]
- Changed the default value of Max Wait Time to 60. [Max Wait Time]
- Added “SCA” action type to filtering CDR options. [Action Type]
- Added a prompt asking whether to delete all recording files or not on CDR. [CDR]
- Added a 64-character limit to Conference → Extension field. [Extension]
- Added a 3-character limit to Extension Incrementation field. [Extension Incrementation]
- Added a maximum limit of 500 entries to the IVR blacklist and whitelist. [Black/White List in IVR]
- Added character restrictions to LDAP Number Attributes field to ensure correct input. [LDAP Client Configurations]
- Added maximum limit of 99,999 to parking timeout. [Parking Timeout]





- Added a 63-character limit to the Room Number field. [Room Number]
- Editing an already executed wakeup service will automatically change the service's status to "Programmed". [Action Status]
- Adding UCM disconnection when an SSH client changes the password for the connected user.
- LCD display will show "Recovery Mode" instead of "No Provision" when the UCM is in recovery mode. [System Recovery]
- Capture files saved on external devices will have "capture" prepended to filenames. [Ethernet Capture]
- Added dashes and + characters to Fax, Home Number, and Mobile Phone Number fields.
- Updated external number to accept only letters, numbers and special characters. [External Number]
- Added a 64-character limit to Custom Presence Status field. [Custom Presence Status]
- Increased the character limit for provider name field to 64 characters during creation. [Provider Name]
- Updated From Domain field to accept special characters. [From Domain]
- Username Prompt filename character limit changed to 18. [Username Prompt Customization]
- Increased the character limit for the wakeup service name field to 64 characters. [Wakeup Service]
- Users can now upload custom prompts directly to pages without having to get redirected to the Voice Prompt → Custom Prompt page. [Custom Prompt]
- Added character restrictions to device firmware file names. [firmware file name]
- Added Email-to-Fax support. [Email-to-Fax]

### **Firmware Version 1.0.16.20**

- Added an option to enable/disable remote voicemail access function. [Voicemail Remote Access]

### **Firmware Version 1.0.16.18**

- Added support for emergency calls. [EMERGENCY]
- Added support for vTigerCRM [vTigerCRM]
- Added support for ZohoCRM [ZohoCRM v2 ]
- Added support for HSC PMS [HSC PMS]
- Added support for Direct Callback. [Direct Callback] [Direct Callback] [Direct Callback]
- Added option to disable call waiting under extensions. [Enable Call Waiting]
- Added option to passthrough SIP PAI Header. [Passthrough PAI Header]
- Added option to copy device settings in zero config. [Managing discovered devices]
- Added support for call failure prompts customization. [General Call Failure Tone]
- Added option to automatically request HTTP server certificate. [Certificate Options]
- Added support to include more menus on user custom privilege. [Custom Privilege]
- Added possibility to delete CDR records based on search result. [CDR]
- Added option to send caller to specific destination when queue is empty. [CALL QUEUE]
- Added XML file type support for importing phonebooks. [LDAP Phonebook]
- Added ability to customize the LDAP name and number attributes in the Phonebook Download Configurations page [LDAP Client Configurations]
- Added option to download Call Queue Statistics. [Queue Statistics]



- Added option to choose DNS mode during calls. [DNS mode]
- Added ability to batch add devices by importing a CSV file [PROVISIONING]
- Added a Copy Device Configuration option that allows users to copy the configuration of one device to another device [PROVISIONING]
- Added ability to divide a parking lot extension into multiple lots [Parking Lot]
- Added option to reset extension settings. [Reset single extension]

### **Firmware Version 1.0.15.16**

- Added support to announce name in Dial By Name feature. [DIAL BY NAME]
- Increased maximum number of call queue static agents. [Static Agents limitation]
- Added Queue Log Cleaner. [Cleaner]
- Added operation logs details and remarks. [Operation Log]
- Added extension level voicemail-to-Email setting. [Send Voicemail to Email] [Keep Voicemail after Emailing]
- Added support for reset certificate. [Reset Certificates]
- Added support for GXP21xx color phone queue login/logout softkey. [Enable Agent Login]
- Added support to add comments to inbound/outbound route patterns. [Pattern] [Pattern]
- Added PPI mode option under SIP trunk advanced settings [PPI Mode]
- Added ability to import PIN groups from CSV files. [Importing PIN Groups from CSV files]
- Added support for wakeup groups. [WAKEUP SERVICE]
- Added support for SYN Flood defense. [SYN-Flood Defense Enable]
- Added Queue Log option to Backup/Restore page. [Backup/Restore]
- Modified Switchboard UI to offer easier access to call Options. [Switchboard]
- Further optimized Call Queue Statistics page to provide a more user-friendly experience and improve performance. [Queue Statistics]
- Added image uploading support to Email templates [Email Templates]
- Added IPv6 gateway support. [Static Routes]
- Improved Voice Message Responses for failed SIP trunk calls. [voice message responses]
- Added Fail2Ban to support TCP/TLS beside the already-supported UDP. [Fail2ban]
- Restored ability to sort the ringing order of Follow Me numbers. [FOLLOW ME]
- Added option to restrict users from changing their SIP credentials through user portal [Consumer]
- Added option to enable or disable fax-to-mail feature on extension level. [Fax to Email]
- Added IP address whitelist to web GUI configuration. [Enable IP Whitelist]
- Added CDR records separation [CDR separation]

### **Firmware Version 1.0.14.24**

- Added protection to prevent HTTP rogue login.

### **Firmware Version 1.0.14.23**

- Restored ability to view voicemail count in the Extension/Trunk overview. [Extension Voicemail Count]



- Restored the ability to set custom numbers for call forwarding settings. [Call Forward Unconditional] [Call Forward No Answer] [Call Forward Busy]
- Restored previous format for entering multiple dial plans (one pattern per line) for inbound/outbound rules. [Pattern] [Pattern]
- Restored Zero Config's sorting by column and introduced a search bar. [Managing discovered devices]

### **Firmware Version 1.0.14.21**

- Added support for SIP Presence. [PRESENCE]
- Added support for Call Center feature/Virtual Call Queue. [Call Center Settings and Enhancements]
- Added support for Call Queue position announcement. [Call Center Settings and Enhancements]
- Added support for Call Queue Statistics. [Queue Statistics]
- Added support for Call Queue Auto-Fill. [Queue Auto fill enhancement]
- Added switchboard for call queue monitoring. [Switchboard]
- Added ability to restore blind transfer call to transferrer. [Allow callback when blind transfer fails]
- Added support for external disk cleaner. [System Cleanup/Reset]
- Added option to enable DOD when call is being diverted/forwarded. [Use callee DOD on FWD or Ring Simultaneously]
- Change follow me settings to extension level settings. [FOLLOW ME]
- Added support for call forward whitelist. [FWD Whitelist]
- Added Fail2Ban defense from web login attack. [Login Attack Defense]
- Added limitation for maximum number of call queue static agents. [Static Agents limitation]
- Added support for wakeup service module in Custom privilege. [Custom Privilege]
- Added IPv6 support for T.38.
- Added DAHDI settings. [DAHDI Settings]
- Added ability to pass through SIP Call-Info header to support GXP phone JPEG\_Over\_HTTP with encryption and authentication to open door for GDS3710. [Transparent Call-Info header]

### **Firmware Version 1.0.13.14**

- Added extension whitelist/blacklist for IVR dialing[IVR]
- Added ability to include DOD in PPI Header for SIP trunk. [PPI Mode]
- Added advanced IPv6 support including IPv6-to-IPv4 SIP calls, IPv6 router, IPv6 iptables/Static defense.
- Added ability to customize PAI Header. [PAI Header]
- Added blacklist for outbound calls. [Outbound Blacklist]
- Added support to upload/download MOH package from Web GUI. [Music On Hold]
- Added support to download custom prompts from Web GUI. [Download All Custom Prompt]
- Added option to configure prompt timeout in Dial By Name. [DIAL BY NAME]
- Added description field in ZeroConfig settings to configure Softkey/Line/MPK for GXP series phones. [PROVISIONING]
- Improved seamless transfer privilege control. [Seamless transfer privilege control]
- Added RTP Keep-alive support. [RTP Keep-alive]



- Added Email Send Log. [Email Send Log]
- Added support for Mitel simulation/protocol interfaces for PMS module. [PMS]
- Added support for up to 10 failover trunks. [Use Failover Trunk]

## **Firmware Version 1.0.12.19**

- Added support for binding a mobile phone number to extension. [Mobile Phone Number]
- Added support OPUS codec.
- Added support call-barging privilege settings based on extensions. [Monitor privilege control]
- Added support for Seamless Transfer. [Seamless Transfer]
- Added support for Custom Call-Info for Auto Answer. [Custom Call-info for AutoAnswer]
- Added support for DND Whitelist. [Do Not Disturb]
- Add the Field Description on Softkey, Line keys and MPK from Zero Config.
- Added support to select interval for numbers on Batch add extension. [Extension Incrementation]
- Added support for Batch Add CallerID Number. [CallerID Number]
- Added support for Search Extensions Using CallerID Name.
- Added support to Enable/Disable Inbound and Outbound Route. [Disable This Route/Disable This Route]
- Added support for Outbound Route Time Condition. [Time Condition]
- Added support for IPv6. [IPv6 Address]
- Added Support for MTU configurable. [MTU]
- Added support of CRM. [CRM]
- Added support for Custom Privilege in User Management. [Custom Privilege]
- Added Hotline support for FXS Extension. [Hotline]
- Added support for Separate Wakeup Service. [WAKEUP SERVICE]
- Added ability to provision phones from different network subnets using zero config. [Subnet Whitelist]
- One-key-dial is replaced by Speed Dial to support more than one digit. [SPEED DIAL]
- Added Append extension number in the end of DOD. [Direct Outward Dialing (DOD)]
- Support Japan CID NTT Detect.
- Added support for Ethernet Capture Auto Sync to SFTP Server. [Enable SFTPData Sync]
- Added support for Ethernet Capture saved to External Storage Device. [Storage to External Device]
- Added support for Disable Extension Range on the Setup Wizard. [Setup Wizard]
- Added more support for Port Forwarding. [Port Forwarding]
- Added support for USB/SD Card Files Cleanup. [USB/SD Card Files Cleanup]
- Added support for A Key Dial-up FXO. [A key Dial-up FXO]
- Added support for ACIM Detect Option for FXO. [INTERFACESETTINGS]
- Added support for some special character on the file name of FW. [Upgrading Via Local Upload]
- Added more search criteria of CDR. [CDR]
- Added support of "Allow outgoing calls if registration failure" for register trunks. [Allow outgoing calls if registration failure]
- Added support for music on hold playback from webGUI. [Music On Hold]
- Added support to enable delete recording files for user privilege. [Consumer]
- Added support disk Inode usage in "Storage Usage" page. [Storage Usage]



- Added support for Ring Group/Call Queue/IVR Display Option for Caller ID. [Replace Display Name | Replace | Replace Caller ID]
- Added support for compatibility between backup package from UCM61xx and UCM62xx. [Backup/Restore]
- Added support to Detect talking users in conference. [CONFERENCE]
- Added Support of Mini Bar for PMS. [Mini Bar]

### **Firmware Version 1.0.11.27**

- Added ability to sort extension status on Web GUI.
- Added one click enable / disable feature code. [Feature Codes]
- Added Uruguay time zone support. [Auto time updating]
- Added distinctive ring tone support. [Configure Call Queue] [Configure IVR] [Create New SIP Extension]
- Added special character support for SFTP client account. [Data Sync]
- Added destination directory support for data sync. [Data Sync]
- Added ring group music on hold. [Configure Ring Group]
- Added CDR multi-email / time condition support. [CDR]
- Added blacklist anonymous call block. [Blacklist Configurations]
- Added ability to sort selected extension in Eventlist. [Event List]
- Added Banned User list for Web GUI login attempts. [Login Settings]
- Added Email template support. [Email Templates]
- Added outbound route country restriction.
- Added external disk usage alert option. [Alert Events List]
- Added range IP input support for dynamic defense white list. [Dynamic Defense]
- Added blacklist support for Fail2ban. [Fail2ban]
- Added ability to reboot device from zero config page. [Discovery]
- Added GXP1628B template for zero config. [Model Update]
- Added PIN group support. [PIN Groups]
- Added PMS support. [PMS]
- Added call queue custom prompt support. [Configure Call Queue]
- Added call queue retry time support. [Configure Call Queue]
- Added Support for DHCP Client List. [DHCP Client List]

### **Firmware Version 1.0.0.7**

- This is the initial version.



## WELCOME

Thank you for purchasing Grandstream UCM6200 series IP PBX appliance. The UCM6200 series IP PBX appliance is designed to bring enterprise-grade voice, video, data, and mobility features to small-to-medium businesses (SMBs) in an easy-to-manage fashion. This IP PBX series allows businesses to unify multiple communication technologies, such comprehensive voice, video calling, video conferencing, video surveillance, data tools and facility access management onto one common network that that can be managed and/or accessed remotely. The UCM6200 series supports a dual core 1GHz ARM Cortex™ A9 and 400Mhz VINETIC™ A8 processors, 1GB RAM and 4GB flash. The secure and reliable UCM6200 series delivers enterprise-grade features without any licensing fees, costs-per-feature or recurring fees.

---

 **Caution:**

Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

 **Warning:**

Please do not use a different power adaptor with the UCM6200 as it may cause damage to the product and void the manufacturer warranty.

---

This document is subject to change without notice. The latest electronic version of this user manual is available for download here:

<http://www.grandstream.com/support>

Reproduction or transmittal of the entire or any part, in any form or by any means, electronic or print, for any purpose without the express written permission of Grandstream Networks, Inc. is not permitted.



## PRODUCT OVERVIEW

### Technical Specifications

The following table resumes all the technical specifications including the protocols / standards supported, voice codecs, telephony features, languages and upgrade/provisioning settings for UCM6200 series.

**Table 1: Technical Specifications**

Interfaces	
<b>Analog Telephone FXS Ports</b>	2x RJ11 ports with lifeline support Each port supports 2 REN
<b>PSTN Line FXO Ports</b>	<ul style="list-style-type: none"> <li>UCM6202: 2 ports</li> <li>UCM6204: 4 ports</li> <li>UCM6208: 8 ports</li> </ul>
<b>Network Interfaces</b>	<ul style="list-style-type: none"> <li>UCM6202/6204/6208: Dual Gigabit RJ45 ports with integrated PoE Plus (IEEE 802.3at-2009)</li> </ul>
<b>NAT Router</b>	Yes
<b>Peripheral Ports</b>	USB, SD
<b>LED Indicators</b>	Power/Ready, Network, PSTN Line, USB, SD
<b>LCD Display</b>	128x32 graphic LCD with DOWN and OK button
<b>Reset Switch</b>	Yes
Voice/Video Capabilities	
<b>Voice-over-Packet Capabilities</b>	128ms tail-length carrier-grade Line Echo Cancellation with NLP Packetized Voice Protocol Unit, dynamic jitter buffer, modem detection, and auto-switch to G.711.
<b>Voice and Fax Codecs</b>	G.711 A-law/U-law, G.722, G.723.1 5.3K/6.3K, G.726, G.729A/B, iLBC (30ms only), GSM, AAL2-G.726-32, RTX, ADPCM; T.38
<b>Video Codecs</b>	H.264, H.265, H.263, H.263+, VP8
<b>QoS</b>	Layer 3 QoS, Layer 2 QoS
Signaling and Control	
<b>DTMF Methods</b>	Inband, RFC4733, and SIP INFO
<b>Provisioning Protocol and Plug-and-Play</b>	TFTP/HTTP/HTTPS, auto-discovery and auto-provisioning of Grandstream IP endpoints via ZeroConfig (DHCP Option 66/multicast SIP SUBSCRIBE/mDNS), Eventlist between local and remote trunk
<b>Network Protocols</b>	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, SIP (RFC3261), STUN, SRTP, TLS, LDAP/LDAPS
<b>Disconnect Methods</b>	Call Progress Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect, Busy Tone





Security	
<b>Media</b>	SRTP, TLS1.2, HTTPS, SSH
Physical	
<b>Universal Power Supply</b>	<ul style="list-style-type: none"> <li>Output: 12VDC, 1.5A</li> <li>Input: 100-240VAC, 50-60Hz</li> </ul>
<b>Dimensions</b>	<ul style="list-style-type: none"> <li>UCM6202/6204: 226mm (L) x 155mm (W) x 34.5mm (H)</li> <li>UCM6208: 440mm (L) x 185mm (W) x 44mm (H)</li> </ul>
<b>Environmental</b>	<ul style="list-style-type: none"> <li>Operating: 32 - 104°F / 0 - 40°C, 10-90% (non-condensing)</li> <li>Storage: 14 - 140°F / -10 - 60°C</li> </ul>
<b>Mounting</b>	<ul style="list-style-type: none"> <li>UCM6202/6204: Wall mount and Desktop</li> <li>UCM6208: Rack mount and Desktop</li> </ul>
<b>Weight</b>	<ul style="list-style-type: none"> <li>UCM6202: Unit weight 0.51kg, Package weight 0.94kg</li> <li>UCM6204: Unit weight 0.51kg, Package weight 0.94kg</li> <li>UCM6208: Unit weight 2.23kg, Package weight 3.09kg</li> </ul>
Additional Features	
<b>Multi-language Support</b>	English/Simplified Chinese/Traditional Chinese/Spanish/French/Portuguese/German/Russian/Italian/Polish/Czech for Web GUI; Customizable IVR/voice prompts for English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic; Customizable language pack to support any other languages
<b>Caller ID</b>	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 - BT
<b>Polarity Reversal/ Wink</b>	Yes, with enable/disable option upon call establishment and termination
<b>Call Center</b>	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability busy level, in-queue announcement
<b>Customizable Auto Attendant</b>	Up to 5 layers of IVR (Interactive Voice Response)
<b>Maximum Call Capacity</b>	<ul style="list-style-type: none"> <li><b>UCM6202:</b> Concurrent audio calls up to 30, concurrent WebRTC calls up to 25.</li> <li><b>UCM6204:</b> Concurrent audio calls up to 45, concurrent WebRTC calls up to 35.</li> <li><b>UCM6208:</b> Concurrent audio calls up to 100, concurrent WebRTC calls up to 50. Or up to 66% performance if calls are SRTP encrypted</li> </ul>
<b>SIP Devices</b>	<ul style="list-style-type: none"> <li>UCM6208 up to 100 registered SIP endpoints</li> <li>UCM6204 up to 300 registered SIP endpoints.</li> <li>UCM6208 up to 800 registered SIP endpoints.</li> </ul>
<b>Conference Rooms</b>	<ul style="list-style-type: none"> <li>UCM6202/6204: Up to 3 password-protected conference rooms allowing up to 25 simultaneous PSTN or IP participants</li> </ul>





<b>Call Features</b>	<ul style="list-style-type: none"> <li>• UCM6208: Up to 6 password-protected conference rooms allowing up to 32 simultaneous PSTN or IP participants</li> </ul> <p>Call park, call forward, call transfer, DND, ring/hunt group, paging/intercom and etc.</p>
<b>Compliance</b>	<ul style="list-style-type: none"> <li>• FCC: Part 15 (CFR 47) Class B, Part 68</li> <li>• CE: EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1, TBR21, RoHS</li> <li>• A-TICK: AS/NZS CISPR 22 Class B, AS/NZS CISPR 24, AS/NZS 60950, AS/ACIF S002 and ITU-T K.21 (Basic Level)</li> <li>• UL 60950 (power adapter)</li> </ul>



**Note:**

- UCM6200 FXS ports lifeline functionality:  
 The UCM6200 FXS interfaces are metallic through to the FXO interfaces. If there is power outage, FXS1 port will fail over to FXO 1 port, FXS 2 port will fail over to FXO 2 port. The user can still access the PSTN connected with the FXO interfaces from FXS interfaces.
- 



## INSTALLATION

Before deploying and configuring the UCM6200 series, the device needs to be properly powered up and connected to a network. This section describes detailed information on installation, connection and warranty policy of the UCM6200 series.

### Equipment Packaging

Table 2: UCM6200 Equipment Packaging

<b>Main Case</b>	1
	X
<b>Power Adaptor</b>	1
	X
<b>Ethernet Cable</b>	1
	X
<b>Quick Installation Guide</b>	1
	X
<b>GPL License</b>	1
	X

### Connect Your UCM6200

#### Connect The UCM6202

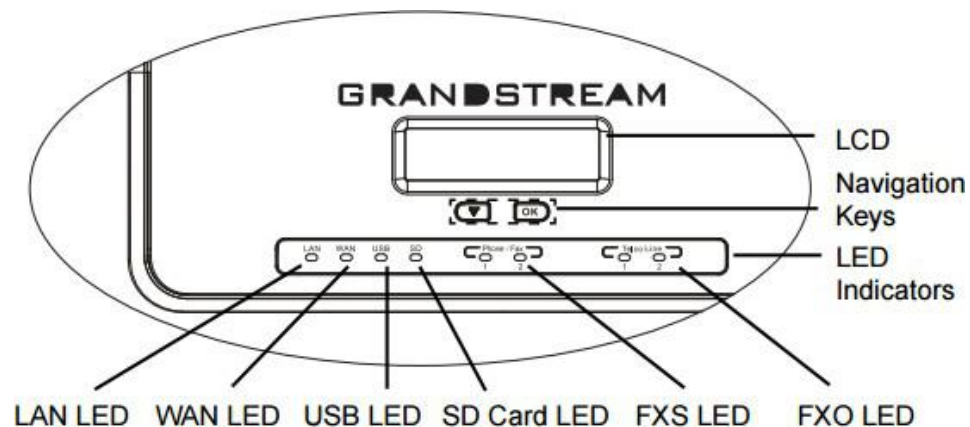


Figure 1: UCM6202 Front View



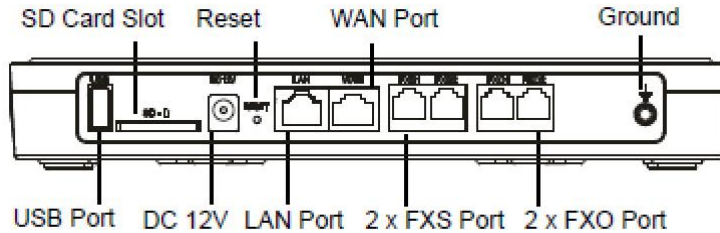


Figure 2: UCM6202 Back View

To set up the UCM6202, follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the WAN port of the UCM6202.
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub.
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the UCM6202. Insert the main plug of the power adapter into a surge-protected power outlet.
4. Wait for the UCM6202 to boot up. The LCD in the front will show the device hardware information when the boot process is done.
5. Once the UCM6202 is successfully connected to network, the LED indicator for WAN in the front will be solid green and the LCD will display the IP address.
6. (Optional) Connect PSTN lines from the wall jack to the FXO ports; connect analog lines (phone and Fax) to the FXS ports.

**Note:** The ground screw needs to be connected.

## Connect The UCM6204

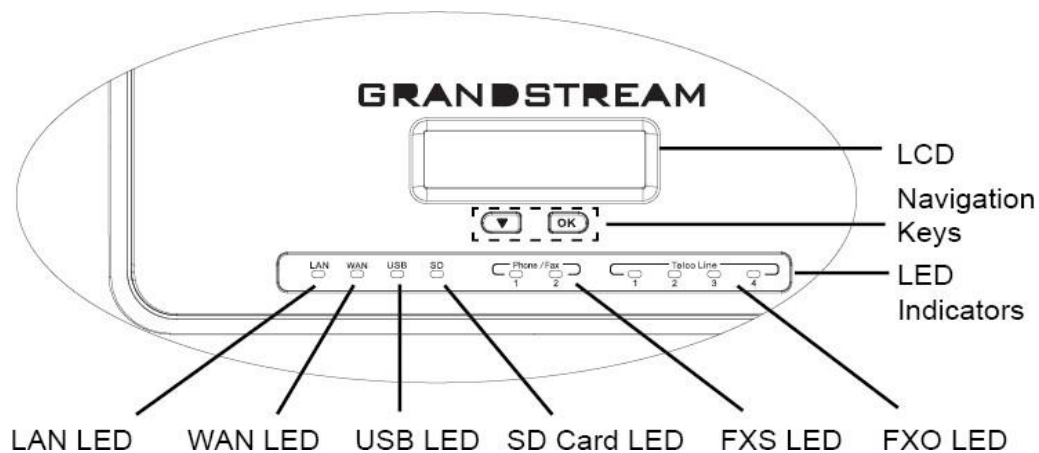
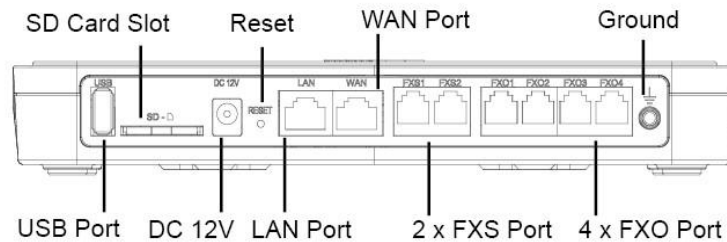


Figure 3: UCM6204 Front View





**Figure 4: UCM6204 Back View**

To set up the UCM6204, follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the WAN port of the UCM6204.
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub.
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the UCM6204. Insert the main plug of the power adapter into a surge-protected power outlet.
4. Wait for the UCM6204 to boot up. The LCD in the front will show the device hardware information when the boot process is done.
5. Once the UCM6204 is successfully connected to network, the LED indicator for WAN in the front will be solid green and the LCD will display the IP address.
6. (Optional) Connect PSTN lines from the wall jack to the FXO ports; connect analog lines (phone and Fax) to the FXS ports.
7. **Note:** The ground screw needs to be connected.

## Connect The UCM6208

To set up the UCM6208, follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the WAN port of the UCM6208.
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub.
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the UCM6208. Insert the main plug of the power adapter into a surge-protected power outlet.
4. Wait for the UCM6208 to boot up. The LCD in the front will show the device hardware information when the boot process is done.
5. Once the UCM6208 is successfully connected to network, the LED indicator for NETWORK in the front will be solid green and the LCD will display the IP address.
6. (Optional) Connect PSTN lines from the wall jack to the FXO ports; connect analog lines (phone and Fax) to the FXS ports.

**Note:** The ground screw needs to be connected.



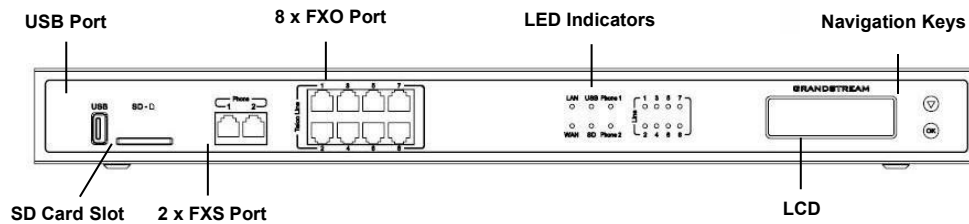


Figure 5: UCM6208 Front View



Figure 6: UCM6208 Back View

## Safety Compliances

The UCM6200 series IP PBX complies with FCC/CE and various safety standards. The UCM6200 power adapter is compliant with the UL standard. Use the universal power adapter provided with the UCM6200 package only. The manufacturer's warranty does not cover damages to the device caused by unsupported power adapters.

## Warranty

If the UCM6200 series IP PBX was purchased from a reseller, please contact the company where the device was purchased for replacement, repair or refund. If the device was purchased directly from Grandstream, contact our Technical Support Team for an RMA (Return Materials Authorization) number before the product is returned. Grandstream reserves the right to remedy warranty policy without prior notification.

---

 **Warning:**

Use the power adapter provided with the UCM6200 series IP PBX. Do not use a different power adapter as this may damage the device. This type of damage is not covered under warranty.

---



## GETTING STARTED

To get started with the UCM6200 setup process, use the following available interfaces: LCD display, LED indicators, and web portal.

- The LCD display shows hardware, software, and network information and can be navigated via the DOWN and OK buttons next to the display. From here, users can configure basic network settings, run diagnostic tests, and factory reset.
- The LED indicators at the front of the device provides interface connection and activity status.
- The web portal (may also be referred to as web UI in this guide) is the primary method of configuring the UCM.

This section will provide step-by-step instructions on how to use these interfaces to quickly set up the UCM and start making and receiving calls with it.

### Use the LCD Menu

- **Idle Screen**  
Once the device has booted up completely, the LCD will show the UCM model, hardware version (e.g., V1.4A), and IP address. Upon menu key press timeout (15 seconds), the screen will default back to this information. Pressing the DOWN button will show the system time.
- **Menu**  
Pressing the OK button will show the main menu. All available menu options are found in [Table 3: LCD Menu Options].
- **Menu Navigation**  
Pressing the DOWN button will scroll through the menu options. Press the OK button to select an option.
- **Exit**  
Selecting the Back option will return to the previous menu. For the Device Info, Network Info, and Web Info screens that have no Back option, pressing the OK button will return to the previous menu.
- **LCD Backlight**  
The LCD backlight will turn on upon button press and will go off when idle for 30 seconds.



The following table summarizes the layout of the LCD menu of UCM.

**Table 3: LCD Menu Options**

<b>View Events</b>	<ul style="list-style-type: none"> <li>• <b>Critical Events</b></li> <li>• <b>Other Events</b></li> </ul>
<b>Device Info</b>	<ul style="list-style-type: none"> <li>• <b>Hardware:</b> Hardware version number</li> <li>• <b>Software:</b> Software version number</li> <li>• <b>P/N:</b> Part number</li> <li>• <b>WAN MAC:</b> WAN side MAC address</li> <li>• <b>LAN MAC:</b> LAN side MAC address</li> <li>• <b>Uptime:</b> System uptime</li> </ul>
<b>Network Info</b>	<ul style="list-style-type: none"> <li>• <b>WAN Mode:</b> DHCP, Static IP, or PPPoE</li> <li>• <b>WAN IP:</b> IP address</li> <li>• <b>WAN Subnet Mask</b></li> <li>• <b>LAN IP:</b> IP address</li> <li>• <b>LAN Subnet Mask</b></li> </ul>
<b>Network Menu</b>	<ul style="list-style-type: none"> <li>• <b>WAN Mode:</b> Select WAN mode as DHCP, Static IP or PPPoE</li> <li>• <b>Static Route Reset:</b> Select this to reset static route settings.</li> </ul>
<b>Factory Menu</b>	<ul style="list-style-type: none"> <li>• <b>Reboot</b></li> <li>• <b>Factory Reset</b></li> <li>• <b>LCD Test Patterns</b> Press DOWN and OK buttons to scroll through and select different LCD patterns to test. Once a test is done, press the OK button to return to the previous menu.</li> <li>• <b>Fan Mode</b> Select Auto or On.</li> <li>• <b>LED Test Patterns</b> All On, All Off, and Blinking are the available options. Selecting Back in the menu will revert the LED indicators back to their actual status.</li> <li>• <b>RTC Test Patterns</b> Select either 2022-02-22 22:22 or 2011-01-11 11:11 to start the RTC (Real-Time Clock) test pattern. Check the system time from either the LCD idle screen or in the web portal System Status-&gt;System Information-&gt;General page. To revert back to the correct time, manually reboot the device.</li> <li>• <b>Hardware Testing</b></li> </ul>



	Select Test SVIP to verify hardware connections within the device. The result will display on the LCD when the test is complete.
<b>Web Info</b>	<ul style="list-style-type: none"> <li>• <b>Protocol:</b> Web access protocol (HTTP/ HTTPS). HTTPS is used by default.</li> <li>• <b>Port:</b> Web access port number, which is 8089 by default.</li> </ul>
<b>SSH Switch</b>	<ul style="list-style-type: none"> <li>• <b>Enable SSH</b></li> <li>• <b>Disable SSH</b></li> </ul> SSH access is disabled by default

## Use the LED Indicators

The UCM6200 has LED indicators in the front to display connection status. The following table shows the status definitions.

Table 4: UCM6202/UCM6204 LED Indicators









LED Indicator	LED Status
LAN	
WAN	
USB	 <b>Solid:</b> Connected_
SD	 <b>Flashing:</b> Data Transferring
FXS (Phone/Fax)	 <b>OFF:</b> Not Connected
FXO (Telco Line)	

Table 5: UCM6208 LED Indicators

LED	LED Status
NETWORK	 <b>Solid:</b> Connected  <b>OFF:</b> Not Connected
ACT	
USB	 <b>Solid:</b> Connected
SD	 <b>Flashing:</b> Data Transferring
Phone (FXS) Line (FXO)	 <b>OFF:</b> Not Connected

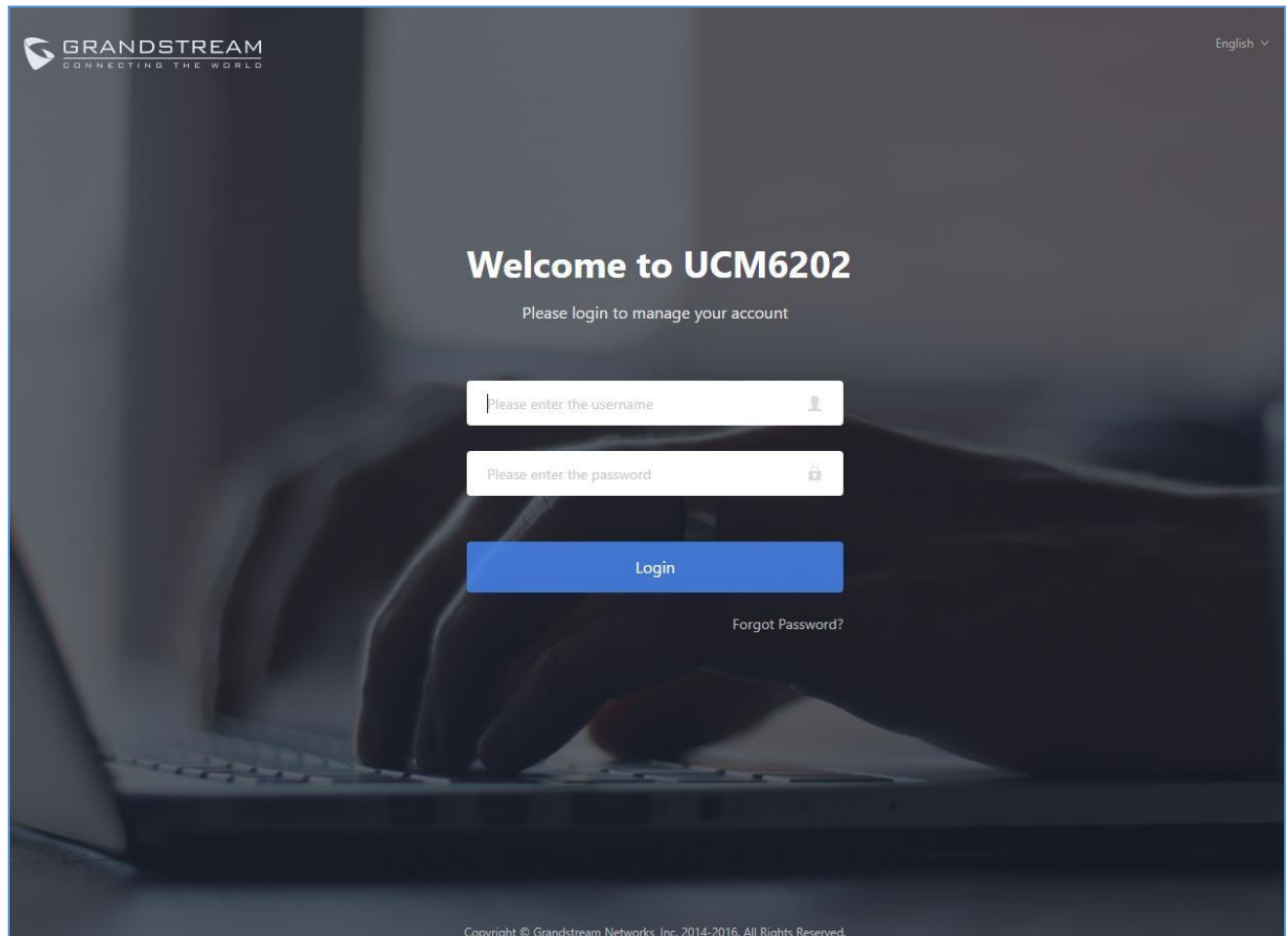




## Using the Web UI

### Accessing the Web UI

The UCM's web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow users to configure the device through a web browser such as Microsoft IE (version 8+), Mozilla Firefox, Google Chrome, etc. To access the UCM's web portal, follow the steps below:



**Figure 7: UCM6202 Web GUI Login Page**

1. Make sure your computer is on the same network as the UCM.
2. Make sure that the UCM's IP address is displayed on its LCD.
3. Enter the UCM's IP address into a web browsers' address bar. The login page should appear (please see the above image).
4. Enter default administrator username "admin" and password.

**Note:** Units manufactured starting January 2017 have a unique random password printed on the sticker located on the back of the unit. It is highly recommended to change the default password after logging in for the first time. Older units have default password "admin".



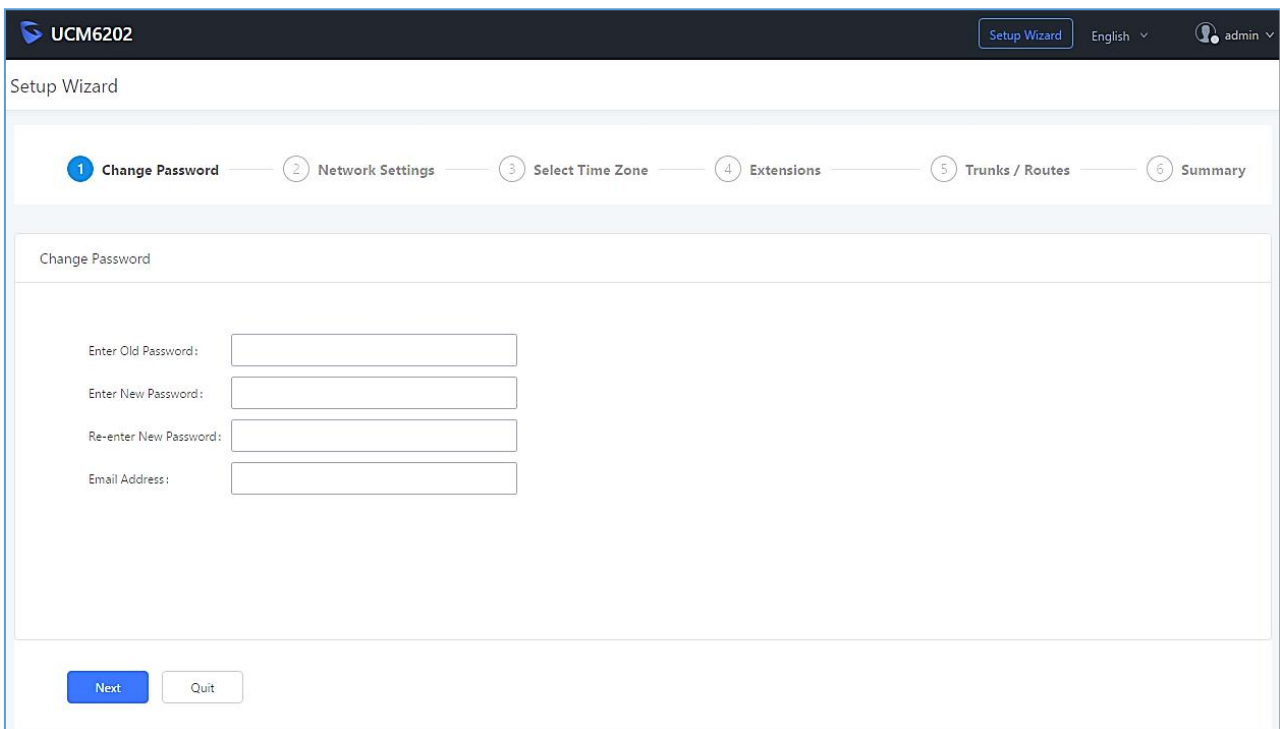
 **Note:**

By default, the UCM6200 has **Redirect From Port 80** enabled. As such, if users type in the UCM6200 IP address in the web browser, the web page will be automatically redirected to the page using HTTPS and port 8089. For example, if the LCD shows 192.168.40.167, and 192.168.40.167 is entered into the web browser, the web page will be redirected to: <https://192.168.40.167:8089>

The option **Redirect From Port 80** can be found under the UCM6200 Web GUI→**System Settings**→**HTTP Server**.

## Setup Wizard

After logging into the UCM web portal for the first time, the setup wizard will guide the user through basic configurations such as time zone, network settings, trunks, and routing rules.



**Figure 8: UCM6200 Setup Wizard**

The setup wizard can be closed and reopened at any time. At the end of the wizard, a summary of the pending configuration changes can be reviewed before applying them.

## Main Settings

There are 8 main sections in the web portal to manage various features of the UCM.



- **System Status:** Displays the dashboard, system information, current active calls, and network status.
- **Extensions/Trunks:** Manages extensions, trunks, and routing rules.
- **Call Features:** Manages various features of the UCM such as the IVR and voicemail.
- **PBX Settings:** Manages the settings related to PBX functionality such as SIP settings and interface settings.
- **System Settings:** Manages the settings related to the UCM system itself such as network and security settings.
- **CDR:** Contains the call detail records, statistics, and audio recordings of calls processed by the UCM.
- **Value-Added Features:** Manages the settings of features unrelated to core PBX functionality such as Zero Config provisioning and CRM/PMS integrations.
- **Maintenance:** Manages settings and logs related to system management and maintenance such as user management, activity logs, backup settings, upgrade settings and troubleshooting tools.

## Web GUI Languages

Currently the UCM6200 series Web GUI supports **English, Simplified Chinese, Traditional Chinese, Spanish, French, Portuguese, Russian, Italian, Polish, German etc.**

Users can select the UCM's web UI display language in the top-right corner of the page.

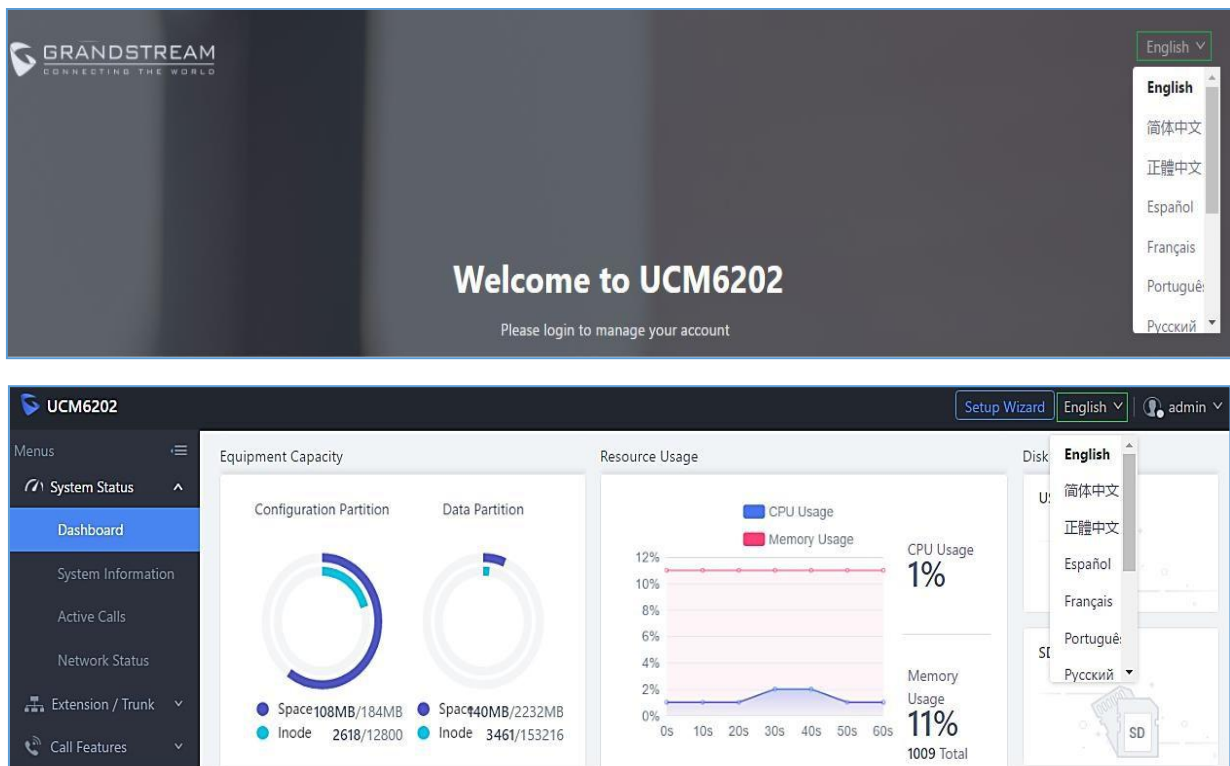


Figure 9: UCM6200 Web GUI Language



## Web GUI Search Bar

Users can search for options in the web portal with the search bar on the top right of the page.

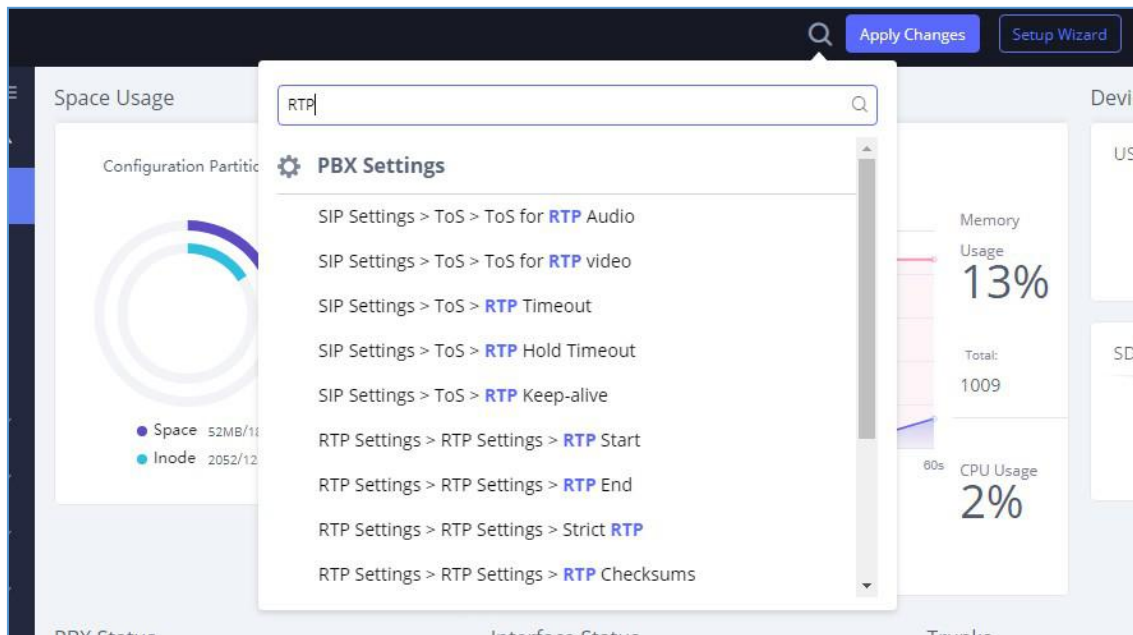


Figure 10: Web GUI Search Bar

## Saving and Applying Changes

After making changes to a page, click on the "Save" button to save them and then the "Apply Changes" button that appears to finalize the changes. If a modification requires a reboot, a prompt will appear asking to reboot the device.

## Setting Up an Extension

Power on the UCM6200 and your SIP endpoint. Connect both devices to the same network and follow the steps below to set up an extension.

1. Log into the UCM web portal and navigate to **Extension/Trunk->Extensions**
2. Click on the "Add" button to start creating a new extension. The Extension and SIP/IAX Password information will be used to register to this extension. To set up voicemail, the Voicemail Password will be required.
3. To register an endpoint to this extension, go into your endpoint's web UI and edit the desired account. Enter the newly created extension's number, SIP user ID, and password into their corresponding fields on the endpoint. Enter the UCM's IP address into the SIP server field. If setting up voicemail, enter \*97 into the Voice Mail Access Number field. This field may be named differently on other devices.



4. To access the extension's voicemail, use the newly registered extension to dial \*97 and access the personal voicemail system. Once prompted, enter the voicemail password. If successful, you will now be prompted with various voicemail options.
5. You have now set up an extension on an endpoint.



## SYSTEM SETTINGS

This section will explain the available system-wide parameters and configuration options on the UCM62xx. This includes settings for the following items: HTTP server, network, OpenVPN, DDNS, LDAP server and email server.

### HTTP Server

The UCM6200's embedded web server responds to HTTP/HTTPS GET/POST requests and allows users to configure the UCM via web browsers such as Microsoft IE, Mozilla Firefox, and Google Chrome. By default, users can access the UCM by just typing its IP address into a browser address bar. The browser will automatically be redirected to HTTPS using port 8089. For example, typing in "192.168.40.50" into the address bar will redirect the browser to "https://192.168.40.50:8089". This behavior can be changed in the **System Settings->HTTP Server page**.

**Table 6: HTTP Server Settings**

<b>Redirect From Port 80</b>	Toggles automatic redirection to UCM's web portal from port 80. If disabled, users will need to manually add the UCM's configured HTTP/HTTPS port to the server address when accessing the UCM web portal via browser. Default is "Enabled".
<b>Protocol Type</b>	Select either HTTP or HTTPS as the protocol to access the UCM's HTTP server. This will also determine what is used when endpoints download config files from the UCM via Zero Config. Default is "HTTPS".
<b>Port</b>	Specifies the port number used to access the UCM HTTP server. Default is "8089".
<b>Enable IP Whitelist</b>	If enabled, only the server addresses in whitelist will be able to access the UCM's web portal. It is highly recommended to add the IP address currently used to access the UCM web page before enabling this option. Default is "Disabled".
<b>Permitted IP(s)</b>	List of addresses that can access the UCM web portal. Ex: 192.168.6.233 / 255.255.255.255
<b>Certificate Options</b>	Selects the method of acquiring SSL certificates for the UCM web server. Two methods are currently available: <ul style="list-style-type: none"> <li>- Upload Certificate: Upload the appropriate files from one's own PC.</li> <li>- Request Certificate: Enter the domain for which to request a certificate for from Let's Encrypt.</li> </ul>



<b>TLS Private Key</b>	Uploads the private key for the HTTP server.  <b>Note:</b> Key file must be under 2MB in file size and in *.pem format. File name will automatically be changed to "private.pem".
<b>TLS Cert</b>	Uploads the certificate for the HTTP server.  <b>Note:</b> Certificate must be under 2MB in file size and in *.pem format. This will be used for TLS connections and contains private key for the client and signed certificate for the server.
<b>Domain</b>	Enter the domain to request the certificate for and click on <a href="#">Request Certificate</a> to request the certificate.

If the protocol or port has been changed, the user will be logged out and redirected to the new URL.

## Network Settings

After successfully connecting the UCM6200 to the network for the first time, users could login the Web GUI and go to **System Settings**→**Network Settings** to configure the network parameters for the device.

- UCM6200 supports Route/Switch/Dual mode functions.

In this section, all the available network setting options are listed for all models. Select each tab in Web GUI→**System Settings**→**Network Settings** page to configure LAN settings, WAN settings, 802.1X and Port Forwarding.

## Basic Settings

Please refer to the following tables for basic network configuration parameters on UCM6202, UCM6204 and UCM6208, respectively.

**Table 7: UCM6200 Network Settings→Basic Settings**

<b>Method</b>	Select "Route", "Switch" or "Dual" mode on the network interface of UCM6200. The default setting is "Route". <ul style="list-style-type: none"> <li>• <b>Route</b> WAN port will be used for uplink connection. LAN port will function similarly to a regular router port.</li> <li>• <b>Switch</b> WAN port will be used for uplink connection. LAN port will be used as a bridge for connections.</li> </ul>
---------------	--



	<ul style="list-style-type: none"> <li> <b>Dual</b>            Both WAN and LAN ports will be used for uplink connections labeled as LAN2 and LAN1, respectively. The port selected as the Default Interface will need to have a gateway IP address configured if it is using a static IP.         </li> </ul>
<b>MTU</b>	Specifies the maximum transmission unit value. Default is 1500.
<b>IPv4 Address</b>	
<b>Preferred DNS Server</b>	If configured, this will be used as the Primary DNS server.
<b>WAN (when "Method" is set to "Route")</b>	
<b>IP Method</b>	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
<b>IP Address</b>	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
<b>Subnet Mask</b>	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
<b>Gateway IP</b>	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
<b>DNS Server 1</b>	Enter the DNS server 1 address for static IP settings. The default setting is 0.0.0.0.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static IP settings.
<b>Username</b>	Enter the username to connect via PPPoE.
<b>Password</b>	Enter the password to connect via PPPoE.
<b>Layer 2 QoS 802.1Q/VLAN Tag</b>	Assign the VLAN tag of the layer 2 QoS packets for WAN port. The default value is 0.
<b>Layer 2 QoS 802.1p Priority Value</b>	Assign the priority value of the layer 2 QoS packets for WAN port. The default value is 0.
<b>LAN (when Method is set to "Route")</b>	
<b>IP Address</b>	Enter the IP address assigned to LAN port. The default setting is 192.168.2.1.
<b>Subnet Mask</b>	Enter the subnet mask. The default setting is 255.255.255.0.
<b>DHCP Server Enable</b>	Enable or disable DHCP server capability. The default setting is "Yes".
<b>DNS Server 1</b>	Enter DNS server address 1. The default setting is 8.8.8.8.
<b>DNS Server 2</b>	Enter DNS server address 2. The default setting is 208.67.222.222.
<b>Allow IP Address From</b>	Enter the DHCP IP Pool starting address. The default setting is 192.168.2.100.





<b>Allow IP Address To</b>	Enter the DHCP IP Pool ending address. The default setting is 192.168.2.254.
<b>Default IP Lease Time</b>	Enter the IP lease time (in seconds). The default setting is 43200.
<b>LAN (when Method is set to "Switch")</b>	
<b>IP Method</b>	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
<b>IP Address</b>	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
<b>Subnet Mask</b>	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
<b>Gateway IP</b>	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
<b>DNS Server 1</b>	Enter the DNS server 1 address for static IP settings. The default setting is 0.0.0.0.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static IP settings.
<b>Username</b>	Enter the username to connect via PPPoE.
<b>Password</b>	Enter the password to connect via PPPoE.
<b>Layer 2 QoS 802.1Q/VLAN Tag</b>	Assign the VLAN tag of the layer 2 QoS packets for LAN port. The default value is 0.
<b>Layer 2 QoS 802.1p Priority Value</b>	Assign the priority value of the layer 2 QoS packets for LAN port. The default value is 0.
<b>LAN 1 / LAN 2 (when Method is set to "Dual")</b>	
<b>Default Interface</b>	If "Dual" is selected as "Method", users will need assign the default interface to be LAN 1 (mapped to UCM6202 WAN port) or LAN 2 (mapped to UCM6202 LAN port) and then configure network settings for LAN 1/LAN 2. The default interface is LAN 2.
<b>IP Method</b>	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
<b>IP Address</b>	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
<b>Subnet Mask</b>	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
<b>Gateway IP</b>	Enter the gateway IP address for static IP settings when the port is assigned as default interface. The default setting is 0.0.0.0.
<b>DNS Server 1</b>	Enter the DNS server 1 address for static IP settings. The default setting is 0.0.0.0.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static IP settings.



<b>Username</b>	Enter the username to connect via PPPoE.
<b>Password</b>	Enter the password to connect via PPPoE.
<b>Layer 2 QoS 802.1Q/VLAN Tag</b>	Assign the VLAN tag of the layer 2 QoS packets for LAN port. The default value is 0.
<b>Layer 2 QoS 802.1p Priority Value</b>	Assign the priority value of the layer 2 QoS packets for LAN port. The default value is 0.

### IPv6 Address

#### WAN (when "Method" is set to "Route")

<b>IP Method</b>	Select Auto or Static. The default setting is Auto
<b>IP Address</b>	Enter the IP address for static IP settings.
<b>IP Prefixlen</b>	Enter the Prefix length for static settings. Default is 64
<b>DNS Server 1</b>	Enter the DNS server 1 address for static settings.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static settings.

#### LAN (when Method is set to "Route")

<b>DHCP Server</b>	Select Disable, Auto or DHCPv6. <b>Disable:</b> the DHCPv6 server is disabled. <b>Auto:</b> Stateless address auto configuration using NDP protocol. <b>DHCPv6:</b> Stateful address auto configuration using DHCPv6 protocol.
<b>DHCP Prefix</b>	Enter DHCP prefix. (Default is 2001:db8:2:2::)
<b>DHCP prefixlen</b>	Enter the Prefix length for static settings. Default is 64
<b>DNS Server 1</b>	Enter the DNS server 1 address for static settings. Default is (2001:4860:4860::8888 )
<b>DNS Server 2</b>	Enter the DNS server 2 address for static settings. Default is (2001:4860:4860::8844 )
<b>Allow IP Address From</b>	Configure starting IP address assigned by the DHCP prefix and DHCP prefixlen.
<b>Allow IP Address To</b>	Configure the ending IP address assigned by the DHCP Prefix and DHCP prefixlen.
<b>Default IP Lease Time</b>	Configure the lease time (in second) of the IP address.

#### LAN (when Method is set to "Switch")

<b>IP Method</b>	Select Auto or Static. The default setting is Auto
------------------	--



<b>IP Address</b>	Enter the IP address for static IP settings.
<b>IP Prefixlen</b>	Enter the Prefix length for static settings. Default is 64
<b>DNS Server 1</b>	Enter the DNS server 1 address for static settings.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static settings.
<b>LAN 1 / LAN 2 (when Method is set to "Dual")</b>	
<b>Default Interface</b>	Users will need assign the default interface to be LAN 1 (mapped to UCM6200 WAN port) or LAN 2 (mapped to UCM6200 LAN port) and then configure network settings for LAN 1/LAN 2. The default interface is LAN 2.
<b>IP Method</b>	Select Auto or Static. The default setting is Auto
<b>IP Address</b>	Enter the IP address for static IP settings.
<b>IP Prefixlen</b>	Enter the Prefix length for static settings. Default is 64
<b>DNS Server 1</b>	Enter the DNS server 1 address for static settings.
<b>DNS Server 2</b>	Enter the DNS server 2 address for static settings.

- **Method: Route**

When the UCM6200 has, method set to Route in network settings, WAN port interface is used for uplink connection and LAN port interface is used as a router. Please see a sample diagram below.

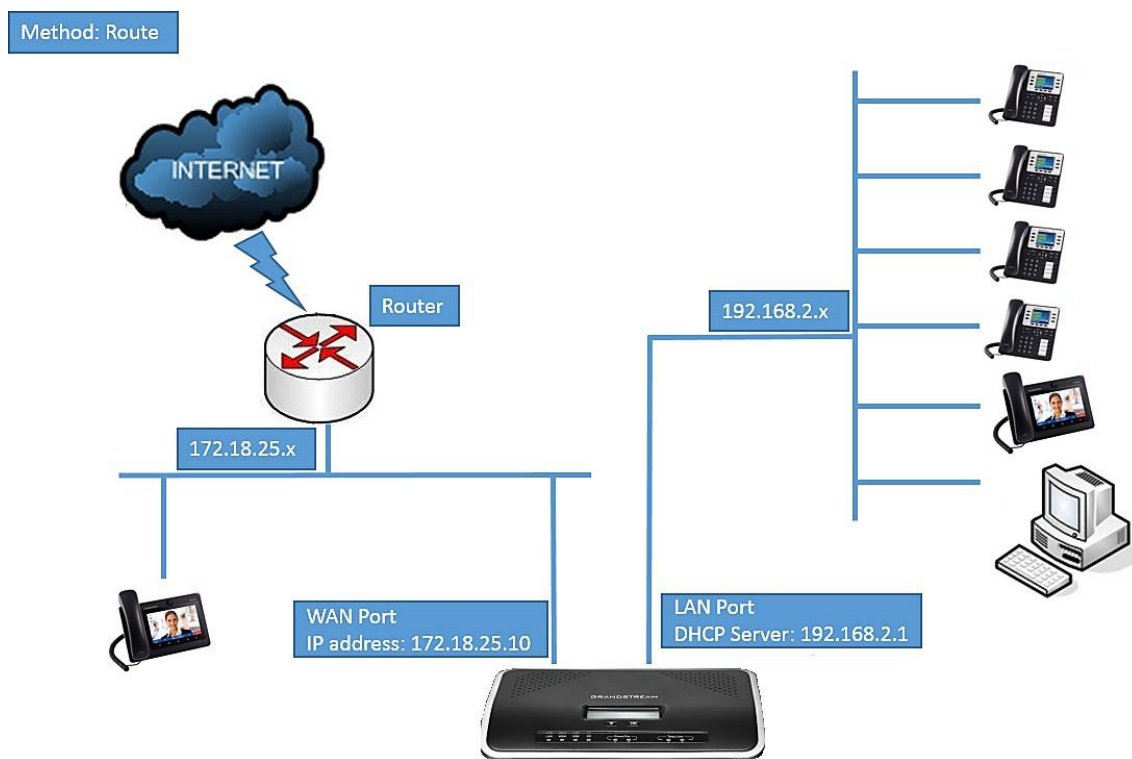


Figure 11: UCM6202 Network Interface Method: Route



- **Method: Switch**

WAN port interface is used for uplink connection; LAN port interface is used as room for PC connection.

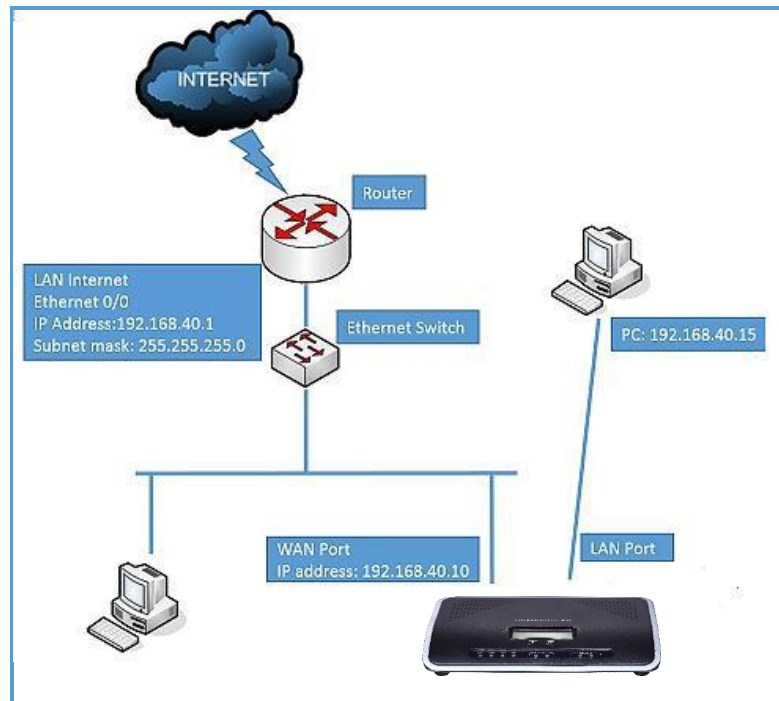


Figure 12: UCM6202 Network Interface Method: Switch

- **Method: Dual**

Both WAN port and LAN port are used for uplink connection. Users will need assign LAN 1 or LAN 2 as the default interface in option "Default Interface" and configure "Gateway IP" if static IP is used for this interface.



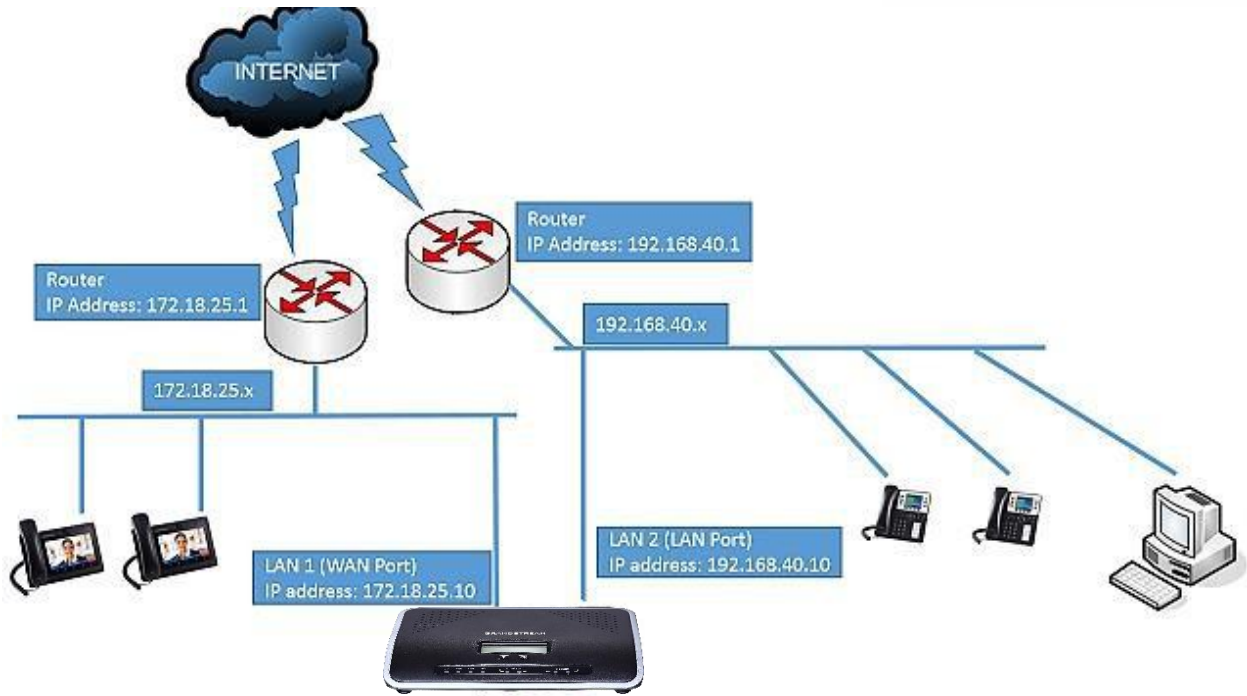


Figure 13: UCM6202 Network Interface Method: Dual

## DHCP Client List

This feature can bind MAC to IP addresses on the LAN port when UCM6200 is set to Route mode.

When devices receive IP addresses from LAN port, they will be listed on the webGUI under “**System Settings**→**Network Settings**→**DHCP Client List**” as shown below.

Network Settings				
Basic Settings	<b>DHCP Client List</b>	802.1X Settings	Static Routes	Port Forwarding
<input type="button" value="+ Add Mac Address Bind"/> <input type="button" value="Batch add MAC addresses to bind"/> <input type="button" value="Batch Release MAC Address Bind"/>				
<input type="checkbox"/>	MAC Address ↕	IP Address ↕	Bind Status ↕	Options
<input type="checkbox"/>	dc4a3e78dd25	192.168.2.100	Unbind	<input type="button" value="↶"/> <input type="button" value="↷"/>

Figure 14: DHCP ClientList

User can bind manually a MAC to an IP address by clicking on , the following figure will pop up.



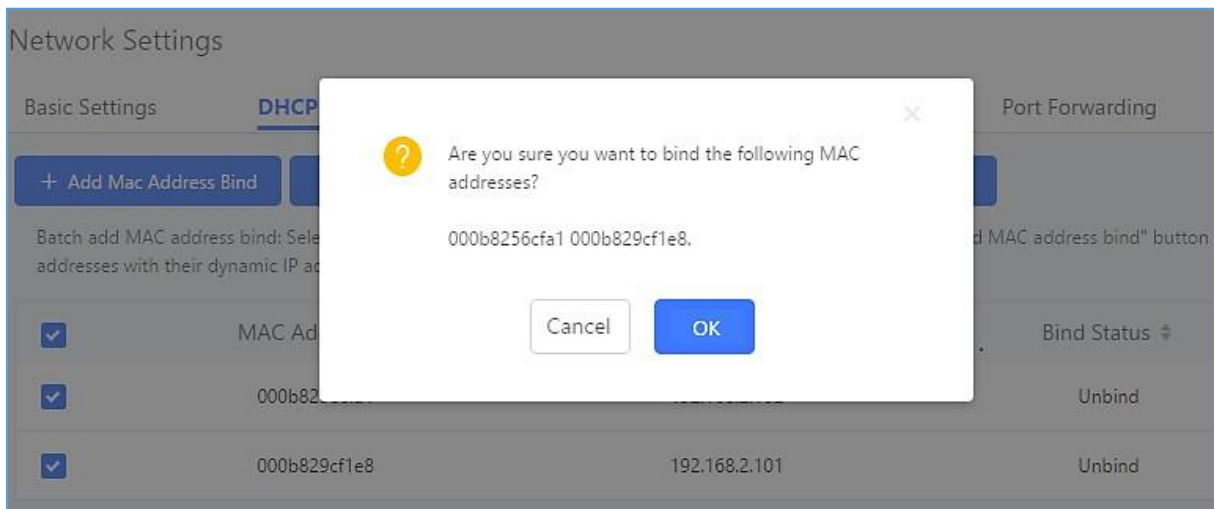


**Figure 15: Add MAC Address Bind**

User needs to set the device MAC address and the IP that will be bound to it (the IP address needs to be within the UCM6200 DHCP range).

To bind a batch of listed MAC addresses, user needs to check first the MAC addresses to bind and click on

**Batch add MAC addresses to bind**. A confirmation popup will be shown, click **OK** to bind the addresses.



**Figure 16: Batch Add MAC Address Bind**

After Clicking “OK” to confirm the binding, the “Bind Status” will change from “Unbind” to “Binding”.

## 802.1X

IEEE 802.1X is an IEEE standard for port-based network access control. It provides an authentication mechanism to device before the device can access Internet or other LAN resources. The UCM6200 supports 802.1X as a supplicant/client to be authenticated. The following diagram and figure show UCM6200 use 802.1X mode “EAP-MD5” on WAN port as client in the network to access Internet.



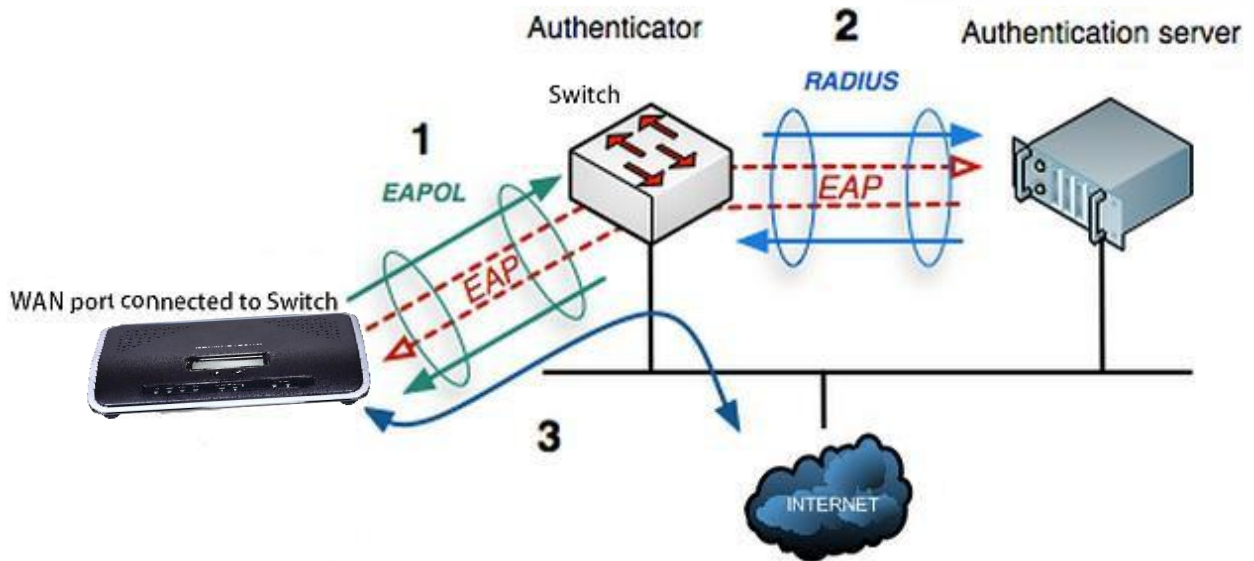


Figure 17: UCM6200 Using 802.1X as Client

Network Settings	
Basic Settings	DHCP Client List
<b>802.1X Settings</b>	
WAN	
802.1X Mode:	EAP-MD5
* Identity:	8021xxUCM6202
* MD5 Password:	.....

Figure 18: UCM6200 Using 802.1X EAP-MD5

The following table shows the configuration parameters for 802.1X on UCM6200. Identity and MD5 password are required for authentication, which should be provided by the network administrator obtained from the RADIUS server. If “EAP-TLS” or “EAP-PEAPv0/MSCHAPv2” is used as the 802.1X mode, users will also need to upload 802.1X CA Certificate and 802.1X Client Certificate, which should be also generated from the RADIUS server.









**Table 8: UCM6200 Network Settings→802.1X**

<b>802.1X Mode</b>	Select 802.1X mode. The default setting is "Disable". The supported 802.1X mode are: <ul style="list-style-type: none"> <li>• EAP-MD5</li> <li>• EAP-TLS</li> <li>• EAP-PEAPv0/MSCHAPv2</li> </ul>
<b>Identity</b>	Enter 802.1X mode Identity information.
<b>MD5 Password</b>	Enter 802.1X mode MD5 password information.
<b>802.1X Certificate</b>	Select 802.1X certificate from local PC and then upload.
<b>802.1X Client Certificate</b>	Select 802.1X client certificate from local PC and then upload.

## Static Routes

The UCM6200 provides users static routing capability that allows the device to use manually configured routes, rather than information only from dynamic routing or gateway configured in the UCM6200 Web GUI→**System Settings**→**Network Settings**→**Basic Settings** to forward traffic. It can be used to define a route when no other routes are available or necessary, or used in complementary with existing routing on the UCM6200 as a failover backup, etc.

- Click on  to create a new IPv4 static route or click on  to create a new IPv6 static route. The configuration parameters are listed in the table below.
- Once added, users can select  to edit the static route.
- Select  to delete the static route.

**Table 9: UCM6200 Network Settings→Static Routes**

<b>Destination</b>	Configure the destination IPv4 address or the destination IPv6 subnet for the UCM6200 to reach using the static route. Example: IPv4 address - <b>192.168.66.4</b> IPv6 subnet - <b>2001:740:D::1/64</b>
<b>Netmask</b>	Configure the subnet mask for the above destination address. If left blank, the default value is 255.255.255.255. Example: <b>255.255.255.0</b>

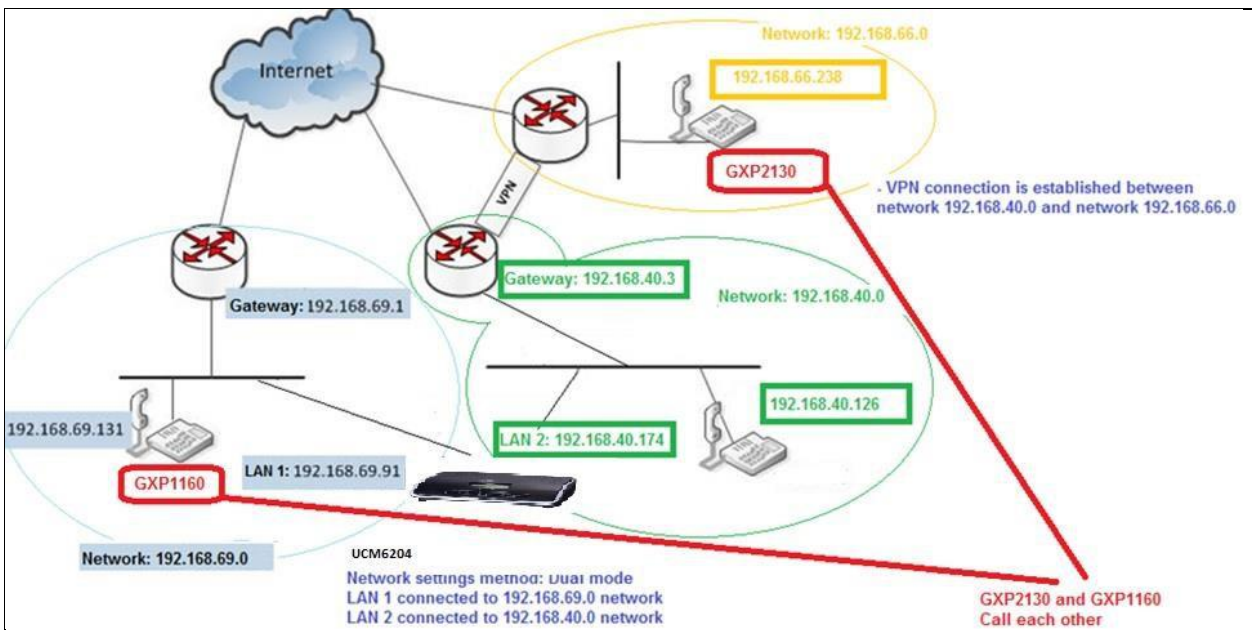




<b>Gateway</b>	Configure the IPv4 or IPv6 gateway address so that the UCM6200 can reach the destination via this gateway. Gateway address is optional. Example: <b>192.168.40.5 or 2001:740:D::1</b>
<b>Interface</b>	Specify the network interface on the UCM6200 to reach the destination using the static route. LAN interface is eth0; WAN interface is eth1.

Static routes configuration can be reset from LCD menu → Network Menu.

The following diagram shows a sample application of static route usage on UCM6204.



**Figure 19: UCM6204 Static Route Sample**

The network topology of the above diagram is as below:

- Network 192.168.69.0 has IP phones registered to UCM6204 LAN 1 address
- Network 192.168.40.0 has IP phones registered to UCM6204 LAN 2 address
- Network 192.168.66.0 has IP phones registered to UCM6204 via VPN
- Network 192.168.40.0 has VPN connection established with network 192.168.66.0



In this network, by default the IP phones in network 192.168.69.0 are unable to call IP phones in network 192.168.66.0 when registered on different interfaces on the UCM6204. Therefore, we need configure a static route on the UCM6204 so that the phones in isolated networks can make calls between each other.



**Create New IPV4 Static Route**

\* Destination:

Subnet Mask:

Gateway:

\* Protocol Type:

**Figure 20: UCM6204 Static Route Configuration**

## Port Forwarding

The UCM network interface supports router function which provides users the ability to do port forwarding. If LAN mode is set to "Route" under Web GUI→**System Settings**→**Network Settings**→**Basic Settings** page, port forwarding is available for configuration.

The port forwarding configuration is under Web GUI→**System Settings**→**Network Settings**→**Port Forwarding** page. Please see related settings in the table below.

**Table 10: UCM6200 Network Settings→Port Forwarding**

<b>WAN Port</b>	Specify the WAN port number or a range of WAN ports. Unlimited number of ports can be configured.  <b>Note:</b> When it is set to a range, WAN port and LAN port must be configured with the same range, such as WAN port: 1000-1005 and LAN port: 1000-1005, and access from WAN port will be forwarded to the LAN port with the same port number, for example, WAN port 1000 will be port forwarding to LAN port 1000.
<b>LAN IP</b>	Specify the LAN IP address.
<b>LAN Port</b>	Specify the LAN port number or a range of LAN ports.  <b>Note:</b>



	When it is set to a range, WAN port and LAN port must be configured with the same range, such as WAN port: 1000-1005 and LAN port: 1000-1005, and access from WAN port will be forwarded to the LAN port with the same port number, for example, WAN port 1000 will be port forwarding to LAN port 1000.
<b>Protocol Type</b>	Select protocol type "UDP Only", "TCP Only" or "TCP/UDP" for the forwarding in the selected port. The default setting is "UDP Only".

The following figures demonstrate a port forwarding example to provide phone's Web GUI access to public side.

- UCM6200 network mode is set to "Route".
- UCM6200 WAN port is connected to uplink switch, with a public IP address configured, e.g. 1.1.1.1.
- UCM6200 LAN port provides DHCP pool that connects to multiple phone devices in the LAN network 192.168.2.x. The UCM6200 is used as a router, with gateway address 192.168.2.1.
- There is a GXP2160 connected under the LAN interface network of the UCM6200. It obtains IP address 192.168.2.100 from UCM6200 DHCP pool.
- On the UCM6200 Web GUI → **System Settings** → **Network Settings** → **Port Forwarding**, configure a port forwarding entry as the figure shows below.
- Click on [+ Create New Port Forwarding](#)

**WAN Port:** This is the port opened on the WAN side for access purpose.

**LAN IP:** This is the GXP2160 IP address, under the LAN interface network of the UCM6200.

**LAN Port:** This is the port opened on the GXP2160 side for access purpose.

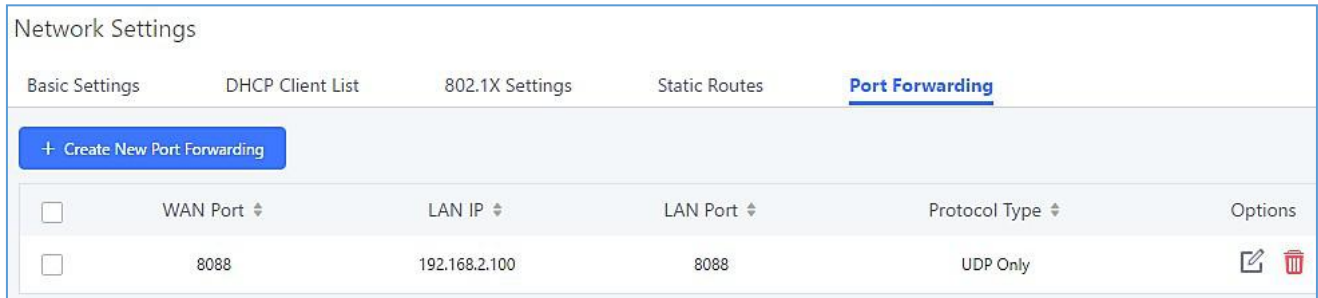
**Protocol Type:** We select TCP here for Web GUI access using HTTP.

The screenshot shows a web form titled "Create New Port Forwarding". It contains the following fields:

- \* WAN Port:** Input field containing "8088".
- \* LAN IP:** Input field containing "192.168.2.100".
- \* LAN Port:** Input field containing "8088".
- \* Protocol Type:** Dropdown menu with "UDP Only" selected.

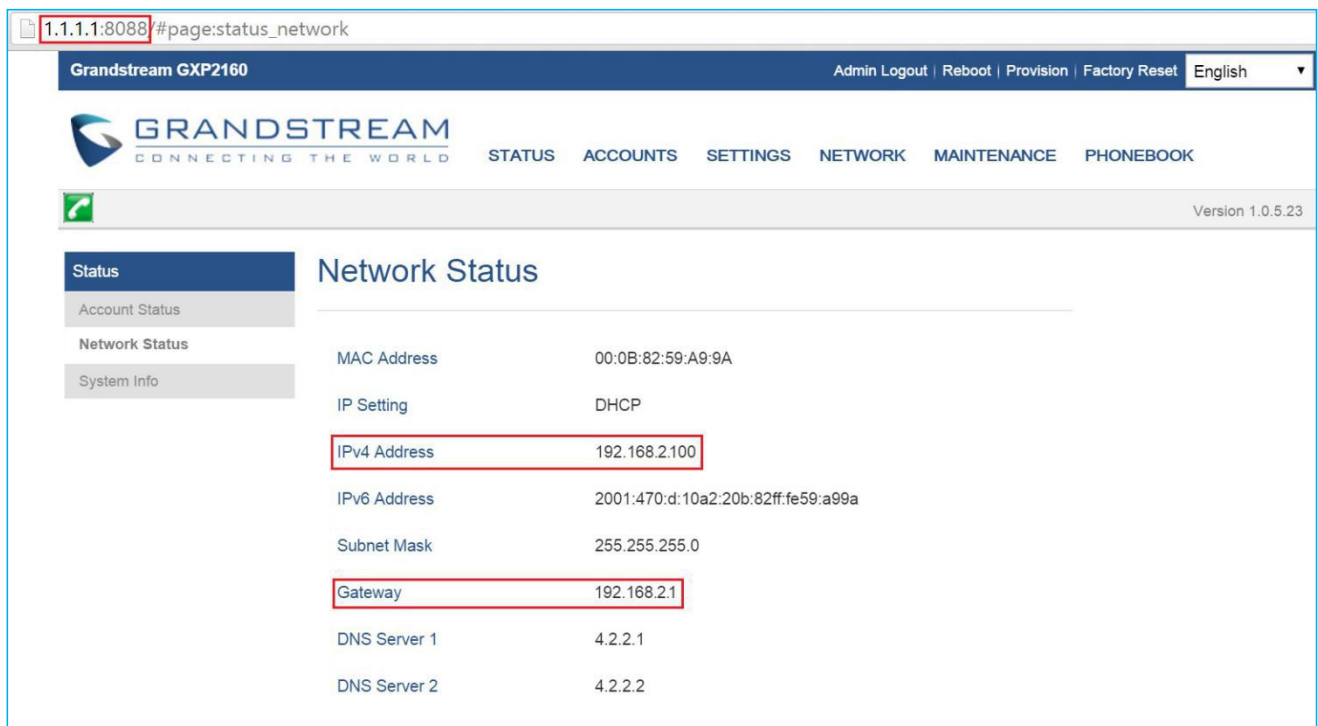
**Figure 21: Create New Port Forwarding**





**Figure 22: UCM6200 Port Forwarding Configuration**

This will allow users to access the GXP2160 Web GUI from public side, by typing in public IP address (example: 1.1.1.1:8088).



**Figure 23: GXP2160 Web Access using UCM6202 Port Forwarding**

## OpenVPN®

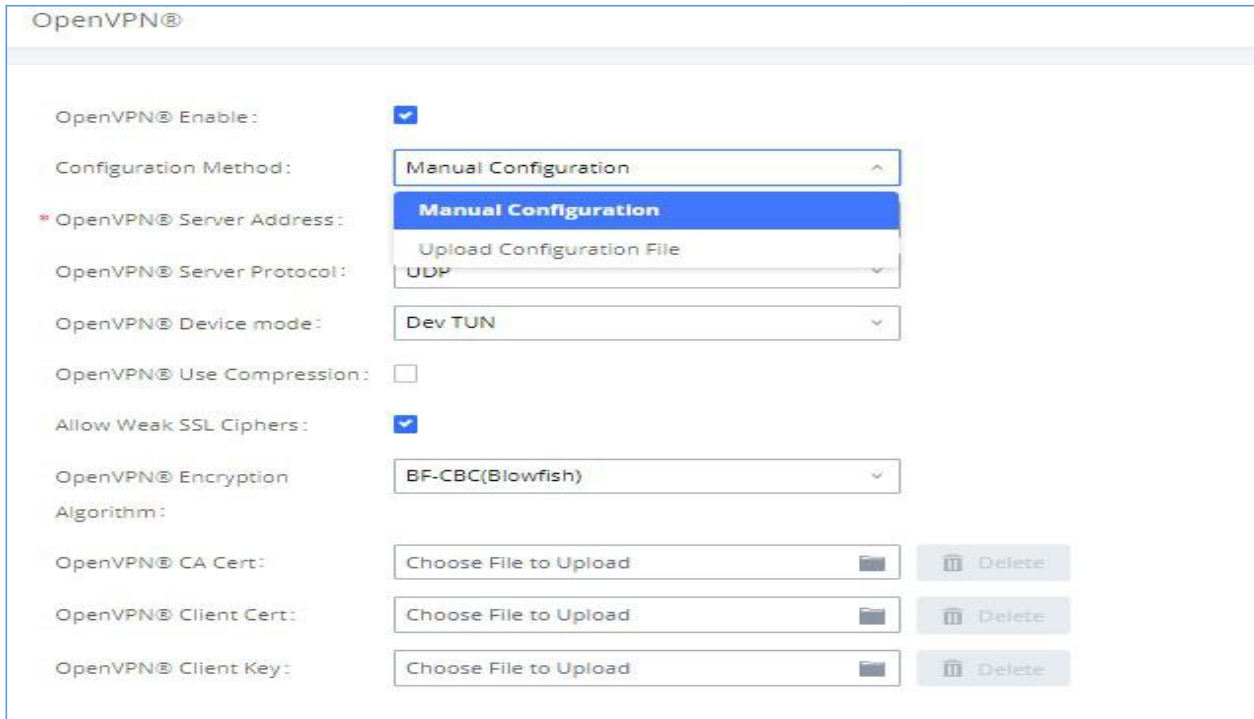
OpenVPN® settings allow the users to configure UCM6200 to use VPN features, the following table gives details about the various options in order to configure the UCM as OpenVPN Client.



**Table 11: UCM6200 System Settings→Network Settings→OpenVPN®**

<b>Enable</b>	Enable / Disable the OpenVPN® feature.
<b>Configuration Method</b>	Select OpenVPN® configuration method. <b>Manual Configuration:</b> Allows to configure OpenVPN settings manually. <b>Upload Configuration File:</b> Allows to upload .ovpn and .conf files to the UCM and to automatically configure OpenVPN settings.
<b>OpenVPN® Server Address</b>	Configures the hostname/IP and port of the server. For example: 192.168.1.2:22
<b>OpenVPN® Server Protocol</b>	Specify the protocol user, user should use the same settings as used on the server
<b>OpenVPN® Device mode</b>	Use the same setting as used on the server. <b>Dev TUN:</b> Create a routed IP tunnel. <b>Dev TAP:</b> Create an Ethernet tunnel.
<b>OpenVPN® Use Compression</b>	Compress tunnel packets using the LZO algorithm on the VPN link. Do not enable this unless it is also enabled in the server config file.
<b>Allow Weak SSL Ciphers</b>	Enable/Disable allowing Weak SSL Ciphers.
<b>OpenVPN® Encryption Algorithm</b>	Specify the cryptographic cipher. Users should make sure to use the same setting that they are using on the OpenVPN server.
<b>OpenVPN® CA Cert</b>	Upload as SSL/TLS root certificate. This file will be renamed as 'ca.crt' automatically.
<b>OpenVPN® Client Cert</b>	Upload a client certificate. This file will be renamed as 'cliend.crt' automatically.
<b>OpenVPN® Client Key</b>	Upload a client private key. This file will be renamed as 'client.key' automatically.





OpenVPN®

OpenVPN® Enable:

Configuration Method: Manual Configuration

\* OpenVPN® Server Address: Manual Configuration

OpenVPN® Server Protocol: UDP

OpenVPN® Device mode: Dev TUN

OpenVPN® Use Compression:

Allow Weak SSL Ciphers:

OpenVPN® Encryption Algorithm: BF-CBC(Blowfish)

OpenVPN® CA Cert: Choose File to Upload

OpenVPN® Client Cert: Choose File to Upload

OpenVPN® Client Key: Choose File to Upload

**Figure 24: Open VPN® feature on the UCM6200**

## DDNS Settings

DDNS setting allows user to access UCM6200 via domain name instead of IP address.

The UCM supports DDNS service from the following DDNS provider:

- dydns.org
- noip.com
- freedns.afraid.org
- zoneedit.com
- oray.net

Here is an example of using noip.com for DDNS.

1. Register domain in DDNS service provider. Please note the UCM6200 needs to have public IP access.










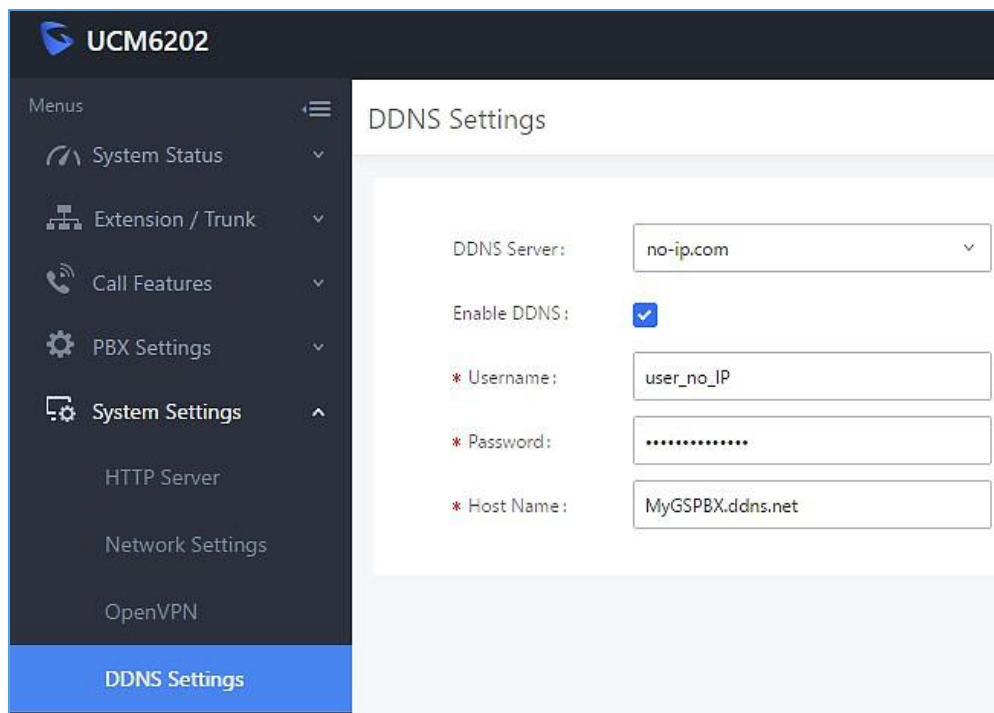
Hostname Information	
Hostname:	haograndstream.ddns.net 
Host Type:	<input checked="" type="radio"/> DNS Host (A) <input type="radio"/> DNS Host (Round Robin) <input type="radio"/> DNS Alias (CNAME)  <input type="radio"/> Port 80 Redirect <input type="radio"/> Web Redirect
IP Address:	<input type="text" value="1.2.3.4"/> Last Update: 2015-01-07 17:29:20 PST 
Assign to Group:	<input type="text" value="- No Group -"/>  <a href="#">Configure Groups</a> 
Enable Wildcard:	Wildcards are a Plus / Enhanced feature. <a href="#">Upgrade Now!</a> 
Advanced Records:	TXT, SPF, and SRV records and the use of some special clients are Plus / Enhanced features. <a href="#">Upgrade now</a> to use them. 

Figure 25: Register Domain Name on noip.com

- On Web GUI→**System Settings**→**Network Settings**→**DDNS Settings**, enable DDNS service and configure username, password and host name.



**UCM6202**

Menus

- System Status
- Extension / Trunk
- Call Features
- PBX Settings
- System Settings**
  - HTTP Server
  - Network Settings
  - OpenVPN
  - DDNS Settings**

### DDNS Settings

DDNS Server:

Enable DDNS:

\* Username:

\* Password:

\* Host Name:

Figure 26: UCM6200 DDNS Setting





3. Now you can use domain name instead of IP address to connect to the UCM6200 Web GUI.

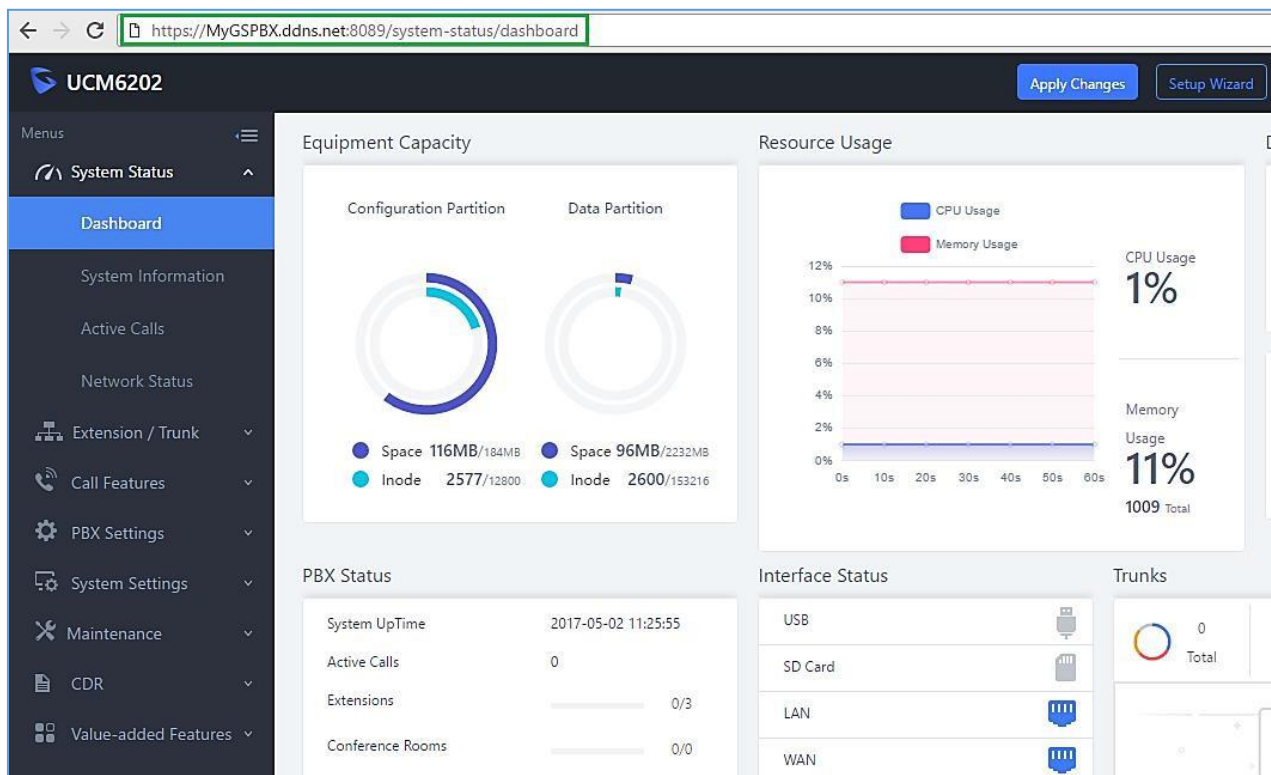


Figure 27: Using Domain Name to Connect to UCM6200

## Security Settings

The UCM6200 provides users firewall security configurations to prevent certain malicious attack to the UCM6200 system. Users could configure to allow, restrict or reject specific traffic through the device for security and bandwidth purpose. The UCM6200 also provides Fail2ban feature for authentication errors in SIP REGISTER, INVITE and SUBSCRIBE. To configure firewall settings in the UCM6200, go to Web GUI→**System Settings**→**Security Settings** page.

## Static Defense

Under Web GUI→**System Settings**→**Security Settings**→**Static Defense** page, users will see the following information:

- Current service information with port, process and type.
- Typical firewall settings.
- Custom firewall settings.





The following table shows a sample current service status running on the UCM6200.

**Table 12: UCM6200 Firewall→Static Defense→Current Service**

Port	Process	Type	Protocol or Service
7777	Asterisk	TCP/IPv4	SIP
389	Slapd	TCP/IPv4	LDAP
2000	Asterisk	TCP/IPv4	SCCP
22	Dropbear	TCP/IPv4	SSH
80	Lighthttpd	TCP/IPv4	HTTP
8089	Lighthttpd	TCP/IPv4	HTTPS
69	Opentftpd	UDP/IPv4	TFTP
9090	Asterisk	UDP/IPv4	SIP
6060	zero_config	UDP/IPv4	UCM6200 zero_config service
5060	Asterisk	UDP/IPv4	SIP
4569	Asterisk	UDP/IPv4	SIP
5353	zero_config	UDP/IPv4	UCM6200 zero_config service
37435	Syslogd	UDP/IPv4	Syslog

For typical firewall settings, users could configure the following options on the UCM6200.

**Table 13: Typical Firewall Settings**

<b>Ping Defense Enable</b>	If enabled, ICMP response will not be allowed for Ping request. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (UCM6200) interface.
<b>SYN-Flood Defense Enable</b>	<p>Allows the UCM6200 to handle excessive amounts of SYN packets from one source and keep the web portal accessible. There are two options available and only one of these options may be enabled at one time.</p> <ul style="list-style-type: none"> <li>eth(0)LAN defends against attacks directed to the LAN IP address of the UCM6200.</li> <li>eth(1)WAN defends against attacks directed to the WAN IP address of the UCM6200.</li> </ul> <p>SYN Flood Defense will limit the amount of SYN packets accepted by the UCM from one source to 10 packets per second. Any excess packets from that source will be discarded.</p>



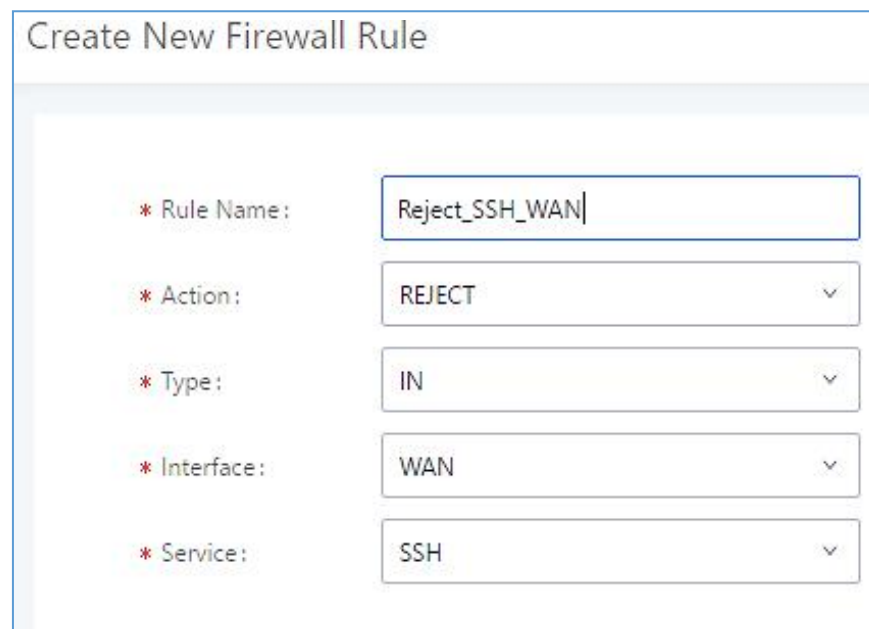
**Ping-of-Death  
Defense Enable**

Enable to prevent Ping-of-Death attack to the device. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (UCM6200) interface.

Under "Custom Firewall Settings", users could create new rules to accept, reject or drop certain traffic going through the UCM6200. To create new rule, click on "Create New Rule" button and a new window will pop up for users to specify rule options.

Right next to "Create New Rule" button, there is a checkbox for option "Reject Rules". If it is checked, all the rules will be rejected except the firewall rules listed below. In the firewall rules, only when there is a rule that meets all the following requirements, the option "Reject Rules" will be allowed to check:

- Action: "Accept"
- Type "In"
- Destination port is set to the system login port (e.g., by default 8089)
- Protocol is not UDP



**Create New Firewall Rule**

\* Rule Name:

\* Action:

\* Type:

\* Interface:

\* Service:



**Figure 28: Create New Firewall Rule**



**Table 14: Firewall Rule Settings**

<b>Rule Name</b>	Specify the Firewall rule name to identify the firewall rule.
<b>Action</b>	Select the action for the Firewall to perform. <ul style="list-style-type: none"> <li>• ACCEPT</li> <li>• REJECT</li> <li>• DROP</li> </ul>
<b>Type</b>	Select the traffic type. <ul style="list-style-type: none"> <li>• <b>IN</b> If selected, users will need specify the network interface "LAN" or "WAN" (for UCM6200) for the incoming traffic.</li> <li>• <b>OUT</b></li> </ul>
<b>Service</b>	Select the service type. <ul style="list-style-type: none"> <li>• <b>FTP</b></li> <li>• <b>SSH</b></li> <li>• <b>Telnet</b></li> <li>• <b>TFTP</b></li> <li>• <b>HTTP</b></li> <li>• <b>LDAP</b></li> <li>• <b>Custom</b> If "Custom" is selected, users will need specify Source (IP and port), Destination (IP and port) and Protocol (TCP, UDP or Both) for the service. Please note if the source or the destination field is left blank, it will be used as "Anywhere".</li> </ul>

Save the change and click on "Apply" button. Then submit the configuration by clicking on "Apply Changes" on the upper right of the web page. The new rule will be listed at the bottom of the page with sequence number, rule name, action, protocol, type, source, destination and operation. More operations below:

- Click on  to edit the rule.
- Click on  to delete the rule.

## Dynamic Defense

Dynamic defense is supported on the UCM6200 series. It can blacklist hosts dynamically when the LAN mode is set to "Route" under Web GUI→**System Settings**→**Network Settings**→**Basic Settings** page. If enabled, the traffic coming into the UCM6200 can be monitored, which helps prevent massive connection attempts or brute force attacks to the device. The blacklist can be created and updated by the UCM6200 firewall, which will then be displayed in the web page. Please refer to the following table for dynamic defense options on the UCM6200.

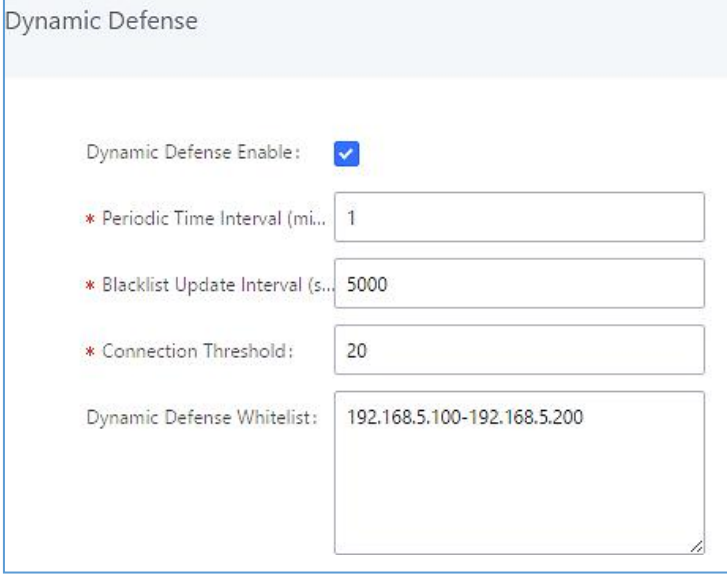


**Table 15: UCM6200 Firewall Dynamic Defense**

<b>Dynamic Defense Enable</b>	Enable dynamic defense. The default setting is disabled.
<b>Blacklist Update Interval</b>	Configure the blacklist update time interval (in seconds). The default setting is 120.
<b>Connection Threshold</b>	Configure the connection threshold. Once the number of connections from the same host reaches the threshold, it will be added into the blacklist. The default setting is 100.
<b>Dynamic Defense Whitelist</b>	Allowed IPs and ports range, multiple IP addresses and port range. For example: <b>192.168.5.100-192.168.5.200 1500:2000</b>

The following figure shows a configuration example like this:

- If a host at IP address 192.168.5.7 initiates more than 20 TCP connections to the UCM6200 within 1 minute, it will be added into UCM6200 blacklist.
- This host 192.168.5.7 will be blocked by the UCM6200 for 500 seconds.
- Since IP range 192.168.5.100-192.168.5.200 is in whitelist, if a host initiates more than 20 TCP connections to the UCM6200 within 1 minute, it will not be added into UCM6200 blacklist. It can still establish TCP connection with the UCM6200.



Dynamic Defense

Dynamic Defense Enable:

\* Periodic Time Interval (mi...: 1

\* Blacklist Update Interval (s...: 5000

\* Connection Threshold: 20

Dynamic Defense Whitelist: 192.168.5.100-192.168.5.200

**Figure 29: Configure Dynamic Defense**


## Fail2ban

Fail2Ban feature on the UCM6200 provides intrusion detection and prevention for authentication errors in SIP REGISTER, INVITE and SUBSCRIBE. Once the entry is detected within "Max Retry Duration", the UCM6200 will act to forbid the host for certain period as defined in "Banned Duration". This feature helps prevent SIP brute force attacks to the PBX system.

### Security Settings

Static Defense
Dynamic Defense
Fail2ban
SSH Access

Global Settings

Enable Fail2Ban:

\* Banned Duration:

\* Max Retry Duration:

\* MaxRetry:

Fail2ban Whitelist:  +

Local Settings

Asterisk Service:

Listening port number:  UDP Port

\* MaxRetry:

Login Attack Defense:

Listening port number:  TCP Port

\* MaxRetry:

**Figure 30: Fail2ban Settings**



**Table 16: Fail2Ban Settings**

Global Settings	
<b>Enable Fail2Ban</b>	Enable Fail2Ban. The default setting is disabled. Please make sure both "Enable Fail2Ban" and "Asterisk Service" are turned on to use Fail2Ban for SIP authentication on the UCM6200.
<b>Banned Duration</b>	Configure the duration (in seconds) for the detected host to be banned. The default setting is 600. If set to 0, the host will be always banned.
<b>Max Retry Duration</b>	Within this duration (in seconds), if a host exceeds the max times of retry as defined in "MaxRetry", the host will be banned. The default setting is 600.
<b>MaxRetry</b>	Configure the number of authentication failures during "Max Retry Duration" before the host is banned. The default setting is 5.
<b>Fail2Ban Whitelist</b>	Configure IP address, CIDR mask or DNS host in the whitelist. Fail2Ban will not ban the host with matching address in this list. Up to 20 addresses can be added into the list.
Local Settings	
<b>Asterisk Service</b>	Enable Asterisk service for Fail2Ban. The default setting is disabled. Please make sure both "Enable Fail2Ban" and "Asterisk Service" are turned on to use Fail2Ban for SIP authentication on the UCM6200.
<b>Listening Port Number</b>	Configure the listening port number for the service. By default, port 5060 will be used for UDP and TCP, and port 5061 will be used for TLS.
<b>MaxRetry</b>	Configure the number of authentication failures during "Max Retry Duration" before the host is banned. The default setting is 10. Please make sure this option is properly configured as it will override the "MaxRetry" value under "Global Settings".
<b>Login Attack Defense</b>	Enables defense against excessive login attacks to the UCM's web GUI. The default setting is disabled.
<b>Listening Port Number</b>	This is the Web GUI listening port number which is configured under <b>System Settings</b> → <b>HTTP Server</b> → <b>Port</b> . The default is 8089.
<b>MaxRetry</b>	When the number of failed login attempts from an IP address exceeds the MaxRetry number, that IP address will be banned from accessing the Web GUI.
Blacklist	
<b>Blacklist</b>	Users will be able to view the IPs that have been blocked by UCM.

## TLS Security

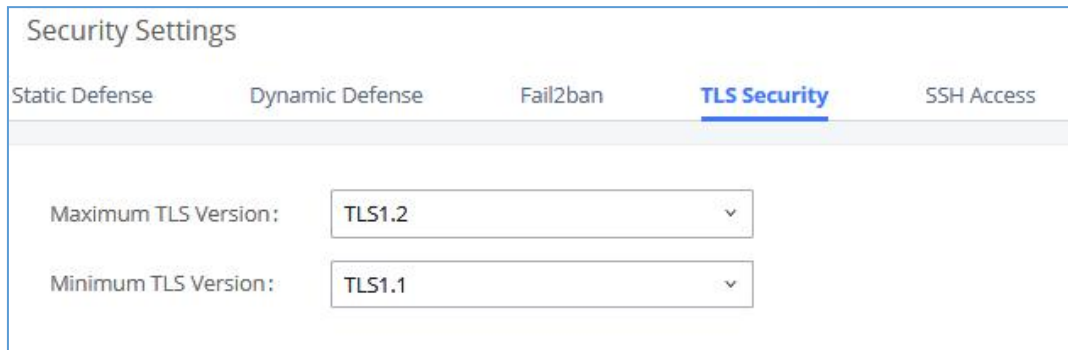
TLS security to specify minimum and maximum TLS versions supported by the UCM.

Please log in UCM62xx web interface and go to **System Settings**→**Security Settings**→**TLS Security**.

By default, minimum TLS version is set to TLS1.1, and maximum TLS version is set to TLS1.2.

Supported versions are 1.0, 1.1 and 1.2

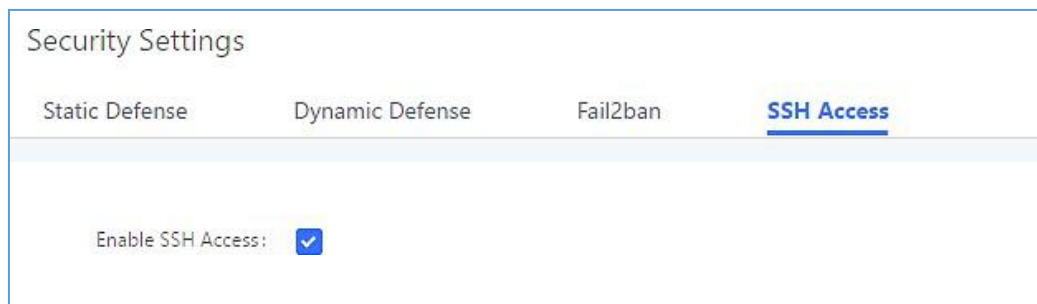




**Figure 31: TLS Security**

## SSH Access

SSH switch now is available via Web GUI and LCD. User can enable or disable SSH access directly from Web GUI or LCD screen. For web SSH access, please log in UCM6200 web interface and go to Web GUI→**System Settings**→**Security Settings**→**SSH Access**. By default, SSH access is disabled for security concerns. It is highly recommended to only enable SSH access for debugging purpose.



**Figure 32: SSH Access**

## LDAP Server

The UCM6200 has an embedded LDAP/LDAPS server for users to manage corporate phonebook in a centralized manner.

- By default, the LDAP server has generated the first phonebook with **PBX DN** "ou=pbx,dc=pbx,dc=com" based on the UCM6200 user extensions already.
- Users could add new phonebook with a different **Phonebook DN** for other external contacts. For example, "ou=people,dc=pbx,dc=com".
- All the phonebooks in the UCM6200 LDAP server have the same **Base DN** "dc=pbx,dc=com".

### Term Explanation:

cn= Common Name  
ou= Organization Unit



dc= Domain Component

These are all parts of the LDAP data Interchange Format, according to RFC 2849, which is how the LDAP tree is filtered.

If users have the Grandstream phone provisioned by the UCM6200, the LDAP directory will be set up on the phone and can be used right away for users to access all phonebooks.

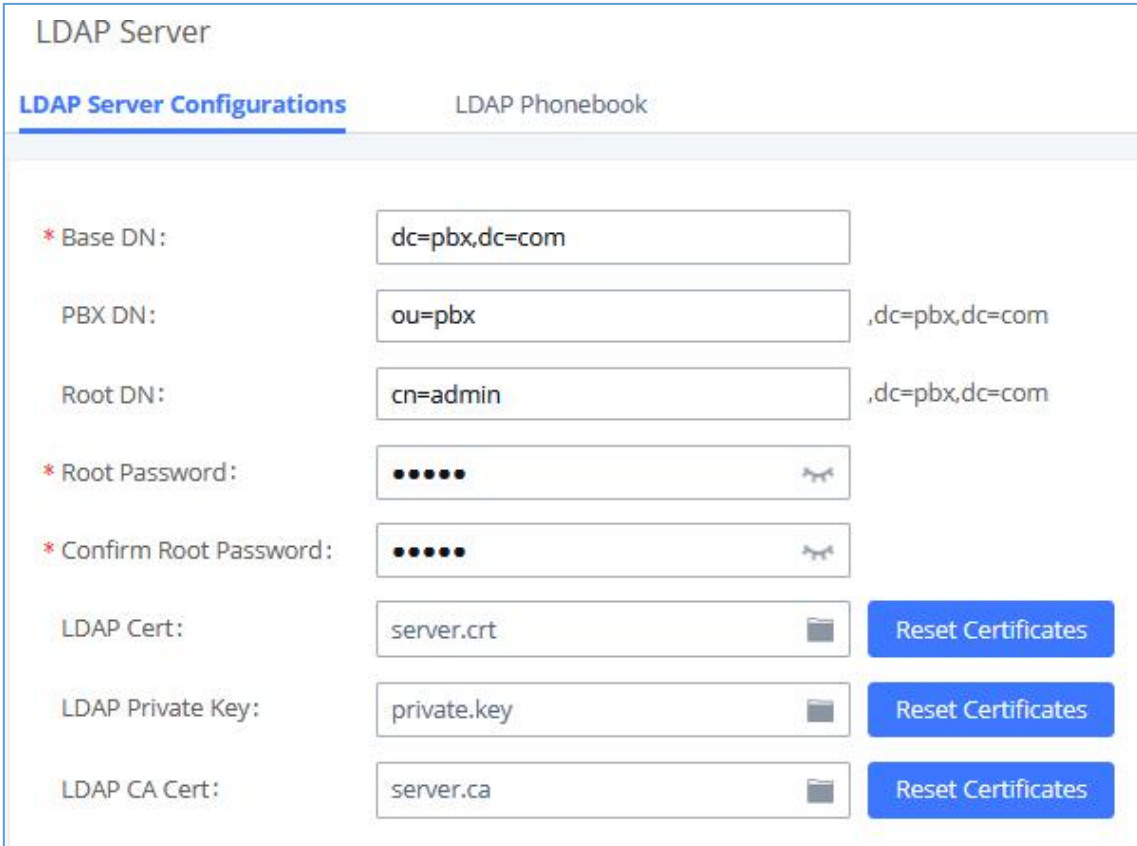
Additionally, users could manually configure the LDAP client settings to manipulate the built-in LDAP server on the UCM6200. If the UCM6200 has multiple LDAP phonebooks created, in the LDAP client configuration, users could use "dc=pbx,dc=com" as Base DN to have access to all phonebooks on the UCM6200 LDAP server, or use a specific phonebook DN, for example "ou=people,dc=pbx,dc=com", to access to phonebook with Phonebook DN "ou=people,dc=pbx,dc=com " only.

UCM can also act as a LDAP client to download phonebook entries from another LDAP server.

To access LDAP server and client settings, go to Web GUI→**Settings**→**LDAP Server**.

## LDAP Server Configurations

The following figure shows the default LDAP server configurations on the UCM6200.




LDAP Server Configurations		LDAP Phonebook
* Base DN:	<input type="text" value="dc=pbx,dc=com"/>	
PBX DN:	<input type="text" value="ou=pbx"/>	,dc=pbx,dc=com
Root DN:	<input type="text" value="cn=admin"/>	,dc=pbx,dc=com
* Root Password:	<input type="password" value="....."/>	
* Confirm Root Password:	<input type="password" value="....."/>	
LDAP Cert:	<input type="text" value="server.crt"/>	<input type="button" value="Reset Certificates"/>
LDAP Private Key:	<input type="text" value="private.key"/>	<input type="button" value="Reset Certificates"/>
LDAP CA Cert:	<input type="text" value="server.ca"/>	<input type="button" value="Reset Certificates"/>

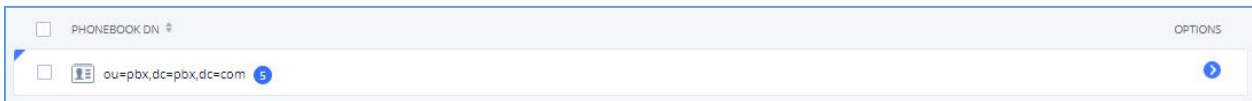
Figure 33: LDAP Server Configurations





The UCM6200 LDAP server supports anonymous access (read-only) by default. Therefore, the LDAP client does not have to configure username and password to access the phonebook directory. The "Root DN" and "Root Password" (limited with 64 and 32 characters respectively) here are for LDAP management and configuration where users will need provide for authentication purpose before modifying the LDAP information.













The default phonebook list in this LDAP server can be viewed and edited by clicking on  for the first phonebook under LDAP Phonebook.



**Figure 34: Default LDAP Phonebook DN**

Edit Phonebook: pbx

+ Add Contact

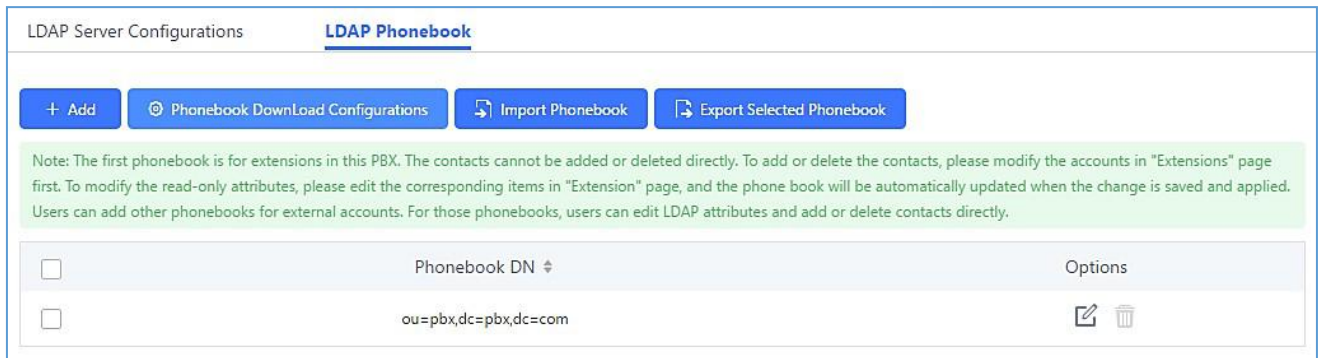
AccountNumber ↕	CallerID Name ↕	Options
1000	John DOE	 
1001		 
1002		 
1003		 
1004		 
1005		 

**Figure 35: Default LDAP Phonebook Attributes**

## LDAP Phonebook

Users could use the default phonebook, edit the default phonebook, add new phonebook, import phonebook on the LDAP server as well as export phonebook from the LDAP server. The first phonebook with default phonebook dn "ou=pbx,dc=pbx,dc=com" displayed on the LDAP server page is for extensions in this PBX. Users cannot add or delete contacts directly. The contacts information will need to be modified via Web GUI→**Extension/Trunk**→**Extensions** first. The default LDAP phonebook will then be updated automatically.

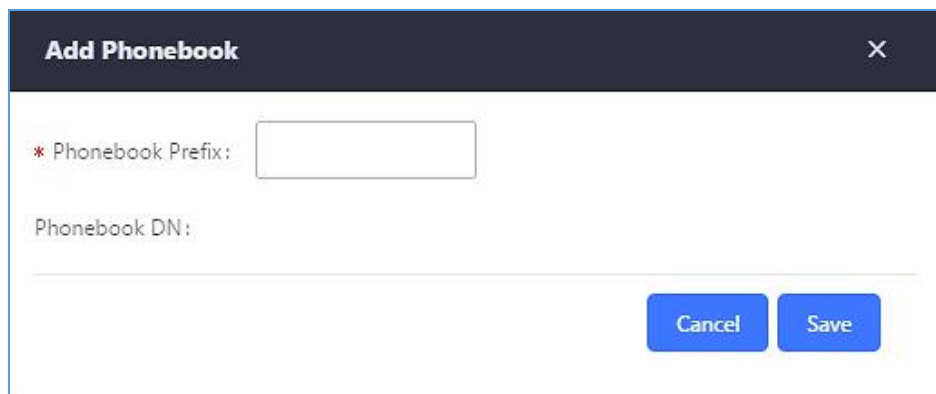






**Figure 36: LDAP Server→LDAP Phonebook**

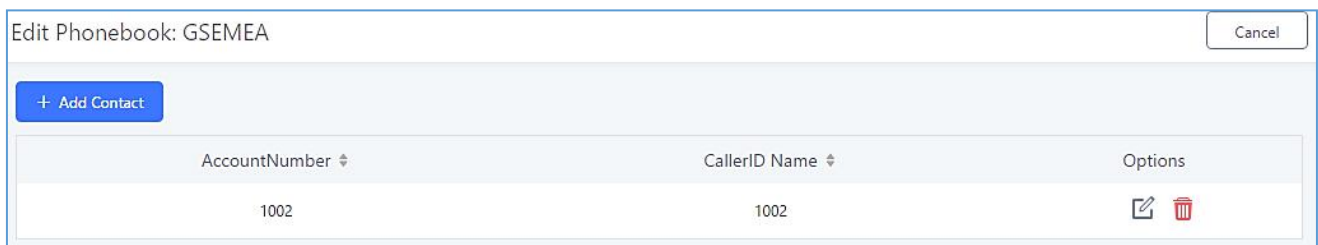
- **Add new phonebook**

A new sibling phonebook of the default PBX phonebook can be added by clicking on "Add" under "LDAP Phonebook" section.



**Figure 37: Add LDAP Phonebook**

Configure the "Phonebook Prefix" first. The "Phonebook DN" will be automatically filled in. For example, if configuring "Phonebook Prefix" as "people", the "Phonebook DN" will be filled with "ou=people,dc=pbx,dc=com". Once added, users can select  to edit the phonebook attributes and contact list (see figure below) or select  to delete the phonebook.

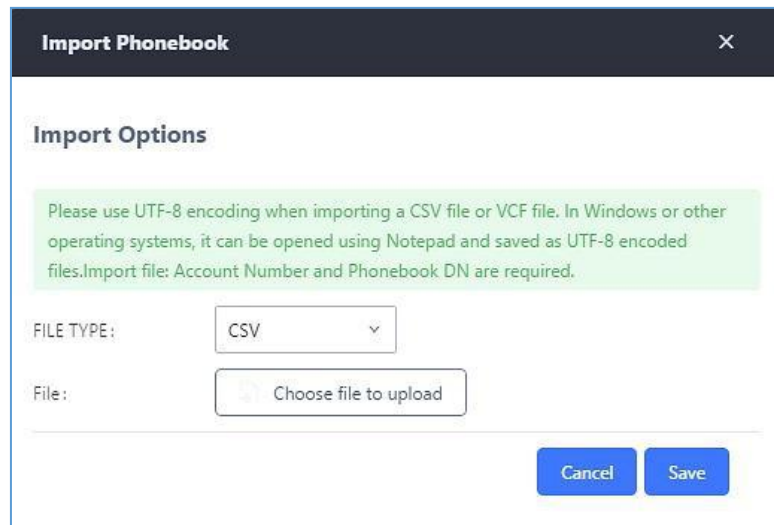


**Figure 38: Edit LDAP Phonebook**



- **Import phonebook from your computer to LDAP server**

Click on “Import Phonebook” and a dialog will prompt as shown in the figure below.



**Figure 39: Import Phonebook**

The file to be imported must be a CSV, VCF or XML file with UTF-8 encoding. Users can open the file with Notepad and save it with UTF-8 encoding.

Here is how a sample file looks like. Please note “Account Number” and “Phonebook DN” fields are required. Users could export a phonebook file from the UCM6200 LDAP phonebook section first and use it as a sample to start with.

	A	B	C	D	E	F	G	H	I	J
1	First Name	Last Name	Account Number	CallerID Name	Email	Department	Mobile Number	Home Number	Fax	Phonebook DN
2	John	Doe	1001	1001		IT	1001000000			phonebook
3	Jane	Doe	1002	1002		Sales	1002000000			phonebook
4	William	Chung	1003	1003		Marketing	1003000000			phonebook
5	Linda	Kuo	1004	1004		Accounting	1004000000			phonebook
6	Steve	Chang	1005	1005		Support	1005000000			others

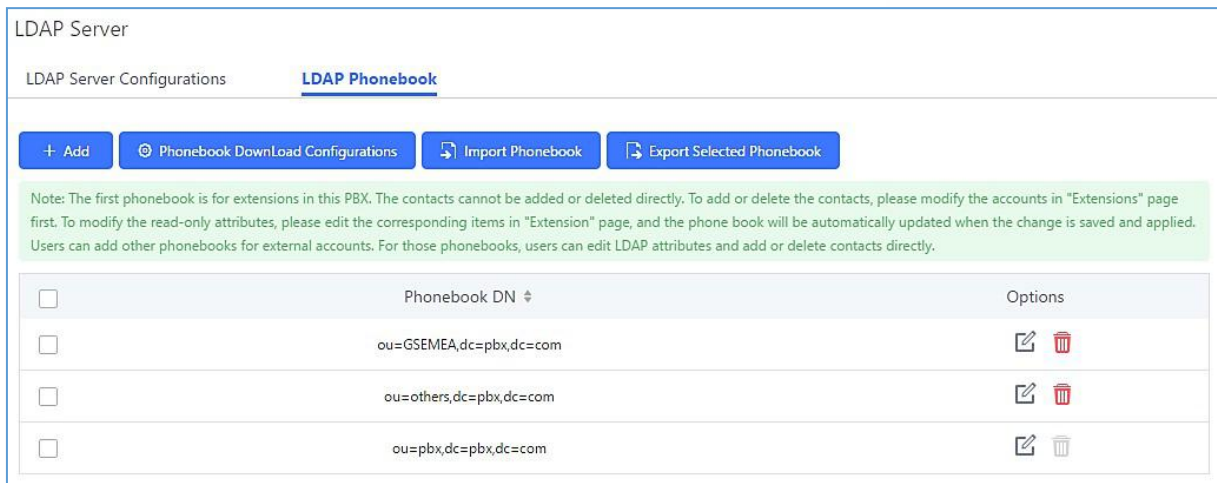
**Figure 40: Phonebook CSV File Format**

The Phonebook DN field is the same “Phonebook Prefix” entry as when the user clicks on “Add” to create a new phonebook. Therefore, if the user enters “phonebook” in “Phonebook DN” field in the CSV file, the actual phonebook DN “ou=phonebook,dc=pbx,dc=com” will be automatically created by the UCM6200 once the CSV file is imported.

In the CSV file, users can specify different phonebook DN fields for different contacts. If the phonebook DN already exists on the UCM6200 LDAP Phonebook, the contacts in the CSV file will be added into the existing phonebook. If the phonebook DN does not exist on the UCM6200 LDAP Phonebook, a new phonebook with this phonebook DN will be created.



The sample phonebook CSV file in above picture will result in the following LDAP phonebook in the UCM6200.



LDAP Server

LDAP Server Configurations **LDAP Phonebook**

+ Add Phonebook DownLoad Configurations Import Phonebook Export Selected Phonebook

Note: The first phonebook is for extensions in this PBX. The contacts cannot be added or deleted directly. To add or delete the contacts, please modify the accounts in "Extensions" page first. To modify the read-only attributes, please edit the corresponding items in "Extension" page, and the phone book will be automatically updated when the change is saved and applied. Users can add other phonebooks for external accounts. For those phonebooks, users can edit LDAP attributes and add or delete contacts directly.

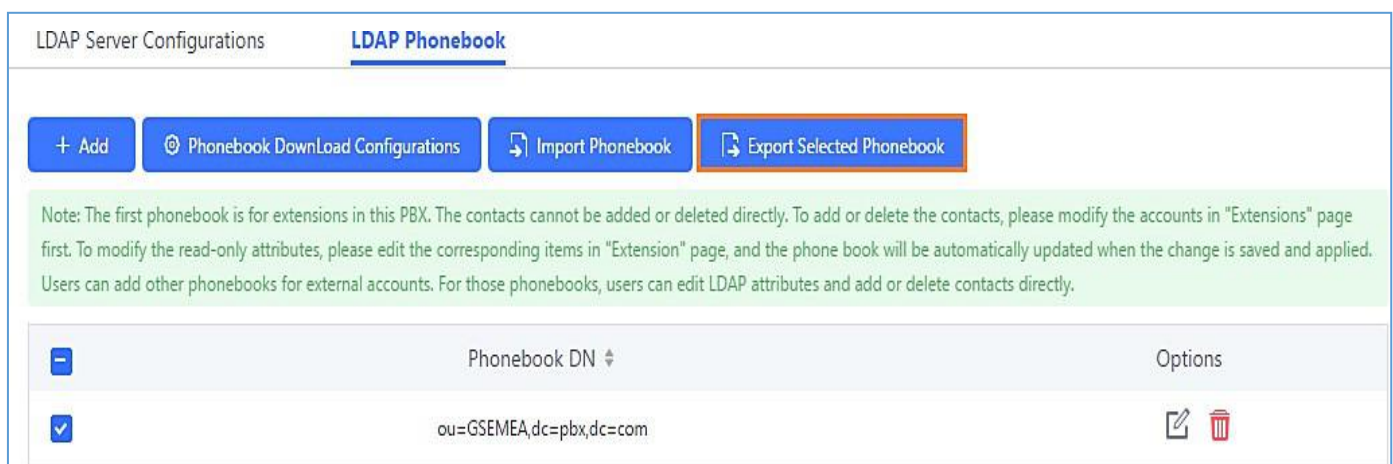
<input type="checkbox"/>	Phonebook DN ↕	Options
<input type="checkbox"/>	ou=GSEMEA,dc=pbx,dc=com	
<input type="checkbox"/>	ou=others,dc=pbx,dc=com	
<input type="checkbox"/>	ou=pbx,dc=pbx,dc=com	

**Figure 41: LDAP Phonebook After Import**

As the default LDAP phonebook with DN “ou=pbx,dc=pbx,dc=com” cannot be edited or deleted in LDAP phonebook section, users cannot import contacts with Phonebook DN field “pbx” if existed in the CSV file.

- **Export phonebook to your computer from UCM6200 LDAP server**

Select the checkbox for the LDAP phonebook and then click on “Export Selected Phonebook” to export the selected phonebook. The exported phonebook can be used as a record or a sample CSV, VFC or XML file for the users to add more contacts in it and import to the UCM6200 again.



LDAP Server Configurations **LDAP Phonebook**

+ Add Phonebook DownLoad Configurations Import Phonebook **Export Selected Phonebook**

Note: The first phonebook is for extensions in this PBX. The contacts cannot be added or deleted directly. To add or delete the contacts, please modify the accounts in "Extensions" page first. To modify the read-only attributes, please edit the corresponding items in "Extension" page, and the phone book will be automatically updated when the change is saved and applied. Users can add other phonebooks for external accounts. For those phonebooks, users can edit LDAP attributes and add or delete contacts directly.

<input type="checkbox"/>	Phonebook DN ↕	Options
<input checked="" type="checkbox"/>	ou=GSEMEA,dc=pbx,dc=com	

**Figure 42: Export Selected LDAP Phonebook**



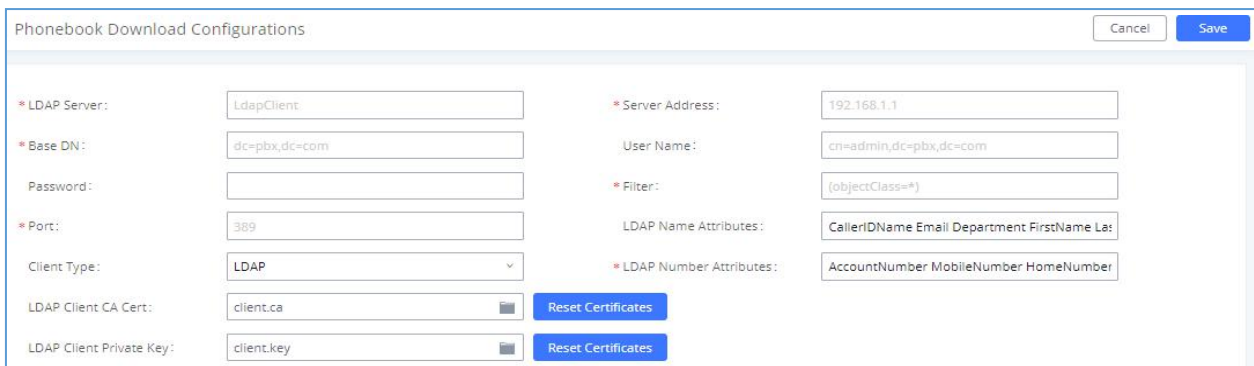
## LDAP Client Configurations

The configuration on LDAP client is useful when you use other LDAP servers. Here we provide an example on how to configure the LDAP client on the UCM.

Assuming the remote server base dn is “**dc=pbx,dc=com**”, configure the LDAP client as follows:

- **LDAP Server:** Enter a name for the remote LDAP server
- **Server Address:** Enter the IP address or domain name for remote LDAP server.
- **Base DN:** dc=pbx,dc=com
- **Username:** Enter username if authentication is required. This field cannot exceed 64 characters and can contain space.
- **Password:** Enter password if authentication is required.
- **Filter:** Enter the filter. Ex: ((CallerIDName=%)(AccountNumber=%))
- **Port:** Enter the port number. Ex:389
- **LDAP Name Attributes:** Enter the name attributes for remote server
- **LDAP Number Attributes:** Enter the number attributes for remote server
- **Client type:** Protocol of LDAP or LDAPS.
- **LDAP Client CA cert:** Upload LDAP client CA certificate, The following file types are supported: .crt .der and .pem.
- **LDAP Client Private Key:** Upload LDAP client private key.

The following figure gives a sample configuration for UCM acting as a LDAP client.



**Figure 43: LDAP Client Configurations**

To configure Grandstream IP phones as the LDAP clients for UCM, please refer to the following example:

- **Server Address:** The IP address or domain name of the UCM6200
- **Base DN:** dc=pbx,dc=com
- **Username:** Please leave this field empty



- **Password:** Please leave this field empty
- **LDAP Name Attribute:** CallerIDName Email Department FirstName LastName
- **LDAP Number Attribute:** AccountNumber MobileNumber HomeNumber Fax
- **LDAP Number Filter:** (AccountNumber=%)
- **LDAP Name Filter:** (CallerIDName=%)
- **LDAP Display Name:** AccountNumber CallerIDName
- **LDAP Version:** If existed, please select LDAP Version 3
- **Port:** 389

The following figure shows the configuration information on a Grandstream GXP2170 to successfully use the LDAP server as configured in **[Figure 33: LDAP Server Configurations]**.



## LDAP

LDAP protocol	<input type="text" value="LDAP"/>
Server Address	<input type="text" value="192.168.40.134"/>
Port	<input type="text" value="389"/>
Base	<input type="text" value="dc=pbx,dc=com"/>
User Name	<input type="text"/>
Password	<input type="password"/>
LDAP Number Filter	<input type="text" value="(AccountNumber=%)"/>
LDAP Name Filter	<input type="text" value="(CallerIDName=%)"/>
LDAP Version	<input type="radio"/> Version 2 <input checked="" type="radio"/> Version 3
LDAP Name Attributes	<input type="text" value="CallerIDName"/>
LDAP Number Attributes	<input type="text" value="AccountNumber"/>
LDAP Display Name	<input type="text" value="AccountNumber CallerIDName"/>
Max. Hits	<input type="text" value="50"/>
Search Timeout	<input type="text" value="30"/>
Sort Results	<input checked="" type="radio"/> No <input type="radio"/> Yes
LDAP Lookup	<input checked="" type="checkbox"/> Incoming Calls <input checked="" type="checkbox"/> Outgoing Calls
Lookup Display Name	<input type="text"/>

Figure 44: GXP2170 LDAP Phonebook Configuration





## Time settings

### Auto time updating

The current system time on the UCM6200 can be found under Web GUI→**System Status**→**Dashboard**→**PBX Status**.

To configure the UCM6200 to update time automatically, go to Web GUI→**System Settings**→**Time Settings**→**Auto Time Updating**.

 **Note:**

The configurations under Web GUI→**Settings**→**Time Settings**→**Auto Time Updating** page require reboot to take effect. Please consider configuring auto time updating related changes when setting up the UCM6200 for the first time to avoid service interrupt after installation and deployment in production.

**Table 17: Time Auto Updating**

<b>Remote NTP Server</b>	Specify the URL or IP address of the NTP server for the UCM6200 to synchronize the date and time. The default NTP server is ntp.ipvideotalk.com.
<b>Enable DHCP Option 2</b>	If set to "Yes", the UCM6200 can get provisioned for Time Zone from DHCP Option 2 in the local server automatically. The default setting is "Yes".
<b>Enable DHCP Option 42</b>	If set to "Yes", the UCM6200 can get provisioned for NTP Server from DHCP Option 42 in the local server automatically. This will override the manually configured NTP Server. The default setting is "Yes".
<b>Time Zone</b>	Select the proper time zone option so the UCM6200 can display correct time accordingly.  If "Self-Defined Time Zone" is selected, please specify the time zone parameters in "Self-Defined Time Zone" field as described in below option.
<b>Self-Defined Time Zone</b>	If "Self-Defined Time Zone" is selected in "Time Zone" option, users will need define their own time zone following the format below.  The syntax is: <b><i>std offset dst [offset], start [/time], end [/time]</i></b>  Default is set to: MTZ+6MDT+5,M4.1.0,M11.1.0





### MTZ+6MDT+5

This indicates a time zone with 6 hours offset and 1 hour ahead for DST, which is U.S central time. If it is positive (+), the local time zone is west of the Prime Meridian (A.K.A: International or Greenwich Meridian); If it is negative (-), the local time zone is east.

### M4.1.0,M11.1.0

The 1st number indicates Month: 1,2,3..., 12 (for Jan, Feb, ..., Dec).

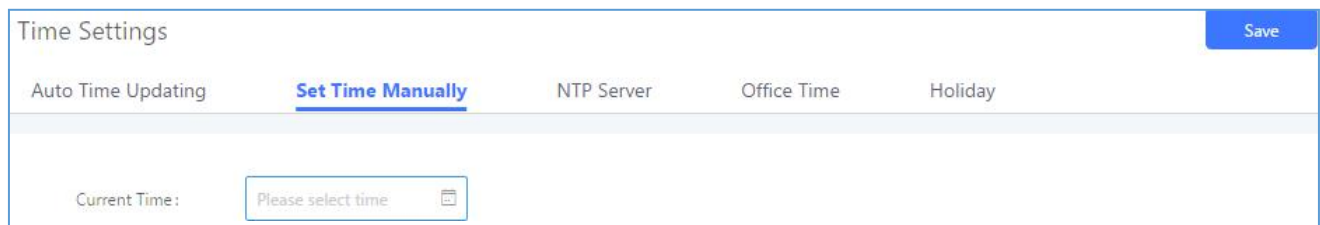
The 2nd number indicates the nth iteration of the weekday: (1st Sunday, 3rd Tuesday...). Normally 1, 2, 3, 4 are used. If 5 is used, it means the last iteration of the weekday.

The 3rd number indicates weekday: 0,1,2,...,6 ( for Sun, Mon, Tues, ..., Sat).

Therefore, this example is the DST which starts from the First Sunday of April to the 1st Sunday of November.

## Set Time Manually

To manually set the time on the UCM6200, go to Web GUI→**System Settings**→**Time Settings**→**Set Time Manually**. The format is YYYY-MM-DD HH:MM:SS.



Time Settings Save

Auto Time Updating    **Set Time Manually**    NTP Server    Office Time    Holiday

---

Current Time:

Figure 45: Set Time Manually

 **Note:**

Manually setup time will take effect immediately after saving and applying change in the Web GUI. If users would like to reboot the UCM6200 and keep the manually setup time setting, please make sure "Remote NTP Server", "Enable DHCP Option 2" and "Enable DHCP Option 42" options under Web GUI→**Settings**→**Time Settings**→**Auto Time Updating** page are unchecked or set to empty. Otherwise, time auto updating settings in this page will take effect after reboot.

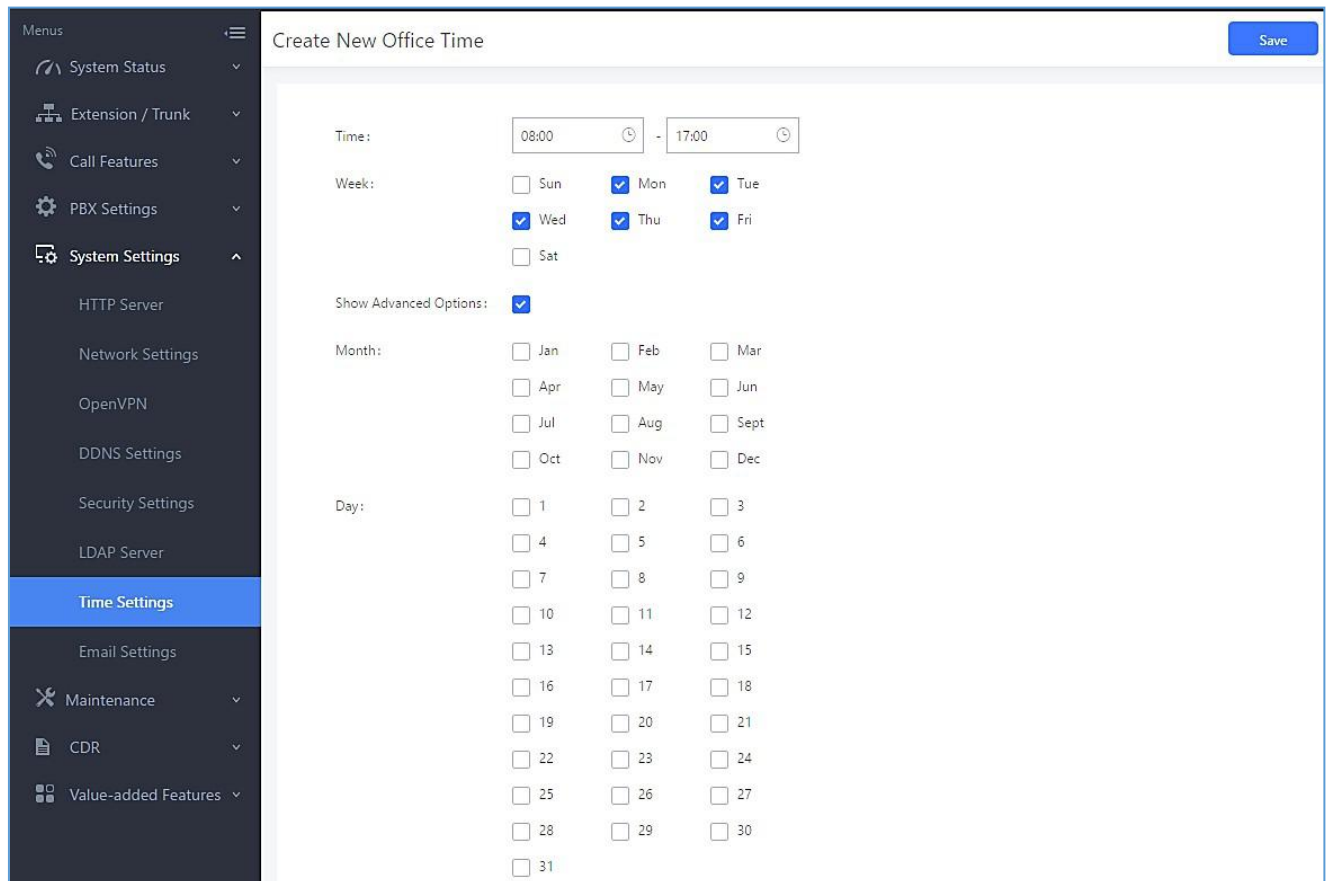


## NTP Server

The UCM6200 can be used as an NTP server for the NTP clients to synchronize their time with. To configure the UCM6200 as the NTP server, set "Enable NTP server" to "Yes" under Web GUI→**System Settings**→**Time Settings**→**NTP Server**. On the client side, point the NTP server address to the UCM6200 IP address or host name to use the UCM6200 as the NTP server.

## Office Time

On the UCM6200, the system administrator can define "office time", which can be used to configure time condition for extension call forwarding schedule and inbound rule schedule. To configure office time, go to Web GUI→**System Settings**→**Time Settings**→**Office Time**. Click on "Create New Office Time" to create an office time.



**Figure 46: Create New Office Time**

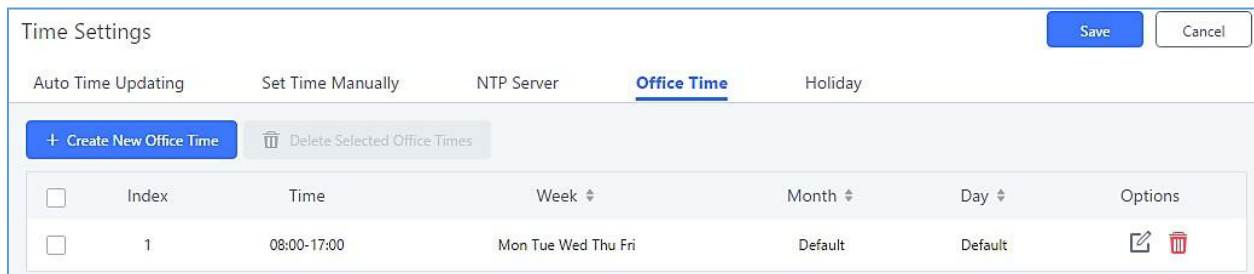
**Table 18: Create New Office Time**

<b>Start Time</b>	Configure the start time for office hour.
<b>End Time</b>	Configure the end time for office hour





<b>Week</b>	Select the workdays in one week.
<b>Show Advanced Options</b>	Check this option to show advanced options. Once selected, please specify "Month" and "Day" below.
<b>Month</b>	Select the months for office time.
<b>Day</b>	Select the workdays in one month.

Select "Start Time", "End Time" and the day for the "Week" for the office time. The system administrator can also define month and day of the month as advanced options. Once done, click on "Save" and then "Apply Change" for the office time to take effect. The office time will be listed in the web page as the figure shows below.



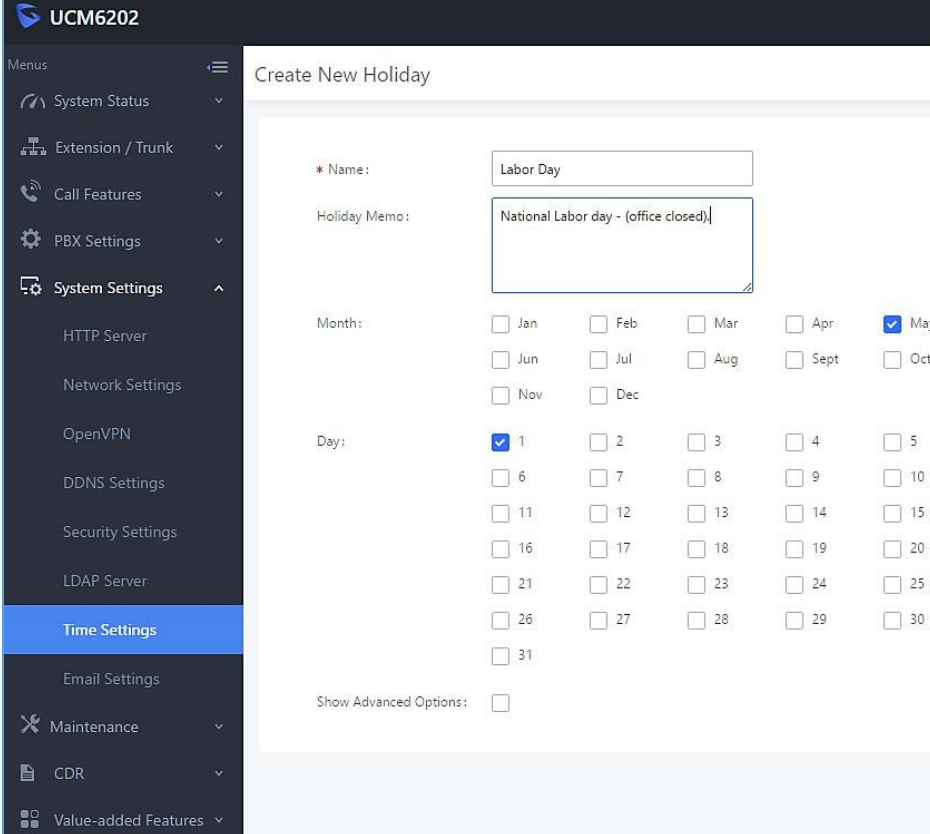
**Figure 47: Settings→Time Settings→Office Time**

- Click on  to edit the office time.
- Click on  to delete the office time.
- Click on "Delete Selected Office Times" to delete multiple selected office times at once.

## Holiday

On the UCM6200, the system administrator can define "holiday", which can be used to configure time condition for extension call forwarding schedule and inbound rule schedule. To configure holiday, go to Web GUI→**System Settings→Time Settings→Holiday**. Click on "Create New Holiday" to create holiday time.





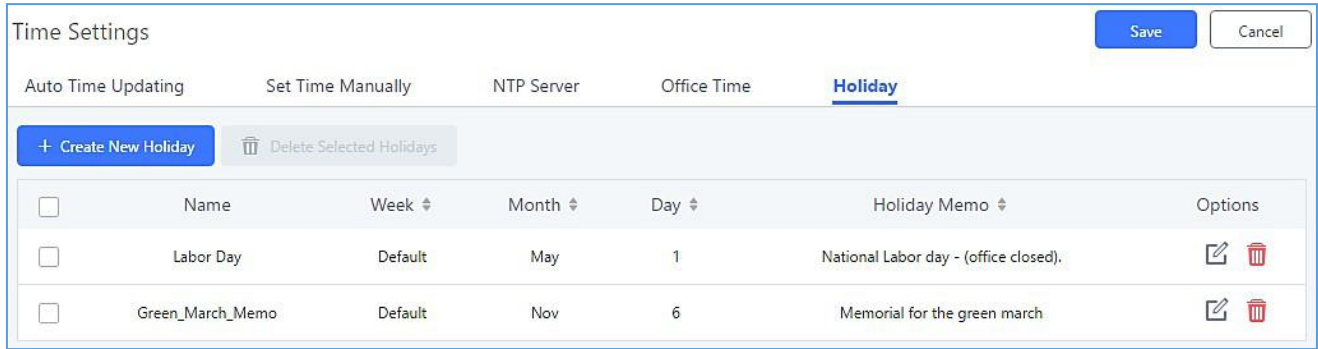
**Figure 48: Create New Holiday**

**Table 19: Create NewHoliday**



<b>Name</b>	Specify the holiday name to identify this holiday.
<b>Holiday Memo</b>	Create a note for the holiday.
<b>Year</b>	Select the year of the Holiday. <b>Note:</b> Users can now configure holidays for the next 4 years.
<b>Month</b>	Select the month for the holiday.
<b>Day</b>	Select the day for the holiday.
<b>Show Advanced Options</b>	Check this option to show advanced options. If selected, please specify the days as holiday in one week below.
<b>Week</b>	Select the days as holiday in one week.

Enter holiday "Name" and "Holiday Memo" for the new holiday. Then select "Month" and "Day". The system administrator can also define days in one week as advanced options. Once done, click on "Save" and then "Apply Change" for the holiday to take effect. The holiday will be listed in the web page as the figure shows below.





**Figure 49: Settings→Time Settings→Holiday**

- Click on  to edit the holiday.
- Click on  to delete the holiday.
- Click on "Delete Selected Holidays" to delete multiple selected holidays at once.

 **Note:**

For more details on how to use office time and holiday, please refer to the link below:

[http://www.grandstream.com/sites/default/files/Resources/office\\_time\\_and\\_holiday\\_on\\_ucm6xxx.pdf](http://www.grandstream.com/sites/default/files/Resources/office_time_and_holiday_on_ucm6xxx.pdf)

## Email Settings

### Email settings

The Email application on the UCM6200 can be used to send out alert event Emails, Fax (Fax-To-Email), Voicemail (Voicemail-To-Email) etc. The configuration parameters can be accessed via Web GUI→**System Settings**→**Email Settings**→**Email Settings**.

**Table 20: Email Settings**

<b>TLS Enable</b>	Enable or disable TLS during transferring/submitted your Email to another SMTP server. The default setting is "Yes".
<b>Type</b>	<p>Select Email type.</p> <ul style="list-style-type: none"> <li>• <b>MTA:</b> Mail Transfer Agent. The Email will be sent from the configured domain. When MTA is selected, there is no need to set up SMTP server for it or no user login is required. However, the Emails sent from MTA might be considered as spam by the target SMTP server.</li> <li>• <b>Client:</b> Submit Emails to the SMTP server. A SMTP server is required, and users need login with correct credentials.</li> </ul>




<b>Domain</b>	Specify the domain name to be used in the Email when using type "MTA".
<b>Server</b>	Specify the SMTP server when using type "Client".
<b>Username</b>	Username is required when using type "Client". Normally it is the Email address.
<b>Password</b>	Password to login for the above Username (Email address) is required when using type "Client".
<b>Enable Email-to-Fax</b>	<p>Enable or Disable Email-to-Fax feature.</p> <p>If enabled, the UCM will monitor the configured email inbox (using provided [Username] and [Password]) for emails with the subject "<b>SendFaxMail To XXX</b>" or "<b>XXX</b>".</p> <p>Subject example: SendFaxMail To 7200 Or 7200.</p> <p>The UCM will extract the attachments of detected emails and send it to the <b>XXX</b> extension by fax. The attachment must be in PDF/TIF/TIFF format.</p> <p><b>Note:</b> This field will appear when using Type "Client".</p>
<b>Email-to-Fax Blacklist/Whitelist</b>	Enables the Email to fax Blacklist/Whitelist functionality.
<b>Email-to-Fax Subject</b>	Specify the Email subject format for fax sending, the subject can be either " <b>SendFaxMail To XXX</b> " or " <b>XXX</b> " with XXX the fax number.
<b>Internal Blacklist/Whitelist</b>	Specify the Email address blacklist/whitelist for local extensions. This feature prevents faxing from unauthorized email addresses. The internal list includes only contacts with local extensions.
<b>External Blacklist/Whitelist</b>	<p>Specify the Email address blacklist/whitelist for non-local contacts. This feature prevents faxing from unauthorized email addresses. The external list is for non-local contacts.</p> <p><b>Note:</b> Multiple addresses can be separated with semicolon (;) i.e. "<b>XXX;YYY</b>".</p>
<b>POP/POP3 Server Address</b>	Configure the POP/POP3 server address for the configured username Example: pop.gmail.com
<b>POP/POP3 Server Port</b>	Configure the POP/POP3 server port for the configured username Example: 995
<b>Display Name</b>	Specify the display name in the FROM header in the Email.
<b>Sender</b>	Specify the sender's Email address. For example: pbx@example.mycompany.com.

The following figure shows a sample Email setting on the UCM6200, assuming the Email is using 192.168.6.202 as the SMTP server.



### Email Settings

Email Settings
Email Template
Email Send Log

TLS Enable:	<input checked="" type="checkbox"/>
Type:	<input type="text" value="Client"/>
Email Template Sending	<input type="text" value="HTML"/>
Format:	
* SMTP Server:	<input type="text" value="192.168.6.202:587"/>
* Enable SASL Authentication:	<input checked="" type="checkbox"/>
* Username:	<input type="text" value="adminSntp"/>
* Password:	<input type="password" value="....."/> 
Enable Email-to-Fax:	<input checked="" type="checkbox"/>
Email-to-Fax	<input type="text" value="Disable"/>
Blacklist/Whitelist:	
Email-to-Fax Subject	<input type="text" value="XXX"/>
Format:	
* POP/POP3 Server Address:	<input type="text" value="192.168.6.202"/>
* POP/POP3 Server Port:	<input type="text" value="991"/>
* Display Name:	<input type="text" value="Branch_PBX"/>
* Sender:	<input type="text" value="Branch1@domain.local"/>

















**Figure 50: UCM6200 Email Settings**

Once the configuration is finished, click on "Test". In the prompt, fill in a valid Email address to send a test Email to verify the Email settings on the UCM6200.




## Email Templates


The Email templates on the UCM6200 can be used for email notification the configuration parameters can be accessed via Web GUI→**Settings**→**Email Settings**→**Email Templates**.

Email Settings				
Email Settings		<b>Email Template</b>	Email Send Log	
				
TYPE	NAME	TIME	OPTIONS	
Alert Events	alert_template.html	2019-11-14 15:20:52 UTC-05:00		
PMS	pms_template.html	2019-11-14 15:20:52 UTC-05:00		
Emergency Calls	emergency_template.html	2019-11-14 15:20:52 UTC-05:00		
Extension	account_template.html	2019-11-14 15:20:52 UTC-05:00		
Fax	fax_template.html	2019-11-14 15:20:52 UTC-05:00		
Video Conference	mcm_template.html	2019-11-14 15:20:52 UTC-05:00		
Conference Report	conferencereport_template.html	2019-11-14 15:20:52 UTC-05:00		
Call Queue Statistics	callqueuestatistics_template.html	2019-11-14 15:20:52 UTC-05:00		
Fax Sending	sendfax_template.html	2019-11-14 15:20:52 UTC-05:00		
CDR	cdr_template.html	2019-11-14 15:20:52 UTC-05:00		
Missed Calls	missedcall_template.html	2019-11-14 15:20:52 UTC-05:00		
Voicemail	voicemail_template.html	2019-11-14 15:20:52 UTC-05:00		
User Password	password_template.html	2019-11-14 15:20:52 UTC-05:00		
Conference Schedule	conference_template.html	2019-11-14 15:20:52 UTC-05:00		

**Figure 51: Email Templates**

Press on  to upload pictures to be used on email templates.

Press  to reset all email templates to default ones.

To configure the email template, click the  button under Options column, and edit the template as desired.





Edit Email Template: Conference Schedule
Save

---

\* Subject:

\* Message in Text Format: 

This is the information of the schedule conference which you will attendee.  
  
 Topic:  
 {CNFR\_TOPIC}  
  
 Description:  
 {CNFR\_DESCRIPTION}  
  
 Schedule Time:  
 {CNFR\_WHEN}

Restore Default Template

Message in HTML Format: 

B I U ABC X<sup>2</sup> X<sub>2</sub> A
段落 宋体 16px

\${HELLO}  
 {CNFR\_MSG}  
  
 Conference Schedule Details  
 This is the information of the schedule conference which you will attendee.  
  
 Topic:  
 {CNFR\_TOPIC}  
  
 Description:  
 {CNFR\_DESCRIPTION}

Preview
Restore Default Template
Upload

**Figure 52: Conference Schedule Template**

- Users have the ability to preview mail sample by clicking on Preview.
- Click on Restore Default Template in order to restore the default email template.
- Finally, users can click on Upload to upload a custom picture to the email template to display their own logo in the sent mails for example

## Email Send Log

Under UCM Web GUI→**System Settings**→**Email Settings**→**Email Send Log**, users could search, filter and check whether the Email is sent out successfully or not. This page will also display the corresponding error message if the Email is not sent out successfully.



Email Settings

Email Settings    Email Template    **Email Send Log**

Email Send Log Filter

Show All Logs    Delete All Logs

**250** Mail sent successfully.  
**501** Address format parsing error. In MTA mode, If the recipient's email address contains unsupported characters, a 501 message will be returned. Please check if the format of the recipient's email address is correct. In Client mode, some servers also return 501 when the sender and mail accounts do not match. Please correct "Sender" for your "Mail Account".  
**535** There was a problem with account/password verification in client mode. Please check that "account and password" are configured correctly.  
**550** Possible Causes: (1)The recipient's email address does not exist or is in a disabled state. Please check the recipient's email address for errors.  
 (2)The number of destination addresses sent by the sender exceeds the maximum daily limit and is temporarily blacklisted. Please decrease the sending frequency or try again the next day.  
 (3)The sending IP does not pass the SPF permission detection of the sending domain. Messages sent in MTA mode may still return the error code even if they are sent successfully.  
**552** The message sent is too large, or the message attachment type is disabled.  
**553** Sender and mail account inconsistencies. Please configure the "Sender" for your "Mail Account".  
**554** The message is identified as spam. Please decrease the sending frequency or retry the next day.  
**none** Means no return code. If the "sending result" is deferred, there may be a problem with the mail server configuration, Please check to see if the "server" configuration is correct. If the result is bounced, there may be a problem with the domain name of the recipient's email address. Please check the message's "recipient" to make sure it is correct. If in MTA mode, please make sure that "Domain" is configured to be in the same domain as the recipient.

In MTA mode, you cannot receive SPF authentication. Therefore, even if mail is sent successfully, the return code of 550 will still be returned. Many mail servers will place non-SPF-certified mail into the trash or quarantine mailbox. If the recipient has not received sent mail, please check to see if the sent mail was placed in the recipient's trash or quarantine mailbox.

In Client mode, a 250 return code means that the Email has been sent successfully from the UCM to your proxy mail server. The Email still fails to be sent due to invalid destination address or other reasons. Please login in your configuration mail account and check whether there is System bounce notification to confirm the cause of the failure.

**Figure 53: Email Send log**



**Table 21: Email Log**

Field	Description
<b>Start Time</b>	Enter the start time for filter
<b>End Time</b>	Enter the end time for filter
<b>Receivers</b>	Enter the email recipient, while searching for multiple recipients, please separate then with comma and no spaces.
<b>Send Result</b>	Enter the status of the send result to filter with
<b>Return Code</b>	Enter the email code to filter with
<b>Email Send Module</b>	Select the email module to filter with from the drop-down list, which contains: <ul style="list-style-type: none"> <li>• All Modules</li> <li>• Extension</li> <li>• Voicemail</li> <li>• Conference Schedule</li> <li>• Emergency calls</li> <li>• Video conference Schedule</li> <li>• User Password</li> <li>• Alert Events</li> <li>• CDR</li> </ul>



- Fax
- Fax sending
- Call Queue Statistics
- Conference Reports
- PMS
- Test
- Missed calls

Email logs will be shown on bottom of the “Email Send Log” page, as shown on the following figure.

Email Generated Time	Email Send Module	Receivers	Last Send Time	Last Send Address	Send Result	Return Code	Options
2017-05-03 03:43:16	Test	mbaomar@grandstream.com	05-03 03:43:18	mbaomar@grandstream.com	sent	250	
2017-05-03 03:43:10	Test	mbaomar@grandstream.com	05-03 03:43:13	mbaomar@grandstream.com	sent	250	

**Figure 54: Email Logs**

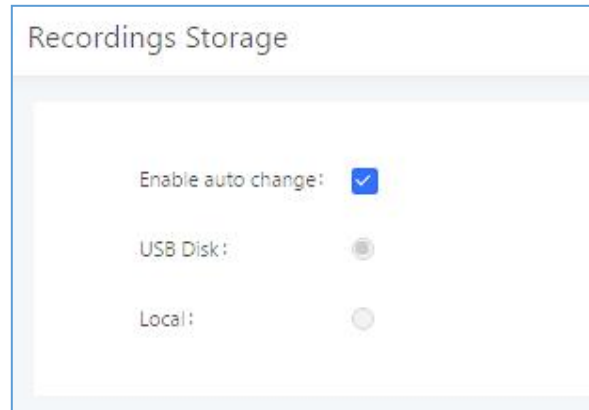
## NTP Server

The UCM62xx can be used as an NTP server for the NTP clients to synchronize their time with. To configure the UCM62xx as the NTP server, set “Enable NTP server” to “Yes” under Web GUI→**System Settings**→**Time Settings**→**NTP Server**. On the client side, point the NTP server address to the UCM62xx IP address or host name to use the UCM6xx as the NTP server.

## Recordings Storage

The UCM62xx supports call recordings automatically or manually and the recording files can be saved in external storage plugged in the UCM62xx. To manage the recording storage, users can go to UCM62xx Web GUI→**PBX Settings**→**Recordings Storage** page and select whether to store the recording files in USB Disk, SD card or locally on the UCM62xx.

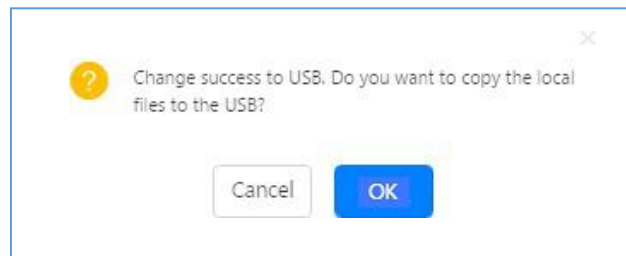




**Figure 55: PBX Settings→Recordings Storage**

- If **“Enable Auto Change”** is selected, the recording files will be automatically saved in the available USB Disk or SD card plugged into the UCM62xx. If both USB Disk and SD card are plugged in, the recording files will be always saved in the USB Disk.
- If **“Local”** is selected, the recordings will be stored in UCM62xx internal storage.
- If **“USB Disk”** or **“SD Card”** is selected, the recordings will be stored in the corresponding plugged in external storage device. Please note the options “USB Disk” and “SD Card” will be displayed only if they are plugged into the UCM62xx.

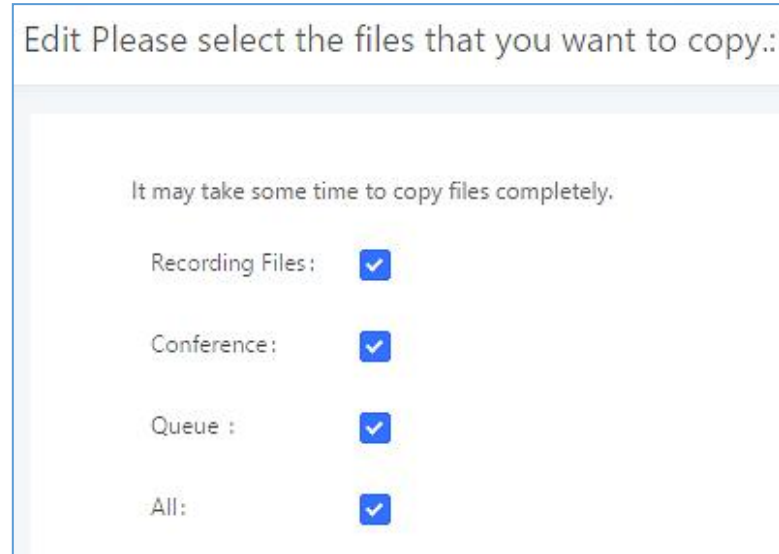
Once “USB Disk” or “SD Card” is selected, click on “OK”. The user will be prompted to confirm to copy the local files to the external storage device.



**Figure 56: Recordings Storage Prompt Information**

Click on “OK” to continue. The users will be prompted a new dialog to select the categories for the files to be copied over.





**Figure 57: Recording Storage Category**

On the UCM62xx, recording files are generated and exist in 3 categories: normal call recording files, conference recording files, and call queue recording files. Therefore, users have the following options when select the categories to copy the files to the external device:

- **Recording Files:** Copy the normal recording files to the external device.
- **Conference:** Copy the conference recording files to the external device.
- **Queue:** Copy the call queue recording files to the external device.
- **All:** Copy all recording files to the external device.



# PROVISIONING

## Overview

Grandstream SIP Devices can be configured via Web interface as well as via configuration file through TFTP/HTTP/HTTPS download. All Grandstream SIP devices support a proprietary binary format configuration file and XML format configuration file. The UCM6200 provides a Plug and Play mechanism to auto-provision the Grandstream SIP devices in a zero-configuration manner by generating XML config file and having the phone to download it within LAN area. This allows users to finish the installation with ease and start using the SIP devices in a managed way.

To provision a phone, three steps are involved, i.e., discovery, configuration and provisioning. This section explains how Zero Config works on the UCM6200. The settings for this feature can be accessed via Web GUI→**Value-added Features**→**Zero Config**.

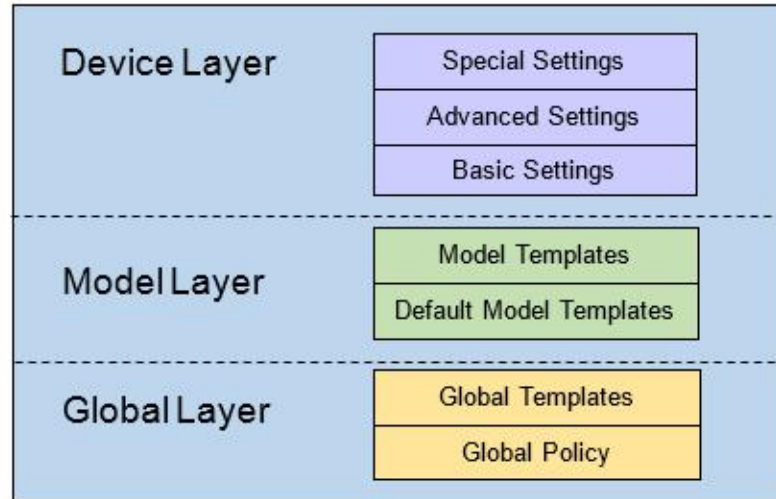
## Configuration Architecture for End Point Device

Started from firmware version 1.0.7.10, the end point device configuration in zero config is divided into the following three layers with priority from the lowest to the highest:

- **Global**  
This is the lowest layer. Users can configure the most basic options that could apply to all Grandstream SIP devices during provisioning via Zero config.
- **Model**  
In this layer, users can define model-specific options for the configuration template.
- **Device**  
This is the highest layer. Users can configure device-specific options for the configuration for individual device here.

Each layer also has its own structure in different levels. Please see figure below. The details for each layer are explained in sections **[Global configuration]**, **[Model configuration]** and **[Device Configuration]**.





**Figure 58: Zero Config Configuration Architecture for End Point Device**

The configuration options in model layer and device layer have all the option in global layers already, i.e., the options in global layer is a subset of the options in model layer and device layer. If an option is set in all three layers with different values, the highest layer value will override the value in lower layer. For example, if the user selects English for Language setting in Global Policy and Spanish for Language setting in Default Model Template, the language setting on the device to be provisioned will use Spanish as model layer has higher priority than global layer. To sum up, **configurations in higher layer will always override the configurations for the same options/fields in the lower layer when presented at the same time.**

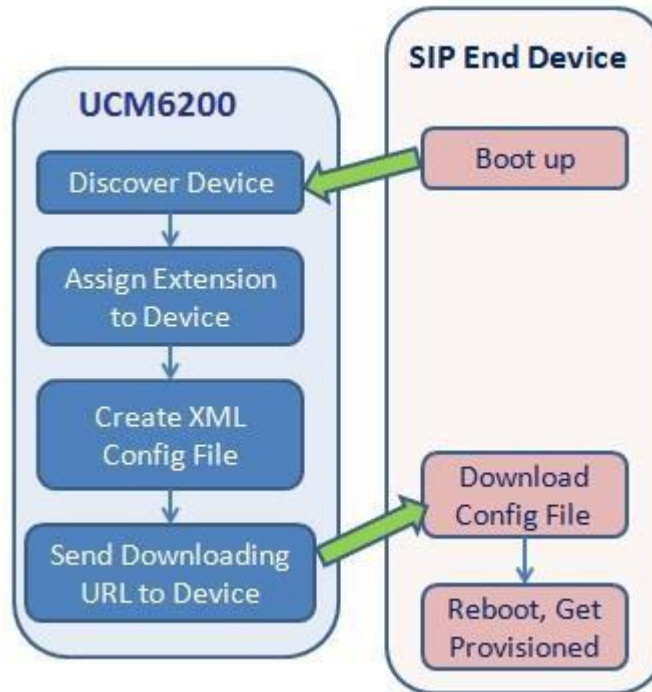
After understanding the zero-config configuration architecture, users could configure the available options for end point devices to be provisioned by the UCM6200 by going through the three layers. This configuration architecture allows users to set up and manage the Grandstream end point devices in the same LAN area in a centralized way.

## Auto Provisioning Settings

By default, the Zero Config feature is enabled on the UCM6200 for auto provisioning. Three methods of auto provisioning are used.







**Figure 59: UCM6200 Zero Config**

- **SIP SUBSCRIBE**

When the phone boots up, it sends out SUBSCRIBE to a multicast IP address in the LAN. The UCM6200 discovers it and then sends a NOTIFY with the XML config file URL in the message body. The phone will then use the path to download the config file generated in the UCM6200 and take the new configuration.

- **DHCP OPTION 66**

Route mode need to be set to use this feature. When the phone restarts (by default DHCP Option 66 is turned on), it will send out a DHCP DISCOVER request. The UCM6200 receives it and returns DHCP OFFER with the config server path URL in Option 66, for example, <https://192.168.2.1:8089/zccgi/>. The phone will then use the path to download the config file generated in the UCM6200.

- **mDNS**

When the phone boots up, it sends out mDNS query to get the TFTP server address. The UCM6200 will respond with its own address. The phone will then send TFTP request to download the XML configfile from the UCM6200.

To start the auto provisioning process, under Web GUI→**Value-added Features**→**Zero Config**→**Zero Config Settings**, fill in the auto provision information.





### Zero Config

Zero Config
Global Policy
Global Templates
Model Templates
Model Update
Zero Config Settings

#### Basic Settings

Enable Zero Config:

Enable Automatic Configuration Assignment:

#### Extension Assignment

Auto provision automatically provides an extension to the device. There are two methods of auto provision: SIP SUBSCRIBE and DHCP Option 66.

For example, when the device boots up, it will send SIP SUBSCRIBE multicast in the LAN. The PBX will find it, create an account and return a URL of the config file for the device to download.

Auto Assign Extension:

Zero Config Extension Segment: 5000 - 6299 Zero Config Extension Segment

Enable Pick Extension:

Pick Extension Segment: 4000 - 4999 Pick Extension Segment

Pick Extension Period (hour):

#### Network Settings

Subnet Whitelist:  ⊕

[Save](#)

**Figure 60: Auto Provision Settings**

**Table 22: Auto Provision Settings**

<b>Enable Zero Config</b>	Enable or disable the zero-config feature on the PBX. The default setting is enabled.
<b>Enable Automatic Configuration Assignment</b>	By default, this is disabled. If disabled, when SIP device boots up, the UCM6200 will not send the SIP device the URL to download the config file and therefore the SIP device will not be automatically provisioned by the UCM6200.  <b>Note:</b> When disabled, SIP devices can still be provisioned by manually sending NOTIFY from the UCM6200 which will include the XML config file URL for the SIP device to download.
<b>Automatically Assign Extension</b>	If enabled, when the device is discovered, the PBX will automatically assign an extension within the range defined in "Zero Config Extension Segment" to the device. The default setting is disabled.



<b>Zero Config Extension Segment</b>	Click on the link "Zero Config Extension Segment" to specify the extension range to be assigned if "Automatically Assign Extension" is enabled. The default range is 5000-6299. Zero Config Extension Segment range can be defined in Web GUI→ <b>PBX Settings</b> → <b>General Settings</b> → <b>General page</b> →Extension Preference section: "Auto Provision Extensions".
<b>Enable Pick Extension</b>	If enabled, the extension list will be sent out to the device after receiving the device's request. This feature is for the GXP series phones that support selecting extension to be provisioned via phone's LCD. The default setting is disabled.
<b>Pick Extension Segment</b>	Click on the link "Pick Extension Segment" to specify the extension list to be sent to the device. The default range is 4000 to 4999. Pick Extension Segment range can be defined in Web GUI→ <b>PBX Settings</b> → <b>General Settings</b> → <b>General page</b> →Extension Preference section: "Pick Extensions".
<b>Pick Extension Period (hour)</b>	Specify the number of minutes to allow the phones being provisioned to pick extensions.
<b>Subnet Whitelist</b>	This feature allows the UCM to provision devices in different subnets other than UCM network. Enter subnets IP addresses to allow devices within these subnets to be provisioned. The syntax is <b>&lt;IP&gt;/&lt;CIDR&gt;</b> . <u>Examples:</u> 10.0.0.1/8 192.168.6.0/24 <b>Note:</b> Only private IP ranges (10.0.0.0   172.16.0.0   192.168.0.0) are supported.

Please make sure an extension is manually assigned to the phone or "Automatically Assign Extension" is enabled during provisioning. After the configuration on the UCM6200 Web GUI, click on "Save" and "Apply Changes". Once the phone boots up and picks up the config file from the UCM6200, it will take the configuration right away.

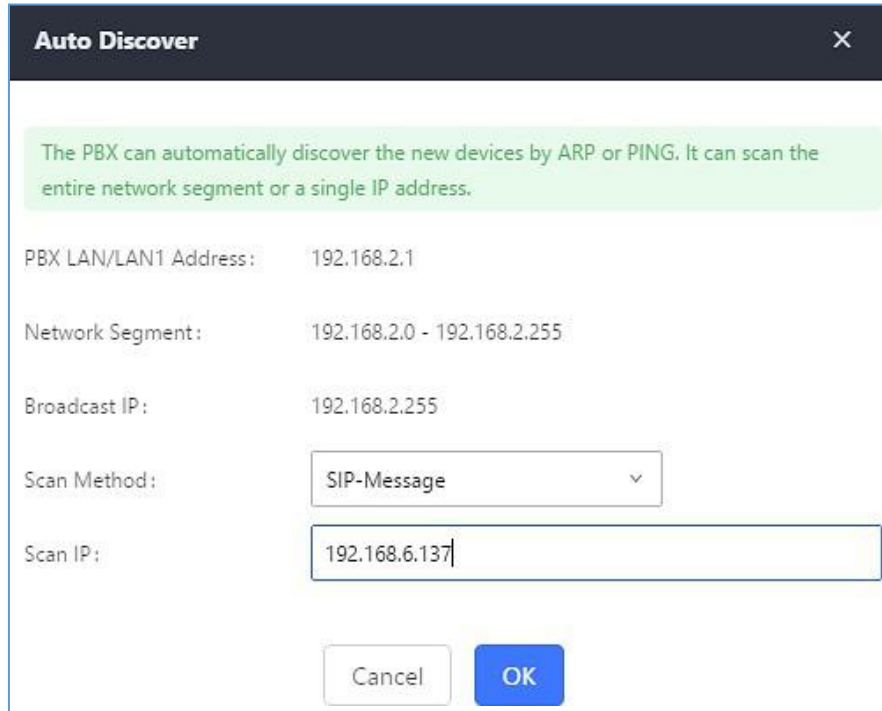
## Discovery

Grandstream endpoints are automatically discovered after bootup. Users could also manually discover device by specifying the IP address or scanning the entire LAN network. Three methods are supported to scan the devices.

- PING
- ARP
- SIP Message (NOTIFY)



Click on "Auto Discover" under Web GUI→**Value-added Features**→**Zero Config**→**Zero Config**, fill in the "Scan Method" and "Scan IP". The IP address segment will be automatically filled in based on the network mask detected on the UCM6200. If users need scan the entire network segment, enter 255 (for example, 192.168.40.255) instead of a specific IP address. Then click on "Save" to start discovering the devices within the same network. To successfully discover the devices, "Zero Config" needs to be enabled on the UCM6200 Web GUI→**Value-added Features**→**Zero Config**→**Auto Provisioning Settings**.

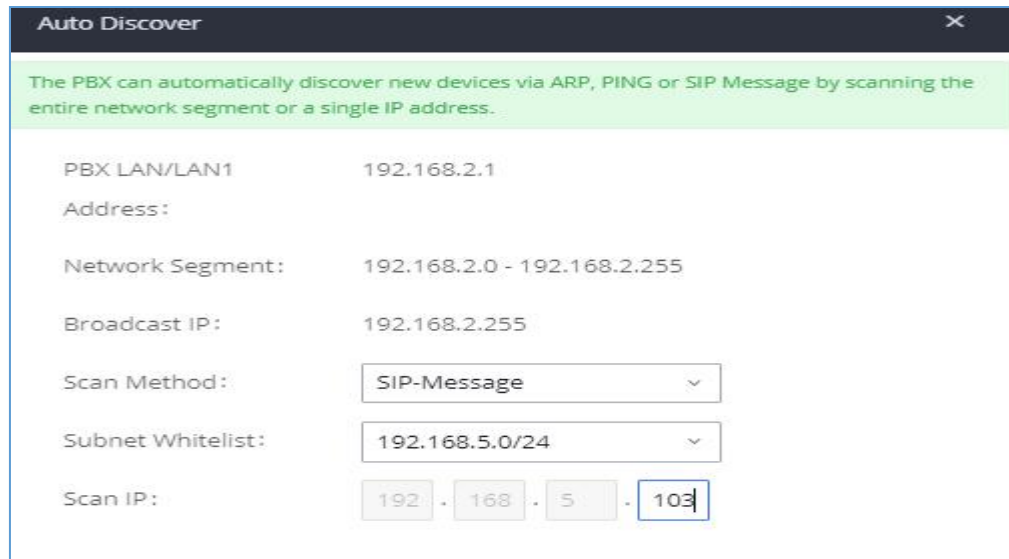


**Figure 61: Auto Discover**







The following figure shows a list of discovered phones. The MAC address, IP Address, Extension (if assigned), Version, Vendor, Model, Connection Status, Create Config, Options (Edit /Delete /Update /Reboot /Access Device WebGUI) are displayed in the list.

Auto Discover can also search for devices located on other subnets, in condition for the subnet to be added under **Zero Config Settings> Subnet Whitelist**. The method allowed to auto discover other subnets then the UCM's is **SIP-Message** like shown below.





**Figure 62: Auto Discover other subnets**

<input type="checkbox"/>	MAC Address ↓	IP Address ↓	Extension	Version ↓	Vendor ↓	Model ↓	Create Config ↓	Options
<input type="checkbox"/>	000B825C6926	192.168.2.104	--	1.0.9.17	GRANDSTREAM	GXP2160	--	    
<input type="checkbox"/>	000B82836616	192.168.6.175	--	1.0.9.14	GRANDSTREAM	GXP2160	--	    
<input type="checkbox"/>	000B82866015	192.168.2.101	--	1.0.9.11	GRANDSTREAM	GXP2170	--	    
<input type="checkbox"/>	000B82A206D8	192.168.6.241	--	1.0.8.50	GRANDSTREAM	GXP2160	--	    
<input type="checkbox"/>	000B8275C888	192.168.6.137	--	1.0.8.50	GRANDSTREAM	GXP2130	--	    

**Figure 63: Discovered Devices**

## Uploading Devices List

Besides the built-in discovery method on the UCM, users could prepare a list of devices on .CSV file and upload it by clicking on the button **Choose File to Upload**, after which a success message prompt should be displayed. Users need to make sure that the CSV file respects the format as shown on the following figure and that the entered information is correct (valid IP address, valid MAC address, device model and an existing account), otherwise the UCM will reject the file and the operation will fail:

	A	B	C	D	E
1	mac	model	ip	version	account
2	000b82ec21a5	GXP1630	192.168.2.103	1.0.3.132	5555
3	000b825c6927	GXP2160	192.168.5.108	1.0.9.135	5551

**Figure 64: Device list - CSV file sample**

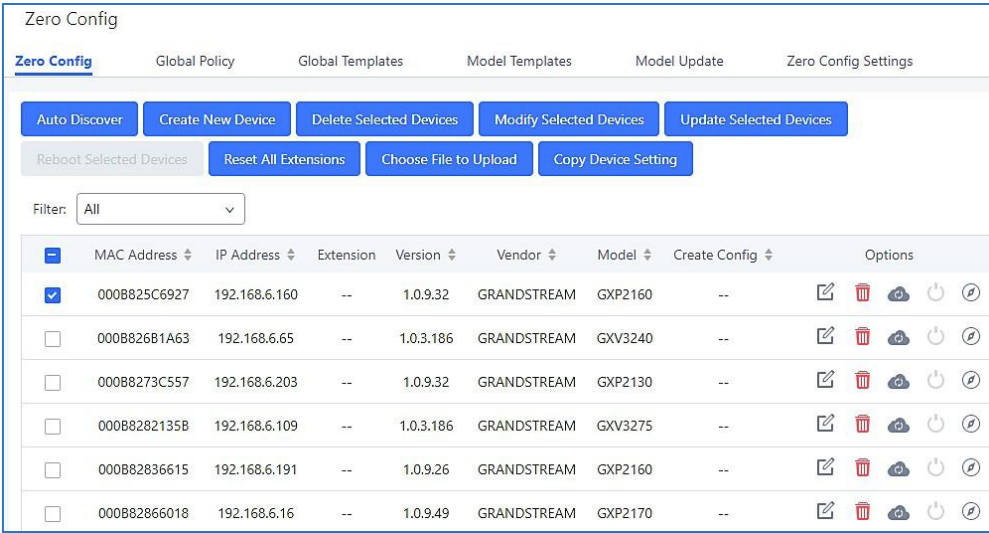


## Managing discovered devices

- **Sorting:** Press ▲ or ▼ per MAC Address, IP Address, Version, Vendor, Model or Create Config columns from lower to higher or higher to lower, respectively.

Filter:

- **Filter:** Select a filter to display corresponding results.
  - **All:** Display all discovered devices.
  - **Scan Results:** Display only manually discovered devices. [Discovery]
  - **IP Address:** Enter device IP and press **Search** button.
  - **MAC Address:** Enter device MAC and press **Search** button.
  - **Model:** Enter a model name and press **Search** button. Example: GXP2130.



	MAC Address	IP Address	Extension	Version	Vendor	Model	Create Config	Options
<input checked="" type="checkbox"/>	000B825C6927	192.168.6.160	--	1.0.9.32	GRANDSTREAM	GXP2160	--	
<input type="checkbox"/>	000B826B1A63	192.168.6.65	--	1.0.3.186	GRANDSTREAM	GXV3240	--	
<input type="checkbox"/>	000B8273C557	192.168.6.203	--	1.0.9.32	GRANDSTREAM	GXP2130	--	
<input type="checkbox"/>	000B8282135B	192.168.6.109	--	1.0.3.186	GRANDSTREAM	GXV3275	--	
<input type="checkbox"/>	000B82836615	192.168.6.191	--	1.0.9.26	GRANDSTREAM	GXP2160	--	
<input type="checkbox"/>	000B82866018	192.168.6.16	--	1.0.9.49	GRANDSTREAM	GXP2170	--	

**Figure 65: Managing Discovered Devices**

From the main menu of zero config, users can perform the following operations:

- Click on **Auto Discover** in order to access to the discovery menu as shown on [Discovery] section.
- Click on **Create New Device** to add a new device to zero config database using its MAC address.
- Click on **Delete Selected Devices** to delete selected devices from the zero-config database.
- Click on **Modify Selected Devices** to modify selected devices.
- Click on **Update Selected Devices** to batch update a list of devices, the UCM on this case will send SIP NOTIFY message to all selected devices in order to update them at once.
- Click on **Reboot Selected Devices** to reboot selected devices (the selected devices, should have been provisioned with extensions since the phone will authenticate the server which is trying to send it reboot command).



- Click on **Reset All Extensions** to clear all devices configurations.
- Click on **Choose File to Upload** to upload CSV file containing list of devices.
- Click on **Copy Device Setting** to copy configuration from one device to another. This can be useful for easily replace devices and note that this feature works only between devices of same model.

All these operations will be detailed on the next sections.

## Global configuration

### Global policy

Global configuration will apply to all the connected Grandstream SIP end point devices in the same LAN with the UCM6200 no matter what the Grandstream device model it is. It is divided into two levels:

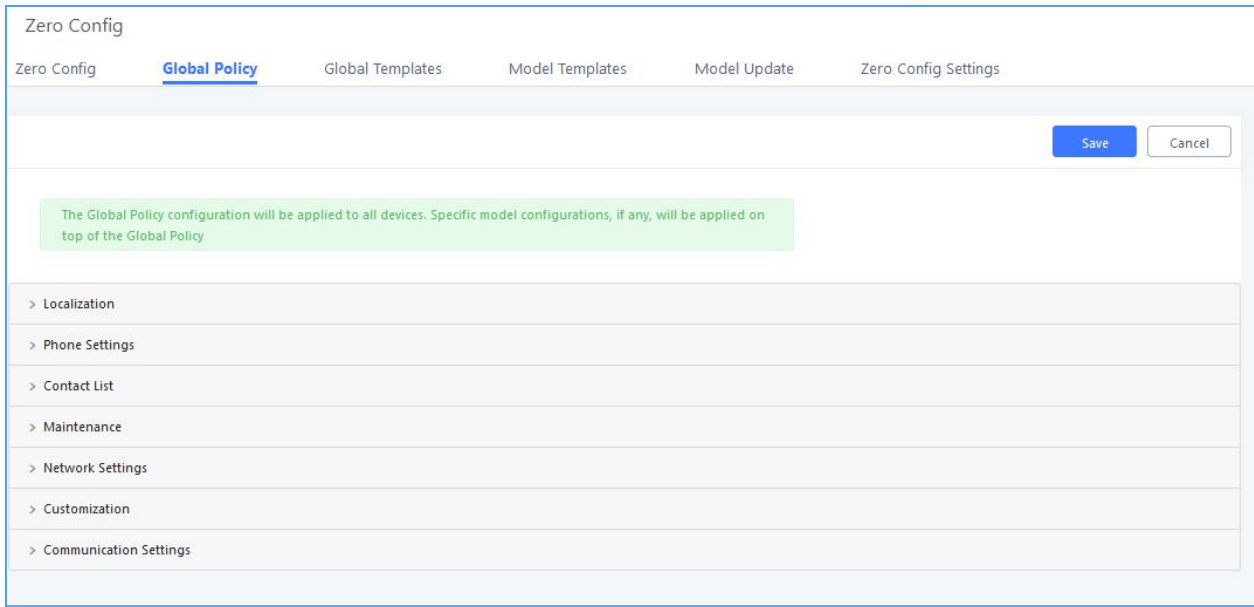
- Web GUI→**Value-added Features**→**Zero Config**→**Global Policy**
- Web GUI→**Value-added Features**→**Zero Config**→**Global Templates**.
- **Global Templates** configuration has higher priority to **Global Policy** configuration.

Global Policy can be accessed in Web GUI→**Value-added Features**→**Zero Config**→**Global Policy** page. On the top of the configuration table, users can select category in the “Options” dropdown list to quickly navigate to the category. The categories are:

- **Localization**: configure display language, data and time.
- **Phone Settings**: configure dial plan, call features, NAT, call progress tones and etc.
- **Contact List**: configure LDAP and XML phonebook download.
- **Maintenance**: configure upgrading, web access, Telnet/SSH access and syslog.
- **Network Settings**: configure IP address, QoS and STUN settings.
- **Customization**: customize LCD screen wallpaper for the supported models.
- **Communication Settings**: configure Email and FTP settings

Select the checkbox on the left of the parameter you would like to configure to activate the dropdown list for this parameter.





**Figure 66: Global Policy Categories**

The following tables list the Global Policy configuration parameters for the SIP end device.

**Table 23: Global Policy Parameters – Localization**

Language settings	
<b>Language</b>	Select the LCD display language on the SIP end device.
Date and Time	
<b>Date Format</b>	Configure the date display format on the SIP end device's LCD.
<b>Time Format</b>	Configure the time display in 12-hour or 24-hour format on the SIP end device's LCD.
<b>NTP Server</b>	Configure the URL or IP address of the NTP server. The SIP end device may obtain the date and time from the server.
<b>Time Zone</b>	Configure the time zone used on the SIP end device.

**Table 24: Global Policy Parameters – Phone Settings**

Default Call Settings	
<b>Dial Plan</b>	Configure the default dial plan rule. For syntax and examples, please refer to user manual of the SIP devices to be provisioned for more details.
<b>Enable Call Features</b>	When enabled, "Do Not Disturb", "Call Forward" and other call features can be used via the local feature code on the phone. Otherwise, the ITSP feature code will be used.
<b>Use # as Dial Key</b>	If set to "Yes", pressing the number key "#" will immediately dial out the input digits.





<b>Auto Answer by Call-info</b>	If set to “Yes”, the phone will automatically turn on the speaker phone to answer incoming calls after a short reminding beep, based on the SIP Call-Info header sent from the server/proxy. The default setting is enabled.
<b>NAT Traversal</b>	Configure if NAT traversal mechanism is activated.
<b>User Random Port</b>	If set to “Yes”, this parameter will force random generation of both the local SIP and RTP ports.
<b>General Settings</b>	
<b>Call Progress Tones</b>	Configure call progress tones including ring tone, dial tone, second dial tone, message waiting tone, ring back tone, call waiting tone, busy tone and reorder tone using the following syntax: f1=val, f2=val[, c=on1/ off1[- on2/ off2[- on3/ off3]]]; <ul style="list-style-type: none"> <li>Frequencies are in Hz and cadence on and off are in 10ms).</li> <li>“on” is the period (in ms) of ringing while “off” is the period of silence. Up to three cadences are supported.</li> <li>Please refer to user manual of the SIP devices to be provisioned for more details</li> </ul>
<b>HEADSET Key Mode</b>	Select “Default Mode” or “Toggle Headset/Speaker” for the Headset key. Please refer to user manual of the SIP devices to be provisioned for more details.

**Table 25: Global Policy Parameters – Contact List**

<b>LDAP Phonebook</b>	
<b>Source</b>	Select “Manual” or “PBX” as the LDAP configuration source. <ul style="list-style-type: none"> <li>If “Manual” is selected, the LDAP configuration below will be applied to the SIP end device.</li> <li>If “PBX” is selected, the LDAP configuration built-in from UCM6200 Web GUI→<b>System Settings</b>→<b>LDAP Server</b> will be applied.</li> </ul>
<b>Address</b>	Configure the IP address or DNS name of the LDAP server.
<b>Port</b>	Configure the LDAP server port. The default value is 389.
<b>Base DN</b>	This is the location in the directory where the search is requested to begin. Example: <ul style="list-style-type: none"> <li>dc=grandstream, dc=com</li> <li>ou=Boston, dc=grandstream, dc=com</li> </ul>
<b>Username</b>	Configure the bind “Username” for querying LDAP servers. The field can be left blank if the LDAP server allows anonymous binds.
<b>Password</b>	Configure the bind “Password” for querying LDAP servers. The field can be left blank if the LDAP server allows anonymous binds.





<b>Number Filter</b>	Configure the filter used for number lookups. Please refer to user manual for more details.
<b>Name Filter</b>	Configure the filter used for name lookups. Please refer to user manual for more details.
<b>Version</b>	Select the protocol version for the phone to send the bind requests. The default value is 3.
<b>Name Attribute</b>	Specify the “name” attributes of each record which are returned in the LDAP search result. <u>Example:</u> gn cn sn description
<b>Number Attribute</b>	Specify the “number” attributes of each record which are returned in the LDAP search result. <u>Example:</u> telephoneNumber telephoneNumber Mobile
<b>Display Name</b>	Configure the entry information to be shown on phone’s LCD. Up to 3 fields can be displayed. <u>Example:</u> %cn %sn %telephoneNumber
<b>Max Hits</b>	Specify the maximum number of results to be returned by the LDAP server. Valid range is 1 to 3000. The default value is 50.
<b>Search Timeout</b>	Specify the interval (in seconds) for the server to process the request and client waits for server to return. Valid range is 0 to 180. Default value is 30.
<b>Sort Results</b>	Specify whether the searching result is sorted or not. Default setting is No.
<b>Incoming Calls</b>	Configure to enable LDAP number searching when receiving calls. The default setting is No.
<b>Outgoing Calls</b>	Configure to enable LDAP number searching when making calls. The default setting is No.
<b>Lookup Display Name</b>	Configures the display name when LDAP looks up the name for incoming call or outgoing call. It must be a subset of the LDAP Name Attributes.
<b>XML Phonebook</b>	
<b>Phonebook XML Server</b>	Select the source of the phonebook XML server. <ul style="list-style-type: none"> <li>• <b>Disable</b> Disable phonebook XML downloading.</li> <li>• <b>Manual</b> Once selected, users need specify downloading protocol HTTP, HTTPS or TFTP and the server path to download the phonebook XML</li> </ul>



	<p>file. The server path could be IP address or URL, with up to 256 characters.</p> <ul style="list-style-type: none"> <li> <b>Local UCM Server</b>            Once selected, click on the Server Path field to upload the phonebook XML file. Please note after uploading the phonebook XML file to the server, the original file name will be used as the directory name and the file will be renamed as phonebook.xml under that directory.         </li> </ul>
<b>Phonebook Download Interval</b>	Configure the phonebook download interval (in Minute). If set to 0, automatic download will be disabled. Valid range is 5 to 720.
<b>Remove manually-edited entries on download</b>	If set to “Yes”, when XML phonebook is downloaded, the entries added manually will be automatically removed.

**Table 26: Global Policy Parameters – Maintenance**

Upgrade and Provision	
<b>Firmware Source</b>	<p>Firmware source via ZeroConfig provisioning could a URL for external server address, local UCM directory or USB media if plugged in to the UCM6200.</p> <p>Select a source to get the firmware file:</p> <ul style="list-style-type: none"> <li> <b>URL</b>            If select to use URL to upgrade, complete the configuration for the following four parameters: “Upgrade Via”, “Server Path”, “File Prefix” and “File Postfix”.         </li> <li> <b>Local UCM Server</b>            Firmware can be uploaded to the UCM6200 internal storage for firmware upgrade. If selected, click on “Manage Storage” icon next to “Directory” option, upload firmware file and select directory for the end device to retrieve the firmware file.         </li> <li> <b>Local USB Media</b>            If selected, the USB storage device needs to be plugged into the UCM6200 and the firmware file must be put under a folder named “ZC_firmware” in the USB storage root directory.         </li> <li> <b>Local SD Card Media</b>            If selected, an SD card needs to be plugged into the UCM6200 and the firmware file must be put under a folder named “ZC_firmware” in the USB storage root directory.         </li> </ul>
<b>Upgrade via</b>	When URL is selected as firmware source, configure upgrade via TFTP, HTTP or HTTPS.



<b>Server Path</b>	When URL is selected as firmware source, configure the firmware upgrading server path.
<b>File Prefix</b>	When URL is selected as firmware source, configure the firmware file prefix. If configured, only the firmware with the matching encrypted prefix will be downloaded and flashed into the phone, if URL is selected as firmware source.
<b>File Postfix</b>	When URL is selected as firmware source, configure the firmware file postfix. If configured, only the configuration file with the matching encrypted postfix will be downloaded and flashed into the phone.
<b>Allow DHCP Option 43/66</b>	If DHCP option 43 or 66 is enabled on the LAN side, the TFTP server can be redirected.
<b>Automatic Upgrade</b>	<p>If enabled, the end point device will automatically upgrade if a new firmware is detected. Users can select automatic upgrading by day, by week or by minute.</p> <ul style="list-style-type: none"> <li> <b>By week</b>            Once selected, specify the day of the week to check HTTP/TFTP server for firmware upgrades or configuration files changes.         </li> <li> <b>By day</b>            Once selected, specify the hour of the day to check the HTTP/TFTP server for firmware upgrades or configuration files changes.         </li> <li> <b>By minute</b>            Once selected, specify the interval <b>X</b> that the SIP end device will request for new firmware every <b>X</b> minutes.         </li> </ul>
<b>Firmware Upgrade Rule</b>	Specify how firmware upgrading and provisioning request to be sent.
<b>Web Access</b>	
<b>Admin Password</b>	Configure the administrator password for admin level login.
<b>End-User Password</b>	Configure the end-user password for the end user level login.
<b>Web Access Mode</b>	Select HTTP or HTTPS as the web access protocol.
<b>Web Server Port</b>	Configure the port for web access. The valid range is 1 to 65535.
<b>Security</b>	
<b>Disable Telnet/SSH</b>	Enable Telnet/SSH access for the SIP end device. If the SIP end device supports Telnet access, this option controls the Telnet access of the device; if the SIP end device supports SSH access, this option controls the SSH access of the device.
<b>Syslog</b>	
<b>Syslog Server</b>	Configure the URL/IP address for the syslog server.
<b>Syslog Level</b>	Select the level of logging for syslog.
<b>Send SIP Log</b>	Configure whether the SIP log will be included in the syslog message.



**Table 27: Global Policy Parameters – Network Settings**

Basic Settings	
<b>IP Address</b>	Configure how the SIP end device shall obtain the IP address. DHCP or PPPoE can be selected. <ul style="list-style-type: none"> <li>• <b>DHCP</b> Once selected, users can specify the Host Name (option 12) of the SIP end device as DHCP client, and Vendor Class ID (option 60) used by the client and server to exchange vendor class ID information.</li> <li>• <b>PPPoE</b> Once selected, users need specify the Account ID, Password and Service Name for PPPoE.</li> </ul>
Advanced Setting	
<b>Layer 3 QoS</b>	Define the Layer 3 QoS parameter. This value is used for IP Precedence, Diff-Serv or MPLS. Valid range is 0-63.
<b>Layer 2 QoS Tag</b>	Assign the VLAN Tag of the Layer 2 QoS packets. Valid range is 0 -4095.
<b>Layer 2 QoS Priority Value</b>	Assign the priority value of the Layer 2 QoS packets. Valid range is 0-7.
<b>STUN Server</b>	Configure the IP address or Domain name of the STUN server. Only non-symmetric NAT routers work with STUN.
<b>Keep Alive Interval</b>	Specify how often the phone will send a blank UDP packet to the SIP server in order to keep the “ping hole” on the NAT router to open. Valid range is 10-160.

**Table 28: Global Policy Parameters – Customization**

Wallpaper	
<b>Screen Resolution 1024 x 600</b>	Check this option if the SIP end device shall use 1024 x 600 resolution for the LCD screen wallpaper. <ul style="list-style-type: none"> <li>• <b>Source</b> Configure the location where wallpapers are stored.</li> <li>• <b>File</b> If “URL” is selected as source, specify the URL of the wallpaper file. If “Local UCM Server” is selected as source, click to upload wallpaper file to the UCM6200.</li> </ul>
<b>Screen Resolution 800 x 400</b>	Check this option if the SIP end device shall use 800 x 400 resolution for the LCD screen wallpaper. <ul style="list-style-type: none"> <li>• <b>Source</b> Configure the location where wallpapers are stored.</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>File</b> If “URL” is selected as source, specify the URL of the wallpaper file. If “Local UCM Server” is selected as source, click to upload wallpaper file to the UCM6200.</li> </ul>
<b>Screen Resolution 480 x 272</b>	<p>Check this option if the SIP end device shall use 480 x 272 resolution for the LCD screen wallpaper.</p> <ul style="list-style-type: none"> <li>• <b>Source</b> Configure the location where wallpapers are stored.</li> <li>• <b>File</b> If “URL” is selected as source, specify the URL of the wallpaper file. If “Local UCM Server” is selected as source, click to upload wallpaper file to the UCM6200.</li> </ul>
<b>Screen Resolution 320 x 240</b>	<p>Check this option if the SIP end device supports 320 x 240 resolution for the LCD screen wallpaper.</p> <ul style="list-style-type: none"> <li>• <b>Source</b> Configure the location where wallpapers are stored.</li> <li>• <b>File</b> If “URL” is selected as source, specify the URL of the wallpaper file. If “Local UCM Server” is selected as source, click to upload wallpaper file to the UCM6200.</li> </ul>

**Table 29: Global Policy Parameters – Communication Settings**

<b>Email Settings</b>	
<b>SMTP Settings</b>	<p>Check this option to configure the email settings that will be sent to the provisioned phones:</p> <ul style="list-style-type: none"> <li>• <b>Server</b> IP address of the SMTP server</li> <li>• <b>Port</b> SMTP server port</li> <li>• <b>From E-Mail address</b> Email address</li> <li>• <b>Sender Username</b> Username of the sender</li> <li>• <b>Password Recovery Email</b> Email where recovered password will be sent</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>Alarm receive Email 1</b> Email address where alarms notifications will be sent</li> <li>• <b>Alarm receive Email 1</b> Email address where alarms notifications will be sent</li> <li>• <b>Enable SSL</b> Enable SSL protocol for SMTP</li> </ul>
<b>FTP</b>	
<b>FTP</b>	<p>Check this option to configure the FTP settings that will be sent to the provisioned phones:</p> <ul style="list-style-type: none"> <li>• <b>Storage Server Type</b> Either FTP or Central Storage</li> <li>• <b>Server</b> FTP server address</li> <li>• <b>Port</b> FTP port to be used</li> <li>• <b>Username</b> FTP username</li> <li>• <b>Path</b> FTP Directory path</li> </ul>

## Global Templates

Global Templates can be accessed in Web GUI→**Value-added Features**→**Zero Config**→**Global Templates**. Users can create multiple global templates with different sets of configurations and save the templates. Later on, when the user configures the device in Edit Device dialog→Advanced Settings, the user can select to use one of the global templates for the device. Please refer to section **[Manage Devices]** for more details on using the global templates.


When creating global template, users can select the categories and the parameters under each category to be used in the template. The global policy and the selected global template will both take effect when generating the config file. However, the selected global template has higher priority to the global policy when it comes to the same setting option/field. If the same option/field has different value configured in the global policy and the selected global template, the value for this option/field in the selected global template will override the value in global policy.



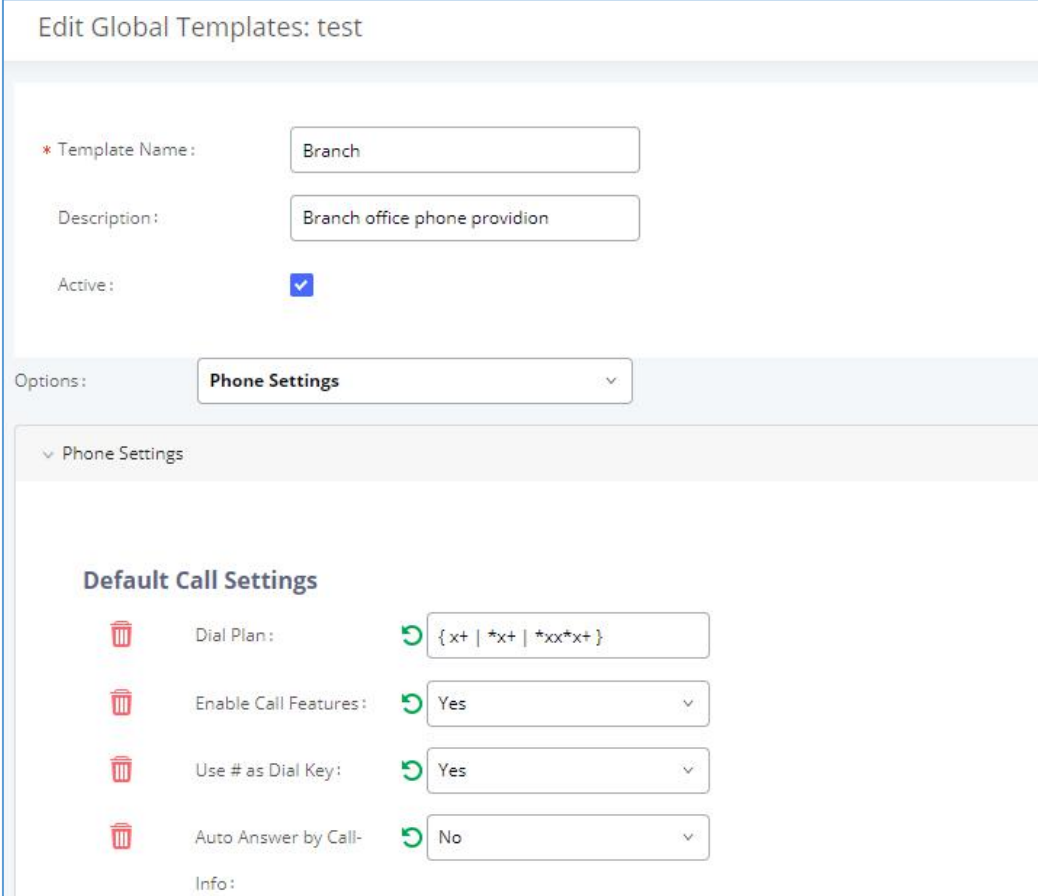
Click on "Create New Template" to add a global template. Users will see the following configurations.

**Table 30: Create New Template**

<b>Template Name</b>	Create a name to identify this global template.
<b>Description</b>	Provide a description for the global template. This is optional.
<b>Active</b>	Check this option to enable the global template.

- Click on  to edit the global template.

The window for editing global template is shown in the following figure. In the "Options" field, after entering the option name key word, the options containing the key word will be listed. Users could then select the options to be modified and click on "Add Option" to add it into the global template.



**Edit Global Templates: test**

\* Template Name:



Description:



Active:



Options:



▼ Phone Settings

**Default Call Settings**

 Dial Plan: 



 Enable Call Features: 

 Use # as Dial Key: 

 Auto Answer by Call: 


Info:

**Figure 67: Edit Global Template**

The added options will show in the list. Users can then enter or select value for each option to be used in the global template. On the left side of each added option, users can click on  to remove this option from the template. On the right side of each option, users can click on  to reset the option value to the default value.

Click on "Save" to save this global template.



- The created global templates will show in the Web GUI→**Value-added Features**→**Zero Config**→**Global Templates** page. Users can click on  to delete the global template or click on “Delete Selected Templates” to delete multiple selected templates at once.
- Click on “Toggle Selected Template(s)” to toggle the status between enabled/disabled for the selected templates.

## Model configuration

### Model templates

Model layer configuration allows users to apply model-specific configurations to different devices. Users could create/edit/delete a model template by accessing Web GUI, page **Value-added Features**→**Zero Config**→**Model Templates**. If multiple model templates are created and enabled, when the user configures the device in Edit Device dialog→Advanced Settings, the user can select to use one of the model templates for the device. Please refer to section **[Manage Devices]** for more details on using the model template.

For each created model template, users can assign it as default model template. If assigned as default model template, the values in this model template will be applied to all the devices of this model. There is always only one default model template that can be assigned at one time on the UCM6200.

The selected model template and the default model template will both take effect when generating the config file for the device. However, the model template has higher priority to default model template when it comes to the same setting option/field. If the same option/field has different value configured in the default model template and the selected model template, the value for this option/field in the selected model template will override the value in default model template.

- Click on “Create New Template” to add a model template.


**Table 31: Create New Model Template**

<b>Model</b>	Select a model to apply this template. The supported Grandstream models are listed in the dropdown list for selection.
<b>Template Name</b>	Create a name for the model template.
<b>Description</b>	Enter a description for the model template. This is optional.







<b>Default Model Template</b>	Select to assign this model template as the default model template. The value of the option in default model template will be overridden if other selected model template has a different value for the same option.
<b>Active</b>	Check this option to enable the model template.

- Click on  to edit the model template.


The editing window for model template is shown in the following figure. In the “Options” field, enter the option name key word, the option that contains the key word will be listed. User could then select the option and click on “Add Option” to add it into the model template.

Once added, the option will be shown in the list below. On the left side of each option, users can click on  to remove this option from the model template. On the right side of each option, users can click on  to reset the option to the default value.

User could also click on “Add New Field” to add a P value number and the value to the configuration. The following figure shows setting P value “P1362” to “en”, which means the display language on the LCD is set to English. For P value information of different models, please refer to configuration template here <http://www.grandstream.com/support/tools>.



### Edit Model Templates: GXV3370



\* Model: GRANDSTREAM GXV3370

\* Template Name: GXV3370

Description:

Default Model

Template:

Active:

Options: MPK1

Custom Parameters

| Custom Parameters

Please enter the alias name or P-values without the "P" into the Name fields. Example: To configure Account 1's DN "103" into the Name field and "ARecord", or "0" into the Value field.

	P1362	en	Description
--	-------	----	-------------

+ Add New Field

**Figure 68: Edit Model Template**

- Click on Save when done. The model template will be displayed on Web GUI→**Value-added Features**→**Zero Config**→**Model Templates** page.
- Click on to delete the model template or click on “Delete Selected Templates” to delete multiple selected templates at once.
- Click on “Toggle Selected Template(s)” to toggle the status between enabled/disabled for the selected model templates.


### **Model Update**

UCM6200 zero config feature supports provisioning all models of Grandstream SIP end devices including OEM device models.



## OEM Models

Users can associate OEM device models with their original Grandstream-branded models, allowing these OEM devices to be provisioned appropriately.

- Click on  button.
- In the *Source Model* field, select the Grandstream device that the OEM model is based on from the dropdown list.
- For the *Destination Model* and *Destination Vendor* field, enter the custom OEM model name and vendor name.
- The newly added OEM model should now be selectable as an option in *Model* fields.

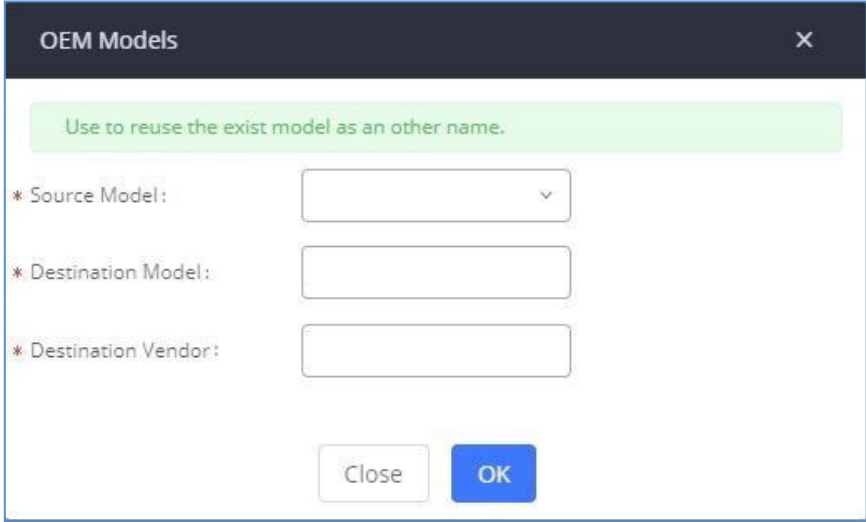




Figure 69: OEM Models

## Model Template Package List

Templates for most of the Grandstream models are built in with the UCM6200 already. Templates for GS Wave and Grandstream surveillance products require users to download and install under Web GUI→**Value-added Features**→**Zero Config**→**Model Update** first before they are available in the UCM6200 for selection. After downloading and installing the model template to the UCM6200, it will show in the dropdown list for “Model” selection when editing the model template.

- Click on  to download the template.
- Click on  to upgrade the model template. Users will see this icon available if the device model has template updated in the UCM6200.













Model Template Package List				
Vendor	Model	Version (Remote / Local)	Size	Options
Grandstream	DP750	1.0/-	271K	
Grandstream	GAC2500	1.0/-	25K	
Grandstream	GDS3705	1.0/-	55K	
Grandstream	GDS3710	1.0/-	97K	
Grandstream	GSWave	1.0/-	20K	
Grandstream	GVC3200	1.0/-	18K	
Grandstream	GVC3202	1.0/-	13K	
Grandstream	GVC3210	1.0/-	63K	
Grandstream	GXP1100	1.0/-	729K	
Grandstream	GXP1105	1.0/-	297K	

Figure 70: Template Management

## Upload Model Template Package

In case the UCM6200 is placed in the private network and Internet access is restricted, users will not be able to get packages by downloading and installing from the remote server. Model template package can be manually uploaded from local device through Web GUI. Please contact Grandstream customer support if the model package is needed for manual uploading.

**Upload Model Template Package**

Choose Model Package to

Upload:

Figure 71: Upload Model Template Manually

## Device Configuration

On Web GUI, page **Value-added Features**→**Zero Config**→**Zero Config**, users could create new device, delete existing device(s), make special configuration for a single device, or send NOTIFY to existing device(s).

### Create New Device

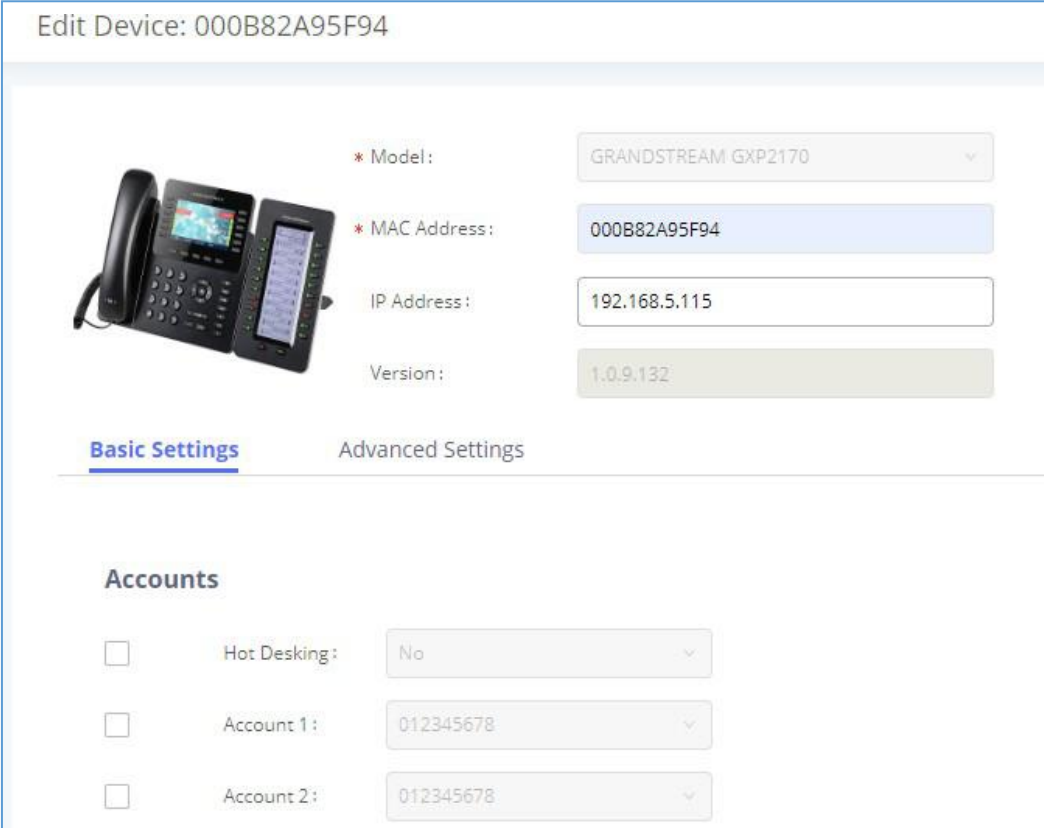
Besides configuring the device after the device is discovered, users could also directly create a new device and configure basic settings before the device is discovered by the UCM6200. Once the device is plugged in, it can




then be discovered and provisioned. This gives the system administrator adequate time to set up each device beforehand.

Click on "Create New Device" and the following dialog will show. Follow the steps below to create the configurations for the new device.

1. Firstly, select a model for the device to be created and enter its MAC address, IP address and firmware version (optional) in the corresponding field.
2. Basic settings will show a list of settings based on the model selected in step 1. Users could assign extensions to accounts, assign functions to Line Keys and Multiple-Purposed Keys if supported on the selected model.
3. Click on "Create New Device" to save the configuration for this device.



Edit Device: 000B82A95F94



\* Model: GRANDSTREAM GXP2170

\* MAC Address: 000B82A95F94

IP Address: 192.168.5.115

Version: 1.0.9.132

Basic Settings    Advanced Settings

**Accounts**

Hot Desking: No

Account 1: 012345678

Account 2: 012345678

**Figure 72: Create New Device**



## Manage Devices

The device manually created or discovered from Auto Discover will be listed in the Web GUI→**Value-added Features**→**Zero Config**→**Zero Config** page. Users can see the devices with their MAC address, IP address, vendor, model etc.

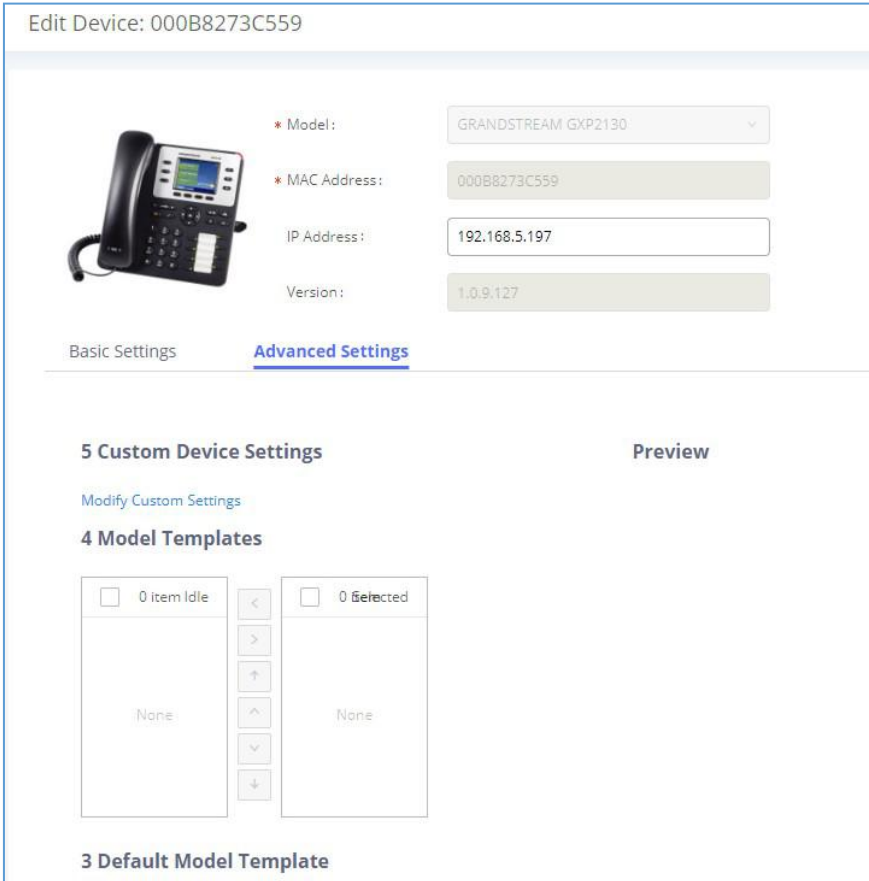


<input type="checkbox"/>	000B825C6927	192.168.6.162	1.0.9.9	Grandstream	GXP2160	1					
<input type="checkbox"/>	000B826B1958	192.168.6.115	1.0.3.167	Grandstream	GXV3240	1					
<input type="checkbox"/>	000B826B1FF7	192.168.6.158	1.0.3.171	Grandstream	GXV3240	1					


**Figure 73: Manage Devices**

- Click on  to access the Web GUI of the phone.
- Click on  to edit the device configuration.

A new dialog will be displayed for the users to configure “Basic” settings and “Advanced” settings. “Basic” settings have the same configurations as displayed when manually creating a new device, i.e., account, line key and MPK settings; “Advanced” settings allow users to configure more details in a five-level structure.



Edit Device: 000B8273C559



\* Model: GRANDSTREAM GXP2130

\* MAC Address: 000B8273C559

IP Address: 192.168.5.197

Version: 1.0.9.127

Basic Settings **Advanced Settings**

**5 Custom Device Settings** Preview

[Modify Custom Settings](#)

**4 Model Templates**

0 Item Idle  0 Selected

None

**3 Default Model Template**

**Figure 74: Edit Device**





A preview of the “Advanced” settings is shown in the above figure. There are five levels configurations as described in (1) (2) (3) (4) (5) below, with priority from the lowest to the highest. The configurations in all levels will take effect for the device. If there are same options existing in different level configurations with different value configured, the higher-level configuration will override the lower level configuration.



(1) Global Policy

This is the lowest level configuration. The global policy configured in Web GUI→**Value-added Features**→**Zero Config**→**Global Policy** will be applied here. Clicking on “Modify Global Policy” to redirect to page **Value-added Features**→**Zero Config**→**Global Policy**.





(2) Global Templates

Select a global template to be used for the device and click on  to add. Multiple global templates can be selected, and users can arrange the priority by adjusting orders via  and . All the selected global templates will take effect. If the same option exists on multiple selected global templates, the value in the template with higher priority will override the one in the template with lower priority. Click on  to remove the global template from the selected list.

(3) Default Model Template

Default Model Template will be applied to the devices of this model. Default model template can be configured in model template under Web GUI→**Value-added Features**→**Zero Config**→**Model Templates** page. Please see default model template option in [\[Table 31: Create New Model Template\]](#).

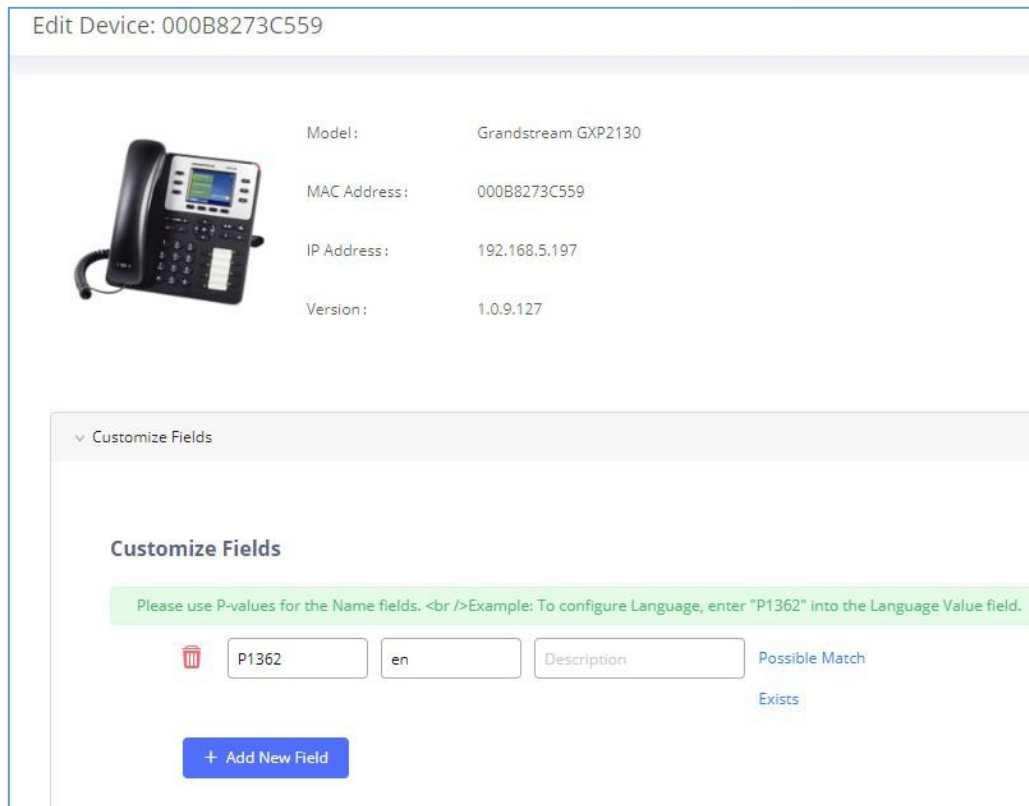
(4) Model Templates

Select a model template to be used for the device and click on  to add. Multiple global templates can be selected, and users can arrange the priority by adjusting orders via  and . All the selected model templates will take effect. If the same option exists on multiple selected model templates, the value in the template with higher priority will override the one in the template with lower priority. Click on  to remove the model template from the selected list.

(5) Customize Device Settings


This is the highest-level configuration for the device. Click on “Modify Customize Device Settings” and following dialog will show.





**Figure 75: Edit Customize Device Settings**

Scroll down in the dialog to view and edit the device-specific options. If the users would like to add more options which are not in the pre-defined list, click on “Add New Field” to add a P value number and the value to the configuration. The above figure shows setting P value “P1362” to “en”, which means the display language on the LCD is set to English. The warning information on right tells that the option matching the P value number exists and clicking on it will lead to the matching option. For P value information of different models, please refer to configuration template here <http://www.grandstream.com/sites/default/files/Resources/config-template.zip>.

- Select multiple devices that need to be modified and then click on  to batch modify devices.


If selected devices are of the same model, the configuration dialog is like the following figure. Configurations in five levels are all available for users to modify.





### Modify Selected Devices

**WARNING: Performing a batch operation will override all the existing device configurations on this page.**



\* Model: GXP2130

MAC Address: 000B8273C556 ×  
000B8273C559 ×


**Basic Settings**    Advanced Settings

**Accounts**

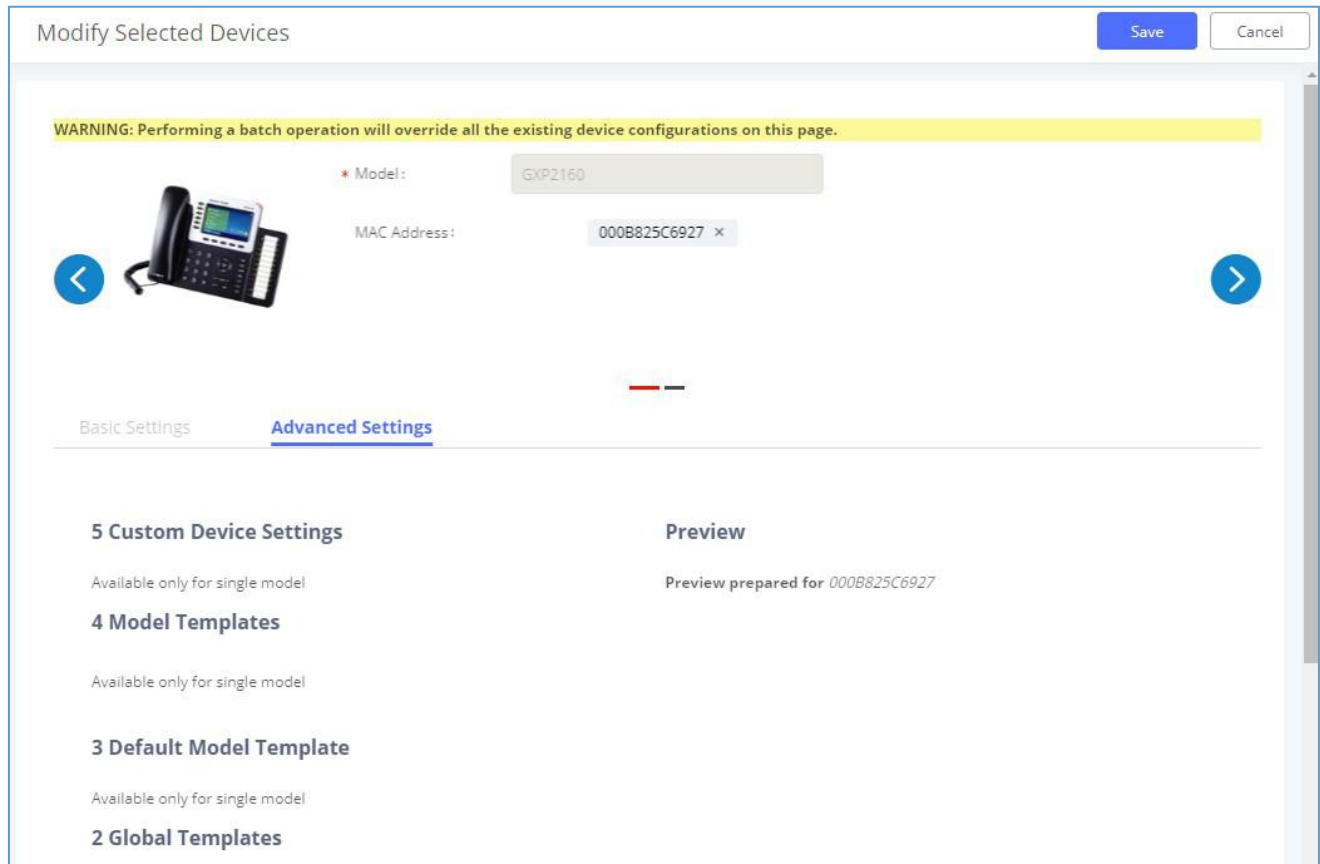
Hot Desking: No

**Virtual Multi-Purpose Key Settings**

**Figure 76: Modify Selected Devices - Same Model**

If selected devices are of different models, the configuration dialog is like the following figure. Click on  to view more devices of other models. Users are only allowed to make modifications in Global Templates and Global Policy level.






**Figure 77: Modify Selected Devices - Different Models**

**⚠ Note:**

Performing batch operation will override all the existing device configuration on the page.

After the above configurations, save the changes and go back to Web GUI → **Value-added Features** → **Zero Config** → **Zero Config** page. Users could then click on  to send NOTIFY to the SIP end point device and trigger the provisioning process. The device will start downloading the generated configuration file from the URL contained in the NOTIFY message.



Zero Config

[Zero Config](#)   [Global Policy](#)   [Global Templates](#)   [Model Templates](#)   [Model Update](#)   [Zero Config Settings](#)

Filter:

<input type="checkbox"/>	MAC Address ↕	IP Address ↕	Extension	Version ↕	Vendor ↕	Model ↕	Create Config ↕	Options
<input type="checkbox"/>	000B823FBCB0	192.168.6.176	--	1.0.14.100	GRANDSTREAM	HT503	--	
<input type="checkbox"/>	000B82415D27	192.168.6.72	--	1.0.7.80	GRANDSTREAM	GXV3140	--	
<input type="checkbox"/>	000B825C5806	192.168.6.218	--	1.0.8.50	GRANDSTREAM	GXP2140	--	
<input type="checkbox"/>	000B825C6926	192.168.2.104	--	1.0.9.17	GRANDSTREAM	GXP2160	--	
<input type="checkbox"/>	000B826B1052	192.168.6.146	--	1.0.3.177	GRANDSTREAM	GXV3240	--	
<input type="checkbox"/>	000B826B1FF7	192.168.6.144	--	1.0.3.144	GRANDSTREAM	GXV3240	--	
<input type="checkbox"/>	000B826B24CD	192.168.6.45	--	1.0.3.177	GRANDSTREAM	GXV3275	--	
<input type="checkbox"/>	000B8271B249	192.168.6.119	--	1.0.4.60	GRANDSTREAM	GXP1625	--	
<input type="checkbox"/>	000B8271B419	192.168.6.195	--	1.0.4.56	GRANDSTREAM	GXP1610	--	
<input type="checkbox"/>	000B8275CB88	192.168.6.137	--	1.0.8.50	GRANDSTREAM	GXP2130	--	
<input type="checkbox"/>	000B827846B1	192.168.6.224	--	0.6.9.61	GRANDSTREAM	GXP1628	--	

**Figure 78: Device List in Zero Config**

In this web page, users can also click on “Reset All Extensions” to reset the extensions of all the devices.

## Sample Application

Assuming in a small business office where there are 8 GXP2140 phones used by customer support and 1 GXV3370 phone used by customer support supervisor. 3 of the 8 customer support members speak Spanish and the rest speak English. We could deploy the following configurations to provisioning the office phones for the customer support team.

1. Go to Web GUI→**Value-added Features**→**Zero Config**→**Zero Config Settings**, select “Enable Zero Config”.
2. Go to Web GUI→**Value-added Features**→**Zero Config**→**Global Policy**, configure Date Format, Time Format and Firmware Source as follows.



v Localization

### Language Settings

\* Language: ↻ English ▼

### Date and Time

Date Format: ↻ yyyy-mm-dd ▼

Time Format: ↻ 24-hour Clock ▼

Enable NTP: ↻ Disabled ▼

NTP Server: ↻

NTP Update Interval: ↻ 1440

Time Zone: ↻ GMT+08:00 (Beijing, Taipei, ... ▼

Enable Daylight Saving Time: ↻ Disabled ▼

> Phone Settings

> Contact List

v Maintenance

### Upgrade and Provision

Firmware Source: ↻ URL ▼

Upgrade via: ↻ TFTP ▼

Server Path: ↻ fm.grandstream.com/gs

File Prefix: ↻

File Postfix: ↻

Config Server Path: ↻ 192.168.2.1:8089/zccg/

Allow DHCP Option 43/66: ↻ No ▼


Automatic Upgrade: ↻ Disabled ▼

Periodic option: ↻ Disabled ▼

Firmware Upgrade Rule: ↻ Check new firmware only w... ▼

**Figure 79: Zero Config Sample - Global Policy**



3. Go to Web GUI→**Value-added Features**→**Zero Config**→**Model Templates**, create a new model template “English Support Template” for GXP2140. Add option “Language” and set it to “English”. Then select the option “Default Model Template” to make it the default model template.
4. Go to Web GUI→**Value-added Features**→**Zero Config**→**Model Templates**, create another model template “Spanish Support Template” for GXP2140. Add option “Language” and set it to “Español”.
5. After 9 devices are powered up and connected to the LAN network, use “Auto Discover” function or “Create New Device” function to add the devices to the device list on Web GUI→**Value-added Features**→**Zero Config**→**Zero Config**.
6. On Web GUI→**Value-added Features**→**Zero Config**→**Zero Config** page, users could identify the devices by their MAC addresses or IP addresses displayed on the list. Click on  to edit the device settings.
7. For each of the 5 phones used by English speaking customer support, in “Basic” settings select an available extension for account 1 and click on “Save”. Then click on “Advanced” settings tab to bring up the following dialog. Users will see the English support template is applied since this is the default model template. A preview of the device settings will be listed on the right side.

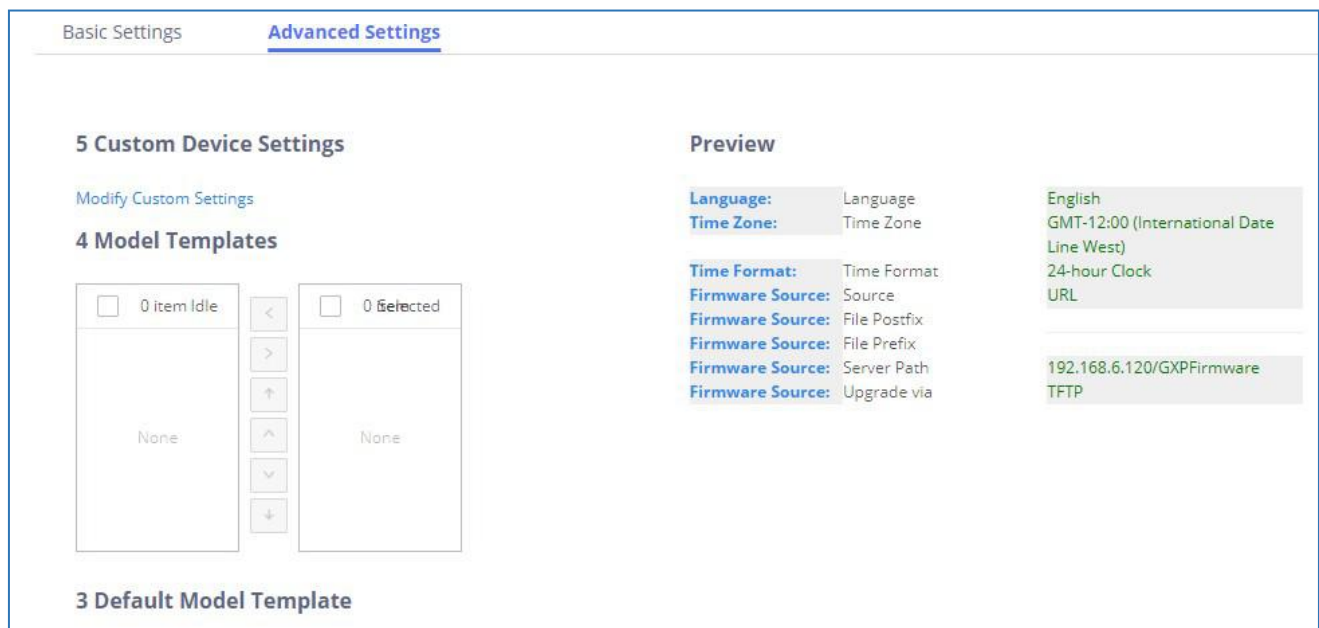
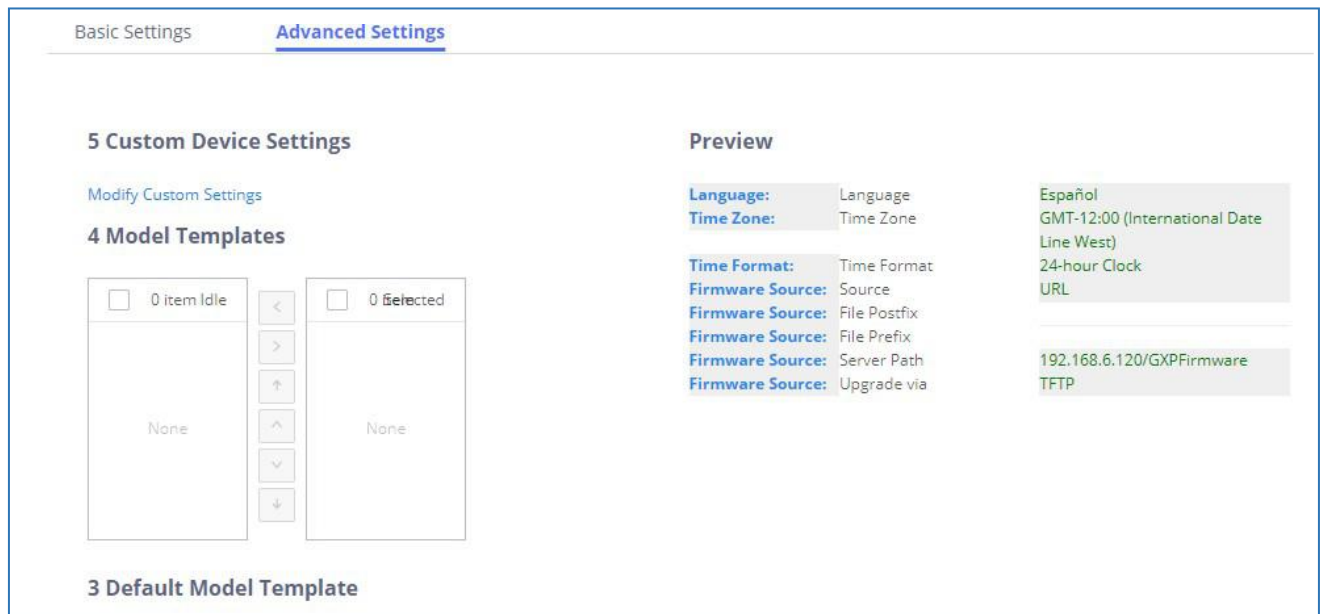


Figure 80: Zero Config Sample - Device Preview 1

- For the 3 phones used by Spanish support, in “Basic” settings select an available extension for account 1 and click on “Save”. Then click on “Advanced” settings tab to bring up the following dialog.



**Figure 81: Zero Config Sample - Device Preview 2**

Select “Spanish Support Template” in “Model Template”. The preview of the device settings is displayed on the right side and we can see the language is set to “Español” since Model Template has the higher priority for the option “Language”, which overrides the value configured in default model template.

- For the GXV3370 used by the customer support supervisor, select an available extension for account 1 on “Basic” settings and click on “Save”. Users can see the preview of the device configuration in “Advanced” settings. There is no model template configured for GXV3370.



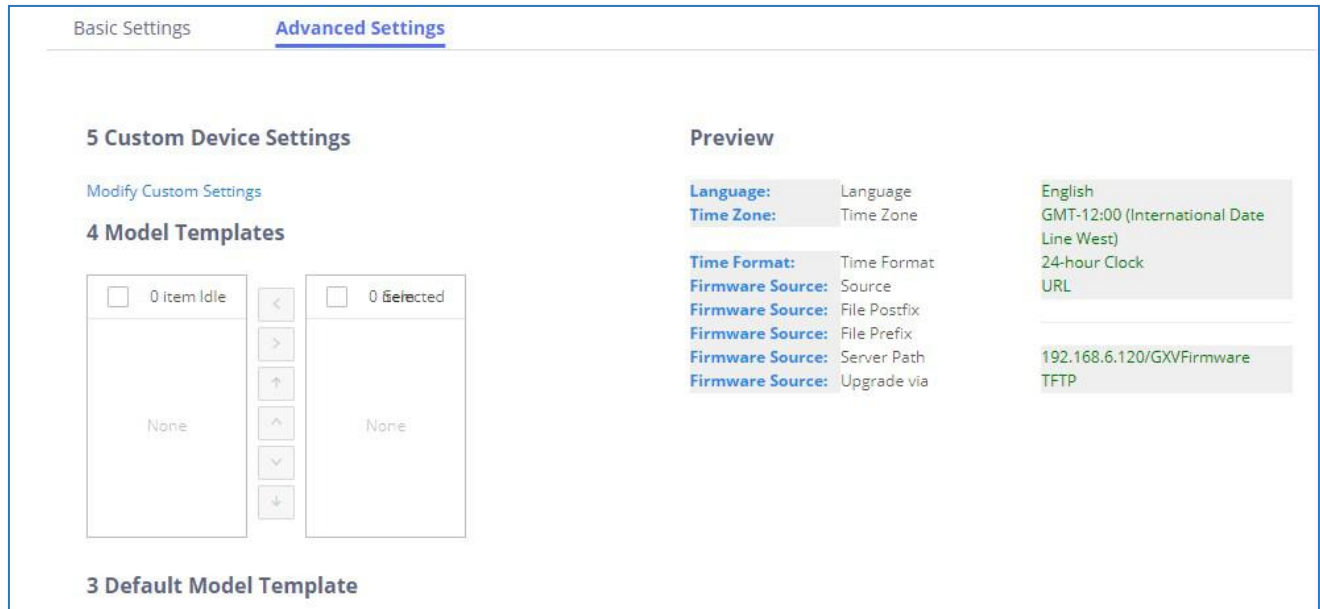



Figure 82: Zero Config Sample - Device Preview 3

10. Click on “Apply Changes” to apply saved changes.
11. On the Web GUI → **Value-added Features** → **Zero Config** → **Zero Config** page, click on  to send NOTIFY to trigger the device to download config file from UCM6200.

Now all the 9 phones in the network will be provisioned with a unique extension registered on the UCM6200. 3 of the phones will be provisioned to display Spanish on LCD and the other 5 will be provisioned to display English on LCD. The GXV3370 used by the supervisor will be provisioned to use the default language on LCD display since it is not specified in the global policy.



# EXTENSIONS

## Create New User

### Create New SIP Extension

To manually create new SIP user, go to Web GUI→**Extension/Trunk**→**Extensions**. Click on "Add" and a new dialog window will show for users to fill in the extension information.

Create New Extension
Save

Basic Settings
Media
Features
Specific Time
Follow Me

\* Select Extension Type:

Select Add Method:

**General**

<p>* Extension: <input type="text" value="1010"/></p> <p>* Permission: <input type="text" value="Internal"/></p> <p>AuthID: <input type="text"/></p> <p>* Voicemail Password: <input type="text" value="5850164"/></p> <p>Send Voicemail to Email: <input type="text" value="Default"/></p> <p>Enable Keep-alive: <input type="checkbox"/></p> <p>Disable This Extension: <input type="checkbox"/></p> <p>Emergency Calls CID: <input type="text"/></p>	<p>CallerID Number: <input type="text"/></p> <p>* SIP/IAX Password: <input type="text" value="X@mBWD!3"/></p> <p>Voicemail: <input type="text" value="Enable Local Voicemail"/></p> <p>Skip Voicemail Password Verification: <input type="checkbox"/></p> <p>Keep Voicemail after E-mailing: <input type="text" value="Default"/></p> <p>* Keep-alive Frequency: <input type="text" value="60"/></p> <p>Enable SCA: <input type="checkbox"/></p>
---	--

**User Settings**

<p>First Name: <input type="text"/></p> <p>Email Address: <input type="text"/></p> <p>* Language: <input type="text" value="Default"/></p>	<p>Last Name: <input type="text"/></p> <p>* User Password: <input type="text" value="EB579d"/></p> <p>* Concurrent Registrations: <input type="text" value="1"/></p>
--	--

**Figure 83: Create New Device**

Extension options are divided into four categories:

- Basic Settings
- Media
- Features
- Specific Time
- Follow me





Select first which type of extension: SIP Extension, IAX Extension or FXS Extension. The configuration parameters are as follows.

**Table 32: SIP Extension Configuration Parameters – Basic Settings**

General	
<b>Extension</b>	The extension number associated with the user. <b>Note:</b> This field supports (+) sign.
<b>CallerID Number</b>	Configure the CallerID Number that would be applied for outbound calls from this user. <b>Note:</b> The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
<b>Permission</b>	Assign permission level to the user. The available permissions are “Internal”, “Local”, “National” and “International” from the lowest level to the highest level. The default setting is “Internal”. <b>Note:</b> Users need to have the same level as or higher level than an outbound rule’s privilege to make outbound calls using this rule.
<b>SIP/IAX Password</b>	Configure the password for the user. A random secure password will be automatically generated. It is recommended to use this password for security purpose.
<b>Auth ID</b>	Configure the authentication ID for the user. If not configured, the extension number will be used for authentication. <b>Note:</b> This field supports (+) sign.
<b>Voicemail</b>	Configure Voicemail. There are three valid options and the default option is "Enable Local Voicemail". <ul style="list-style-type: none"> <li>• <b>Disable Voicemail:</b> Disable Voicemail.</li> <li>• <b>Enable Local Voicemail:</b> Enable voicemail for the user.</li> <li>• <b>Enable Remote Voicemail:</b> Forward the notify message from remote voicemail system for the user, and the local voicemail will be disabled. <b>Note:</b> Remote voicemail feature is used only for Infomatec (Brazil).</li> </ul>
<b>Voicemail Password</b>	Configure voicemail password (digits only) for the user to access the voicemail box. A random numeric password is automatically generated. It is recommended to use the random generated password for security purpose.
<b>Skip Voicemail Password Verification</b>	When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default, this option is disabled.
<b>Send Voicemail to Email</b>	Allows users to send voicemail recording as .wav file attachment to specified email addresses, providing more customizable user experience.



	<p><b>Note:</b> When set to “Default”, the global settings in <b>Call Features → Voicemail → Voicemail Email Settings</b> will be used.</p>
<b>Keep Voicemail after Emailing</b>	<p>Whether to keep the local voicemail recording after sending them. If set to “Default”, the global settings will be used. Default is “No”.</p> <p><b>Note:</b> When set to “Default”, the global settings in <b>Call Features → Voicemail → Voicemail Email Settings</b> will be used.</p>
<b>Enable Keep-alive</b>	<p>If enabled, empty SDP packet will be sent to the SIP server periodically to keep the NAT port open. The default setting is “Yes”.</p>
<b>Keep-alive Frequency</b>	<p>Configure the Keep-alive interval (in seconds) to check if the host is up. The default setting is 60 seconds.</p>
<b>Disable This Extension</b>	<p>If selected, this extension will be disabled on the UCM6200.</p> <p><b>Note:</b> The disabled extension still exists on the PBX but cannot be used on the end device.</p>
<b>User Settings</b>	
<b>First Name</b>	<p>Configure the first name of the user. The first name can contain characters, letters, digits and _.</p>
<b>Last Name</b>	<p>Configure the last name of the user. The last name can contain characters, letters, digits and _.</p>
<b>Email Address</b>	<p>Fill in the Email address for the user. Voicemail will be sent to this Email address.</p>
<b>User Password</b>	<p>Configure the password for user portal access. A random numeric password is automatically generated. It is recommended to use the randomly generated password for security purpose.</p>
<b>Language</b>	<p>Select the voice prompt language to be used for this extension. The default setting is “Default” which is the selected voice prompt language under Web GUI→<b>PBX Settings→Voice Prompt→Language Settings</b>. The dropdown list shows all the current available voice prompt languages on the UCM6200. To add more languages in the list, please download voice prompt package by selecting “Check Prompt List” under Web GUI→<b>PBX Settings→Voice Prompt→Language Settings</b>.</p>
<b>Concurrent Registrations</b>	<p>The maximum endpoints which can be registered into this extension. For security concerns, the default value is 1.</p>
<b>Mobile Phone Number</b>	<p>Configure the phone number for the extension, user can type the related star code for phone number followed by the extension number to directly call this number.</p> <p><u>Example:</u> user can type *881000 to call the mobile number associated with extension 1000.</p>



**Table 33: SIP Extension Configuration Parameters – Media**

SIP Settings	
<b>NAT</b>	Use NAT when the UCM6200 is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is one-way audio issue, usually it is related to NAT configuration or Firewall's support of SIP and RTP ports. The default setting is enabled.
<b>Can Direct Media</b>	By default, the UCM6200 will route the media steams from SIP endpoints through itself. If enabled, the PBX will attempt to negotiate with the endpoints to route the media stream directly. It is not always possible for the UCM6200 to negotiate endpoint-to-endpoint media routing. The default setting is "No".
<b>DTMF Mode</b>	Select DTMF mode for the user to send DTMF. The default setting is "RFC4733". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit PCMU and PCMA are required. When "Auto" is selected, RFC4733 will be used if offered, otherwise "Inband" will be used.
<b>TEL URI</b>	If the phone has an assigned PSTN telephone number, this field should be set to "User=Phone". "User=Phone" parameter will be attached to the Request-Line and "TO" header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel" will be used instead of "SIP" in the SIP request.
<b>Alert-Info</b>	Configure the Alert-Info, when UCM6200 receives an INVITE request, the Alert-Info header field specifies an alternative ring tone to the UAS.
<b>Enable T.38 UDPTL</b>	Enable or disable T.38 UDPTL support.
<b>SRTP</b>	Enable SRTP for the call. The default setting is disabled.
<b>Fax Mode</b>	<p>Select Fax mode. The default setting is "None".</p> <ul style="list-style-type: none"> <li>• <b>None:</b> Disable Fax.</li> <li>• <b>Fax Detect:</b> Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under Web GUI→Call Features→Fax/T.38.</li> </ul>
<b>Fax to Email</b>	<ul style="list-style-type: none"> <li>• <b>Yes</b> – Allow Fax to Email for this extension. Faxes will be sent to the user's email address configured in the extension's <i>Basic Settings</i>.</li> <li>• <b>No</b> – Do not send any faxes to the user's email address configured in the extension's <i>Basic Settings</i>.</li> </ul>



<b>ACL Policy</b>	<p>Access Control List manages the IP addresses that can register to this extension.</p> <ul style="list-style-type: none"> <li>• <b>Allow All:</b> Any IP address can register to this extension.</li> <li>• <b>Local Network Address:</b> Only IP addresses in the configured network segments can register to this extension.</li> </ul>
<b>Local Network Address</b>	<p>Specifies allowed IP address or networks from where the extension can be registered. Up to 10 entries are allowed.</p> <p>Format: "xxx.xxx.xxx.xxx", "xxx.xxx.xxx.xxx/32", "[::]" or "[::]/128".</p>
<b>Codec Preference</b>	<p>Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, OPUS, iLBC, ADPCM, H.264, H.265, H.263, H.263p and RTX.</p>

**Table 34: SIP Extension Configuration Parameters – Features**

Call Transfer	
<b>Presence Status</b>	<p>Select which presence status to set for the extension and configure call forward conditions for each status. Six possible options are possible: <b>“Available”</b>, <b>“Away”</b>, <b>“Chat”</b>, <b>“Custom”</b>, <b>“DND”</b> and <b>“Unavailable”</b>.</p> <p>More details at [PRESENCE].</p>
<b>Call Forward Unconditional</b>	<p>Enable and configure the Call Forward Unconditional target number. Available options for target number are:</p> <ul style="list-style-type: none"> <li>• <b>“None”</b>: Call forward deactivated.</li> <li>• <b>“Extension”</b>: Select an extension from dropdown list as CFU target.</li> <li>• <b>“Custom Number”</b>: Enter a customer number as target. For example: *97.</li> <li>• <b>“Voicemail”</b>: Select an extension from dropdown list. Incoming calls will be forwarded to voicemail of selected extension.</li> <li>• <b>“Ring Group”</b>: Select a ring group from dropdown list as CFU target.</li> <li>• <b>“Queues”</b>: Select a queue from dropdown list as CFU target.</li> <li>• <b>“Voicemail Group”</b>: Select a voicemail group from dropdown list as CFU target.</li> </ul> <p>The default setting is “None”.</p>
<b>CFU Time Condition</b>	<p>Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> </ul>



	<ul style="list-style-type: none"> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings</b>→<b>Time Settings</b>→<b>Office Time/Holiday</b> page.</li> </ul>
<b>Call Forward No Answer</b>	<p>Configure the Call Forward No Answer target number. Available options for target number are:</p> <ul style="list-style-type: none"> <li>• <b>“None”</b>: Call forward deactivated.</li> <li>• <b>“Extension”</b>: Select an extension from dropdown list as CFN target.</li> <li>• <b>“Custom Number”</b>: Enter a customer number as target. For example: *97.</li> <li>• <b>“Voicemail”</b>: Select an extension from dropdown list. Incoming calls will be forwarded to voicemail of selected extension.</li> <li>• <b>“Ring Group”</b>: Select a ring group from dropdown list as CFN target.</li> <li>• <b>“Queues”</b>: Select a queue from dropdown list as CFN target.</li> <li>• <b>“Voicemail Group”</b>: Select a voicemail group from dropdown list as CFN target.</li> </ul>
<b>CFN Time Condition</b>	<p>Select time condition for Call Forward No Answer. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings</b>→<b>Time Settings</b>→<b>Office Time/Holiday</b> page.</li> </ul>
<b>Call Forward Busy</b>	<p>Configure the Call Forward Busy target number. Available options for target number are:</p> <ul style="list-style-type: none"> <li>• <b>“None”</b>: Call forward deactivated.</li> <li>• <b>“Extension”</b>: Select an extension from dropdown list as CFB target.</li> <li>• <b>“Custom Number”</b>: Enter a customer number as target. For example: *97.</li> <li>• <b>“Voicemail”</b>: Select an extension from dropdown list. Incoming calls will be forwarded to voicemail of selected extension.</li> <li>• <b>“Ring Group”</b>: Select a ring group from dropdown list as CFB target.</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>“Queues”</b>: Select a queue from dropdown list as CFB target.</li> <li>• <b>“Voicemail Group”</b>: Select a voicemail group from dropdown list as CFB target.</li> </ul>
<b>CFB Time Condition</b>	<p>Select time condition for Call Forward Busy. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday</b> page.</li> </ul>
<b>Do Not Disturb</b>	If enabled the extension will ignore all incoming calls
<b>DND Time Condition</b>	<p>Select time condition for Do Not Disturb. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday</b> page.</li> </ul>
<b>DND Whitelist</b>	<p>If DND is enabled, all calls to this extension will be rejected except the numbers listed on this list.</p> <p><b>Note:</b> The maximum number on the Whitelist is 10.</p>
<b>FWD Whitelist</b>	<p>If call forward is enabled, all calls to this extension will be forwarded except the calls coming from the specified numbers on this list.</p> <p><b>Note:</b> The Maximum number on the whitelist is 10.</p>
<b>CC Settings</b>	
<b>Enable CC</b>	If enabled, UCM6200 will automatically alert this extension when a called party is available, given that a previous call to that party failed for some reason. By default, it is disabled.



<b>CC Mode</b>	<p>Two modes for Call Completion are supported:</p> <ul style="list-style-type: none"> <li>• <b>Normal:</b> This extension is used as ordinary extension.</li> <li>• <b>For Trunk:</b> This extension is registered from a PBX.</li> </ul> <p>The default setting is “Normal”.</p> <p><b>Note:</b> The number of CC agents (for “Normal” mode) will now be limited by the extensions’ allowed number of concurrent registrations.</p>
<b>CC Max Agents</b>	<p>Configure the maximum number of CCSS agents which may be allocated for this channel (when CC Mode is set to “For Trunk”). In other words, this number serves as the maximum number of CC requests this channel is allowed to make. Min. value is 1.</p>
<b>CC Max Monitors</b>	<p>Configure the maximum number of monitor structures which may be created for this device (when CC Mode is set to “For Trunk”). This number tells how many callers may request CC services for a specific device at one time. The minimum value is 1.</p>
<b>Ring Simultaneously</b>	
<b>Ring Simultaneously</b>	<p>Enable this option to have an external number ring simultaneously along with the extension. If a register trunk is used for outbound, the register number will be used to be displayed for the external number as caller ID number.</p>
<b>External Number</b>	<p>Set the external number to be rang simultaneously. ‘-’ is the connection character which will be ignored.</p> <p>This field accepts only letters, numbers, and special characters + = * #.</p>
<b>Time Condition for Ring Simultaneously</b>	<p>Ring the external number simultaneously along with the extension based on this time condition.</p>
<b>Use callee DOD on FWD or Ring Simultaneously</b>	<p>Use the DOD number when calls are being diverted/forwarded to external destinations or when ring simultaneous is configured.</p>
<b>Monitor privilege control</b>	
<b>Allowed to call-barging</b>	<p>Add members from “Available Extensions” to “Selected Extensions” so that the selected extensions can spy on the used extension using feature code.</p>
<b>Seamless transfer privilege control</b>	
<b>Allowed to seamless transfer</b>	<p>Any extensions on the UCM can perform seamless transfer. When using Pickup Incall feature, only extensions available on the “Selected Extensions” list can perform seamless transfer to the edited extension.</p>
<b>Other Settings</b>	
<b>Ring Timeout</b>	<p>Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the</p>





	<p>UCM6200, which can be configured in the global ring timeout setting under Web GUI→<b>PBX Settings</b>→<b>General Settings</b>: General Preference. The valid range is between 5 seconds and 600 seconds.</p> <p><b>Note:</b> If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
<b>Auto Record</b>	Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under Web GUI→ <b>CDR</b> → <b>Recording Files</b> .
<b>Skip Trunk Auth</b>	<ul style="list-style-type: none"> <li>• If set to <b>'Yes'</b>, users can skip entering the password when making outbound calls.</li> <li>• If set to <b>'By Time'</b>, users can skip entering the password when making outbound calls during the selected time condition.</li> <li>• If set to <b>'No'</b>, users will be asked to enter the password when making outbound calls.</li> </ul>
<b>Time Condition for Skip Trunk Auth</b>	If 'Skip Trunk Auth' is set to 'By Time', select a time condition during which users can skip entering password when making outbound calls.
<b>Dial Trunk Password</b>	Configure personal password when making outbound calls via trunk.
<b>Support Hot-Desking Mode</b>	If enabled, SIP Password will accept only alphabet characters and digits. Auth ID will be changed to the same as Extension.
<b>Enable LDAP</b>	If enabled, the extension will be added to LDAP Phonebook PBX list.
<b>Enable WebRTC Support</b>	Enable registration and call from Web RTC.
<b>Music On Hold</b>	Specify which Music On Hold class to suggest to the bridged channel when putting them on hold.
<b>Enable Seamless Transfer</b>	Enable the seamless transfer for this extension.
<b>Permission</b>	Set the permission for this extension when using the seamless transfer
<b>Call Duration Limit</b>	The maximum duration of call-blocking.
<b>Maximum Call Duration</b>	The maximum call duration (in seconds). The default value 0 means no limit.
<b>Custom Call-info for Auto Answer</b>	If enabled, when a call is sent to this extension from UCM, the SIP INVITE message will contain a Call-info header indicating auto answer.
<b>Enable Call Waiting</b>	<p>If disabled, UCM will not invite the extension when it is already in a call and will do the same work as the user is busy.</p> <p><b>Note:</b> The option only works when the caller dials the extension directly.</p>

Table 35: SIP Extension Configuration Parameters – Specific Time

Specific Time	
<b>Time Condition</b>	Click to add Time Condition to configure specific time for this extension.





## Create New IAX Extension

The UCM6200 supports Inter-Asterisk eXchange (IAX) protocol. IAX is used for transporting VoIP telephony sessions between servers and terminal devices. IAX is like SIP but also has its own characteristic. For more information, please refer to RFC 5465.

To manually create new IAX user, go to Web GUI→**Extension/Trunk**→**Extensions**. Click on "Add" and a new dialog window will show for users which need to make sure first to select the extension type to be IAX Extension before proceeding to fill in the extension information. The configuration parameters are as follows.

**Table 36: IAX Extension Configuration Parameters→Basic Settings**

General	
<b>Extension</b>	The extension number associated with the user.
<b>CallerID Number</b>	Configure the CallerID Number that would be applied for outbound calls from this user. <b>Note:</b> The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
<b>Permission</b>	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". <b>Note:</b> Users need to have the same level as or higher level than an outbound rule's privilege to make outbound calls using this rule.
<b>SIP/IAX Password</b>	Configure the password for the user. A random secure password will be automatically generated. It is recommended to use this password for security purpose.
<b>Voicemail</b>	Configure Voicemail. There are three valid options and the default option is "Enable Local Voicemail". <ul style="list-style-type: none"> <li>• <b>Disable Voicemail:</b> Disable Voicemail.</li> <li>• <b>Enable Local Voicemail:</b> Enable voicemail for the user.</li> </ul>
<b>Voicemail Password</b>	Configure voicemail password (digits only) for the user to access the voicemail box. A random numeric password is automatically generated. It is recommended to use the random generated password for security purpose.
<b>Skip Voicemail Password Verification</b>	When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default, this option is disabled.
<b>Disable This Extension</b>	If selected, this extension will be disabled on the UCM6200. <b>Note:</b> The disabled extension still exists on the PBX but cannot be used on the end device.
User Settings	
<b>First Name</b>	Configure the first name of the user. The first name can contain characters, letters, digits and _.



<b>Last Name</b>	Configure the last name of the user. The last name can contain characters, letters, digits and _.
<b>Email Address</b>	Fill in the Email address for the user. Voicemail will be sent to this Email address.
<b>User Password</b>	Configure the password for user portal access. A random numeric password is automatically generated. It is recommended to use the randomly generated password for security purpose.
<b>Language</b>	Select the voice prompt language to be used for this extension. The default setting is "Default" which is the selected voice prompt language under Web GUI→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> . The dropdown list shows all the current available voice prompt languages on the UCM6200. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under Web GUI→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> .

Table 37: IAX Extension Configuration Parameters→Media

IAX Settings	
<b>Max Number of Calls</b>	Configure the maximum number of calls allowed for each remote IP address.
<b>Require Call Token</b>	Configure to enable/disable requiring call token. If set to "Auto", it might lock out users who depend on backward compatibility when peer authentication credentials are shared between physical endpoints. The default setting is "Yes".
<b>SRTP</b>	Enable SRTP for the call. The default setting is disabled.
<b>Fax Mode</b>	Select Fax Mode. The default setting is "None". <ul style="list-style-type: none"> <li>• <b>None:</b> Disable Fax. This is the default setting.</li> <li>• <b>Fax Detect:</b> Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under Web GUI→<b>Call Features</b>→<b>Fax/T.38</b>.</li> </ul>
<b>Strategy</b>	This option controls how the extension can be used on devices within different types of network. <ul style="list-style-type: none"> <li>• Allow All Device in any network can register this extension.</li> <li>• Local Subnet Only Only the user in specific subnet can register this extension. Up to three subnet addresses can be specified.</li> <li>• A Specific IP Address Only the device on the specific IP address can register this extension.</li> </ul> The default setting is "Allow All".



<b>Codec Preference</b>	Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.265, H.263, H.263p, RTX and VP8.
-------------------------	--

**Table 38: IAX Extension Configuration Parameters→Features**

Call Transfer	
<b>Call Forward Unconditional</b>	Configure the Call Forward Unconditional target number. If not configured, the Call Forward Unconditional feature is deactivated. The default setting is deactivated.
<b>CFU Time Condition</b>	Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”. Note: <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday</b> page.</li> </ul>
<b>Call Forward No Answer</b>	Configure the Call Forward No Answer target number. If not configured, the Call Forward No Answer feature is deactivated. The default setting is deactivated.
<b>CFN Time Condition</b>	Select time condition for Call Forward No Answer. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”. <b>Notes:</b> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday</b> page.</li> </ul>
<b>Call Forward Busy</b>	Configure the Call Forward Busy target number. If not configured, the Call Forward Busy feature is deactivated. The default setting is deactivated.
<b>CFB Time Condition</b>	Select time condition for Call Forward Busy. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”. <b>Notes:</b> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> </ul>



	<ul style="list-style-type: none"> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday page</b>.</li> </ul>
<b>Ring Simultaneously</b>	
<b>Ring Simultaneously</b>	Enable this option to have an external number ring simultaneously along with the extension. If a register trunk is used for outbound, the register number will be used to be displayed for the external number as caller ID number.
<b>External Number</b>	Set the external number to be rang simultaneously. '-' is the connection character which will be ignored.
<b>Time Condition for Ring Simultaneously</b>	Ring the external number simultaneously along with the extension on the basis of this time condition.
<b>Other Settings</b>	
<b>Ring Timeout</b>	<p>Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the UCM6200, which can be configured in the global ring timeout setting under Web GUI→PBX Settings→Voice Prompt→Custom Prompt: General Preference. The valid range is between 5 seconds and 600 seconds.</p> <p><b>Note:</b> If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
<b>Auto Record</b>	Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under Web GUI→CDR→Recording Files.
<b>Skip Trunk Auth</b>	<ul style="list-style-type: none"> <li>• If set to "Yes", users can skip entering the password when making outbound calls.</li> <li>• If set to "By Time", users can skip entering the password when making outbound calls during the selected time condition.</li> <li>• If set to "No", users will be asked to enter the password when making outbound calls.</li> </ul>
<b>Time Condition for Skip Trunk Auth</b>	If "Skip Trunk Auth" is set to "By Time", select a time condition during which users can skip entering password when making outbound calls.
<b>Dial Trunk Password</b>	Configure personal password when making outbound calls via trunk.
<b>Enable LDAP</b>	If enabled, the extension will be added to LDAP Phonebook PBX lists.
<b>Music On Hold</b>	Configure the Music On Hold class to suggest to the bridged channel when putting them on hold.
<b>Call Duration Limit</b>	The maximum duration of call-blocking.



**Table 39: IAX Extension Configuration Parameters→Specific Time**

Specific Time	
Time Condition	Click to add Time Condition to configure specific time for this extension.

### Create New FXS Extension

The UCM6200 supports Foreign eXchange Subscriber (FXS) interface. FXS is used when user needs to connect analog phone lines or FAX machines to the UCM6200.

To manually create new FXS user, go to Web GUI→**Extension/Trunk**→**Extensions**. Click on "Add" and a new dialog window will show for users which need to make sure first to select the extension type to be FXS Extension before proceeding to fill in the extension information. The configuration parameters are as follows.

**Table 40: FXS Extension Configuration Parameters→Basic Settings**

General	
<b>Extension</b>	The extension number associated with the user.
<b>Analog Station</b>	Select the FXS port to be assigned for this extension.
<b>Caller ID Number</b>	Configure the CallerID Number that would be applied for outbound calls from this user. <b>Note:</b> The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
<b>Permission</b>	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". <b>Note:</b> Users need to have the same level as or higher level than an outbound rule's privilege to make outbound calls using this rule.
<b>Voicemail</b>	Configure Voicemail. There are three valid options and the default option is "Enable Local Voicemail". <ul style="list-style-type: none"> <li>• <b>Disable Voicemail:</b> Disable Voicemail.</li> <li>• <b>Enable Local Voicemail:</b> Enable voicemail for the user.</li> </ul>
<b>Voicemail Password</b>	Configure voicemail password (digits only) for the user to access the voicemail box. A random numeric password is automatically generated. It is recommended to use the random generated password for security purpose.
<b>Skip Voicemail Password Verification</b>	When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default, this option is disabled.
<b>Disable This Extension</b>	If selected, this extension will be disabled on the UCM6200. <b>Note:</b> The disabled extension still exists on the PBX but cannot be used on the end device.



User Settings	
<b>First Name</b>	Configure the first name of the user. The first name can contain characters, letters, digits and _.
<b>Last Name</b>	Configure the last name of the user. The last name can contain characters, letters, digits and _.
<b>Email Address</b>	Fill in the Email address for the user. Voicemail will be sent to this Email address.
<b>User Password</b>	Configure the password for user portal access. A random numeric password is automatically generated. It is recommended to use the randomly generated password for security purpose.
<b>Language</b>	Select the voice prompt language to be used for this extension. The default setting is "Default" which is the selected voice prompt language under Web GUI→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> . The dropdown list shows all the current available voice prompt languages on the UCM6200. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under Web GUI→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> .

Table 41: FXS Extension Configuration Parameters→Media

Analog Settings	
<b>Call Waiting</b>	Configure to enable/disable call waiting feature. The default setting is "No".
<b>User '#' as SEND</b>	If configured, the # key can be used as SNED key after dialing the number on the analog phone. The default setting is "Yes".
<b>RX Gain</b>	Configure the RX gain for the receiving channel of analog FXS port. The valid range is -30dB to +6dB. The default setting is 0.
<b>TX Gain</b>	Configure the TX gain for the transmitting channel of analog FXS port. The valid range is -30dB to +6dB. The default setting is 0.
<b>MIN RX Flash</b>	Configure the minimum period of time (in milliseconds) that the hook-flash must remain unpressed for the PBX to consider the event as a valid flash event. The valid range is 30ms to 1000ms. The default setting is 200ms.
<b>MAX RX Flash</b>	Configure the maximum period of time (in milliseconds) that the hook-flash must remain unpressed for the PBX to consider the event as a valid flash event. The minimum period of time is 256ms and it cannot be modified. The default setting is 1250ms.
<b>Enable Polarity Reversal</b>	If enabled, a polarity reversal will be marked as received when an outgoing call is answered by the remote party. For some countries, a polarity reversal is used for signaling the disconnection of a phone line and the call will be considered as Hangup on a polarity reversal. The default setting is "Yes".
<b>Echo Cancellation</b>	Specify "ON", "OFF" or a value (the power of 2) from 32 to 1024 as the number of taps of cancellation.



	<p><b>Note:</b> When configuring the number of taps, the number 256 is not translated into 256ms of echo cancellation. Instead, 256 taps mean <math>256/8 = 32</math> ms. The default setting is "ON", which is 128 taps.</p>
<b>3-Way Calling</b>	Configure to enable/disable 3-way calling feature on the user. The default setting is enabled.
<b>Send CallerID After</b>	Configure the number of rings before sending CID. Default setting is 1.
<b>Fax Mode</b>	<p>For FXS extension, there are three options available in Fax Mode. The default setting is "None".</p> <ul style="list-style-type: none"> <li>• <b>None:</b> Disable Fax.</li> <li>• <b>Fax Detect:</b> Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under Web GUI→<b>Call Features</b>→<b>Fax/T.38</b>.</li> <li>• <b>Fax Gateway:</b> If selected, the UCM6200 can support conversation and processing of Fax data from T.30 to T.38 or T.38 to T.30. only for FXS ports.</li> </ul>

**Table 42: FXS Extension Configuration Parameters→Features**

<b>Call Transfer</b>	
<b>Call Forward Unconditional</b>	Configure the Call Forward Unconditional target number. If not configured, the Call Forward Unconditional feature is deactivated. The default setting is deactivated.
<b>CFU Time Condition</b>	<p>Select time condition for Call Forward Unconditional. CFU takes effect only during the selected time condition. The available time conditions are "Office Time", "Out of Office Time", "Holiday", "Out of Holiday", "Out of Office Time or Holiday" and "Specific".</p> <p>Note:</p> <ul style="list-style-type: none"> <li>• "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings</b>→<b>Time Settings</b>→<b>Office Time/Holiday</b> page.</li> </ul>
<b>Call Forward No Answer</b>	Configure the Call Forward No Answer target number. If not configured, the Call Forward No Answer feature is deactivated. The default setting is deactivated.
<b>CFN Time Condition</b>	<p>Select time condition for Call Forward No Answer. The available time conditions are "Office Time", "Out of Office Time", "Holiday", "Out of Holiday", "Out of Office Time or Holiday" and "Specific".</p> <p>Notes:</p> <ul style="list-style-type: none"> <li>• "Specific" has higher priority to "Office Times" if there is a conflict in terms of time period.</li> </ul>





	<ul style="list-style-type: none"> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday</b> page.</li> </ul>
<b>Call Forward Busy</b>	Configure the Call Forward Busy target number. If not configured, the Call Forward Busy feature is deactivated. The default setting is deactivated.
<b>CFB Time Condition</b>	<p>Select time condition for Call Forward Busy. The available time conditions are “Office Time”, “Out of Office Time”, “Holiday”, “Out of Holiday”, “Out of Office Time or Holiday” and “Specific”.</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• “Specific” has higher priority to “Office Times” if there is a conflict in terms of time period.</li> <li>• Specific time can be configured on the bottom of the extension configuration dialog. Scroll down the add Time Condition for specific time.</li> <li>• Office Time and Holiday could be configured on page <b>System Settings→Time Settings→Office Time/Holiday</b> page.</li> </ul>
<b>CC Settings</b>	
<b>Enable CC</b>	If enabled, UCM6200 will automatically alert this extension when a called party is available, given that a previous call to that party failed for some reason.
<b>Ring Simultaneously</b>	
<b>Ring Simultaneously</b>	Enable this option to have an external number ring simultaneously along with the extension. If a register trunk is used for outbound, the register number will be used to be displayed for the external number as caller ID number.
<b>External Number</b>	Set the external number to be rang simultaneously. ‘-’ is the connection character which will be ignored.
<b>Time Condition for Ring Simultaneously</b>	Ring the external number simultaneously along with the extension on the basis of this time condition.
<b>Hotline</b>	
<b>Enable Hotline</b>	If enabled, hotline dialing plan will be activated, a pre-configured number will be used according to the selected Hotline Type.
<b>Hotline Number</b>	Configure the Hotline Number





<b>Hotline Type</b>	Configure the Hotline Type: <ul style="list-style-type: none"> <li>• <b>Immediate Hotline:</b> When the phone is off-hook, UCM6200 will immediately dial the preset number</li> <li>• <b>Delay Hotline:</b> When the phone is off-hook, if there is no dialing within 5 seconds, UCM6200 will dial the preset number.</li> </ul>
<b>Other Settings</b>	
<b>Ring Timeout</b>	Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the UCM6200, which can be configured in the global ring timeout setting under Web GUI→PBX Settings→Voice Prompt→Custom Prompt: General Preference. The valid range is between 5 seconds and 600 seconds. <b>Note:</b> If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.
<b>Auto Record</b>	Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under Web GUI→CDR→Recording Files.
<b>Skip Trunk Auth</b>	<ul style="list-style-type: none"> <li>• If set to “Yes”, users can skip entering the password when making outbound calls.</li> <li>• If set to “By Time”, users can skip entering the password when making outbound calls during the selected time condition.</li> <li>• If set to “No”, users will be asked to enter the password when making outbound calls.</li> </ul>
<b>Time Condition for Skip Trunk Auth</b>	If “Skip Trunk Auth” is set to “By Time”, select a time condition during which users can skip entering password when making outbound calls.
<b>Dial Trunk Password</b>	Configure personal password when making outbound calls via trunk.
<b>Enable LDAP</b>	If enabled, this extension will be added to LDAP Phonebook PBX list; if disabled, this extension will be skipped when creating LDAP Phonebook.
<b>Music On Hold</b>	Select which Music On Hold class to suggest to extension when putting the active call on hold.
<b>Call Duration Limit</b>	Configure the maximum duration of call-blocking.

Table 43: FXS Extension Configuration Parameters→Specific Time

<b>Specific Time</b>	
<b>Time Condition</b>	Click to add Time Condition to configure specific time for this extension.



## Batch Add Extensions

### Batch Add SIP Extensions

To add multiple SIP extensions, BATCH add can be used to create standardized SIP extension accounts.

However, unique extension username cannot be set using BATCH add.

Under Web GUI→**Extension/Trunk**→**Extensions**, click on "Add" and select extension type as SIP extension, and "add method" as Batch.

**Table 44: Batch Add SIP Extension Parameters**

General	
<b>Create Number</b>	Specify the number of extensions to be added. The default setting is 5.
<b>Extension Incrementation</b>	Select how much to increment successive extensions. For example, if the value is 2, the extensions will be 1000,1002,1004,..... <b>Note:</b> Up to 3 characters.
<b>Extension</b>	Configure the starting extension number of the batch of extensions to be added.
<b>Permission</b>	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". <b>Note:</b> Users need to have the same level as or higher level than an outbound rule's privilege to make outbound calls from this rule.
<b>Voicemail</b>	Configure Voicemail. There are three valid options and the default option is "Enable Local Voicemail". <ul style="list-style-type: none"> <li>• <b>Disable Voicemail:</b> Disable Voicemail.</li> <li>• <b>Enable Local Voicemail:</b> Enable voicemail for the user.</li> <li>• <b>Enable Remote Voicemail:</b> Forward the notify message from remote voicemail system for the user, and the local voicemail will be disabled.</li> </ul> <b>Note:</b> Remote voicemail feature is used only for Infomatec (Brazil).
<b>Enable WebRTC Support</b>	Enable WebRTC support.
<b>SIP/IAX Password</b>	Configure the SIP/IAX password for the users. Three options are available to create password for the batch of extensions. <ul style="list-style-type: none"> <li>• User Random Password. A random secure password will be automatically generated. It is recommended to use this password for security purpose.</li> <li>• Use Extension as Password.</li> <li>• Enter a password to be used on all the extensions in the batch.</li> </ul>
<b>Voicemail Password</b>	Configure Voicemail password (digits only) for the users. <ul style="list-style-type: none"> <li>• User Random Password. A random password in digits will be automatically generated. It is recommended to use this password for security purpose.</li> </ul>



<b>CallerID Number</b>	<ul style="list-style-type: none"> <li>• Use Extension as Password.</li> <li>• Enter a password to be used on all the extensions in the batch.</li> </ul> <p>Configure CallerID Number when adding Batch Extensions.</p> <ul style="list-style-type: none"> <li>• Use Extension as Number Users can choose to use the extension number as CallerID</li> <li>• Use as Number Users can choose to set a specific number instead of using the extension number.</li> </ul>
<b>Ring Timeout</b>	<p>Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the UCM6200, which can be configured in the global ring timeout setting under Web GUI→<b>PBX Settings</b>→<b>Voice Prompt</b>→<b>Custom Prompt</b>: General Preference. The valid range is between 5 seconds and 600 seconds.</p> <p><b>Note:</b></p> <p>If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
<b>Auto Record</b>	<p>Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under Web GUI→<b>CDR</b>→<b>Recording Files</b>.</p>
<b>Skip Voicemail Password Verification</b>	<p>When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default, this option is disabled.</p>
<b>Music On Hold</b>	<p>Select which Music On Hold class to suggest to extensions when putting them on hold.</p>
<b>Enable LDAP</b>	<p>If enabled, the batch added extensions will be added to LDAP Phonebook PBX list; if disabled, the batch added extensions will be skipped when creating LDAP Phonebook.</p>
<b>Enable WebRTC Support</b>	<p>If enabled, extensions will be able to login to user portal and use Web RTC features.</p>
<b>Call Duration Limit</b>	<p>Configure the maximum duration of call-blocking.</p>
<b>SIP Settings</b>	
<b>NAT</b>	<p>Use NAT when the PBX is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is one-way audio issue, usually it is related to NAT configuration or Firewall's support of SIP and RTP ports.</p> <p>The default setting is enabled.</p>



<b>Can Direct Media</b>	By default, the PBX will route the media streams from SIP endpoints through itself. If enabled, the PBX will attempt to negotiate with the endpoints to route the media stream directly. It is not always possible for the PBX to negotiate endpoint-to-endpoint media routing. The default setting is "No".
<b>DTMF Mode</b>	Select DTMF mode for the user to send DTMF. The default setting is "RFC4733". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit codec PCMU and PCMA are required. When "Auto" is selected, RFC4733 will be used if offered, otherwise "Inband" will be used.
<b>Enable Keep-alive</b>	If enabled, empty SDP packet will be sent to the SIP server periodically to keep the NAT port open. The default setting is "Yes".
<b>Keep-alive Frequency</b>	Configure the number of seconds for the host to be up for Keep-alive. The default setting is 60 seconds.
<b>TEL URI</b>	If the end device/phone has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled.
<b>Concurrent Registrations</b>	The maximum endpoints which can be registered into this extension. For security concerns, the default value is 1.
<b>Other Settings</b>	
<b>SRTP</b>	Enable SRTP for the call. The default setting is "No".
<b>Fax Mode</b>	Select Fax mode for this user. The default setting is "None". <ul style="list-style-type: none"> <li>• <b>None:</b> Disable Fax.</li> <li>• <b>Fax Detect:</b> Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under Web GUI→<b>Call Features</b>→<b>Fax/T.38</b>.</li> </ul>
<b>Strategy</b>	This option controls how the extension can be used on devices within different types of network. The default setting is "Allow All". <ul style="list-style-type: none"> <li>• <b>Allow All</b> Device in any network can register this extension.</li> <li>• <b>Local Subnet Only</b> Only the user in specific subnet can register this extension. Up to three subnet addresses can be specified.</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>A Specific IP Address</b> Only the device on the specific IP address can register this extension.</li> </ul>
<b>Enable T.38 UDPTL</b>	Enable or disable T.38 UDPTL Support.
<b>Skip Trunk Auth</b>	If enable "All", users do not need to enter password when making an outbound call. If enable "Follow Me", the user can dial out via follow me without password.
<b>Codec Preference</b>	Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.722, G.729, G.723, iLBC, ADPCM, LPC10, H.264, H.265, H.263, H.263p and VP8.

## Batch Add IAX Extensions

Under Web GUI→**Extension/Trunk**→**Extensions**, click on "Add", then select extension type as IAX Extension and the add method to be Batch.

Table 45: Batch Add IAX Extension Parameters

General	
<b>Start Extension</b>	Configure the starting extension number of the batch of extensions to be added.
<b>Create Number</b>	Specify the number of extensions to be added. The default setting is 5.
<b>Permission</b>	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". <b>Note:</b> Users need to have the same level as or higher level than an outbound rule's privilege in order to make outbound calls from this rule.
<b>Voicemail</b>	Configure Voicemail. There are three valid options and the default option is "Enable Local Voicemail". <ul style="list-style-type: none"> <li>• <b>Disable Voicemail:</b> Disable Voicemail.</li> <li>• <b>Enable Local Voicemail:</b> Enable voicemail for the user.</li> </ul>
<b>SIP/IAX Password</b>	Configure the SIP/IAX password for the users. Three options are available to create password for the batch of extensions. <ul style="list-style-type: none"> <li>• User Random Password. A random secure password will be automatically generated. It is recommended to use this password for security purpose.</li> <li>• Use Extension as Password.</li> <li>• Enter a password to be used on all the extensions in the batch.</li> </ul>
<b>Voicemail Password</b>	Configure Voicemail password (digits only) for the users. <ul style="list-style-type: none"> <li>• User Random Password. A random password in digits will be automatically generated. It is recommended to use this password for security purpose.</li> <li>• Use Extension as Password.</li> <li>• Enter a password to be used on all the extensions in the batch.</li> </ul>




<b>Ring Timeout</b>	<p>Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the UCM6200, which can be configured in the global ring timeout setting under Web GUI→<b>PBX Settings</b>→<b>Voice Prompt</b>→<b>Custom Prompt</b>: General Preference. The valid range is between 5 seconds and 600 seconds.</p> <p><b>Note:</b> If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.</p>
<b>Auto Record</b>	<p>Enable automatic recording for the calls using this extension. The default setting is disabled. The recording files can be accessed under Web GUI→<b>CDR</b>→<b>Recording Files</b>.</p>
<b>Skip Voicemail Password Verification</b>	<p>When user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default, this option is disabled.</p>
<b>Music On Hold</b>	<p>Select which Music On Hold class to suggest to extensions when putting them on hold.</p>
<b>Enable LDAP</b>	<p>If enabled, the batch added extensions will be added to LDAP Phonebook PBX list; if disabled, the batch added extensions will be skipped when creating LDAP Phonebook.</p>
<b>Call Duration Limit</b>	<p>Configure the maximum duration of call-blocking.</p>
<b>IAX Settings</b>	
<b>Max Number of Calls</b>	<p>Configure the maximum number of calls allowed for each remote IP address.</p>
<b>Require Call Token</b>	<p>Configure to enable/disable requiring call token. If set to "Auto", it might lock out users who depend on backward compatibility when peer authentication credentials are shared between physical endpoints. The default setting is "Yes".</p>
<b>Other Settings</b>	
<b>SRTP</b>	<p>Enable SRTP for the call. The default setting is "No".</p>
<b>Fax Mode</b>	<p>Select Fax Mode for this user. The default setting is "None".</p> <ul style="list-style-type: none"> <li>• <b>None:</b> Disable Fax.</li> <li>• <b>Fax Detect:</b> Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under Web GUI→<b>Call Features</b>→<b>Fax/T.38</b>.</li> </ul>
<b>Strategy</b>	<p>This option controls how the extension can be used on devices within different types of network.</p>






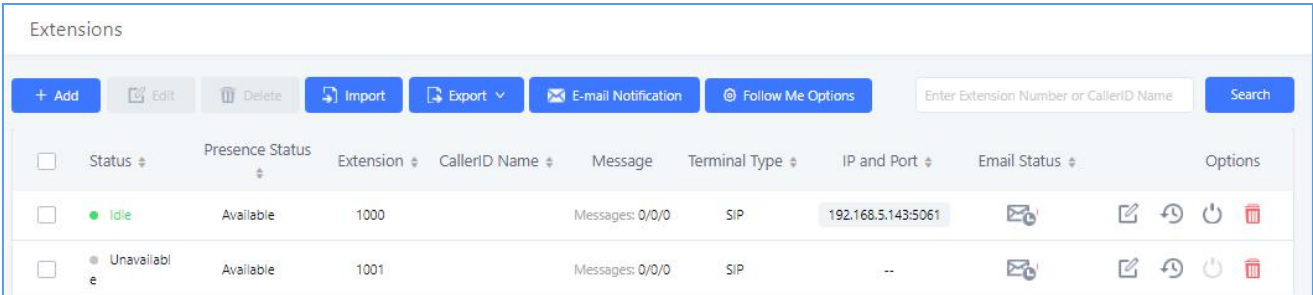
	<ul style="list-style-type: none"> <li>• <b>Allow All</b> Device in any network can register this extension.</li> <li>• <b>Local Subnet Only</b> Only the user in specific subnet can register this extension. Up to three subnet addresses can be specified.</li> <li>• <b>A Specific IP Address</b> Only the device on the specific IP address can register this extension.</li> </ul> <p>The default setting is "Allow All".</p>
<b>Skip Trunk Auth</b>	<p>If enable "All", users do not need to enter password when making an outbound call.</p> <p>If enable "Follow Me", the call can dial out via follow me without password.</p>
<b>Codec Preference</b>	<p>Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.722, G.729, G.723, iLBC, ADPCM, LPC10, H.264, H.265, H.263, H.263p and VP8.</p>

## Batch Extension Resetting Functionality

Users can select multiple extensions and reset their settings to default by pressing the reset button  and confirm the reset functionality. Once done, all settings in Basic Settings page will be restored to default values with the exception of Concurrent Registrations. User voicemail password will be reset to Random Password. User voicemail prompts and recordings will be deleted. User Management settings will also be restored to default with the exception of usernames and custom privileges

## Search and Edit Extension

All the UCM6200 extensions are listed under Web GUI → **Extension/Trunk** → **Extensions**, with status, Extension, CallerID Name, Technology (SIP, IAX and FXS), IP and Port. Each extension has a checkbox for users to "Modify Selected Extensions" or "Delete Selected Extensions". Also, options "Edit" , "Reboot"  and "Delete"  are available per extension. User can search an extension by specifying the extension number to find an extension quickly.



Status	Presence Status	Extension	CallerID Name	Message	Terminal Type	IP and Port	Email Status	Options
<input type="checkbox"/>	● Idle	Available	1000	Messages: 0/0/0	SIP	192.168.5.143:5061		
<input type="checkbox"/>	● Unavailable	Available	1001	Messages: 0/0/0	SIP	--		

Figure 84: Manage Extensions




- **Status**


Users can see the following icon for each extension to indicate the SIP status.

- Green: Idle
- Blue: Ringing
- Yellow: In Use
- Grey: Unavailable (the extension is not registered or disabled on the PBX)

- **Edit single extension**


Click on  to start editing the extension parameters.

- **Reset single extension**


Click on  to reset the extension parameters to default (except concurrent registration).

Other settings will be restored to default in **Maintenance**→**User Management**→**User Information** except username and permissions and delete the user voicemail prompt and voice messages.

- **Reboot the user**

Click on  to send NOTIFY reboot event to the device which has an UCM6200 extension already registered. To successfully reboot the user, "Zero Config" needs to be enabled on the UCM6200 Web GUI→**Value-added Features**→**Zero Config**→**Zero Config Settings**.

- **Delete single extension**

Click on  to delete the extension. Or select the checkbox of the extension and then click on "Delete Selected Extensions".

- **Modify selected extensions**

Select the checkbox for the extension(s). Then click on "Edit" to edit the extensions in a batch.

- **Delete selected extensions**

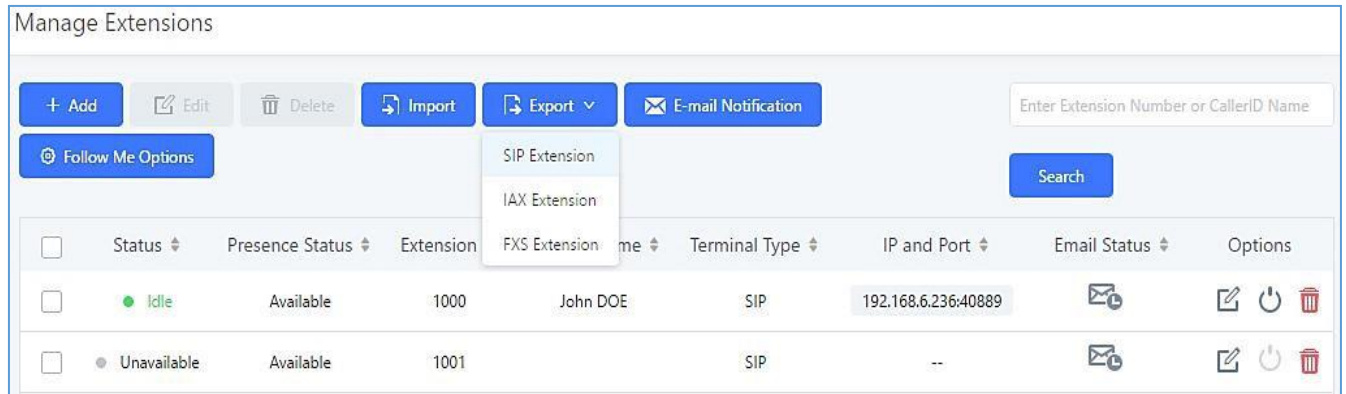
Select the checkbox for the extension(s). Then click on "Delete " to delete the extension(s).

## Export Extensions

The extensions configured on the UCM6200 can be exported to csv format file with selected technology "SIP", "IAX" or "FXS". Click on "Export Extensions" button and select technology in the prompt below.







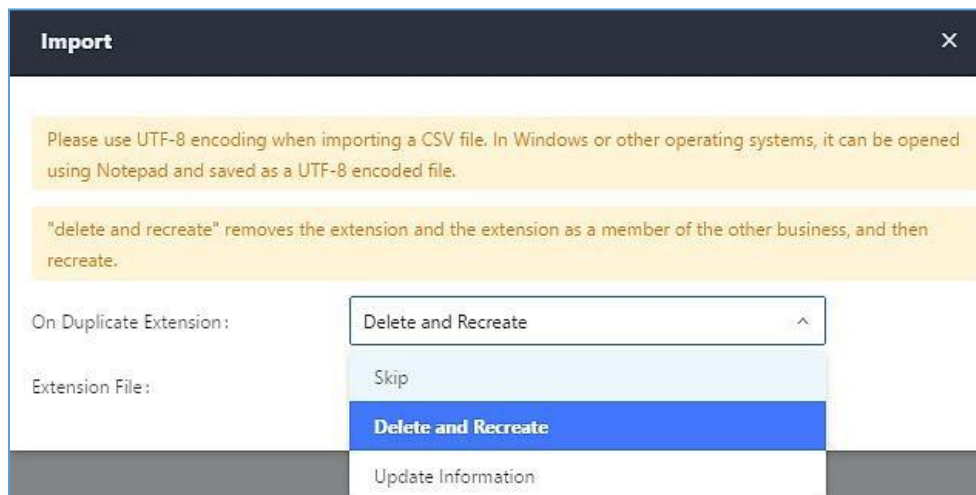
**Figure 85: Export Extensions**

The exported csv file can serve as a template for users to fill in desired extension information to be imported to the UCM6200.

## Import Extensions

The capability to import extensions to the UCM6200 provides users flexibility to batch add extensions with similar or different configuration quickly into the PBX system.

- Export extension csv file from the UCM6200 by clicking on "Export Extensions" button.
- Fill up the extension information you would like in the exported csv template.
- Click on "Import Extensions" button. The following dialog will be prompted.



**Figure 86: Import Extensions**

- Select the option in "On Duplicate Extension" to define how the duplicate extension(s) in the imported csv file should be treated by the PBX.
  - **Skip:** Duplicate extensions in the csv file will be skipped. The PBX will keep the current extension information as previously configured without change.
  - **Delete and Recreate:** The current extension previously configured will be deleted and the duplicate



extension in the csv file will be loaded to the PBX.

- **Update Information:** The current extension previously configured in the PBX will be kept. However, if the duplicate extension in the csv file has different configuration for any options, it will override the configuration for those options in the extension.
- Click on “Choose file to upload” to select csv file from local directory in the PC.
- Click on "Apply Changes" to apply the imported file on the UCM6200.

Example of file to import:

A	B	C	D	E	F	G	H	I	J	K	L	M	N
Extension	Technology	Enable Voicemail	CallerID	SIP/IAX Password	Voicema	Skip Voicemail Password Verification	Ring Timeout	Auto Record	S RTP	Fax Mode	Strategy	Local Subnet 1	Local Sub
1000	SIP	yes	1000	admin123	61783	no		no	no	None	Allow All		
1001	SIP	yes	1001	admin123	955921	no		no	no	None	Allow All		
1002	SIP	yes	1002	admin123	269824	no		no	no	None	Allow All		
1003	SIP	yes	1003	admin123	363196	no		no	no	None	Allow All		
1004	SIP	yes	1004	admin123	12860	no		no	no	None	Allow All		

Figure 87: Import File

Table 46: SIP extensions Imported File Example

Field	Supported values
Extension	Digits
Technology	SIP/SIP(WebRTC)
Enable Voicemail	yes/no/remote
CallerID Number	Digits
SIP/IAX Password	Alphanumeric characters
Voicemail Password	Digits
Skip Voicemail Password Verification	yes/no
Ring Timeout	Empty/ 3 to 600 (in second)
S RTP	yes/no
Fax Mode	None/Fax Detection
Strategy	Allow All/Local Subnet Only/A Specific IP Address
Local Subnet 1	IP address/Mask
Local Subnet 2	IP address/Mask
Local Subnet 3	IP address/Mask
Local Subnet 4	IP address/Mask
Local Subnet 5	IP address/Mask
Local Subnet 6	IP address/Mask
Local Subnet 7	IP address/Mask
Local Subnet 8	IP address/Mask
Local Subnet 9	IP address/Mask
Local Subnet 10	IP address/Mask
Specific IP Address	IP address
Skip Trunk Auth	yes/no/bytime



<b>Codec Preference</b>	PCMU,PCMA,GSM,G.726,G.722,G.729,H.264, H.265,ILBC,AAL2-G.726-32,ADPCM,G.723,H.263,H.263p,vp8,opus
<b>Permission</b>	Internal/Local/National/International
<b>NAT</b>	yes/no
<b>DTMF Mode</b>	RFC4733/info/inband/auto
<b>Insecure</b>	Port
<b>Enable Keep-alive</b>	Yes/no
<b>Keep-alive Frequency</b>	Value from 1-3600
<b>AuthID</b>	Alphanumeric value without special characters
<b>TEL URI</b>	Disabled/user=phone/enabled
<b>Call Forward Busy</b>	Digits
<b>Call Forward No Answer</b>	Digits
<b>Call Forward Unconditional</b>	Digits
<b>Support Hot-Desking Mode</b>	Yes/no
<b>Dial Trunk Password</b>	Digits
<b>Disable This Extension</b>	Yes/no
<b>CFU Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFN Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFB Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Music On Hold</b>	Default/ringbacktone_default
<b>CC Agent Policy</b>	If CC is disabled use: never If CC is set to normal use: generic If CC is set to trunk use: native
<b>CC Monitor Policy</b>	Generic/never
<b>CCBS Available Timer</b>	3600/4800
<b>CCNR Available Timer</b>	3600/7200
<b>CC Offer Timer</b>	60/120
<b>CC Max Agents</b>	Value from 1-999
<b>CC Max Monitors</b>	Value from 1-999
<b>Ring simultaneously</b>	Yes/no
<b>External Number</b>	Digits
<b>Time Condition for Ring Simultaneously</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time



<b>Time Condition for Skip Trunk Auth</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Enable LDAP</b>	Yes/no
<b>Enable T.38 UDPTL</b>	Yes/no
<b>Max Contacts</b>	Values from 1-10
<b>Enable WebRTC</b>	Yes/no
<b>Alert-Info</b>	None/Ring 1/Ring2/Ring3/Ring 4/Ring 5/Ring 6/Ring 7/ Ring 8/Ring 9/Ring 10/bellcore-dr1/bellcore-dr2/ bellcore-dr3/ bellcore-dr4/ bellcore-dr5/custom
<b>Do Not Disturb</b>	Yes/no
<b>DND Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Custom Auto answer</b>	Yes/no
<b>Do Not Disturb Whitelist</b>	Empty/digits
<b>User Password</b>	Alphanumeric characters.
<b>First Name</b>	Alphanumeric without special characters.
<b>Last Name</b>	Alphanumeric without special characters.
<b>Email Address</b>	Email address
<b>Language</b>	Default/en/zh
<b>Phone Number</b>	Digits
<b>Call-Barging Monitor</b>	Extensions allowed to call barging
<b>Seamless Transfer Members</b>	Extensions allowed to seamless transfer

Table 47: IAX extensions Imported File Example

Field	Supported values
<b>Extension</b>	Digits
<b>Technology</b>	IAX
<b>Enable Voicemail</b>	yes/no
<b>CallerID Number</b>	Digits
<b>SIP/IAX Password</b>	Alphanumeric characters
<b>Voicemail Password</b>	Digits
<b>Skip Voicemail Password Verification</b>	yes/no
<b>Ring Timeout</b>	Empty/ 3 to 600 (in second)
<b>S RTP</b>	yes/no
<b>Fax Mode</b>	None/Fax Detection
<b>Strategy</b>	Allow All/Local Subnet Only/A Specific IP Address
<b>Local Subnet 1</b>	IP address/Mask



<b>Local Subnet 2</b>	IP address/Mask
<b>Local Subnet 3</b>	IP address/Mask
<b>Local Subnet 4</b>	IP address/Mask
<b>Local Subnet 5</b>	IP address/Mask
<b>Local Subnet 6</b>	IP address/Mask
<b>Local Subnet 7</b>	IP address/Mask
<b>Local Subnet 8</b>	IP address/Mask
<b>Local Subnet 9</b>	IP address/Mask
<b>Local Subnet 10</b>	IP address/Mask
<b>Specific IP Address</b>	IP address
<b>Skip Trunk Auth</b>	yes/no/bytime
<b>Codec Preference</b>	PCMU,PCMA,GSM,G.726,G.722,G.729,H.264, H.265,ILBC,AAL2-G.726- 32,ADPCM,G.723,H.263,H.263p,vp8,opus
<b>Permission</b>	Internal/Local/National/International
<b>NAT</b>	yes/no
<b>Call Forward Busy</b>	Digits
<b>Call Forward No Answer</b>	Digits
<b>Call Forward Unconditional</b>	Digits
<b>Require Call Token</b>	Yes/no/auto
<b>Max Number of Calls</b>	Values from 1-512
<b>Dial Trunk Password</b>	Digits
<b>Disable This Extension</b>	Yes/no
<b>CFU Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFN Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFB Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Music On Hold</b>	Default/ringbacktone_default
<b>Ring simultaneously</b>	Yes/no
<b>External Number</b>	Digits
<b>Time Condition for Ring Simultaneously</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Time Condition for Skip Trunk Auth</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Enable LDAP</b>	Yes/no
<b>Limit Max time (s)</b>	empty
<b>Do Not Disturb</b>	Yes/no



<b>DND Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Do Not Disturb Whitelist</b>	Empty/digits
<b>User Password</b>	Alphanumeric characters.
<b>First Name</b>	Alphanumeric without special characters.
<b>Last Name</b>	Alphanumeric without special characters.
<b>Email Address</b>	Email address
<b>Language</b>	Default/en/zh
<b>Phone Number</b>	Digits
<b>Call-Barging Monitor</b>	Extensions allowed to call barging
<b>Seamless Transfer Members</b>	Extensions allowed to seamless transfer

Table 48: FXS Extensions Imported File Example

<b>Field</b>	<b>Supported values</b>
<b>Extension</b>	Digits
<b>Technology</b>	FXS
<b>Analog Station</b>	FXS1/FXS2
<b>Enable Voicemail</b>	yes/no
<b>CallerID Number</b>	Digits
<b>Voicemail Password</b>	Digits
<b>Skip Voicemail Password Verification</b>	yes/no
<b>Ring Timeout</b>	Empty/ 3 to 600 (in second)
<b>Auto Record</b>	yes/no
<b>Fax Mode</b>	None/Fax Detection
<b>Skip Trunk Auth</b>	Yes/no/bytime
<b>Permission</b>	Internal/Local/National/International
<b>Call Forward Busy</b>	Digits
<b>Call Forward No Answer</b>	Digits
<b>Call Forward Unconditional</b>	Digits
<b>Call Waiting</b>	Yes/no
<b>Use # as SEND</b>	Yes/no
<b>RX Gain</b>	Values from -30→6
<b>TX Gain</b>	Values from -30→6
<b>MIN RX Flash</b>	Values from: 30 – 1000
<b>MAX RX Flash</b>	Values from: 40 – 2000



<b>Enable Polarity Reversal</b>	Yes/no
<b>Echo Cancellation</b>	On/Off/32/64/128/256/512/1024
<b>3-Way Calling</b>	Yes/no
<b>Send CallerID After</b>	1/2
<b>Dial Trunk Password</b>	digits
<b>Disable This Extension</b>	Yes/no
<b>CFU Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFN Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>CFB Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Music On Hold</b>	Default/ringbacktone_default
<b>CC Agent Policy</b>	If CC is disabled use: never If CC is set to normal use: generic If CC is set to trunk use: native
<b>CC Monitor Policy</b>	Generic/never
<b>CCBS Available Timer</b>	3600/4800
<b>CCNR Available Timer</b>	3600/7200
<b>CC Offer Timer</b>	60/120
<b>CC Max Agents</b>	Value from 1-999
<b>CC Max Monitors</b>	Value from 1-999
<b>Ring simultaneously</b>	Yes/no
<b>External Number</b>	Digits
<b>Time Condition for Ring Simultaneously</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Time Condition for Skip Trunk Auth</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Enable LDAP</b>	Yes/no
<b>Enable Hotline</b>	Yes/no
<b>Hotline Type</b>	Immediate hotline/delay hotline
<b>Hotline Number</b>	digits
<b>Limit Max time (s)</b>	empty
<b>Do Not Disturb</b>	Yes/no
<b>DND Time Condition</b>	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
<b>Do Not Disturb Whitelist</b>	Empty/digits
<b>User Password</b>	Alphanumeric characters.
<b>First Name</b>	Alphanumeric without special characters.



<b>Last Name</b>	Alphanumeric without special characters.
<b>Email Address</b>	Email address
<b>Language</b>	Default/en/zh
<b>Phone Number</b>	Digits
<b>Call-Barging Monitor</b>	Extensions allowed to call barging
<b>Seamless Transfer Members</b>	Extensions allowed to seamless transfer

The CSV file should contain all the above fields, if one of them is missing or empty, the UCM6200 will display the following error message for missing fields.

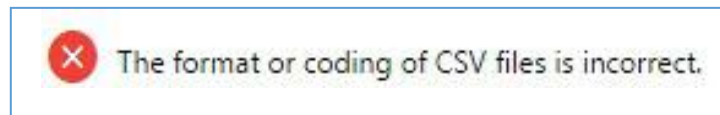


Figure 88: Import Error

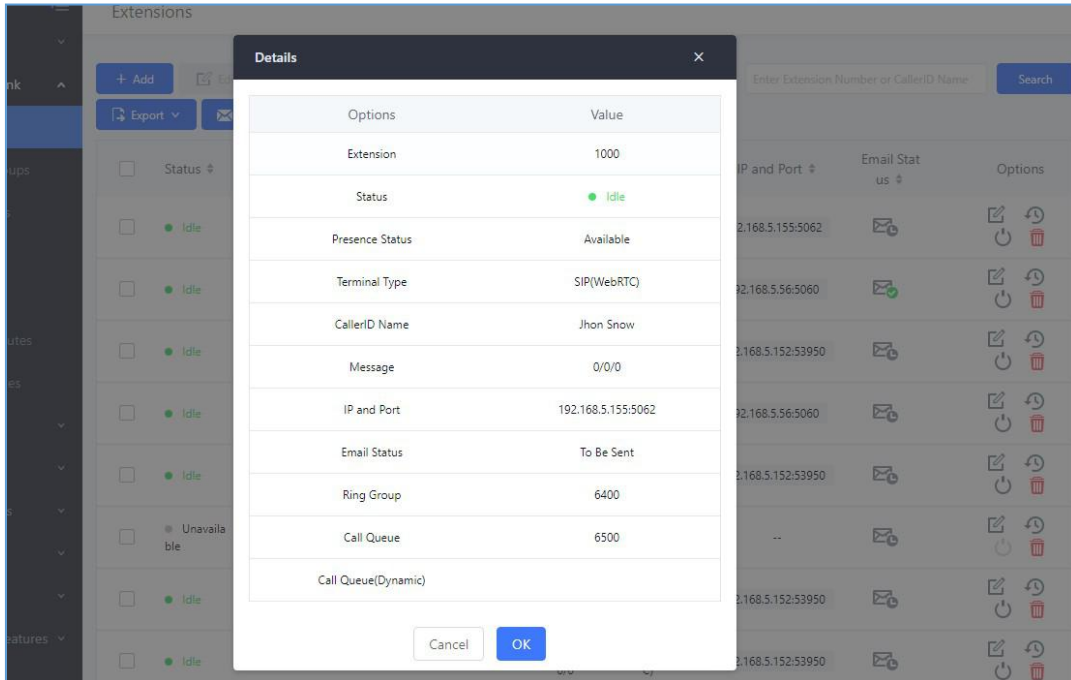
## Extension Details

Users can click on an extension number in the *Extensions* list page and quickly view information about it such as:

- **Extension:** Shows the Extension number.
- **Status:** Shows the status of the extension.
- **Presence status:** Indicates the Presence Status of this extension.
- **Terminal Type:** Shows the Type of the terminal using this extension (SIP, FXS...etc.).
- **Caller ID Name:** Reveals the Caller ID Name configured on the extension.
- **Messages:** Shows the messages stats.
- **IP and Port:** The IP address and the ports of the device using the extension.
- **Email status:** Show the Email status (sent, to be sent...etc.).
- **Ring Group:** Indicates the ring groups that this extension belongs to.
- **Call Queue:** Indicates the Cal Queues that this extension belongs to.
- **Call Queue (Dynamic):** Indicates the Call Queues that this extension belongs to as a dynamic agent.





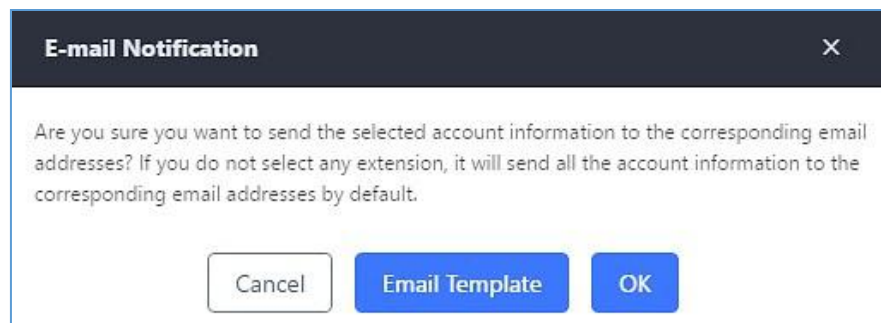


**Figure 89: Extension Details**

## E-mail Notification

Once the extensions are created with Email addresses, the PBX administrator can click on button “E-mail Notification” to send the account registration and configuration information to the user. Please make sure Email setting under Web GUI→**System Settings**→**Email Settings** is properly configured and tested on the UCM6200 before using “E-mail Notification”.

When click on “E-mail Notification” button, the following message will be prompted in the web page. Click on OK to confirm sending the account information to all users’ Email addresses.



**Figure 90: E-mail Notification - Prompt Information**

The user will receive Email including account registration information and LDAP configuration. A QR code is also generated for Mobile applications to scan it and get automatically provisioned. QR code provisioning is supported on Grandstream Softphone GS Wave Android™ application and iOS application.



Account Name : 1001  
SIP Server : 192.168.2.1  
SIP User ID : 1001  
Authenticate ID : 1001  
Authenticate Password : t\*297eoS1h  
Name :

This is the QR code of this account.



Figure 91: Account Registration Information and QR Code

Server Address : 192.168.2.1  
Port : 389  
Base : dc=pbx,dc=com  
This is the QR code of this LDAP config.



Figure 92: LDAP Client Information and QR Code

## Multiple Registrations per Extension

UCM6200 supports multiple registrations per extension so that users can use the same extension on devices in different locations.





**Figure 93: Multiple Registrations per Extension**

This feature can be enabled by configuring option “Concurrent Registrations” under Web GUI→**Extension/Trunk**→**Edit Extension**. The default value is set to 1 for security purpose. Maximum is 10.

The screenshot shows the 'Edit Extension: 1000' configuration page in the Grandstream Web GUI. The 'Basic Settings' tab is selected. The 'General' section includes fields for Extension (1000), Permission (Internal), AuthID, Voicemail Password (791020), Enable Keep-alive, and Disable This Extension. The 'User Settings' section includes fields for First Name (John), Last Name (DOE), Email Address (mbaomar@grandstream.com), Language (Default), and Mobile Phone Number. The 'Concurrent Registration' field is highlighted with a green box and set to the value 3.

**Figure 94: Extension - Concurrent Registration**

## SMS Message Support

The UCM6200 provides built-in SIP SMS message support. For SIP end devices such as Grandstream GXP or GXV phones that supports SIP message, after an UCM6200 account is registered on the end device, the user



can send and receive SMS message. Please refer to the end device documentation on how to send and receive SMS message.

SMS Message support is a new feature added since firmware 1.0.10.x which is built with Asterisk 13.






**Figure 95: SMS Message Support**

## EXTENSION GROUPS

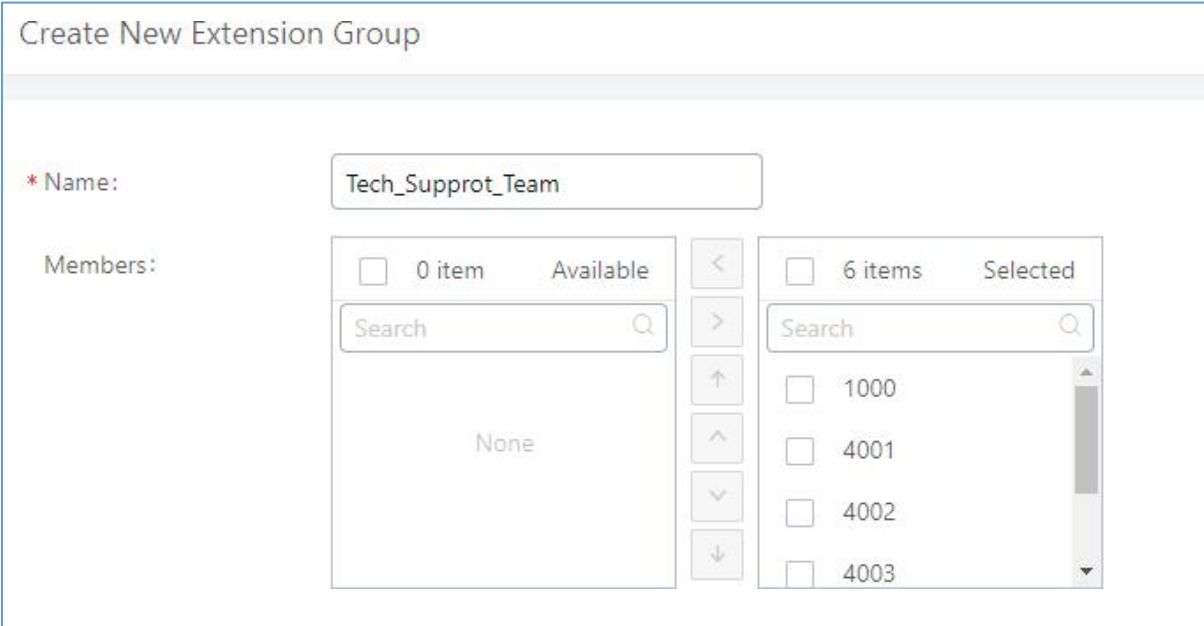
The UCM6200 extension group feature allows users to assign and categorize extensions in different groups to better manage the configurations on the UCM6200. For example, when configuring "Enable Filter on Source Caller ID", users could select a group instead of each person's extension to assign. This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

### Configure Extension Groups

Extension group can be configured via Web GUI→**Extension/Trunk**→**Extension Groups**.

- Click on  to create a new extension group.
- Click on  to edit the extension group.
- Click on  to delete the extension group.

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



**Figure 96: Edit Extension Group**

Click on  in order to change the ringing priority of the members selected on the group.



## Using Extension Groups

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

Edit Outbound Rule: National
Save

<p>* Calling Rule Name: <input style="width: 80%;" type="text" value="National"/></p> <p>Disable This Route: <input type="checkbox"/></p> <p>Password: <input style="width: 80%;" type="text"/></p>	<p>* Pattern: <input style="width: 80%;" type="text" value="_xxxxxxxx"/></p> <p>PIN Groups: <input style="border-bottom: 1px solid #ccc; border-top: 1px solid #ccc; border-right: 1px solid #ccc; border-left: 1px solid #ccc; text-align: right; font-size: 0.8em; color: #666; background-color: #f9f9f9; padding: 2px 5px; border: none; cursor: pointer; width: 100%;" type="text" value="None"/></p> <p>Privilege Level: <input style="border-bottom: 1px solid #ccc; border-top: 1px solid #ccc; border-right: 1px solid #ccc; border-left: 1px solid #ccc; text-align: right; font-size: 0.8em; color: #666; background-color: #f9f9f9; padding: 2px 5px; border: none; cursor: pointer; width: 100%;" type="text" value="Disable"/></p>
---	--

Enable Filter on Source Caller ID

Enable Filter on Source:

Available Extensions/E...



\* Custom Dynamic Ro...

**Figure 97: Select Extension Group in Outbound Route**



## ANALOG TRUNKS

Go to Web GUI→**Extension/Trunk**→**Analog Trunks** to add and edit analog trunks.

- Click on "Create New Analog Trunk" to add a new analog trunk.
- Click on  to edit the analog trunk.
- Click on  to delete the analog trunk.

### Analog Trunk Configuration

The analog trunk options are listed in the table below.

**Table 49: Analog Trunk Configuration Parameters**

<b>Channels</b>	Select the channel for the analog trunk. <ul style="list-style-type: none"> <li>• UCM6202: 2 channels</li> <li>• UCM6204: 4 channels</li> <li>• UCM6208: 8 channels</li> </ul>
<b>Trunk Name</b>	Specify a unique label to identify the trunk when listed in outbound rules, incoming rules and etc.
<b>Advanced Options</b>	
<b>SLA Mode</b>	Enable this option to satisfy two primary use cases, which include emulating a simple key system and creating shared extensions on a PBX. Enable SLA Mode will disable polarity reversal.
<b>Barge Allowed</b>	The barge option specifies whether other stations can join a call in progress on this trunk. If enabled, the other stations can press the line button to join the call. The default setting is Yes.
<b>Hold Access</b>	The hold option specifies hold permissions for this trunk. If set to "Open", any station can place this trunk on hold and any other station is allowed to retrieve the call. If set to "Private", only the station that places the call on hold can retrieve the call. The default setting is Yes.
<b>Enable Polarity Reversal</b>	If enabled, a polarity reversal will be marked as received when an outgoing call is answered by the remote party. For some countries, a polarity reversal is used for signaling the disconnection of a phone line and the call will be considered as "Hangup" on a polarity reversal. The default setting is "No".



<b>Polarity on Answer Delay</b>	When FXO port answers the call, FXS may send a Polarity Reversal. If this interval is shorter than the value of “Polarity on Answer Delay”, the Polarity Reversal will be ignored. Otherwise, the FXO will Onhook to disconnect the call. The default setting is 600ms.
<b>Current Disconnect Threshold (ms)</b>	This is the periodic time (in ms) that the UCM6200 will use to check on a voltage drop in the line. The default setting is 200. The valid range is 50 to 3000.
<b>Ring Timeout</b>	Configure the ring timeout (in ms). Trunk (FXO) devices must have a timeout to determine if there was a Hangup before the line is answered. This value can be used to configure how long it takes before the UCM6200 considers a non-ringing line with Hangup activity. The default setting is 8000.
<b>RX Gain</b>	Configure the RX gain for the receiving channel of analog FXO port. The valid range is from -13.5 (dB) to + 12.0 (dB). The default setting is 0.
<b>TX Gain</b>	Configure the TX gain for the transmitting channel of analog FXO port. The valid range is from -13.5 (dB) to + 12.0 (dB). The default setting is 0.
<b>Use CallerID</b>	Configure to enable CallerID detection. The default setting is “Yes”.
<b>Caller ID Scheme</b>	<p>Select the Caller ID scheme for this trunk.</p> <ul style="list-style-type: none"> <li>• Bellcore/Telcordia.</li> <li>• ETSI-FSK During Ringing</li> <li>• ETSI-FSK Prior to Ringing with DTAS</li> <li>• ETSI-FSK Prior to Ringing with LR</li> <li>• ETSI-FSK Prior to Ringing with RP</li> <li>• ETSI-DTMF During Ringing</li> <li>• ETSI-DTMF Prior to Ringing with DTAS</li> <li>• ETSI-DTMF Prior to Ringing with LR</li> <li>• ETSI-DTMF Prior to Ringing with RP</li> <li>• SIN 227-BT</li> <li>• NTT Japan</li> <li>• Auto Detect</li> </ul> <p>If you are not sure which scheme to choose, please select “Auto Detect”. The default setting is “Bellcore/Telcordia”.</p>
<b>Fax Mode</b>	<p>Enable to detect Fax signal from the trunk during the call and send the received Fax to the default Email address in Fax setting page under Web GUI→<b>Call Features</b>→<b>Fax/T.38</b>. The default setting is “No”.</p> <p><b>Note:</b> If enabled, Fax Pass-through cannot be used.</p>
<b>FXO Dial Delay(ms)</b>	Configure the time interval between off-hook and first dialed digit for outbound calls.
<b>Auto Record</b>	Enable automatic recording for the calls using this trunk. The default setting is disabled. The recording files can be accessed under Web GUI→ <b>CDR</b> → <b>Recording Files</b> .





<p><b>Disable This Trunk</b></p> <p><b>DAHDI Out Line Selection</b></p>	<p>If selected, the trunk will be disabled and incoming/Outgoing calls via this trunk will not be possible.</p> <p>This is to implement analog trunk outbound line selection strategy. Three options are available:</p> <ul style="list-style-type: none"> <li>• <b>Ascend</b> When the call goes out from this analog trunk, it will always try to use the first idle FXO port. The port order that the call will use to go out would be port 1→port 2→port 10→port 16. Every time it will start with port 1 (if it is idle).</li> <li>• <b>Poll</b> When the call goes out from this analog trunk, it will use the port that is not used last time. And it will always use the port in the order of port 1→2→10→16→1→2→10→16→1→2→10→16..., following the last port being used.</li> <li>• <b>Descend</b> When the call goes out from this analog trunk, it will always try to use the last idle FXO port. The port order that the call will use to go out would be port 16→port 10→port 2→port 1. Every time it will start with port 16 (if it is idle).</li> </ul> <p>The default setting is “Ascend” mode.</p>
<p><b>Echo Cancellation Mode</b></p>	<p>The Non-Linear Processing (NLP) in echo cancellation helps to remove/suppress residual echo components that could not be removed by the LEC (Line Echo Canceller). Following modes are supported:</p> <ul style="list-style-type: none"> <li>• <b>Default:</b> The NLP limits the signal level to the background noise level when active, and the background noise level adjustment is low.</li> <li>• <b>Custom mode 0:</b> The NLP limits the signal level to the background noise level when active, and the background noise level adjustment is high.</li> <li>• <b>Custom mode 1:</b> The NLP sends sign noise when active, and the background noise level adjustment is high.</li> <li>• <b>Custom mode 2:</b> The NLP injects white noise when active. The level corresponds to the background noise level at Sin, and the background noise level adjustment is high.</li> </ul>
<p><b>Direct Callback</b></p>	<p>Allows external numbers the option to get directed to the extension that last called them.</p> <p>For Example: User 2002 has dialed external number 061234575 but they were not reachable, once they have received missed call notification on their phone, they would mostly call back the number, if the option “Direct Callback” is enabled then they will be directly bridged to user 2002 regardless of the configured inbound destination.</p>



Tone Settings	
<b>Busy Detection</b>	Busy Detection is used to detect far end Hangup or for detecting busy signal. The default setting is "Yes".
<b>Busy Tone Count</b>	If "Busy Detection" is enabled, users can specify the number of busy tones to be played before hanging up. The default setting is 2. Better results might be achieved if set to 4, 6 or even 8. Please note that the higher the number is, the more time is needed to Hangup the channel. However, this might lower the probability to get random Hangup.
<b>Congestion Detection</b>	Congestion detection is used to detect far end congestion signal. The default setting is "Yes".
<b>Congestion Count</b>	If "Congestion Detection" is enabled, users can specify the number of congestion tones to wait for. The default setting is 2.
<b>Tone Country</b>	Select the country for tone settings. If "Custom" is selected, users could manually configure the values for Busy Tone and Congestion Tone. The default setting is "United States of America (USA)".
<b>Busy Tone</b>	<p><b>Syntax:</b>            f1=val[@level][,f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]];</p> <p>Frequencies are in Hz and cadence on and off are in ms.            Frequencies Range: [0, 4000)            Busy Level Range: (-300, 0)            Cadence Range: [0, 16383].            Select Tone Country "Custom" to manually configure Busy Tone value.</p> <p><b>Default value:</b>            f1=480@-50,f2=620@-50,c=500/500</p>
<b>Congestion Tone</b>	<p><b>Syntax:</b>            f1=val[@level][,f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]];</p> <p>Frequencies are in Hz and cadence on and off are in ms.            Frequencies Range: [0, 4000)            Busy Level Range: (-300, 0)            Cadence Range: [0, 16383].            Select Tone Country "Custom" to manually configure Busy Tone value.</p> <p><b>Default value:</b>            f1=480@-50,f2=620@-50,c=250/250</p>
<b>PSTN Detection</b>	Click on "Detect" to detect the busy tone, Polarity Reversal and Current Disconnect by PSTN. Before the detecting, please make sure there are more than one channel configured and working properly. If the detection has busy tone, the "Tone Country" option will be set as "Custom".



## PSTN Detection

The UCM6200 provides PSTN detection function to help users detect the busy tone, Polarity Reversal and Current Disconnect by making a call from the PSTN line to another destination. The detecting call will be answered and up for about 1 minute. Once done, the detecting result will show and can be used for the UCM6200 settings.

1. Go to UCM6200 Web GUI→**Extension/Trunk**→**Analog Trunks** page.
2. Click to edit the analog trunk created for the FXO port.
3. In the dialog window to edit the analog trunk, go to "Tone Settings" section and there are two methods to set the busy tone.
  - Tone Country. The default setting is "United States of America (USA)".
  - PSTN Detection.




Figure 98: UCM6200 FXO Tone Settings

4. Click on "Detect" to start PSTN detection.

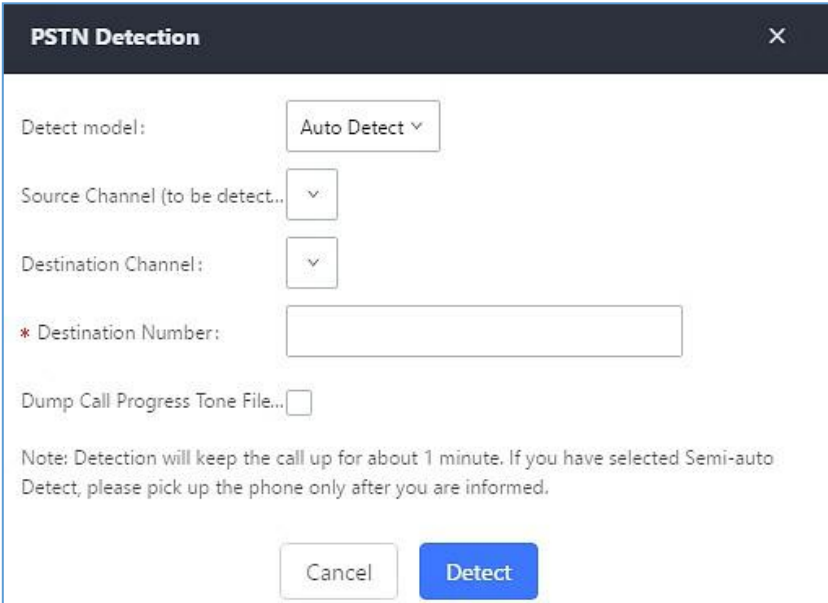


Figure 99: UCM6200 PSTN Detection



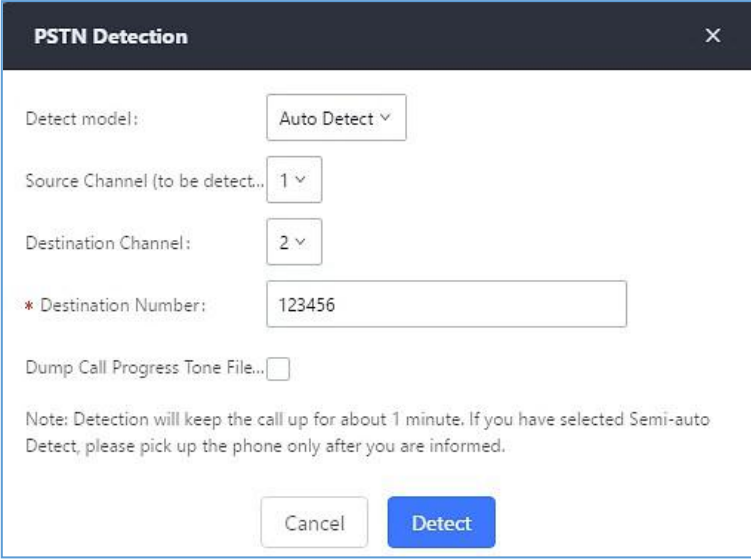
- If there are two FXO ports connected to PSTN lines, use the following settings for auto-detection.

**Detect Model:** Auto Detect.

**Source Channel:** The source channel to be detected.

**Destination Channel:** The channel to help detecting. For example, the second FXO port.

**Destination Number:** The number to be dialed for detecting. This number must be the actual PSTN number for the FXO port used as the destination channel.



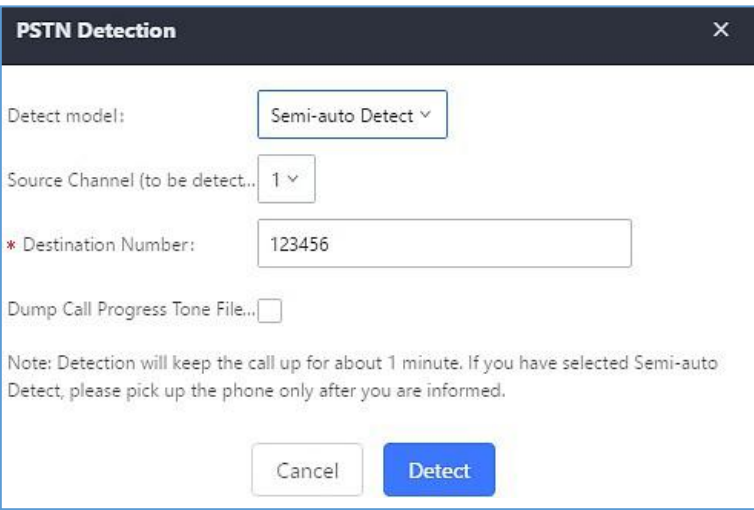
The screenshot shows a dialog box titled "PSTN Detection" with a close button (X) in the top right corner. The settings are as follows:

- Detect model: Auto Detect (dropdown menu)
- Source Channel (to be detect...): 1 (dropdown menu)
- Destination Channel: 2 (dropdown menu)
- \* Destination Number: 123456 (text input field)
- Dump Call Progress Tone File...:

Below the input fields is a note: "Note: Detection will keep the call up for about 1 minute. If you have selected Semi-auto Detect, please pick up the phone only after you are informed." At the bottom are two buttons: "Cancel" and "Detect".

**Figure 100: UCM6200 PSTN Detection: Auto Detect**

- If there is only one FXO port connected to PSTN line, use the following settings for auto-detection.



The screenshot shows a dialog box titled "PSTN Detection" with a close button (X) in the top right corner. The settings are as follows:

- Detect model: Semi-auto Detect (dropdown menu)
- Source Channel (to be detect...): 1 (dropdown menu)
- \* Destination Number: 123456 (text input field)
- Dump Call Progress Tone File...:

Below the input fields is a note: "Note: Detection will keep the call up for about 1 minute. If you have selected Semi-auto Detect, please pick up the phone only after you are informed." At the bottom are two buttons: "Cancel" and "Detect".

**Figure 101: UCM6200 PSTN Detection: Semi-Auto Detect**



**Detect Model:** Semi-auto Detect.

**Source Channel:** The source channel to be detected.

**Destination Number:** The number to be dialed for detecting. This number could be a cell phone number or other PSTN number that can be reached from the source channel PSTN number.

5. Click "Detect" to start detecting. The source channel will initiate a call to the destination number. For "Auto Detect", the call will be automatically answered. For "Semi-auto Detect", the UCM6200 Web GUI will display prompt to notify the user to answer or hang up the call to finish the detecting process.
6. Once done, the detected result will show. Users could save the detecting result as the current UCM6200 settings.

**Table 50: PSTN Detection for Analog Trunk**

<b>Detect Model</b>	<p>Select "Auto Detect" or "Semi-auto Detect" for PSTN detection.</p> <ul style="list-style-type: none"> <li>• <b>Auto Detect</b> Please make sure two or more channels are connected to the UCM6200 and in idle status before starting the detection. During the detection, one channel will be used as caller (Source Channel) and another channel will be used as callee (Destination Channel). The UCM6200 will control the call to be established and hang up between caller and callee to finish the detection.</li> <li>• <b>Semi-auto Detect</b> Semi-auto detection requires answering or hanging up the call manually. Please make sure one channel is connected to the UCM6200 and in idle status before starting the detection. During the detection, source channel will be used as caller and send the call to the configured Destination Number. Users will then need follow the prompts in Web GUI to help finish the detection.</li> </ul> <p>The default setting is "Auto Detect".</p>
<b>Source Channel</b>	Select the channel to be detected.
<b>Destination Channel</b>	Select the channel to help detect when "Auto Detect" is used.
<b>Destination Number</b>	Configure the number to be called to help the detection.

---

 **Note:**

- The PSTN detection process will keep the call up for about 1 minute.
  - If "Semi-auto Detect" is used, please pick up the call only after informed from the Web GUI prompt.
  - Once the detection is successful, the detected parameters "Busy Tone", "Polarity Reversal" and "Current Disconnect by PSTN" will be filled into the corresponding fields in the analog trunk configuration.
- 







# VOIP TRUNKS

## VoIP Trunk Configuration

VoIP trunks can be configured in UCM6200 under Web GUI→**Extension/Trunk**→**VoIP Trunks**. Once created, the VoIP trunks will be listed with Provider Name, Type, Hostname/IP, Username and Options to edit/detect the trunk.

**Note:** UCM6200 supports now 200 VoIP trunks

- Click on "Create New SIP Trunk" or "Create New IAX Trunk" to add a new VoIP trunk.
- Click on  to configure detailed parameters for the VoIP trunk.
- Click on  to configure Direct Outward Dialing (DOD) for the SIP Trunk.
- Click on  to start LDAP Sync.
- Click on  to delete the VoIP trunk.

For VoIP trunk example, please refer to the document in the following link:

[http://www.grandstream.com/sites/default/files/Resources/ucm6xxx\\_sip\\_trunk\\_guide.pdf](http://www.grandstream.com/sites/default/files/Resources/ucm6xxx_sip_trunk_guide.pdf)

The VoIP trunk options are listed in the table below.

**Table 51: Create New SIP Trunk**

<b>Type</b>	Select the VoIP trunk type. <ul style="list-style-type: none"> <li>• Peer SIP Trunk</li> <li>• Register SIP Trunk</li> </ul>
<b>Provider Name</b>	Configure a unique label (up to 64 character) to identify this trunk when listed in outbound rules, inbound rules etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider's server of the trunk.
<b>Keep Original CID</b>	Keep the CID from the inbound call when dialing out. This setting will override "Keep Trunk CID" option. Please make sure that the peer PBX at the other side supports to match user entry using "username" field from authentication line.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".



<b>NAT</b>	Turn on this setting when the PBX is using public IP and communicating with devices behind NAT. If there is one-way audio issue, usually it is related to NAT configuration or SIP/RTP port support on the firewall.
<b>Disable This Trunk</b>	If checked, the trunk will be disabled. <b>Note:</b> If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
<b>TEL URI</b>	If the trunk has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled.
<b>Caller ID</b>	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored.  <b>Important Note:</b> When making outgoing calls, the following priority order rule will be used to determine which CallerID will be set before sending out the call:  From user (Register Trunk Only) → CID from inbound call ( <b>Keep Original CID Enabled</b> ) → Trunk Username/CallerID ( <b>Keep Trunk CID Enabled</b> ) → DOD → Extension CallerID Number → Trunk Username/CallerID ( <b>Keep Trunk CID Disabled</b> ) → Global Outbound CID.
<b>Need Registration</b>	Select whether the trunk needs to register on the external server or not when "Register SIP Trunk" type is selected. The default setting is No.
<b>Username</b>	Enter the username to register to the trunk from the provider when "Register SIP Trunk" type is selected.
<b>Password</b>	Enter the password to register to the trunk from the provider when "Register SIP Trunk" is selected.
<b>Auth ID</b>	Enter the Authentication ID for "Register SIP Trunk" type.
<b>Auto Record</b>	Enable automatic recording for the calls using this trunk (for SIP trunk only). The default setting is disabled. The recording files can be accessed under Web GUI→ <b>CDR</b> → <b>Recording Files</b> .
<b>Direct Callback</b>	Allows external numbers the option to get directed to the extension that last called them.  For Example: User 2002 has dialed external number 061234575 but they were not reachable, once they have received missed call notification on their phone, they would mostly call back the number, if the option "Direct Callback" is enabled then they will be directly bridged to user 2002 regardless of the configured inbound destination.



**Table 52: SIP Register Trunk Configuration Parameters**

Basic Settings	
<b>Provider Name</b>	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider's server of the trunk.
<b>Transport</b>	Configure the SIP transport protocol to be used in this trunk. The default setting is "UDP". <ul style="list-style-type: none"> <li>• UDP</li> <li>• TCP</li> <li>• TLS</li> </ul>
<b>SIP URI Scheme When Using TLS</b>	When TLS is selected as Transport for register trunk, users can select between SIP and SIPS URI scheme
<b>Keep Original CID</b>	Keep the CID from the inbound call when dialing out. This setting will override "Keep Trunk CID" option. Please make sure that the peer PBX at the other side supports to match user entry using "username" field from authentication line.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
<b>NAT</b>	Turn on this option when the PBX is using public IP and communicating with devices behind NAT. If there is one-way audio issue, usually it is related to NAT configuration or SIP/RTP port configuration on the firewall.
<b>Disable This Trunk</b>	If selected, the trunk will be disabled. <b>Note:</b> If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
<b>TEL URI</b>	If the trunk has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled.
<b>Need Registration</b>	Select whether the trunk needs to register on the external server or not when "Register SIP Trunk" type is selected. The default setting is No.
<b>Allow outgoing calls if registration failure</b>	If enabled outgoing calls even if the registration to this trunk fail will still be able to go through. Note that if we uncheck "Need Registration" option, this option will be ignored.
<b>CallerID Name</b>	Configure the new name of the caller when the extension has no CallerID Name configured.





<b>From Domain</b>	<p>Configure the actual domain name where the extension comes from. This can be used to override the "From" Header.</p> <p>For example, "trunk.UCM6200.provider.com" is the From Domain in From Header:</p> <p>sip:1234567@trunk.UCM6200.provider.com.</p>
<b>From User</b>	<p>Configure the actual username of the extension. This can be used to override the "From" Header. There are cases where there is a single ID for registration (single trunk) with multiple DIDs.</p> <p>For example, "1234567" is the From User in From Header:</p> <p>sip:1234567@trunk.UCM6200.provider.com.</p>
<b>Username</b>	Enter the username to register to the trunk from the provider when "Register SIP Trunk" type is selected.
<b>Password</b>	Enter the password to register to the trunk when "Register SIP Trunk" is selected.
<b>Auth ID</b>	Enter the Authentication ID for "Register SIP Trunk" type.
<b>Auth Trunk</b>	If enabled, the UCM will send 401 response to the incoming call to authenticate the trunk.
<b>Auto Record</b>	Enable automatic recording for the calls using this trunk (for SIP trunk only). The default setting is disabled. The recording files can be accessed under Web GUI→ <b>CDR</b> → <b>Recording Files</b> .
<b>Direct Callback</b>	<p>Allows external numbers the option to get directed to the extension that last called them.</p> <p>For Example: User 2002 has dialed external number 061234575 but they were not reachable, once they have received missed call notification on their phone, they would mostly call back the number, if the option "Direct Callback" is enabled then they will be directly bridged to user 2002 regardless of the configured inbound destination.</p>
<b>Advanced Settings</b>	
<b>Codec Preference</b>	<p>Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.265, H.263, H.263p and VP8.</p>
<b>Send PPI Header</b>	<p>If enabled, the SIP INVITE message sent to the trunk will contain PPI (P-Preferred-Identity) header. The default setting is "No".</p> <p><b>Note:</b> "Send PPI Header" and "Send PAI Header" cannot be enabled at the same time. Only one of the two headers can be contained in SIP INVITE message.</p>
<b>PPI Mode</b>	<ul style="list-style-type: none"> <li>• <b>Default</b> – Include the trunk's preferred CID (configured in <i>Basic Settings</i>) in the PPI Header.</li> <li>• <b>Original CID</b> – Include the original CID in the PPI Header.</li> <li>• <b>DOD Number</b> – Include the trunk's DOD number in the PPI Header. If no DOD number has been set, the trunk's preferred CID will be used.</li> </ul>



<b>Send PAI Header</b>	<p>If enabled, the SIP INVITE message sent to the trunk will contain PAI (P-Asserted- Identity) header including configured PAI Header. The default setting is "No".</p> <p><b>Note:</b> "Send PPI Header" and "Send PAI Header" cannot be enabled at the same time. Only one of the two headers can be contained in the SIP INVITE message.</p>
<b>PAI Header</b>	<p>If "Send PAI Header" is enabled and "PAI Header" is configured as "123456" for instance, the PAI header in the SIP message sent from the UCM will contain "123456". If "Send PAI Header" is enabled and "PAI Header" is configured as "empty", the PAI header in the SIP message sent from the UCM will contain the original CID.</p> <p><b>Note:</b> "Send PAI Header" needs to be enabled in order to use this feature. Only alphanumeric characters are allowed and/or special characters #*-_+. with a limit of 64 characters.</p>
<b>DOD As From Name</b>	<p>If enabled and "From User" is configured, the INVITE's From header will contain the DOD number.</p>
<b>Passthrough PAI Header</b>	<p>If checked and option "Send PAI Header" not checked, the PAI header will be passthrough from one side to the other side.</p>
<b>Outbound Proxy Support</b>	<p>Select to enable outbound proxy in this trunk. The default setting is "No".</p>
<b>Outbound Proxy</b>	<p>When outbound proxy support is enabled, enter the IP address or URL of the outbound proxy.</p>
<b>Remove OBP from Route</b>	<p>It is used to set if the phone system will remove outbound proxy URI from the route header. If is set to "Yes", it will remove the route header from SIP requests. The default setting is "No".</p>
<b>DID Mode</b>	<p>Configure where to get the destination ID of an incoming SIP call, from SIP Request-line or To-header. The default is set to "Request-line".</p>
<b>DTMF Mode</b>	<p>Configure the default DTMF mode when sending DTMF on this trunk.</p> <ul style="list-style-type: none"> <li>• <b>Default:</b> The global setting of DTMF mode will be used. The global setting for DTMF Mode setting is under Web GUI→<b>PBX Settings</b>→<b>SIP Settings</b>→<b>ToS</b>.</li> <li>• <b>RFC4733:</b> Send DTMF using RFC4733.</li> <li>• <b>Info:</b> Send DTMF using SIP INFO message.</li> <li>• <b>Inband:</b> Send DTMF using inband audio. This requires 64-bit codec, i.e., PCMU and PCMA.</li> <li>• <b>Auto:</b> Send DTMF using RFC4733 if offered. Otherwise, inband will be used.</li> </ul>
<b>Enable Heartbeat</b>	<p>If enabled, the UCM6200 will regularly send SIP OPTIONS to the device to check</p>



<b>Detection</b>	if the device is still online. The default setting is "No".
<b>Heartbeat Frequency</b>	When "Enable Heartbeat Detection" option is set to "Yes", configure the interval (in seconds) of the SIP OPTIONS message sent to the device to check if the device is still online. The default setting is 60 seconds.
<b>Maximum Number of Call Lines</b>	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limit.
<b>Fax Mode</b>	Select Fax mode. The default setting is "None". <ul style="list-style-type: none"> <li>• <b>None:</b> Disable Fax.</li> <li>• <b>Fax Detect:</b> Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under Web GUI→<b>Call Features</b>→<b>Fax/T.38</b>.</li> </ul>
<b>SRTP</b>	Enable SRTP for the VoIP trunk. The default setting is "No".
<b>CC Settings</b>	
<b>Enable CC</b>	If enabled, the system will automatically alert the user when a called party is available, given that a previous call to that party failed for some reason.
<b>CC Max Agents</b>	Configure the maximum number of CCSS agents which may be allocated for this channel. In other words, this number serves as the maximum number of CC requests this channel is allowed to make. The minimum value is 1.
<b>CC Max Monitors</b>	Configure the maximum number of monitor structures which may be created for this device. In other words, this number tells how many callers may request CC services for a specific device at one time. The minimum value is 1.

**Table 53: SIP Peer Trunk Configuration Parameters**

<b>Basic Settings</b>	
<b>Provider Name</b>	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider's server of the trunk.
<b>Auto Record</b>	Enable automatic recording for the calls using this trunk (for SIP trunk only). The default setting is disabled. The recording files can be accessed under Web GUI→ <b>CDR</b> → <b>Recording Files</b> .
<b>Keep Original CID</b>	Keep the CID from the inbound call when dialing out, this setting will override "Keep Trunk CID" option. Please make sure that the peer PBX at the other side supports to match user entry using "username" field from authentication line.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".



<b>NAT</b>	Turn on this option when the PBX is using public IP and communicating with devices behind NAT. If there is one-way audio issue, usually it is related to NAT configuration or SIP/RTP port configuration on the firewall.
<b>Disable This Trunk</b>	<p>If selected, the trunk will be disabled.</p> <p><b>Note:</b> If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.</p>
<b>TEL URI</b>	If the trunk has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled.
<b>Caller ID</b>	<p>Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored.</p> <p><b>Important Note:</b> When making outgoing calls, the following priority order rule will be used to determine which CallerID will be set before sending out the call :</p> <ul style="list-style-type: none"> <li>• CID from inbound call (<b>Keep Original CID Enabled</b>) → Trunk Username/CallerID (<b>Keep Trunk CID Enabled</b>) → DOD → Extension CallerID Number → Trunk Username/CallerID (<b>Keep Trunk CID Disabled</b>) → Global Outbound CID.</li> </ul>
<b>CallerID Name</b>	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
<b>Transport</b>	<p>Configure the SIP transport protocol to be used in this trunk. The default setting is "UDP".</p> <ul style="list-style-type: none"> <li>• UDP</li> <li>• TCP</li> <li>• TLS</li> </ul>
<b>Direct Callback</b>	<p>Allows external numbers the option to get directed to the extension that last called them.</p> <p>For Example: User 2002 has dialed external number 061234575 but they were not reachable, once they have received missed call notification on their phone, they would mostly call back the number, if the option "Direct Callback" is enabled then they will be directly bridged to user 2002 regardless of the configured inbound destination.</p>
<b>Advanced Settings</b>	
<b>Codec Preference</b>	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM,





<b>DID Mode</b>	Configure where to get the destination ID of an incoming SIP call, from SIP Request-line or To-header. The default is set to "Request-line".
<b>DTMF Mode</b>	<p>Configure the default DTMF mode when sending DTMF on this trunk.</p> <ul style="list-style-type: none"> <li>• <b>Default:</b> The global setting of DTMF mode will be used. The global setting for DTMF Mode setting is under Web GUI→<b>PBX Settings</b>→<b>SIP Settings</b>→<b>ToS</b>.</li> <li>• <b>RFC4733:</b> Send DTMF using RFC4733.</li> <li>• <b>Info:</b> Send DTMF using SIP INFO message.</li> <li>• <b>Inband:</b> Send DTMF using inband audio. This requires 64-bit codec, i.e., PCMU and PCMA.</li> <li>• <b>Auto:</b> Send DTMF using RFC4733 if offered. Otherwise, inband is used.</li> </ul>
<b>Enable Heartbeat Detection</b>	If enabled, the UCM6200 will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
<b>Heartbeat Frequency</b>	When "Enable Heartbeat Detection" option is set to "Yes", configure the interval (in seconds) of the SIP OPTIONS message sent to the device to check if the device is still online. The default setting is 60 seconds.
<b>Maximum Number of Call Lines</b>	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limit.
<b>Fax Mode</b>	<p>Select Fax mode. The default setting is "None".</p> <ul style="list-style-type: none"> <li>• <b>None:</b> Disable Fax.</li> <li>• <b>Fax Detect:</b> Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under Web GUI→<b>Call Features</b>→<b>Fax/T.38</b>.</li> </ul>
<b>SRTP</b>	<p>Enable SRTP for the VoIP trunk.</p> <p>The default setting is "No".</p>
<b>IPVT Mode</b>	<p>If enabled, it will allow SDP passthrough to Grandstream IPVideoTalk therefore it will allow calls between the UCM and IPVideoTalk. The default setting is disabled.</p> <p><b>Note:</b> This will lock out certain UCM features.</p>
<b>Sync LDAP Enable</b>	<p>If enabled, the local UCM6200 will automatically provide and update the local LDAP contacts to the remote UCM6200 SIP peer trunk. In order to ensure successful synchronization, the remote UCM6200 peer also needs to enable this option on the SIP peer trunk.</p> <p>The default setting is "No".</p>
<b>Sync LDAP Password</b>	This is the password used for LDAP contact file encryption and decryption during the LDAP sync process. The password must be the same on both UCM6200 peers to ensure successful synchronization.



<b>Sync LDAP Port</b>	Configure the TCP port used LDAP sync feature between two peer UCM6200.
<b>LDAP Outbound Rule</b>	Specify an outbound rule for LDAP sync feature. The UCM6200 will automatically modify the remote contacts by adding prefix parsed from this rule.
<b>LDAP Dialed Prefix</b>	Specify the prefix for LDAP sync feature. The UCM6200 will automatically modify the remote contacts by adding this prefix.
<b>CC Settings</b>	
<b>Enable CC</b>	If enabled, the system will automatically alert the user when a called party is available, given that a previous call to that party failed for some reason.
<b>CC Max Agents</b>	Configure the maximum number of CCSS agents which may be allocated for this channel. In other words, this number serves as the maximum number of CC requests this channel can make. The minimum value is 1.
<b>CC Max Monitors</b>	Configure the maximum number of monitor structures which may be created for this device. In other words, this number tells how many callers may request CC services for a specific device at one time. The minimum value is 1.

**Table 54: Create New IAX Trunk**

<b>Type</b>	Select the VoIP trunk type. <ul style="list-style-type: none"> <li>Peer IAX Trunk</li> <li>Register IAX Trunk</li> </ul>
<b>Provider Name</b>	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider's server of the trunk.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
<b>Username</b>	Enter the username to register to the trunk from the provider when "Register IAX Trunk" type is selected.
<b>Password</b>	Enter the password to register to the trunk from the provider when "Register IAX Trunk" type is selected.
<b>Disable This Trunk</b>	If selected, the trunk will be disabled.

**Table 55: IAX Register Trunk Configuration Parameters**

<b>Basic Settings</b>	
<b>Provider Name</b>	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider's server of the trunk.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured.



<b>Disable This Trunk</b>	<p>The default setting is "No".</p> <p>If selected, the trunk will be disabled.</p>
<b>Caller ID</b>	<p>Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored.</p> <p><b>Important Note:</b> When making outgoing calls, the following priority order rule will be used to determine which CallerID will be set before sending out the call :</p> <p>From user (Register Trunk Only) → CID from inbound call (<b>Keep Original CID</b> Enabled) → Trunk Username/CallerID (<b>Keep Trunk CID</b> Enabled) → DOD → Extension CallerID Number → Trunk Username/CallerID (<b>Keep Trunk CID</b> Disabled) → Global Outbound CID.</p>
<b>CallerID Name</b>	<p>Configure the name of the caller to be displayed when the extension has no CallerID Name configured.</p>
<b>Username</b>	<p>Enter the username to register to the trunk from the provider.</p>
<b>Password</b>	<p>Enter the password to register to the trunk from the provider.</p>
<b>Advanced Settings</b>	
<b>Codec Preference</b>	<p>Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.265, H.263, H.263p and VP8.</p>
<b>Enable Heartbeat Detection</b>	<p>If enabled, the UCM6200 will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".</p>
<b>Heartbeat Frequency</b>	<p>When "Enable Heartbeat Detection" option is set to "Yes", configure the interval (in seconds) of the SIP OPTIONS message sent to the device to check if the device is still online. The default setting is 60 seconds.</p>
<b>Maximum Number of Call Lines</b>	<p>The maximum number of concurrent calls using the trunk. The default settings 0, which means no limited.</p>
<b>Fax Mode</b>	<p>Select Fax mode. The default setting is "None".</p> <ul style="list-style-type: none"> <li>• <b>None:</b> Disable Fax.</li> <li>• <b>Fax Detect:</b> Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under Web GUI→<b>Call Features</b>→<b>Fax/T.38</b>.</li> </ul>





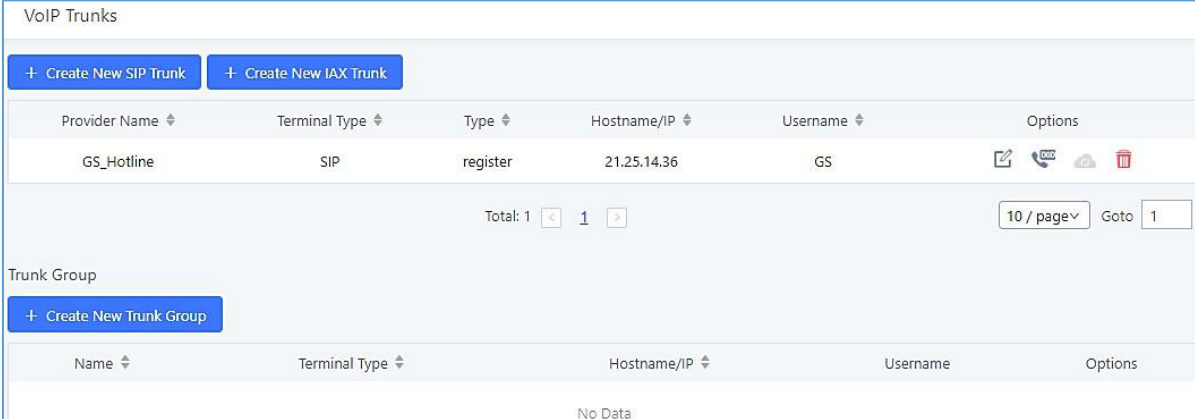
**Table 56: IAX Peer Trunk Configuration Parameters**

Basic Settings	
<b>Provider Name</b>	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules etc.
<b>Host Name</b>	Configure the IP address or URL for the VoIP provider's server of the trunk.
<b>Keep Trunk CID</b>	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
<b>Disable This Trunk</b>	If selected, the trunk will be disabled.
<b>Caller ID</b>	<p>Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored.</p> <p><b>Important Note:</b> When making outgoing calls, the following priority order rule will be used to determine which CallerID will be set before sending out the call :</p> <p>CID from inbound call (<b>Keep Original CID</b> Enabled) → Trunk Username/CallerID (<b>Keep Trunk CID</b> Enabled) → DOD → Extension CallerID Number → Trunk Username/CallerID (<b>Keep Trunk CID</b> Disabled) → Global Outbound CID..</p>
<b>CallerID Name</b>	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
Advanced Settings	
<b>Codec Preference</b>	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.265, H.263, H.263p and VP8.
<b>Enable Heartbeat Detection</b>	If enabled, the UCM6200 will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
<b>Heartbeat Frequency</b>	When "Enable Heartbeat Detection" option is set to "Yes", configure the interval (in seconds) of the SIP OPTIONS message sent to the device to check if the device is still online. The default setting is 60 seconds.
<b>Maximum Number of Call Lines</b>	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limited.
<b>Fax Mode</b>	<p>Select Fax mode. The default setting is "None".</p> <ul style="list-style-type: none"> <li>• <b>None:</b> Disable Fax.</li> <li>• <b>Fax Detect:</b> Fax signal from the user/trunk during the call can be detected and the received Fax will be sent to the Email address configured for this extension. If no Email address can be found for the user, the Fax will be sent to the default Email address configured in Fax setting page under Web</li> </ul>



## Trunk Groups

Users can create VoIP Trunk Groups to register and easily apply the same settings on multiple accounts within the same SIP server. This can drastically reduce the amount of time needed to manage accounts for the same server and improve the overall cleanliness of the web UI.



The screenshot shows two sections: 'VoIP Trunks' and 'Trunk Group'.

**VoIP Trunks Section:**

- Buttons: '+ Create New SIP Trunk', '+ Create New IAX Trunk'
- Table with columns: Provider Name, Terminal Type, Type, Hostname/IP, Username, Options
- Table Row: GS\_Hotline, SIP, register, 21.25.14.36, GS, [Edit] [Call] [Cloud] [Delete]
- Summary: Total: 1 [Previous] 1 [Next]
- Page Controls: 10 / page [Dropdown], Goto 1 [Input]

**Trunk Group Section:**

- Button: '+ Create New Trunk Group'
- Table with columns: Name, Terminal Type, Hostname/IP, Username, Options
- Table Content: No Data

**Figure 102: Trunk Group**

Once creating the new trunk group and configuring the SIP settings, users can add multiple accounts within the configured SIP server by pressing button and configuring the username, password and authentication ID fields.



**Create New Trunk Group**

---

Type: Register SIP Trunk ▾

\* Provider Name: Please select a provider

\* Host Name:

Keep Original CID:

Keep Trunk CID:

NAT:

Disable This Trunk:

TEL URI: Disabled ▾

Need Registration:

Allow outgoing calls if registration fails:

CallerID Name:

Username: Please enter the usernam / Please enter the passwon / AuthID ⊕

AuthTrunk:

Auto Record:

Direct Callback:

**Figure 103: Trunk Group Configuration**

## Direct Outward Dialing (DOD)

The UCM6200 provides Direct Outward Dialing (DOD) which is a service of a local phone company (or local exchange carrier) that allows subscribers within a company's PBX system to connect to outside lines directly.



### Example of how DOD is used:

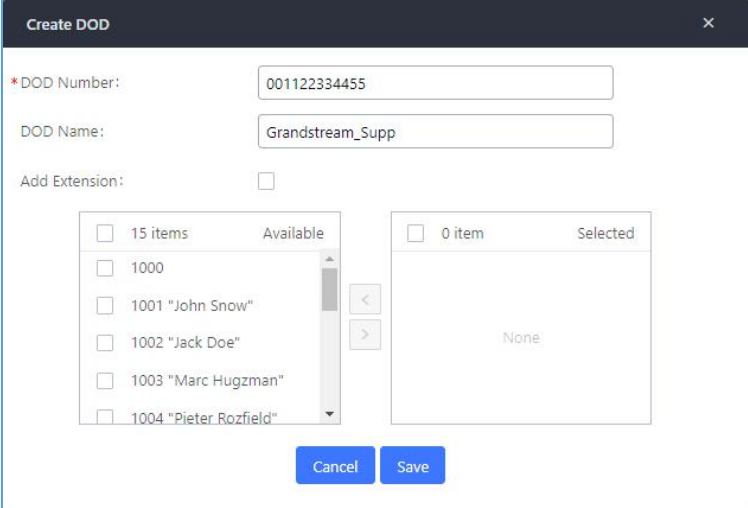
Company ABC has a SIP trunk. This SIP trunk has 4 DIDs associated to it. The main number of the office is routed to an auto attendant. The other three numbers are direct lines to specific users of the company. Now when a user makes an outbound call their caller ID shows up as the main office number. This poses a problem as the CEO would like their calls to come from their direct line. This can be accomplished by configuring DOD for the CEO's extension.

### Steps to configure DOD on the UCM6200:

1. To setup DOD go to UCM6200 Web GUI → **Extension/Trunk** → **VoIP Trunks** page.





2. Click  to access the DOD options for the selected SIP Trunk.
3. Click "Create a new DOD" to begin your DOD setup
4. For "DOD Number" enter one of the numbers (DIDs) from your SIP trunk provider. In the example above Company ABC received 4 DIDs from their provider. ABC will enter in the number for the CEO's direct line.  
**Note:** DOD number cannot exceed 32 characters.
5. Set the DOD name and If extension number need to be appended to the DID number click on "Add Extension".  
**Note:** DOD name cannot exceed 32 characters.
6. Select an extension from the "Available Extensions" list. Users have the option of selecting more than one extension. In this case, Company ABC would select the CEO's extension. After making the selection, click on the  button to move the extension(s) to the "Selected Extensions" list.



**Figure 104: DOD extension selection**

7. Click "Save" at the bottom.

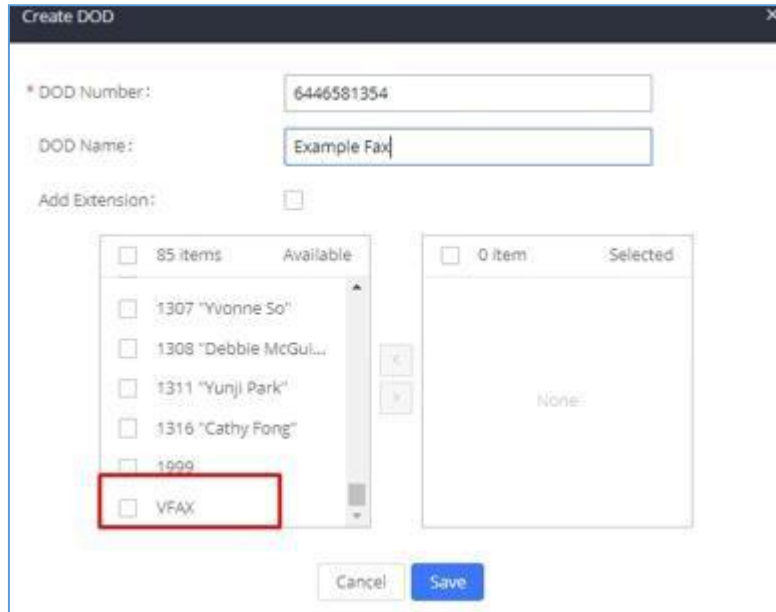
Once completed, the user will return to the **Edit DOD** page that shows all the extensions that are associated to a particular DOD.

DOD		
+ Create a new DOD		
DOD	Extensions	Options
1234568789	1000	 

**Figure 105: Edit DOD**



DOD can also 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 like shown below.



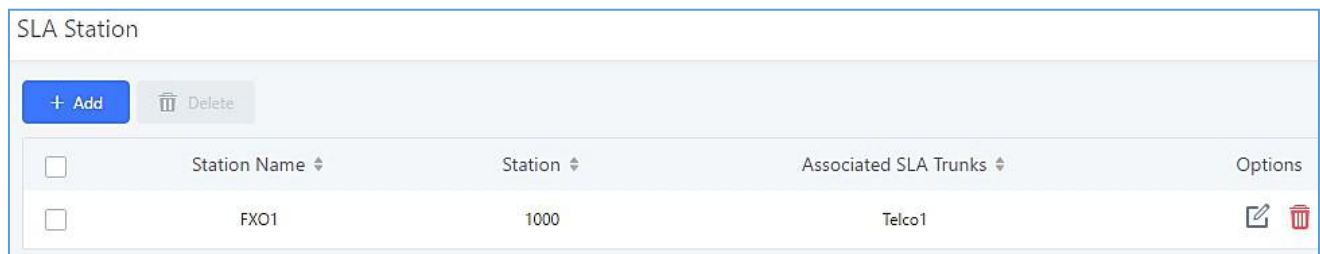
**Figure 106: Fax Sending DOD**

## SLA STATION

The UCM6200 supports SLA that allows mapping the key with LED on a multi-line phone to different external lines. When there is an incoming call and the phone starts to ring, the LED on the key will flash in red and the call can be picked up by pressing this key. This allows users to know if the line is occupied or not. The SLA function on the UCM6200 is like BLF but SLA is used to monitor external line i.e., analog trunk on the UCM6200. Users could configure the phone with BLF mode on the MPK to monitor the analog trunk status or press the line key pick up call from the analog trunk on the UCM6200.

### Create/Edit SLA Station

SLA Station can be configured on Web GUI→**Extension/Trunk**→**SLA Station**.



<input type="checkbox"/>	Station Name ▾	Station ▾	Associated SLA Trunks ▾	Options
<input type="checkbox"/>	FX01	1000	Telco1	

Figure 107: SLA Station






- Click on  to add an SLA Station.
- Click on  to edit the SLA Station. The following table shows the SLA Station configuration parameters.
- Click on  to delete the SLA Station.

Table 57: SLA Station Configuration Parameters

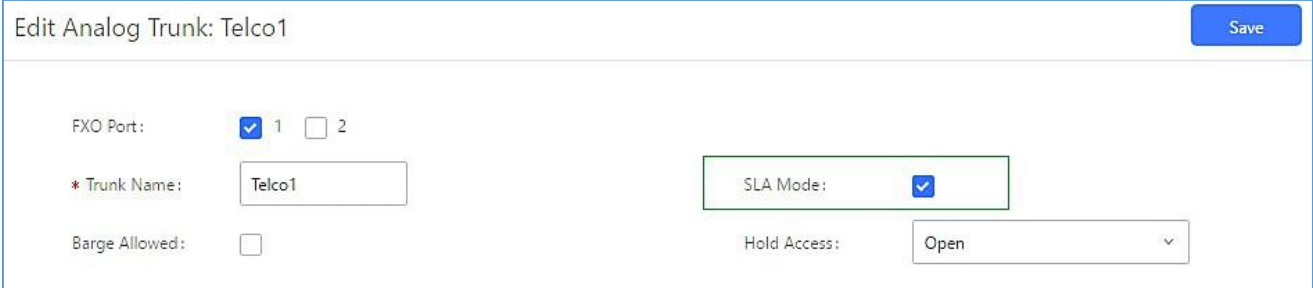
<b>Station Name</b>	Configure a name to identify the SLA Station.
<b>Station</b>	Specify a SIP extension as a station that will be using SLA.
<b>Available SLA Trunks</b>	Existing Analog Trunks with SLA Mode enabled will be listed here.
<b>Selected SLA Trunks</b>	Select a trunk for this SLA from the Available SLA Trunks list. Click on   to arrange the order. If there are multiple trunks selected, when there are calls on those trunks at the same time, pressing the LINE key on the phone will pick up the call on the first trunk here.
<b>SLA Station Options</b>	
<b>Ring Timeout</b>	Configure the time (in seconds) to ring the station before the call is considered unanswered. No timeout is set by default. If set to 0, there will be no timeout.



<b>Ring Delay</b>	Configure the time (in seconds) for delay before ringing the station when a call first coming in on the shared line. No delay is set by default. If set to 0, there will be no delay.
<b>Hold Access</b>	This option defines the competence of the hold action for one particular trunk. If set to “open”, any station could hold a call on that trunk or resume one held session; if set to “private”, only the station that places the trunk call on hold could resume the session. The default setting is “open”.

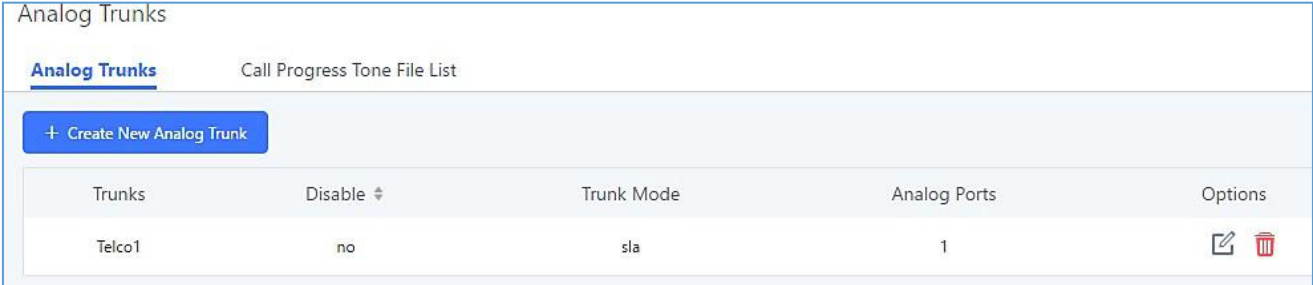
## Sample Configuration



1. On the UCM6200, go to Web GUI→**Extension/Trunk**→**Analog Trunks** page. Create analog trunk or edit the existing analog trunk. Make sure “SLA Mode” is enabled for the analog trunk. Once enabled, this analog trunk will be only available for the SLA stations created under Web GUI→**Extension/Trunk**→**SLA Station** page.



**Figure 108: Enable SLA Mode for Analog Trunk**

2. Click on “Save”. The analog trunk will be listed with trunk mode “SLA”.





Trunks	Disable	Trunk Mode	Analog Ports	Options
Telco1	no	sla	1	 

**Figure 109: Analog Trunk with SLA Mode Enabled**

3. On the UCM6200, go to Web GUI→**Extension/Trunk**→**SLA Station** page, click on “Add”. Please refer to section **[Create/Edit SLA Station]** for the configuration parameters. Users can create one or more SLA stations to monitor the analog trunk. The following figure shows two stations, 1002 and 1005, are configured to be associated with SLA trunk “fxo1”.



<input type="checkbox"/>	Station Name ▾	Station ▾	Associated SLA Trunks ▾	Options
<input type="checkbox"/>	SLA1	1005	Telco1	 

**Figure 110: SLA Example - SLA Station**

- On the SIP phone 1, configure to register UCM6200 extension 1002. Configure the MPK as BLF mode and the value must be set to “extension\_trunkname”, which is 1002\_fxo1 in this case.
- On the SIP phone 2, configure to register UCM6200 extension 1005. Configure the MPK as BLF mode and value must be set to “extension\_trunkname”, which is 1005\_fxo1 in this case.

	Mode	Account	Description	Value
MPK 1	Busy Lamp Field (BLF) ▾	Account 2 ▾	1005_fxo1	1005_fxo1

**Figure 111: SLA Example - MPK Configuration**

Now the SLA station is ready to use. The following functions can be achieved by this configuration.

- Making an outbound call from the station/extension, using LINE key**  
 When the extension is in idle state, pressing the line key for this extension on the phone to off hook. Then dial the station’s extension number, for example, dial 1002 on phone 1 (or dial 1005 on phone 2), to hear the dial tone. Then the users could dial external number for the outbound call.
- Making an outbound call from the station/extension, using BLF key**  
 When the extension is in idle state, pressing the MPK and users could dial external numbers directly.
- Answering call using LINE key**  
 When the station is ringing, pressing the LINE key to answer the incoming call.
- Barging-in active call using BLF key**  
 When there is an active call between an SLA station and an external number using the SLA trunk, other SLA stations monitoring the same trunk could join the call by pressing the BLF key if “Barge Allowed” is enabled for the analog trunk.
- Hold/UnHold using BLF key**  
 If the external line is previously put on hold by an SLA station, another station that monitors the same SLA trunk could UnHold the call by pressing the BLF key if “Hold Access” is set to “open” on the analog trunk and the SLA station.





# CALL ROUTES

## Outbound Routes




In the following sections, we will discuss the steps and parameters used to configure and manage outbound rules in UCM6200, these rules are the regulating points for all external outgoing calls initiated by the UCM through all types of trunks: SIP, Analog and Digital.

### Configuring Outbound Routes

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

**Note:** UCM6200 supports now 500 Outbound routes.

Go to Web GUI→**Extension/Trunk**→**Outbound Routes** to add and edit outbound rules.

- Click on  to add a new outbound route.
- Click on  to edit the outbound route.
- Click on  to delete the outbound route.

On the UCM6200, the outbound route priority is based on “Best matching pattern”. For example, the UCM6200 has outbound route A with pattern 1xxx and outbound route B with pattern 10xx configured. When dialing 1000 for outbound call, outbound route B will always be used first. This is because pattern 10xx is a better match than pattern 1xxx. Only when there are multiple outbound routes with the same pattern configured.

**Table 58: Outbound Route Configuration Parameters**

<b>Calling Rule Name</b>	Configure the name of the calling rule (e.g., local, long_distance, and etc.). Letters, digits, _ and - are allowed.
<b>Pattern</b>	<ul style="list-style-type: none"> <li>• All patterns are prefixed with the "_".</li> <li>• Special characters:  <b>X:</b> Any Digit from 0-9.  <b>Z:</b> Any Digit from 1-9.  <b>N:</b> Any Digit from 2-9.            " . ": Wildcard. Match one or more characters.            " ! ": Wildcard. Match zero or more characters immediately.            Example: [12345-9] - Any digit from 1 to 9.</li> </ul> <p><b>Notes:</b></p>



<b>Disable This Route</b>	<ul style="list-style-type: none"> <li>▪ Multiple patterns can be used. Each pattern should be entered in new line.</li> <li>▪ Users can add comments to the end of patterns to better organize and keep track of complex rules by typing “/” and “*/” before and after each comment, respectively.</li> <li>▪ <u>Example:</u>            _X.            _NNXXNXXXXX /* 10-digit long distance */            _818X. /* Any number with leading 818 */</li> </ul> <p>After creating the outbound route, users can choose to enable and disable it. If the route is disabled, it will not take effect anymore. However, the route settings will remain in UCM. Users can enable it again when it is needed.</p>
<b>Password</b>	<p>Configure the password for users to use this rule when making outbound calls.</p>
<b>Call Duration Limit</b>	<p>Once call duration limit is enabled, it will set the maximum duration of call-blocking.</p>
<b>Maximum Call Duration</b>	<p>User can customize the maximum call duration (in seconds) that is allowed for the outbound call. By default, this value is set to 0 means there is no limit for the call duration.</p>
<b>Warning Time</b>	<p>This option will give caller warning when call duration is approaching to its limit. If the warning time is set to ‘y’, the warning tone will be played to caller when y seconds is left to end the call by UCM.</p>
<b>Warning Repeat Interval</b>	<p>Once this option is set to ‘z’, it will repeatedly be warning caller every z seconds after the first warning.</p>
<b>PIN Groups</b>	<p>If selected, the Password, Privilege Level and Enable Filter on Source Caller ID will not take effect.        For more details, refer to [PIN Groups] section.</p>
<b>PIN Groups with Privilege Level</b>	<p>If enabled and PIN Groups are used, Privilege Levels and Filter on Source Caller ID will also be applied.</p>
<b>Password</b>	<p>Configure the password for users to use this rule when making outbound calls.</p>
<b>Privilege Level</b>	<p>Select privilege level for the outbound rule.</p> <ul style="list-style-type: none"> <li>• <b>Internal:</b>            The lowest level required. All users can use this rule.</li> <li>• <b>Local:</b></li> </ul>





<b>Use Trunk</b>	Select the trunk for this outbound rule.
<b>Strip</b>	<p>Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk.</p> <p><u>Example:</u></p> <p>The users will dial 9 as the first digit of a long-distance calls. However, 9 should not be sent out via analog lines and the PSTN line. In this case, 1 digit should be stripped before the call is placed.</p>
<b>Prepend</b>	<p>Specify the digits to be prepended before the call is placed via the trunk.</p> <p>Those digits will be prepended after the dialing number is stripped.</p>
<b>Use Failover Trunk</b>	
<b>Failover Trunk</b>	<p>Failover trunks can be used to make sure that a call goes through an alternate route when the primary trunk is busy or down. If "Use Failover Trunk" is enabled and "Failover trunk" is defined, the calls that cannot be placed via the regular trunk may have a secondary trunk to go through.</p> <p>UCM6200 support up to 10 failover trunks.</p> <p><u>Example:</u> The user's primary trunk is a VoIP trunk and the user would like to use the PSTN when the VoIP trunk is not available. The PSTN trunk can be configured as the failover trunk of the VoIP trunk.</p>
<b>Strip</b>	<p>Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk.</p> <p><u>Example:</u></p> <p>The users will dial 9 as the first digit of a long-distance calls. However, 9 should not be sent out via analog lines and the PSTN line. In this case, 1 digit should be stripped before the call is placed.</p>
<b>Prepend</b>	<p>Specify the digits to be prepended before the call is placed via the trunk.</p> <p>Those digits will be prepended after the dialing number is stripped.</p>
<b>Time Condition</b>	
<b>Time Condition</b>	<p>Users could customize holiday time, office time, out of office time, out of holiday, out of office time or holiday, office time and out of holiday or a specified time to allow the outbound route to be used.</p>

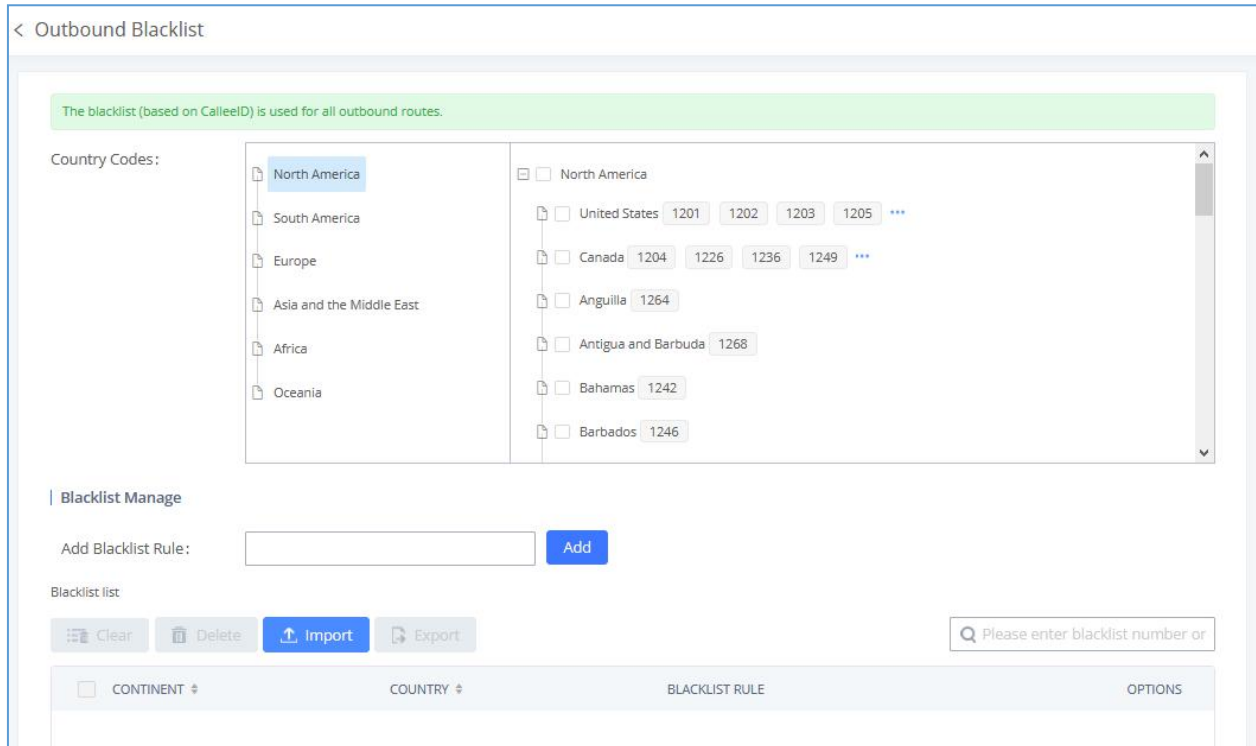
## Outbound Blacklist

The UCM6200 allows users to configure blacklist for outbound routes. If the dialing number matches the blacklist numbers or patterns, the outbound call will not be allowed. The outbound blacklist can be configured under UCM



Web GUI → **Extension/Trunk** → **Outbound Routes**: Outbound Blacklist.

Users can configure numbers, patterns or select country code to add in the blacklist. Please note that the blacklist settings apply to all outbound routes.



**Figure 112: Country Codes**

## Blacklist Manage

### Add Blacklist Rule

Allows to define a rule based on number(s) or pattern(s) as blacklist entry.

Pattern rules:

**N** : Any digit from 2-9

**X** : Any digit from 0-9

**Z** : Any digit from 1-9

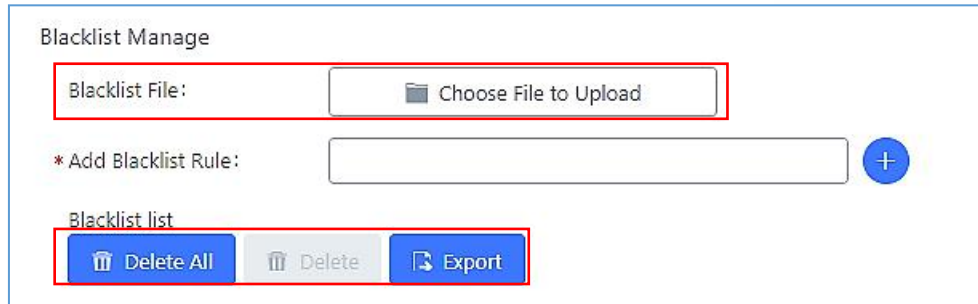
**.** : Wildcard, matching one or more characters

**!** : Wildcard, matching zero or more characters immediately

**-** : Hyphens are used mainly to improve readability and are not involved in pattern matching.

**Note:** Users can export outbound route blacklists and delete all blacklist entries. Additionally, users can also import blacklists for outbound routes.





**Figure 113: Blacklist Import/Export**

## PIN Groups

The UCM6200 supports pin group. Once this feature is configured, users can apply pin group to specific outbound routes. When placing a call on pin protected outbound routes, caller will be asked to input the group pin number, this feature can be found on the webGUI→**Extension/Trunk**→**Outbound Routes**→**PIN Groups**.

**Table 59: Outbound Routes/PIN Group**

<b>Name</b>	Specify the name of the group
<b>Record In CDR</b>	Specify whether to enable/disable record in CDR
<b>PIN Number</b>	Specify the code that will asked once dialing via a trunk
<b>PIN Name</b>	Specify the name of the PIN

Once user click on [PIN Groups](#) the following figure shows to configure the new PIN.



Create New PIN Group
Save

\* Name:

Record in CDR:

**Members**

\* PIN Number:

\* PIN Name:

✓ Save
✗ Cancel

PIN Number: 2020 PIN Name: John

✎
🗑️

**Figure 114: Create New PIN Group**

The following screenshot shows an example of created PIN Groups and members:

PIN Groups

+ Add
Choose file to upload

	Name ↕	Record in CDR ↕	Options
-	GSEMEA	yes	✎ 🗑️

	PIN Number	PIN Name
	2020	John
	2025	Emily
	3009	Jane

**Figure 115: PIN Members**

**Note:**

If PIN group is enabled on outbound route level, password, privilege level and enable filter on source caller ID will be disabled.

Edit Outbound Rule: National
Save

\* Calling Rule Name:

Disable This Route:

Password:

\* Pattern:

PIN Groups:

Privilege Level:

**Figure 116: Outbound PIN**

If PIN group CDR is enabled, the call with PIN group information will be displayed as part of CDR under Account Code field.

CDR

🗑 Delete All
⬇ Download All Records
⬇ Download Search Result (s)
⚙ Automatic Download Settings

Status	Call from	Call to	Action Type	Start Time	Talk Time	Account Code
+	1002	7946541 [Trunk: BranchOffice]	DIAL	2017-05-05 04:59:51	0:00:08	Emily/GSEMEA
+	1002	7654654 [Trunk: BranchOffice]	DIAL	2017-05-05 04:59:12	0:00:06	Jane/GSEMEA
+	1002	7564654 [Trunk: BranchOffice]	DIAL	2017-05-05 04:58:38	0:00:06	John/GSEMEA

**Figure 117: CDR Record**

**- Importing PIN Groups from CSV files:**

User can also import PIN Groups by uploading CSV files for each group. To do this:

1. Navigate to **Extension/Trunk**→**Outbound Routes**→**PIN Groups** and click on the “Choose file to upload” button.





PIN Groups

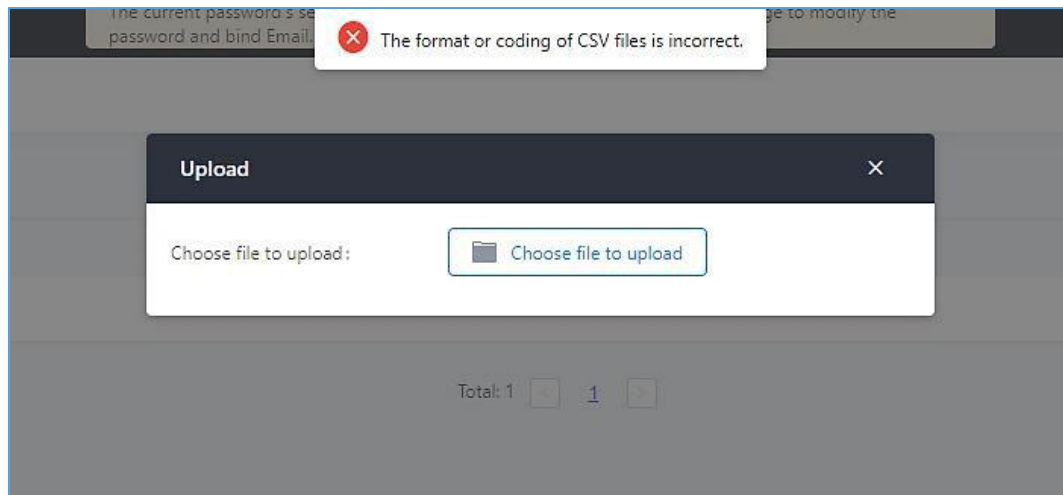
+ Add **Choose file to upload**

Name	Record in CDR	Options
GSEMEA	yes	

PIN Number	PIN Name
2020	John
2025	Emily
3009	Jane

**Figure 118: Importing PIN Groups from CSV files**

2. Select the CSV file to upload. Incorrect file formats and improperly formatted CSV files will result in error messages such as the one below:



**Figure 119: Incorrect CSV File**

3. To ensure a successful import, please follow the format in the sample image below



	A	B	C	D
1	ALPHA			
2	pin	pin_name		
3	1625	test1		
4	9497	test2		
5	5872	test3		
6				
7				

**Figure 120: CSV File Format**


- The top-left value (A1) is the PIN Group name. In this case, it is “ALPHA”.
- Row 2 contains the labels for the modifiable fields: pin and pin\_name. These values should not be changed and will cause an upload error otherwise.
- Rows 3+ contain the user-defined values with Column A holding the PINs and Column B holding the PIN names. PIN values must consist of at least four digits.
- Once the file is successfully uploaded, the entry will be added to the list of PIN Groups.

Name	Record in CDR	Options
ALPHA	no	 

Total: 1 | 1 | 10 / page | Goto 1

**Figure 121: CSV File Successful Upload**


### - Exporting PIN Groups from CSV files:

Press  button under “Options” to download/export PIN Group settings.



## Inbound Routes

Inbound routes can be configured via Web GUI → **Extension/Trunk** → **Inbound Routes**.

**Note:** UCM6200 now supports 5000 Inbound route.

- Click on  to add a new inbound route.



- Click on "Blacklist" to configure blacklist for all inbound routes.
- Click on  to edit the inbound route.
- Click on  to delete the inbound route.

## Inbound Rule Configurations

**Table 60: Inbound Rule Configuration Parameters**

<b>Trunks</b>	Select the trunk to configure the inbound rule.								
<b>Pattern</b>	<ul style="list-style-type: none"> <li>• All patterns are prefixed with the "_".</li> <li>• Special characters:           <ul style="list-style-type: none"> <li><b>X</b>: Any Digit from 0-9.</li> <li><b>Z</b>: Any Digit from 1-9.</li> <li><b>N</b>: Any Digit from 2-9.</li> <li><b>."</b>: Wildcard. Match one or more characters.</li> <li><b>"!"</b>: Wildcard. Match zero or more characters immediately.</li> <li>Example: <b>[12345-9]</b> - Any digit from 1 to 9.</li> </ul> </li> <li>• The pattern can be composed of two parts, <i>Pattern</i> and <i>CallerID Pattern</i>. The first part is used to specify the dialed number while the second part is used to specify the caller ID and it is optional, if set it means only the extension with the specific caller ID can call in or call out. For example, pattern '_2XXX/1234' means the only extension with the caller ID '1234' can use this rule.</li> </ul> <p><u>Notes:</u></p> <ul style="list-style-type: none"> <li>▪ Multiple patterns can be used. Each pattern should be entered in new line.</li> <li>▪ Users can add comments to the end of patterns to better organize and keep track of complex rules by typing "/" and "/" before and after each comment, respectively.</li> <li>▪ Example:</li> </ul> <table border="1" data-bbox="639 1486 1435 1667"> <thead> <tr> <th>Pattern</th> <th>CallerID Pattern</th> </tr> </thead> <tbody> <tr> <td>_X.</td> <td>1000</td> </tr> <tr> <td>_NNXXNXXXXX /* 10-digit long distance */</td> <td>1001</td> </tr> <tr> <td>_818X. /* Any number with leading 818 */</td> <td></td> </tr> </tbody> </table>	Pattern	CallerID Pattern	_X.	1000	_NNXXNXXXXX /* 10-digit long distance */	1001	_818X. /* Any number with leading 818 */	
Pattern	CallerID Pattern								
_X.	1000								
_NNXXNXXXXX /* 10-digit long distance */	1001								
_818X. /* Any number with leading 818 */									
<b>Disable This Route</b>	After creating the inbound route, users can choose to enable and disable it. If the route is disabled, it will not take effect anymore. However, the route settings will remain in UCM. Users can enable it again when it is needed.								



<b>Allowed to seamless transfer</b>	Allows the selected extension to use this function. If an extension is busy, and a mobile phone is bound to that extension, the mobile phone can pick up calls to that extension.
<b>Block the Backward Collect Call</b>	To indicate whether to block the backward collect call. If checked, block the call. Note that the header "P-Asserted-Service-Info: service-code=Backward Collect Call" indicates the backward collect call.
<b>Alert-Info</b>	Configure the Alert-Info, when UCM receives an INVITE request, the Alert-Info header field specifies an alternative ring tone to the UAS.
<b>Prepend Trunk Name</b>	If enabled, the trunk name will be added to the caller id name as the displayed caller id name.
<b>Set Caller ID Info</b>	Manipulates Caller ID (CID) name and/or number within the call flow to help identify who is calling. When enabled two field will show allowing to manipulate the CalleID Number and the Caller ID Name.
<b>CalleID Number</b>	Configure the pattern-matching format to manipulate the numbers of incoming callers or to set a fixed CallerID number for calls that go through this inbound route. <ul style="list-style-type: none"> <li>• <b>`\${CALLERID(num)}`</b>: Default value which indicates the number of an incoming caller (CID). The CID will not be modified.</li> <li>• <b>`\${CALLERID(num):n}`</b>: Skips the first n characters of a CID number, where n is a number.</li> <li>• <b>`\${CALLERID(num):-n}`</b>: Takes the last n characters of a CID number, where n is a number.</li> <li>• <b>`\${CALLERID(num):s:n}`</b>: Takes n characters of a CID number starting from s+1, where n is a number and s is a character position (e.g. <code>`\${CALLERID(num):2:7}`</code> takes 7 characters after the second character of a CID number).</li> <li>• <b>n`\${CALLERID(num)}`</b>: Prepends n to a CID number, where n is a number.</li> </ul>
<b>CallerID Name</b>	Default string is <b>`\${CALLERID(name)}`</b> , which means the name of an incoming caller, it is a pattern-matching syntax format. <b>A`\${CALLERID(name)}B`</b> means Prepend a character 'A' and suffix a character 'B' to <b>`\${CALLERID(name)}`</b> . Not using pattern-matching syntax means setting fix name to incoming caller.
<b>Enable Route-Level Inbound</b>	Gives uses the ability to configure inbound mode per individual route. When enabled two field will show allowing to set the Inbound mode and the Inbound mode Suffix.



<b>Inbound Mode</b>	<p><b>Note:</b> Global inbound mode must be enabled before users can configure route-level inbound mode</p> <p>Choose the inbound mode for this route.</p> <p><b>Note:</b> Toggling the global inbound mode will not affect routes that have Route-level Inbound Mode enabled. If all routes have the option enabled, toggling the global inbound mode via BLF will trigger a voice prompt indicating that none of the routes will be affected by the global inbound mode change.</p>
<b>Inbound Mode Suffix</b>	<p>Dial "Global Inbound Mode feature code + Inbound Mode Suffix" or a route's assigned suffix to toggle the route's inbound mode.</p> <p>The BLF subscribed to the inbound mode suffix can monitor the current inbound mode.</p>
<b>Inbound Multiple Mode</b>	<p>Multiple mode allows user to switch between destinations of the inbound rule by feature codes. Configure related feature codes as described in <b>[Inbound Route: Multiple Mode]</b>. If this option is enabled, user can use feature code to switch between different modes/destinations.</p>
<b>Dial Trunk</b>	<p>This option shows up only when "By DID" is selected. If enabled, the external users dialing in to the trunk via this inbound route can dial outbound call using the UCM's trunk.</p>
<b>Privilege Level</b>	<p>This option shows up only when "By DID" is selected.</p> <ul style="list-style-type: none"> <li>• <b>Disable:</b> Only the selected Extensions or Extension Groups are allowed to use this rule, when enabled Filter on Source Caller ID.</li> <li>• <b>Internal:</b> The lowest level required. All users are allowed to use this rule, check this level might be risky for security purpose.</li> <li>• <b>Local:</b> User with Local level, National or International level are allowed to use this rule.</li> <li>• <b>National:</b> Users with National or International Level are allowed to use this rule.</li> <li>• <b>International:</b> The highest level required. Only users with international level are allowed to use this rule.</li> </ul>
<b>Allowed DID Destination</b>	<p>This option shows up only when "By DID" is selected. This controls the destination that can be reached by the external caller via the inbound route. The DID destination are:</p> <ul style="list-style-type: none"> <li>• Extension</li> <li>• Conference</li> <li>• Call Queue</li> <li>• Ring Group</li> </ul>



	<ul style="list-style-type: none"> <li>• Paging/Intercom Groups</li> <li>• IVR</li> <li>• Voicemail Groups</li> <li>• Fax Extension</li> <li>• Dial By Name</li> <li>• All</li> </ul>
<b>Default Destination</b>	<p>Select the default destination for the inbound call.</p> <ul style="list-style-type: none"> <li>• Extension</li> <li>• Voicemail</li> <li>• Conference Room</li> <li>• Call Queue</li> <li>• Ring Group</li> <li>• Paging/Intercom</li> <li>• Voicemail Group</li> <li>• Fax</li> <li>• DISA</li> <li>• IVR</li> <li>• External Number</li> <li>• By DID</li> </ul> <p>When "By DID" is used, the UCM will look for the destination based on the number dialed, which could be local extensions, conference, call queue, ring group, paging/intercom group, IVR, voicemail groups and Fax extension as configured in "DID destination". If the dialed number matches the DID pattern, the call will be allowed to go through.</p> <ul style="list-style-type: none"> <li>• Dial By Name</li> <li>• Callback</li> <li>• Announcement</li> </ul>
<b>Strip</b>	Specify the number of digits to strip from the beginning of the DID. This is used when "By DID" is selected in "Default Destination".
<b>Prepend</b>	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
<b>Time Condition</b>	
<b>Start Time</b>	Select the start time "hour:minute" for the trunk to use the inbound rule.
<b>End Time</b>	Select the end time "hour:minute" for the trunk to use the inbound rule.
<b>Date</b>	Select "By Week" or "By Day" and specify the date for the trunk to use the inbound rule.
<b>Week</b>	Select the day in the week to use the inbound rule.
<b>Destination</b>	<p>Select the destination for the inbound call under the defined time condition.</p> <ul style="list-style-type: none"> <li>• Extension</li> </ul>



- Voicemail
- Conference Room
- Call Queue
- Ring Group
- Paging/Intercom
- Voicemail Group
- Fax
- DISA
- IVR
- By DID

When "By DID" is used, the UCM will look for the destination based on the number dialed, which could be local extensions, conference, call queue, ring group, paging/intercom group, IVR, voicemail groups and Fax extension as configured in "DID destination". If the dialed number matches the DID pattern, the call will be allowed to go through.

Configure the number of digits to be stripped in "Strip" option.

- Dial By Name
- External Number
- Callback
- Announcement

### Inbound Route: Prepend Example

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

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



Edit Inbound Rule
Save

---

\* Pattern:

Disable This Route:

Alert-info:

Prepend Trunk Name:

Enable Route-Level Inbound:

Mode:

Dial Trunk:

Allowed DID Destination:

Inbound Multiple Mode:

CallerID Pattern:

Allowed to seamless transfer:

Block the Backward Collect:

Call:

Set CallerID Info:

**Default Mode** Mode 1

---

\* Default Destination:

Strip:

Prepend:

**Figure 122: Inbound Route feature: Prepend**

## Inbound Route: Multiple Mode

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





Edit Inbound Rule
Save

---

\* Pattern:

Disable This Route:

Alert-info:

Prepend Trunk Name:

Enable Route-Level Inbound

Mode:

Inbound Multiple Mode:

CallerID Pattern:

Allowed to seamless transfer:

Block the Backward Collect

Call:

Set CallerID Info:

Default Mode:

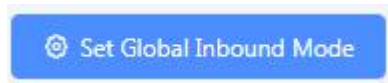
\* Default Destination:

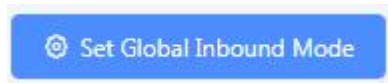
**Figure 123: Inbound Route - Multiple Mode**

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

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

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



To customize feature codes for “Default Mode” and “Mode 1”, click on  under “Inbound Routes” page, check “Enable Inbound Multiple Mode” option and change “Inbound Default Mode” and “Inbound Mode 1” values (By default, \*61 and \*62 respectively).



### Set Global Inbound Mode

Caution: Disabling Inbound Multiple Mode will switch the inbound mode to default mode.

Enable Inbound Multiple

Mode:

Inbound Mode:

\* Inbound Default Mode:

\* Inbound Mode 1:

**Figure 124: Inbound Route - Multiple Mode Feature Codes**

## Inbound Route: Route-Level Mode

In the UCM6200, users can enable Route-Level Inbound Mode to switch between different destinations for each individual inbound route. The inbound Route-Level mode can be enabled under Inbound Route settings.

### Edit Inbound Rule Save

<p>* Pattern: <input type="text" value="_18005251163"/></p> <p>Disable This Route: <input type="checkbox"/></p> <p>Alert-info: <input type="text" value="None"/></p> <p>Prepend Trunk Name: <input type="checkbox"/></p> <div style="border: 2px solid green; padding: 2px;"> <p>Enable Route-Level Inbound <input checked="" type="checkbox"/></p> <p>Mode:</p> </div> <p>Inbound Multiple Mode: <input type="checkbox"/></p>	<p>CallerID Pattern: <input type="text"/></p> <p>Allowed to seamless transfer: <input type="text"/></p> <p>Block the Backward Collect: <input type="checkbox"/></p> <p>Call:</p> <p>Set CallerID Info: <input type="checkbox"/></p> <p>Inbound Mode: <input type="text" value="Default Mode"/></p> <p>* Inbound Mode Suffix: <input type="text" value="5000"/></p>
--	--

**Default Mode** Mode 1

\* Default Destination:

**Figure 125: Inbound Route - Route-Level Mode**



Global inbound mode must be enabled before configuring Route-Level Inbound Mode. Additionally, the Mode 1 must be configured as well.

When Route-Level Inbound Mode is enabled, the user can configure a “Default Destination” and a “Mode 1” destination for each specific route. By default, the call coming into this specific inbound route will be routed to the default destination.

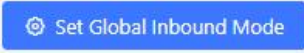
Users can toggle the route’s inbound mode by dialing "Global Inbound Mode feature code + Inbound Mode Suffix" and the current inbound route can be monitored by subscribing a BLF to the Inbound Mode Suffix.

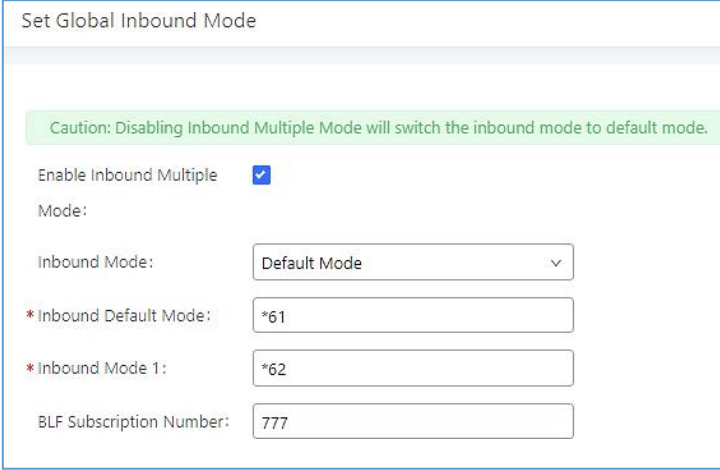
For example, Inbound Default Mode feature code is set to \*61 and the Inbound Mode suffix for route 1 is set to 1010. To switch the mode of route 1 to Default Mode, users can dial \*611010.

**Note:** Toggling the global inbound mode will not affect routes that have *Route-level Inbound Mode* enabled. If all routes have the option enabled, toggling the global inbound mode via BLF will trigger a voice prompt indicating that none of the routes will be affected by the global inbound mode change.

### Inbound Route: Inbound Mode BLF Monitoring

Users can assign MPKs and VPKs to monitor and toggle the current global inbound mode of the UCM. To do this, please refer to the following steps:

1. Access the UCM web GUI and navigate to Extension/Trunk→Inbound Routes.
2. Click on the  button and enable Inbound Multiple Mode.
3. Edit the subscribe number field to the desired BLF value.



**Figure 126: Global Inbound Mode**

4. Configure the BLF value on a phone’s MPK/VPK. As an example, a GXP2140 with the BLF configured will show the Inbound Mode status on its screen once configured. The 777 BLF is lit green, indicating that the current inbound mode is “Default Mode”.





**Figure 127: Inbound Mode - Default Mode**

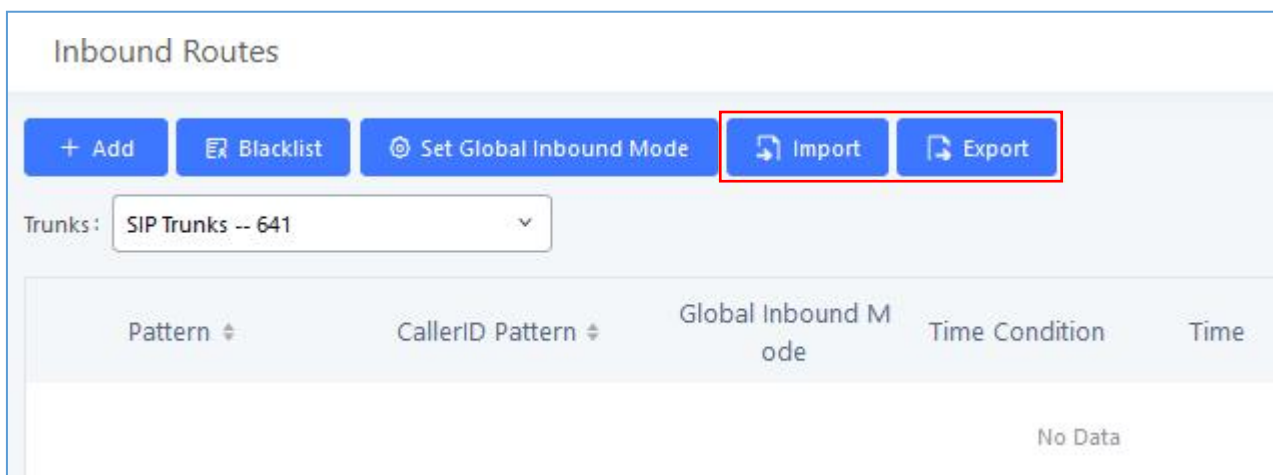
- Pressing the key will toggle the inbound mode to “Mode 1”, and the button’s color will change to red.



**Figure 128: Inbound Mode - Mode 1**

## Inbound Route: Import/Export Inbound Route

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



**Figure 129: Import/Export Inbound Route**

The imported file should be on CSV format and using UTF-8 encoding, the imported file should contain below columns, and each column should be separated by a comma (It is recommended to use Notepad++ for the imported file creation):

- Disable This Route: Yes/No.
- Pattern: Always prefixed with \_
- CallerID Pattern: Always prefixed with \_



- Prepend Trunk Name: Yes/No.
- Prepend User Defined Name Enable: Yes/No.
- Prepend User Defined Name: A string.
- Alert-info: None, Ring 1, Ring 2... User should enter an Alert-info string following the values we have in the Inbound route Alert-Info list.
- Allowed to seamless transfer: [Extension\_number]
- Fax Detection: No, Yes.
- Fax Type: Extension, Fax to Email.
- Fax Destination: [Extension\_number] or [Email address]
- Inbound Multiple Mode: Yes/No.
- Default Destination: By DID, Extension, Voicemail... User should enter a Default Destination string following the values we have in the Inbound route Default Destination list.
- Destination: An Extension number, Ring Group Extension...
- Default Time Condition.
- Mode 1: By DID, Extension, Voicemail... User should enter a Default Destination string following the values we have in the mode 1 Default Destination list.
- Mode 1 Destination: An Extension number, Ring Group Extension...
- Mode 1 Time Condition.

## **FAX Intelligent Route**


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

## **FAX with Two Media**




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

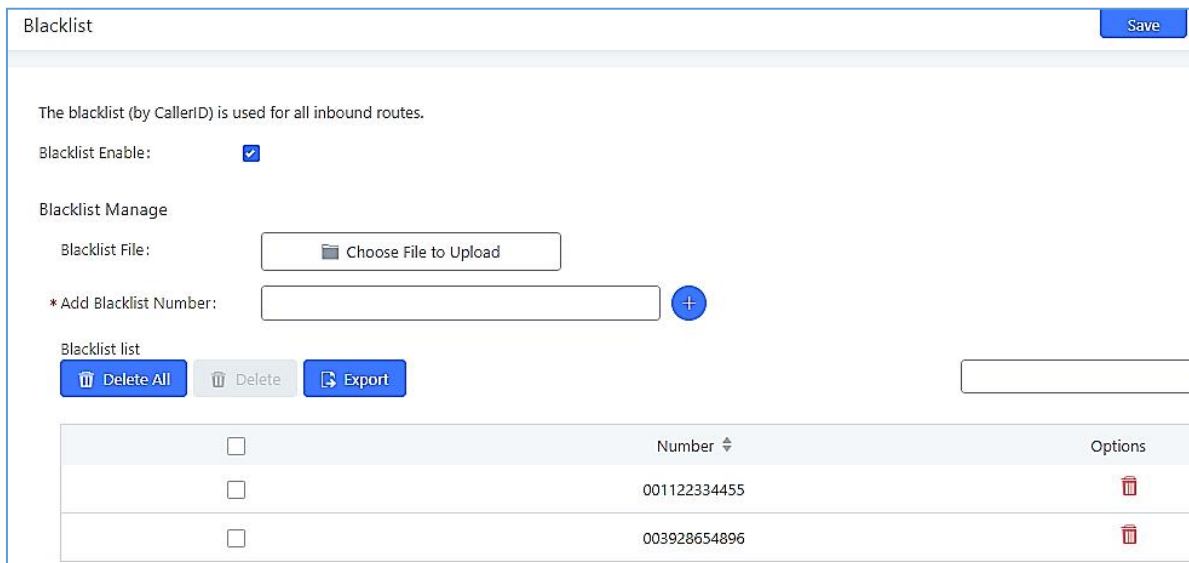
## **Blacklist Configurations**

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

- Select the checkbox for "Blacklist Enable" to turn on Blacklist feature for all inbound routes. Blacklist is disabled by default.
- Enter a number in "Add Blacklist Number" field and then click  to add to the list. Anonymous can also be added as a Blacklist Number.



- To remove a number from the Blacklist, select the number in "Blacklist list" and click on  or click on  button to remove all the numbers on the blacklist.
- User can also export the inbound route blacklist by pressing on  button.



Blacklist Save

The blacklist (by CallerID) is used for all inbound routes.




Blacklist Enable:



Blacklist Manage

Blacklist File:

\*Add Blacklist Number:  +

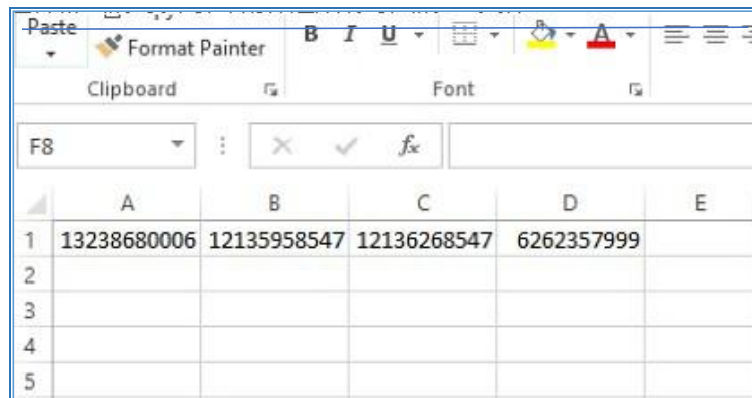
Blacklist list

<input type="checkbox"/>	Number	Options
<input type="checkbox"/>	001122334455	
<input type="checkbox"/>	003928654896	

**Figure 130: Blacklist Configuration Parameters**

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



	A	B	C	D	E
1	13238680006	12135958547	12136268547	6262357999	
2					
3					
4					
5					

**Figure 131: Blacklist csv File**

 **Note:**

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





## CONFERENCE

The UCM6200 supports conference room allowing multiple rooms used at the same time:

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

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

### Conference Room Configurations

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

**Table 61: Conference Room Configuration Parameters**

<b>Extension</b>	Configure the conference number for the users to dial into the conference. <b>Note:</b> Up to 64 characters.
<b>Password</b>	When configured, the users who would like to join the conference call must enter this password before accessing the conference room.  <b>Notes:</b> <ul style="list-style-type: none"> <li>• If "Public Mode" is enabled, the password is not required to join the conference room thus this field is invalid.</li> <li>• The password must be at least 4 characters.</li> <li>• Conference extension number can no longer be used as the conference password if Strong Password is enabled.</li> </ul>
<b>Admin Password</b>	Configure the password to join the conference room as administrator. Conference administrator can manage the conference call via IVR (if "Enable Caller Menu" is enabled) as well as invite other parties to join the conference by dialing "0" (permission required from the invited party) or "1" (permission not required from the invited party) during the conference call.  <b>Notes:</b> <ul style="list-style-type: none"> <li>• If "Public Mode" is enabled, the password is not required to join the conference room thus this field is invalid.</li> </ul>



<b>Enable Caller Menu</b>	<ul style="list-style-type: none"> <li>• Password must be at least 4 characters and different than conference extension number.</li> </ul> <p>If enabled, conference participant could press the * key to access the conference room menu. The default setting is "No".</p>
<b>Record Conference</b>	<p>If enabled, the calls in this conference room will be recorded automatically in a .wav format file. All the recording files will be displayed and can be downloaded in the conference web page. The default setting is "No".</p>
<b>Quiet Mode</b>	<p>If enabled, if there are users joining or leaving the conference, voice prompt or notification tone will not be played. The default setting is "No".</p> <p><b>Note:</b> "Quiet Mode" and "Announce Callers" cannot be enabled at the same time.</p>
<b>Kick Warning Interval (minutes)</b>	<p>If there is only one participant in a conference room, a kick warning prompt will play at the configured interval. If no input from the participant is received after the prompt, he will be automatically kicked out of the conference. The valid range is 1-60 minutes.</p>
<b>Wait For Admin</b>	<p>If enabled, the participants will not hear each other until the conference administrator joins the conference. The default setting is "No".</p> <p><b>Note:</b> If "Quiet Mode" is enabled, the voice prompt for "Wait For Admin" will not be announced.</p>
<b>Enable User Invite</b>	<p>If enabled, users could press 0 to invite other users (with the users' permission) or press 1 to invite other users (without the user's permission) to join the conference. The default setting is "No".</p> <p><b>Note:</b> Conference administrator can always invite other users without enabling this option.</p>
<b>Announce Callers</b>	<p>If enabled, the caller will be announced to all conference participants when there the caller joins the conference. The default setting is "No".</p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>• "Quiet Mode" and "Announce Callers" cannot be enabled at the same time.</li> <li>• Conference participant names will be announced when joining/leaving the conference even when the conference is on hold..</li> </ul>
<b>Public Mode</b>	<p>If enabled, no authentication will be required when joining the conference call. The default setting is "Yes".</p>





<b>Play Hold Music</b>	If enabled, the UCM6200 will play Hold music when there is only one user in the conference. The default setting is "No".
<b>Music On Hold</b>	Select the music on hold class to be played in conference call. Music On Hold class can be set up under Web GUI→ <b>PBX Settings</b> → <b>Music On Hold</b> .
<b>Skip Authentication When Inviting User via Trunk from Web GUI</b>	If enabled, the invitation from Web GUI for a conference room with password will skip the authentication for the invited users. The default setting is "No".

Conference Settings contains the following options:

**Table 62: Conference Settings**

<b>Enable Talk detection</b>	If enabled, the AMI will send the corresponding event when a user starts or ends talking.
<b>DSP Talking Threshold</b>	The time in milliseconds of sound above what the dsp has established as base line silence for a user before a user is considered to be talking. This value affects several operations and should not be changed unless the impact on call quality is fully understood, the default value is 128.
<b>DSP Silence Threshold</b>	The time in milliseconds of sound falling within the what the dsp has established as base line silence before a user is considered to be silent. This value affects several operations and should not be changed unless the impact on call quality is fully understood, the default value is 2500.

Users can check the talking Caller IDs in conference control page (UCM WebUI→Call Features→Conference).

The image will move up and down when the user is talking.



Conference

**Conference**    Conference Schedule    Google Service Settings    Record Conference

+ Create New Conference Bridge    @ Conference Settings    Enable CEI Notify

	Room	Attendee	Administrator	Start Time	Activity	Options
-	6300	2	0	2017-05-03 04:49:01	00:00:15	

User	Caller ID	Caller Name	Channel Name	Activity	Options
1	1000	John DOE	PJSIP/1000-00000000	00:00:15	
2	1001		PJSIP/1001-00000001	00:00:05	

**Figure 132: Conference**

## Conference Call Operations

### Join a Conference Call

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

### Invite Other Parties to Join Conference

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

- **Invite from Web GUI.**

For each conference room in UCM6200 Web GUI → **Call Features** → **Conference**, there is an icon for option "Invite a participant". Click on it and enter the number of the party you would like to invite. Then click on "Add". A call will be sent to this number to join it into the conference.



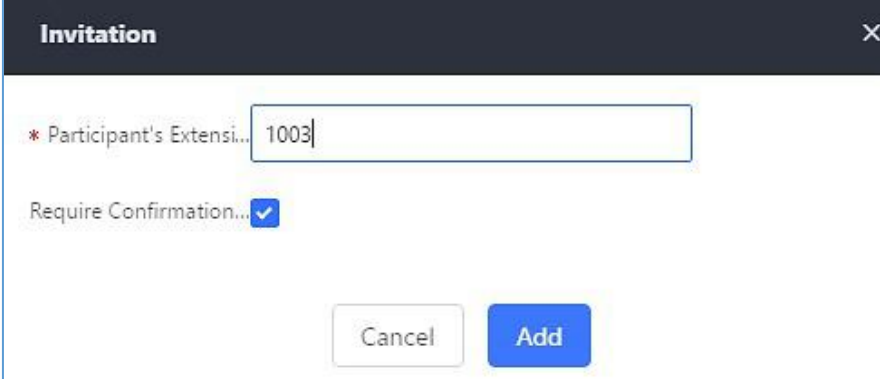


Figure 133: Conference Invitation from Web GUI

- **Invite by dialing 0 or 1 during conference call.**

A conference participant can invite other parties to the conference by dialing from the phone during the conference call. Please make sure option "Enable User Invite" is turned on for the conference room first. Enter 0 or 1 during the conference call. Follow the voice prompt to input the number of the party you would like to invite. A call will be sent to this number to join it into the conference.

**0:** If 0 is entered to invite other party, once the invited party picks up the invitation call, a permission will be asked to "accept" or "reject" the invitation before joining the conference.

**1:** If 1 is entered to invite other party, no permission will be required from the invited party.

---

 **Note:**

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



---

## During The Conference



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

- **Manage the conference call from Web GUI.**

Log in UCM6200 Web GUI during the conference call, the participants in each conference room will be listed.

1. Click on  to kick a participant from the conference.
2. Click on  to mute the participant.



3. Click on  to lock this conference room so that other users cannot join it anymore.
4. Click on  to invite other users into the conference room.

- **Manage the conference call from IVR.**

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

**Table 63: Conference Caller IVR Menu**

Conference Administrator IVR Menu	
1	Mute/unmute yourself.
2	Lock/unlock the conference room.
3	Kick the last joined user from the conference.
4	Decrease the volume of the conference call.
5	Decrease your volume.
6	Increase the volume of the conference call.
7	Increase your volume.
8	More options. <ul style="list-style-type: none"> <li>• 1: List all users currently in the conference call.</li> <li>• 2: Kick all non-Administrator participants from the conference call.</li> <li>• 3: Mute/Unmute all non-Administrator participants from the conference call.</li> <li>• 4: Record the conference call.</li> <li>• 8: Exit the caller menu and return to the conference.</li> </ul>
Conference User IVR Menu	
1	Mute/unmute yourself.
4	Decrease the volume of the conference call.
5	Decrease your volume.
6	Increase the volume of the conference call.
7	Increase your volume.
8	Exit the caller menu and return to the conference.



---

 **Note:**

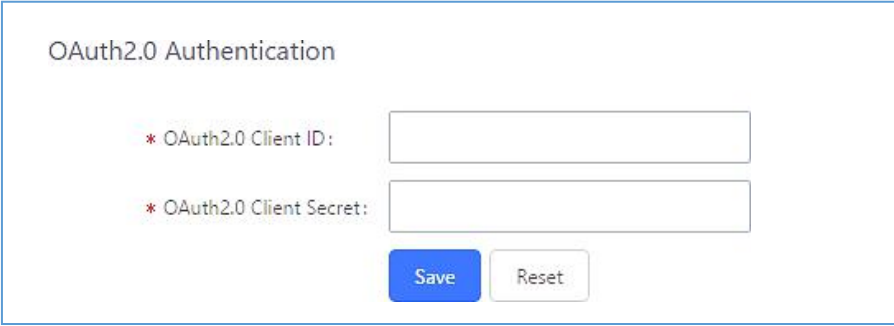
When there is participant in the conference, the conference room configuration cannot be modified.

---

## Google Service Settings Support

UCM6200 now supports Google OAuth 2.0 authentication. This feature is used for supporting UCM6200 conference scheduling system. Once OAuth 2.0 is enabled, UCM6200 conference system can access Google calendar to schedule or update conference.

Google Service Settings can be found under Web GUI→**Call Features**→**Conference**→**Google Service Settings**→**Google Service Settings**.



OAuth2.0 Authentication

\* OAuth2.0 Client ID:

\* OAuth2.0 Client Secret:

Save Reset

Figure 134: Google Service Settings→OAuth2.0 Authentication

If you already have OAuth2.0 project set up on **Google Developers** web page, please use your existing login credential for “OAuth2.0 Client ID” and “OAuth2.0 Client Secret” in the above figure for the UCM6200 to access Google Service.

If you do not have OAuth2.0 project set up yet, please following the steps below to create new project and obtain credentials:

1. Go to Google Developers page <https://console.developers.google.com/start> Create a New Project in Google Developers page.



### New Project

**Project name** ?

Your project ID will be animated-surfer-112001 ? [Edit](#)

[Show advanced options...](#)

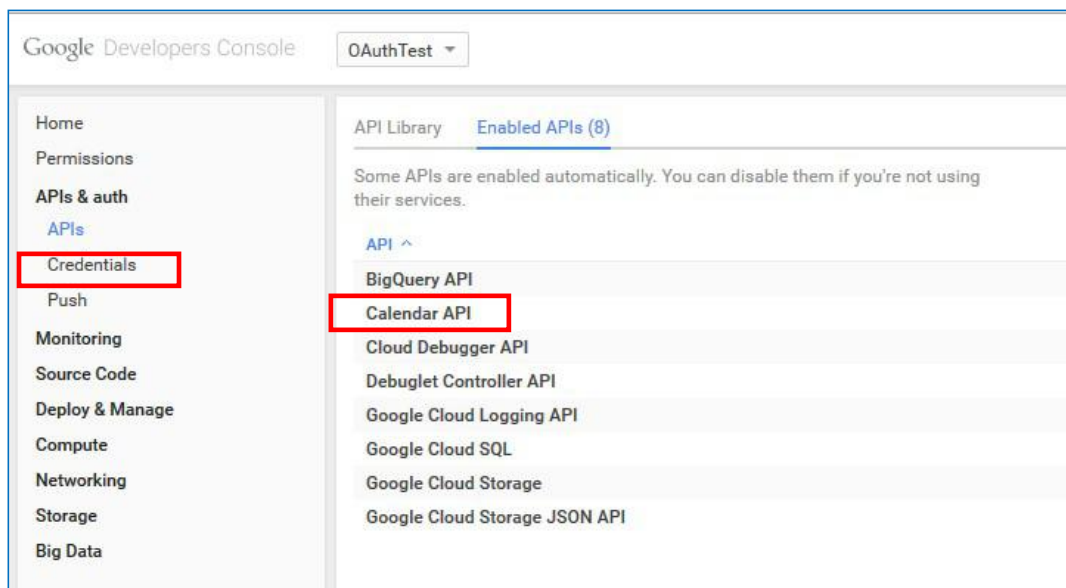
Please email me updates regarding feature announcements, performance suggestions, feedback surveys and special offers.

Yes
  No

I agree that my use of any [services and related APIs](#) is subject to my compliance with the applicable [Terms of Service](#).

**Figure 135: Google Service→New Project**

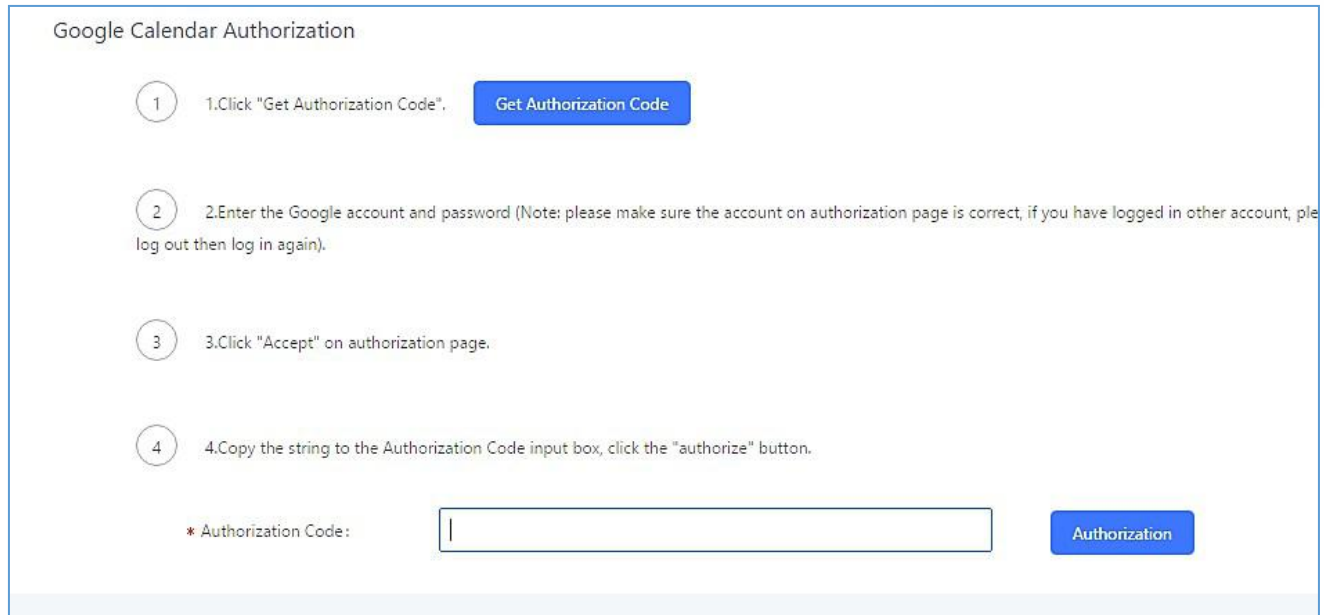
2. Enable Calendar API from API Library.
3. Click “Credentials” on the left drop down menu to create new OAuth2.0 login credentials.



**Figure 136: Google Service→Create New Credential**

4. Use the newly created login credential to fill in “OAuth2.0 Client ID” and “OAuth2.0 Client Secret”.
5. Click “Get Authentication Code” to obtain authentication code from Google Service.





**Figure 137: Google Service→OAuth2.0 Login**

6. Now UCM6200 is connected with Google Service.

You can also configure the Status update, which automatically refresh your Google Calendar with the configured time (m). **Note:** Zero means disable.

## Conference Schedule

Conference Schedule can be found under UCM6200 Web GUI → **Call Features** → **Conference** → **Conference Schedule**. Users can create, edit, view and delete a Conference Schedule.

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

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

**Table 64: Conference Schedule Parameters**

Schedule Options	
<b>Conference Topic</b>	Configure the name of the scheduled conference. Letters, digits, _ and - are allowed.
<b>Conference Room</b>	Select a conference room for this scheduled conference.
<b>Conference Password</b>	Conference login password.
<b>Host Password</b>	Host Password.



<b>Kick Time(m)</b>	<p>Set kick time before conference starts. When kick time is reached, a warning prompt will be played for all attendees in the conference room. After 5 minutes, this conference room will be cleared and locked for the scheduled conference to begin.</p> <p><b>Note:</b> Kick Time cannot be less than 6 minutes in order to clear the conference room.</p>
<b>Wait for Host</b>	<p>If enabled, conference participants will not hear each other until the host joins the conference.</p> <p>Note: If Quiet Mode is enabled, the voice prompt for this option will not be played.</p>
<b>Description</b>	The description of scheduled conference.
<b>Repeat</b>	Repeat interval of scheduled conference. By default, set to single event.
<b>Schedule Time</b>	<p>Configure the beginning date and duration of scheduled conference.</p> <p><b>Note:</b> Please pay attention to avoid time conflict on schedules in the same conference room.</p>
<b>Meeting Duration</b>	<p>Duration of the conference meeting.</p> <p><b>Note:</b> The maximum allowed meeting duration that can be set is 8 hour(s).</p>
<b>Enable Google Calendar</b>	<p>Select this option to synchronize scheduled conference with Google Calendar.</p> <p><b>Note:</b> Google Service Setting OAuth2.0 must be configured on the UCM6200. Please refer to section <b>[Google Service Settings Support]</b>.</p>
<b>Send email notification</b>	Sends Email notification to the extension.
<b>Conference Administrator</b>	<p>Select the administrator of scheduled conference from selected extensions.</p> <p><b>Note:</b> “Public Mode” must be disabled from Conference Room Options tab.</p>
<b>Local Extension</b>	Select available extensions from the list to attend scheduled conference.
<b>Remote Extension</b>	<p>Select available extensions from the remote peer PBX.</p> <p><b>Note:</b> “LDAP Sync” must be enabled on the UCM6200 in order to view remote extensions here.</p>
<b>Special Extension</b>	Add extensions that are not in the list (both local and remote list). If the user wishes to add the special extension, please match the pattern on the outbound route.
<b>Remote Conference</b>	Invite a remote conference.
<b>Conference Room Options</b>	
<b>Password</b>	Configure conference room password. Please note that if “Public Mode” is enabled, this option is automatically disabled.





<b>Admin Password</b>	<p>Configure the password to join as conference administrator.</p> <p>Please note that if “Public Mode” is enabled, this option is automatically disabled.</p>
<b>Enable Caller Menu</b>	<p>If this option is enabled, conference participants will be able to access conference room menu by pressing the * key.</p>
<b>Record Conference</b>	<p>If this option is enabled, conference call will be recorded in .wav format. The recorded file can be found from <b>Conference</b> page.</p>
<b>Quiet Mode</b>	<p>If this option is enabled, the notification tone or voice prompt for joining or leaving the conference will not be played.</p> <p><b>Note:</b> Option “Quiet Mode” and option “Announce Caller” cannot be enabled at the same time.</p>
<b>Wait For Admin</b>	<p>If this option is enabled, the participants in the conference will not be able to hear each other until conference administrator joins the conference.</p> <p><b>Note:</b> If “Quiet Mode” is enabled, voice prompt for this option will not be played.</p>
<b>Enable User Invite</b>	<p>If this option is enabled, the user can:</p> <ul style="list-style-type: none"> <li>• Press ‘0’ to invite others to join the conference with invited party’s permission</li> <li>• Press ‘1’ to invite without invited party’s permission</li> <li>• Press ‘2’ to create a multi-conference room to another conference room</li> <li>• Press ‘3’ to drop all current multi-conference rooms</li> </ul> <p><b>Note:</b> Conference Administrator is always allowed to access this menu.</p>
<b>Announce Callers</b>	<p>If this option is enabled, when a participant joins the conference room, participant’s name will be announced to all members in the conference room.</p> <p><b>Note:</b> Option “Quiet Mode” and option “Announce Caller” cannot be enabled at the same time.</p>
<b>Public Mode</b>	<p>If this option is enabled, no authentication is required for entering the conference room.</p> <p><b>Note:</b> Please be aware of the potential security risks when turning on this option.</p>
<b>Play Hold Music</b>	<p>If this option is enabled, UCM6200 will play Hold Music while there is only one participant in the conference room, or the conference is not yet started.</p>
<b>Skip Authentication When Inviting Users via Trunk from Web GUI</b>	<p>If this option is enabled, the invitation from Web GUI via a trunk with password will not require authentication.</p>



**Note:** Please be aware of the potential security risks when turning on this option.

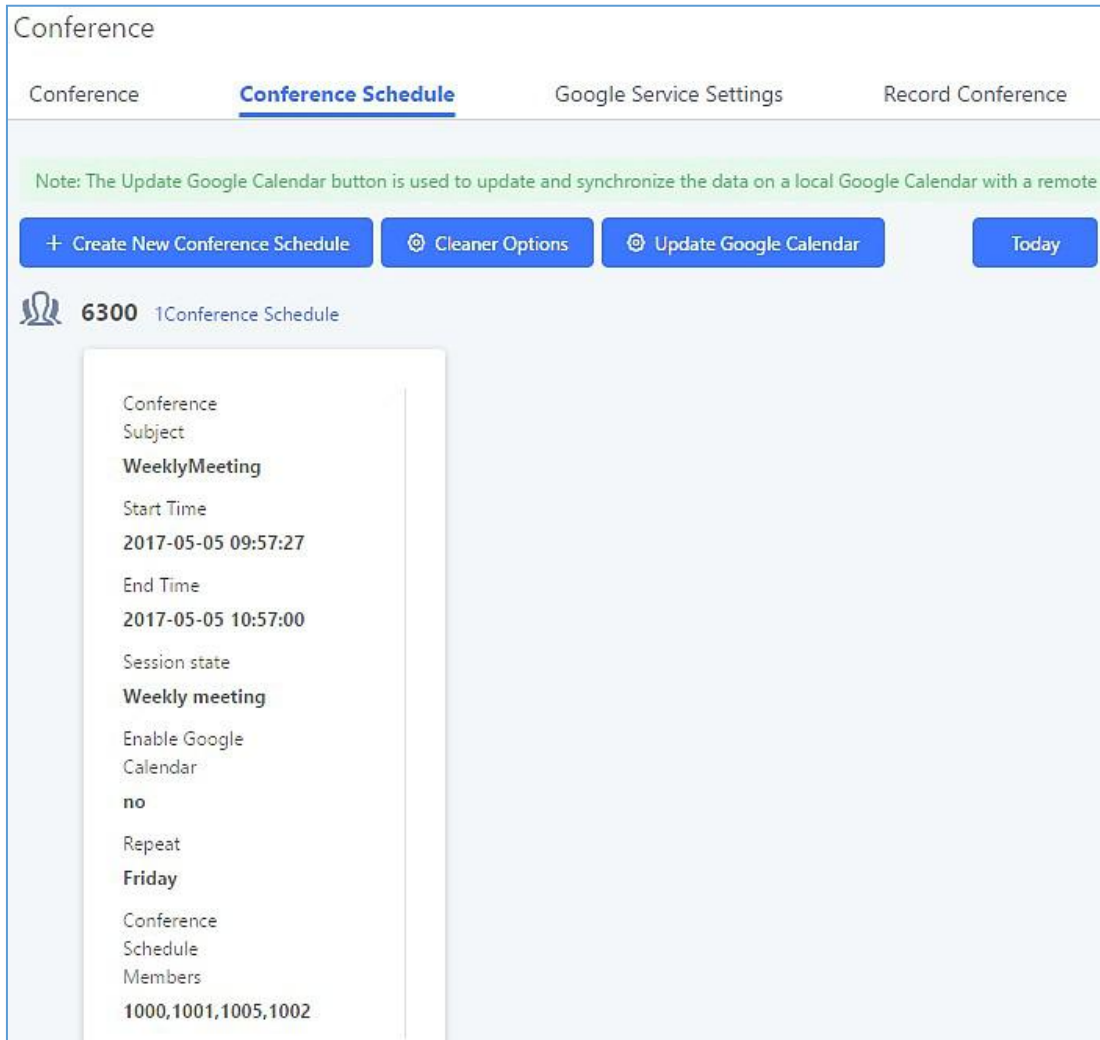
## Cleaner Options

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

## Show/Hide Conference Schedule Table

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





**Figure 138: Conference Schedule**

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

---

**⚠ Note:**

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



## Contact Group

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.

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.

**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"/>

**Figure 139: Contact Group Parameters**

## Contact Group Configurations

- Click on "Create New Contact Group" to add a new Contact Group.
- Click on  to edit the Contact Group.
- Click on  to delete the Contact Group.





**Table 65: Contact Group Parameters**

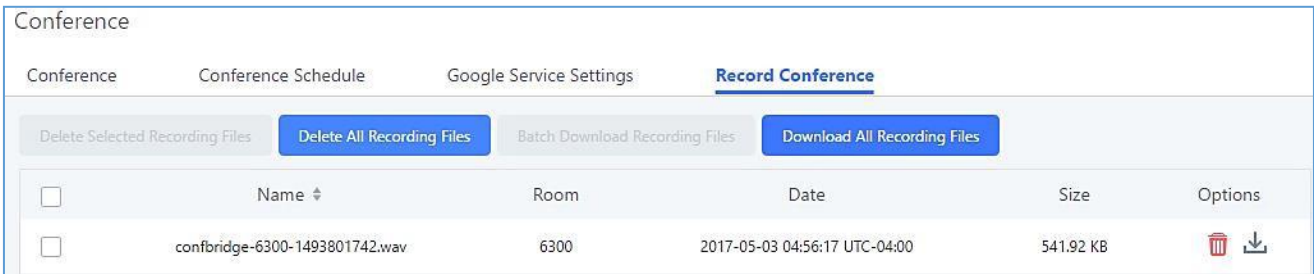
<b>Name</b>	Name associated to the contact group.
<b>Password</b>	Password required to invite the specified contact group to a conference.
<b>Prompt</b>	Audio file that can be uploaded to the group to announce the contact group name such as “Sales” or “Marketing”. The <b>default</b> announcement for each group is “Conference Contact Group 1”, “Conference Contact Group 2”, etc.
<b>Members</b>	Contacts that needs to be added in each group.
<b>Type</b>	Type of the members to be added, it can be either <b>Extensions</b> or a <b>self-defined number</b> .



## Conference Recordings

The UCM6200 allows users to record the conference call and retrieve the recording from Web GUI→**Call Features**→**Conference**→**Record Conference**.

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

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



Conference					
Conference		Conference Schedule	Google Service Settings	<b>Record Conference</b>	
Delete Selected Recording Files		Delete All Recording Files		Batch Download Recording Files	
				Download All Recording Files	
<input type="checkbox"/>	Name	Room	Date	Size	Options
<input type="checkbox"/>	confbridge-6300-1493801742.wav	6300	2017-05-03 04:56:17 UTC-04:00	541.92 KB	 

**Figure 140: Conference Recording**

## Conference Call Statistics

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 141: Conference Call Statistics**

**Figure 142: 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			

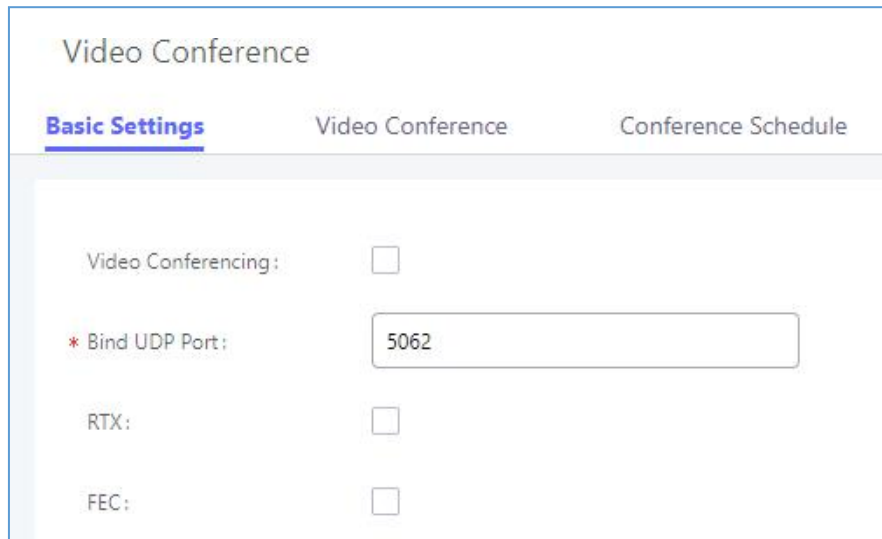
**Figure 143: Conference Report on CSV**



## VIDEO CONFERENCE

With the UCM you can easily create, schedule, manage, and join video conference calls, from your desktop or laptop computer. UCM Video conferencing uses WebRTC technology, so all the participants don't have to download and install any additional software or plugins. UCM Video Conferencing must be enabled by the administrator for the concerned extensions. The video conference configurations can be accessed under Web GUI→**Call Features**→**Video Conference**. In this page, users could enable, set the Basic setting, create, edit, view, manage, delete conference rooms and edit the Conference Schedule.

### Basic Settings





**Figure 144: Video Conference Basic settings**

**Table 66: Video Conference Basic Settings**

Basic Settings	
<b>Video Conferencing</b>	This option should be enabled in order to activate the Video Conference feature.
<b>Bind UDP Port</b>	Configure the UDP port number for MCM. The standard UDP port for MCM is 5062.
<b>RTX</b>	If enabled, the RTX Packet Loss Retransmission will be activated. The default setting is "No".
<b>FEC</b>	If enabled, the Forward Error Correction (FEC) will be activated. The default setting is "No".



## Video Conference Room Configurations

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

**Table 67: Video Conference room Configuration Parameters**

<b>Extension</b>	Configure the conference number for the users to dial into the conference. <b>Note:</b> Up to 64 characters.
<b>Password</b>	When configured, the users who would like to join the conference call must enter this password before accessing the conference room. <b>Note:</b> <ul style="list-style-type: none"> <li>• Only digits are allowed.</li> <li>• The password has to be at least 4 characters. All repetitive and sequential digits (e.g., 0000, 1111, 1234 and 2345) or common digits (e.g., 111222 and 321321) are not allowed.</li> </ul>

## Conference Schedule

Conference Schedule can be found under UCM **Web GUI** → **Call Features** → **Video Conference** → **Conference Schedule**. Users can create, edit, view and delete a Conference Schedule.

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

**Table 68: Video Conference Schedule Parameters**

Schedule Options	
<b>Conference Subject</b>	Configure the name of the scheduled conference. Letters, digits, _ and - are allowed.
<b>Conference Room</b>	Select a conference room for this scheduled conference.
<b>Conference Password</b>	Configure conference room password. Please note that if "Public Mode" is enabled, this option is automatically disabled.
<b>Kick Time(m)</b>	Configure the time before the scheduled conference. When this time is reached, a warning prompt will be played, and all attendees currently in the scheduled conference room will be kicked after 5 mins. The conference room will be locked until the scheduled conference begins. Default value is 10 min.
<b>Schedule Time</b>	Configure the beginning date and duration of scheduled conference. <b>Note:</b> Please pay attention to avoid time conflict on schedules in the same conference room.

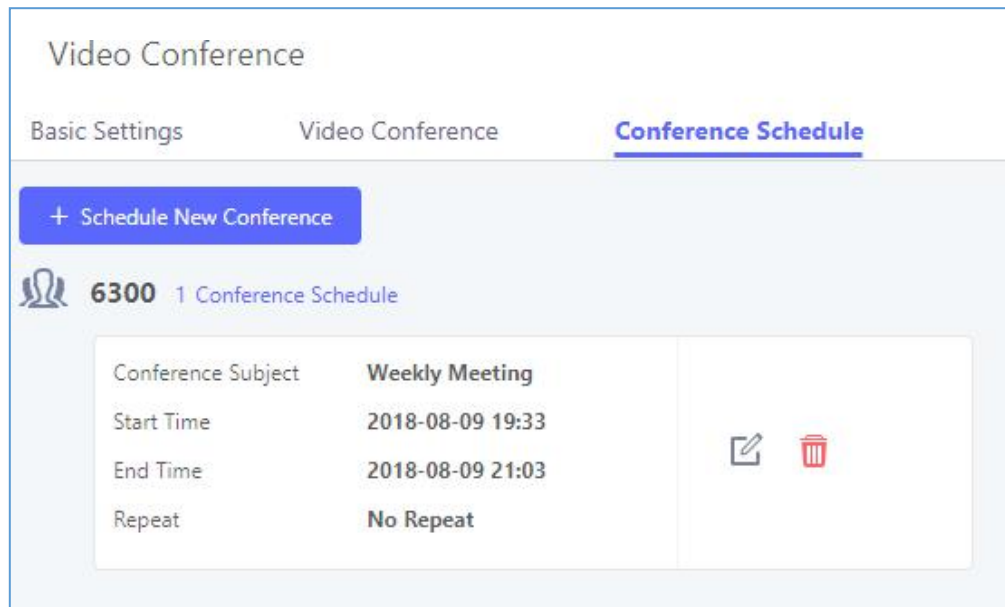




<b>Duration</b>	Configure the time duration of the scheduled conference.
<b>Period</b>	Configure the period of scheduled conference.
<b>Host</b>	Set the admin of this scheduled conference from the following list of members.
<b>Members</b>	Select available extensions from the list to attend scheduled conference.
<b>Special Extension</b>	Add extensions that are not in the list (both local and remote list). If the user wishes to add the special extension, please match the pattern on the outbound route.
<b>Shareable Link</b>	Assign the video conference a public IP address and port to allow anyone with the configured link to participate in the video conference.
<b>Description</b>	Set a description of scheduled conference.

Once created, the Web GUI will display scheduled conference in Conference Schedule.

Please see figure below:



**Figure 145: Video Conference Schedule**

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

---

 **Notes:**

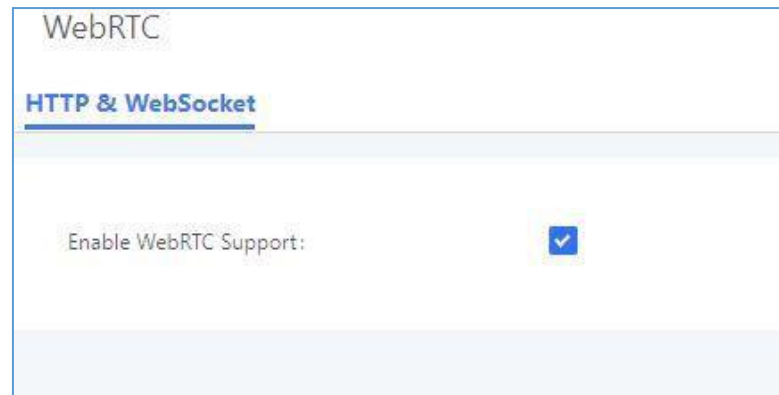
- Video conferencing can be resource-intensive and may cause performance issues with the UCM when used.
  - To ensure the best experience, please use Google Chrome (v67 or higher) or Mozilla Firefox (v60).
- 



## Wave WebRTC Video Calling & Conferencing

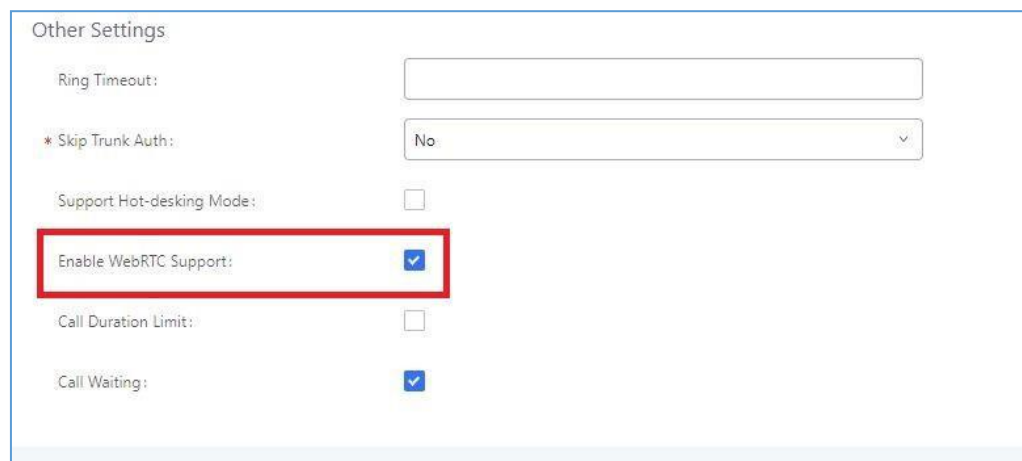
Web audio and video calls and conferencing can now be achieved through the UCM's new WebRTC page. for more details about this feature. To get started with this new feature, please make sure to:

1. Navigate to **Value-Added Features** → **WebRTC** and enable WebRTC support.



**Figure 146: Enabling WebRTC Feature**

2. Enable the WebRTC on the extensions that would use this feature under **Extension / Trunk** → **Extensions** by editing the concerned extensions.

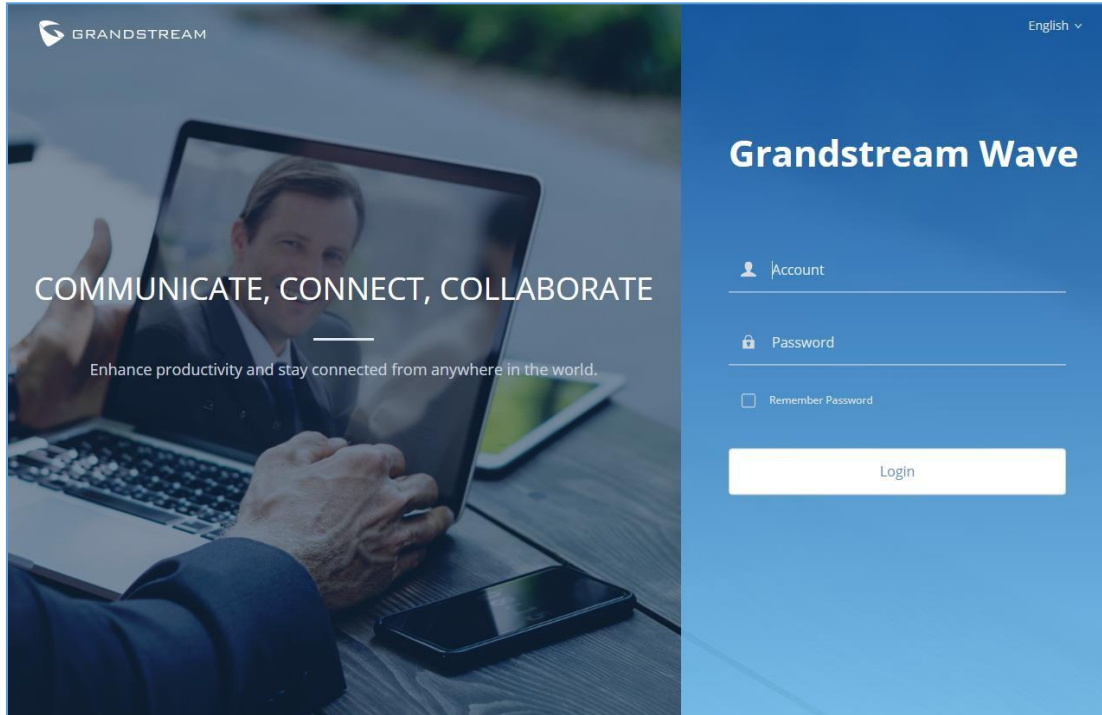


**Figure 147: Enabling WebRTC on Extensions**

The UCM offers the possibility to login to an extension via Grandstream Wave Portal using user portal password in addition to SIP registration password, where it offers a sleek interface to host conferences, receive email reminders for scheduled conferences, manage contacts, initiate calls, call transfer, chat functionality and more.

Access the page by adding “/gswave” after the UCM's server address and port. (e.g. <https://my.ucm.com:8089/gswave>).





**Figure 148: Grandstream Wave Interface**

**Note:** Starting with 1.0.19.27, the registration limit was increased to 300 for UCM62xx. The limit is the same regardless of whether the user is making voice calls or video calls

For more details about the WebRTC feature, please refer to the following guide:

[http://www.grandstream.com/sites/default/files/Resources/wave\\_webrtc\\_guide.pdf](http://www.grandstream.com/sites/default/files/Resources/wave_webrtc_guide.pdf)



## IPVIDEOTALK MEETINGS

UCM extensions can now dial into IPVT (IPVideoTalk) 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.

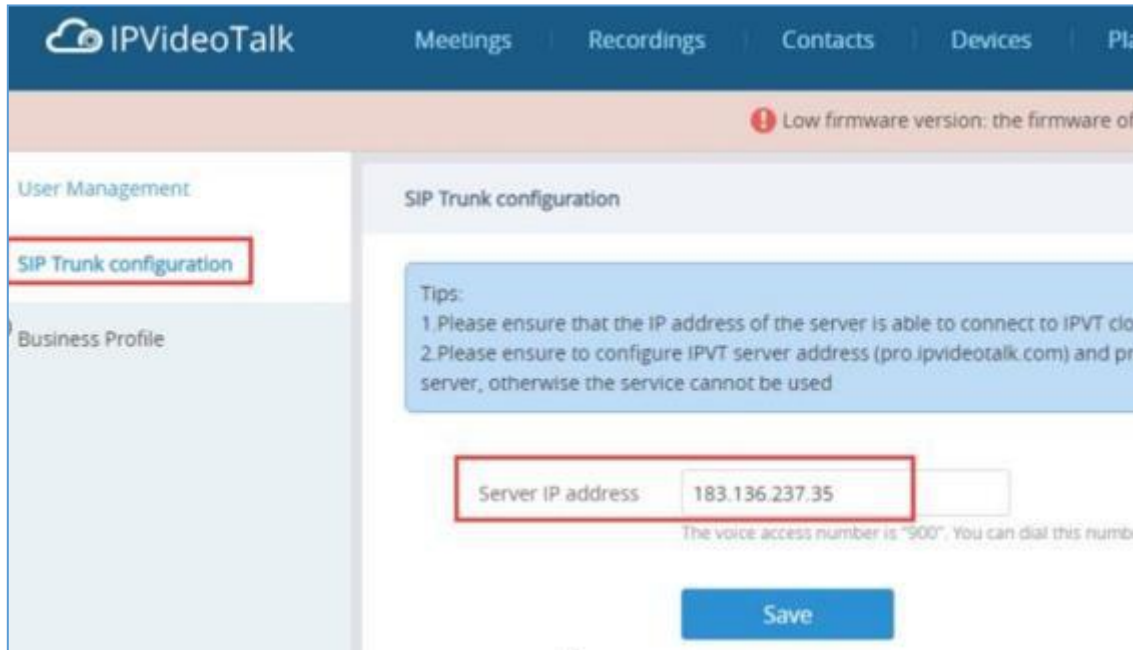


Figure 149: IPVT SIP Trunk page

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

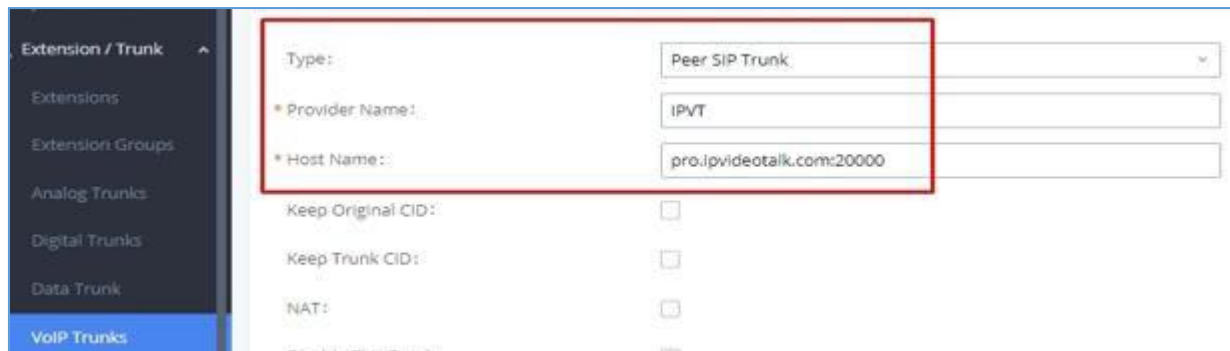
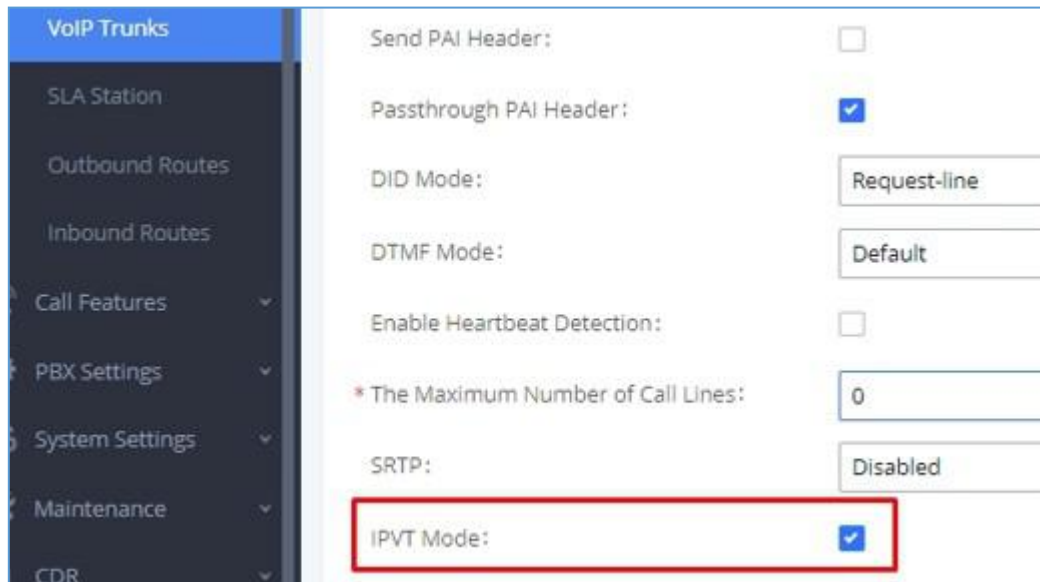


Figure 150 - Peer Trunk to IPVT

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

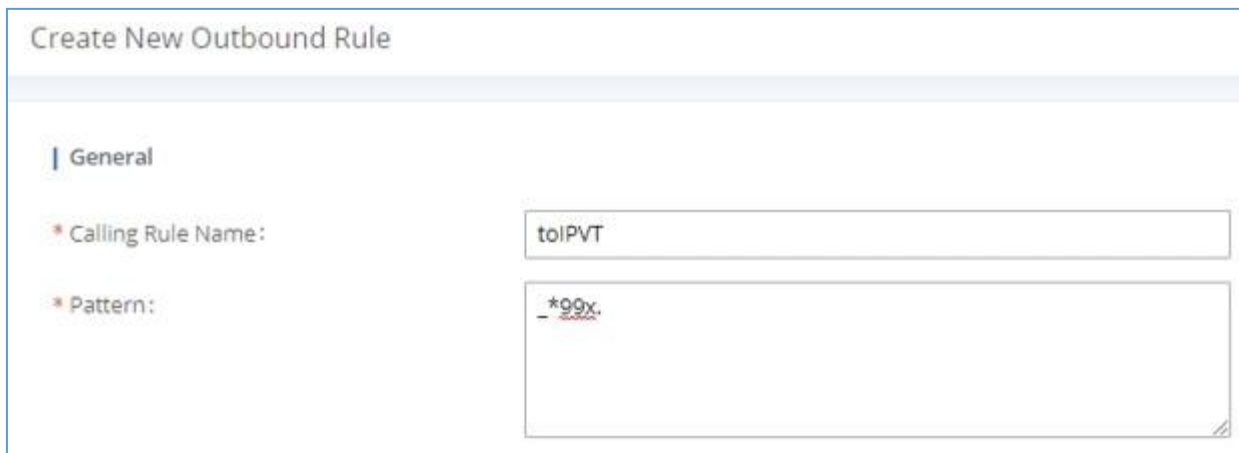


Next, edit the newly created trunk and click on the *Advanced Settings* tab. Make sure the *IPVT Mode* option is checked. Otherwise, you may experience audio issues when dialing into IPVT.



**Figure 151 - 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 152 - 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 153 - 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.



## IVR

### Configure IVR

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

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

Create New IVR

**Basic Settings**      Key Pressing Events

---

\* Name:

\* Extension:

Dial Trunk:

Dial Other Extensions:  All  Extension  Conference  Video Conference  
 Call Queue  Ring Group  Paging/Intercom Groups  
 Voicemail Groups  Fax Extension  Dial By Name


\* IVR Black/Whitelist:

Replace Display Name:

Return to IVR Menu:

Alert-info:

\* Prompt:

[Add Prompt](#) 

\* Digit Timeout:

\* Response Timeout:

\* Response Timeout Prompt:

\* Invalid Input Prompt:

\* Response Timeout Prompt Repeats:

\* Invalid Input Prompt Repeats:

Language:

**Figure 154: Create New IVR**



**Table 69: IVR Configuration Parameters**

Basic Settings	
<b>Name</b>	Configure the name of the IVR. Letters, digits, _ and - are allowed.
<b>Extension</b>	Enter the extension number for users to access the IVR.
<b>DID Destination</b>	<p>This option shows up only when "By DID" is selected. This controls the destination that can be reached by the external caller via the inbound route. The DID destination are:</p> <ul style="list-style-type: none"> <li>• Extension</li> <li>• Conference</li> <li>• Call Queue</li> <li>• Ring Group</li> <li>• Paging/Intercom Groups</li> <li>• Voicemail Groups</li> <li>• Fax Extension</li> <li>• Dial By Name</li> <li>• All</li> </ul>
<b>Dial Trunk</b>	If enabled, all callers to the IVR is allowed to use trunk. The permission must be configured for the users to use the trunk first. The default setting is "No".
<b>Permission</b>	Assign permission level for outbound calls if "Dial Trunk" is enabled. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". If the user tries to dial outbound calls after dialing into the IVR, the UCM6200 will compared the IVR's permission level with the outbound route's privilege level. If the IVR's permission level is higher than (or equal to) the outbound route's privilege level, the call will be allowed to go through.
<b>Replace Caller ID</b>	If enabled, the UCM will replace the caller display name with the IVR name the caller know whether the call is incoming from a direct extension or an IVR.
<b>Alert-Info</b>	When present in an INVITE request, the alert-Info header field specifies and alternative ring tone to the UAS.
<b>Welcome Prompt</b>	<p>Select an audio file to play as the welcome prompt for the IVR. Click on "Prompt" to add additional audio file under Web GUI→<b>PBX Settings</b>→<b>Voice Prompt</b>→<b>Custom Prompt</b>.</p> <p><b>Note:</b> Users can upload more than one welcome prompt. Up to 5.</p>
<b>Digit Timeout</b>	Configure the timeout between digit entries. After the user enters a digit, the user needs to enter the next digit within the timeout. If no digit is detected within the timeout, the UCM6200 will consider the entries complete. The default timeout is 3 seconds.





<b>Response Timeout</b>	After playing the prompts in the IVR, the UCM6200 will wait for the DTMF entry within the timeout (in seconds). If no DTMF entry is detected within the timeout, a timeout prompt will be played. The default setting is 10 seconds.
<b>Response Timeout Prompt</b>	Select the prompt message to be played when timeout occurs.
<b>Invalid Prompt</b>	Select the prompt message to be played when an invalid extension is pressed.
<b>Response Timeout Repeat Loops</b>	Configure the number of times to repeat the prompt if no DTMF input is detected. When the loop ends, it will go to the timeout destination if configured, or hang up. The default setting is 3.
<b>Invalid Repeat Loops</b>	Configure the number of times to repeat the prompt if the DTMF input is invalid. When the loop ends, it will go to the invalid destination if configured, or hang up. The default setting is 3.
<b>Language</b>	Select the voice prompt language to be used for this IVR. The default setting is "Default" which is the selected voice prompt language under Web GUI→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> . The dropdown list shows all the current available voice prompt languages on the UCM6200. To add more languages in the list, please download voice prompt package by selecting "Check Prompt List" under Web GUI→ <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Language Settings</b> .

### Key Pressing Events

<p><b>Key Press Event:</b></p> <p>Press 0</p> <p>Press 1</p> <p>Press 2</p> <p>Press 3</p> <p>Press 4</p> <p>Press 5</p> <p>Press 6</p> <p>Press 7</p> <p>Press 8</p> <p>Press 9</p> <p>Press *</p> <p>Timeout</p> <p>Invalid</p>	<p>Select the event for each key pressing for 0-9, *, Timeout and Invalid. The event options are:</p> <ul style="list-style-type: none"> <li>• Extension</li> <li>• Voicemail</li> <li>• Conference Rooms</li> <li>• Voicemail Group</li> <li>• IVR</li> <li>• Ring Group</li> <li>• Queues</li> <li>• Page Group</li> <li>• Fax</li> <li>• Custom Prompt</li> <li>• Hangup</li> <li>• DISA</li> <li>• Dial By Name</li> <li>• External Number</li> <li>• Callback</li> <li>• Announcement</li> </ul>
---	---



Edit IVR: OfficeOpen

Basic Settings **Key Pressing Events**

Press 0:	Extension ▾	2000 ▾
Press 1:	IVR ▾	Sales ▾
Press 2:	IVR ▾	Support ▾
Press 3:	Select an Op... ▾	
Press 4:	Select an Op... ▾	
Press 5:	Select an Op... ▾	
Press 6:	Select an Op... ▾	
Press 7:	Select an Op... ▾	
Press 8:	Select an Op... ▾	
Press 9:	Select an Op... ▾	
Press *:	Select an Op... ▾	
Timeout:	Custom Pro... ▾	goodbye ▾
Invalid:	Custom Pro... ▾	goodbye ▾

**Figure 155: Key Pressing Events**

## Black/White List in IVR

In some scenarios, the IPPBX administrator needs to restrict the extensions that can be reached from IVR. For example, the company CEO and directors prefer only receiving calls transferred by the secretary, some special extensions are used on IP surveillance end points which shouldn't be reached from external calls via IVR for privacy reason. UCM has now added blacklist and whitelist in IVR settings for users to manage this.

**Note:** up to 500 extensions are allowed on the black/white list.

To use this feature, log in UCM Web GUI and navigate to **Call Features**→**IVR**→Create/Edit IVR: IVR Black/White List.

- If the user selects “Blacklist Enable” and adds extension in the list, the extensions in the list will not be allowed to be reached via IVR.
- If the user selects “Whitelist Enable” and adds extension in the list, only the extensions in the list can be allowed to be reached via IVR.



Create New IVR

\* Name:

\* Extension:

Dial Trunk:

\* Permission:

Dial Other Extensions:  Extension  Conference  Call Queue  
 Ring Group  Paging/Intercom Groups  
 Voicemail Groups  Fax Extension  
 Dial By Name  
 All

\* IVR Black/Whitelist:

Internal Black/Whitelist:

Available	Selected
<input type="checkbox"/> 1 <input type="checkbox"/> 1005 "Marcel LAST"	<input type="checkbox"/> 3 <input type="checkbox"/> 1001 <input type="checkbox"/> 1002 <input type="checkbox"/> 1000 "John DOE"

External Blacklist/Whitelist:

Replace Display Name:

Alert-info:

\* Prompt:  Prompt

\* Digit Timeout:

\* Response Timeout:

\* Response Timeout Promp...:  Prompt

\* Invalid Prompt:  Prompt

\* Response Timeout Repeat...:

\* Invalid Repeat Loops:

Language:

Figure 156: Black/White List

## Create Custom Prompt

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



Alert-info :	None	▼
* Prompt :	welcome	▼

Prompt

**Figure 157: Click on Prompt to Create IVR Prompt**

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



## VOICE PROMPT

The UCM6200 supports multiple languages in Web GUI as well as system voice prompt. The following languages are currently supported in system voice prompt:

English (United States), Arabic, Chinese, Dutch, English (United Kingdom), French, German, Greek, Hebrew, Italian, Polish, Portuguese, Russian, Spanish, Catalan, Swedish and Turkish.

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

Language settings for voice prompt can be accessed under Web GUI→**PBX Settings**→**Voice Prompt**→**Language Settings**. Additionally, UCM6200 allows to customize specific prompt instead of full language package, and it provides ability to upload greeting files for extensions.

### Language Settings

#### Download and Install Voice Prompt Package

To download and install voice prompt package in different languages from UCM6200 Web GUI, click on "Add Voice Prompt Package" button.

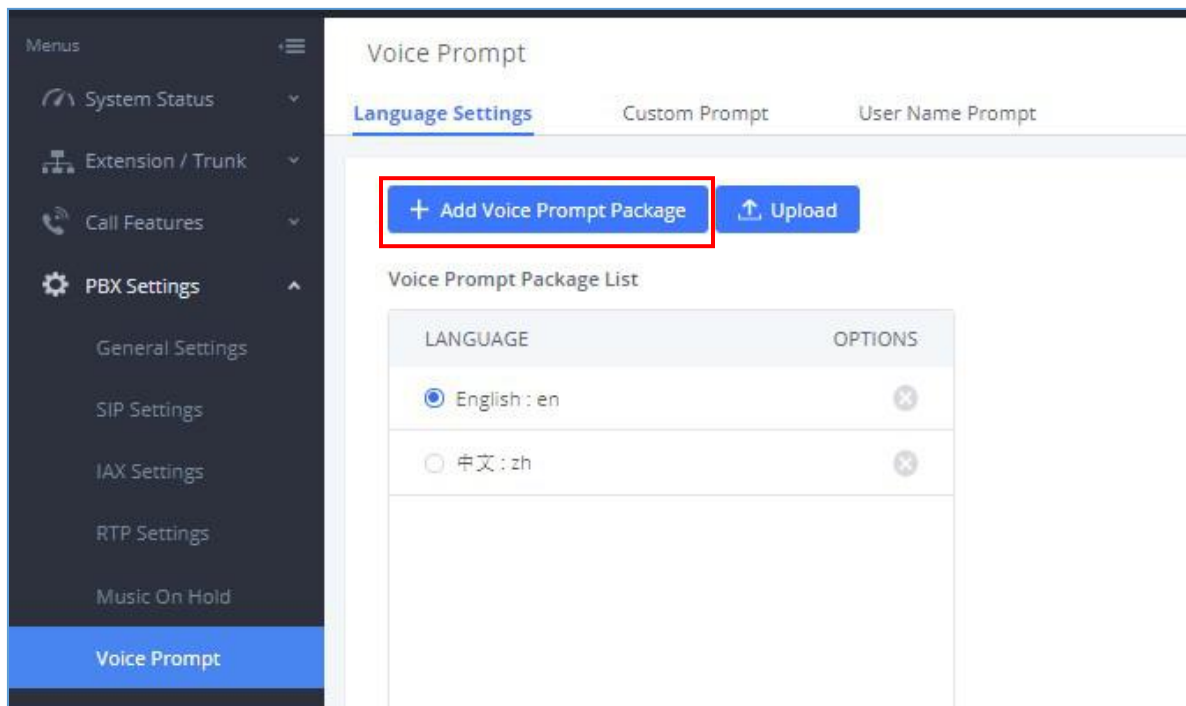


Figure 158: Language Settings for Voice Prompt




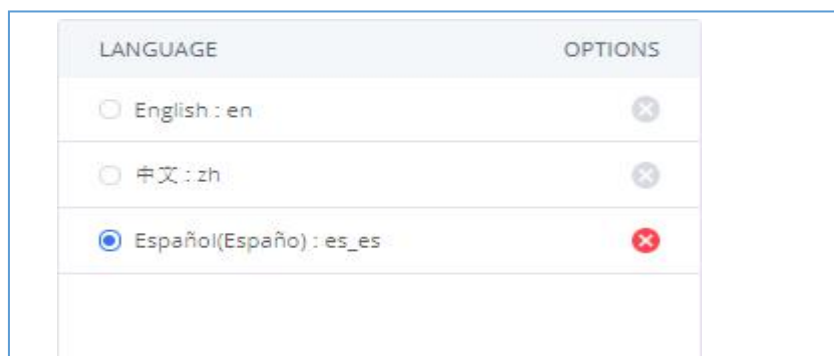
A new dialog window of voice prompt package list will be displayed. Users can see the version number (latest version available V.S. current installed version), package size and options to upgrade or download the language.



VOICE PROMPT PACKAGE LIST	VERSION (REMOTE / LOCAL)	SIZE	OPTIONS
British English	1.9/-	4.2M	↓
Deutsch	1.8/-	4.2M	↓
English	1.11/1.11	6.0M	⊕
Español	1.10/-	4.4M	↓
Español(Català)	1.8/-	3.1M	↓
Español(Español)	1.8/-	4.2M	↓
Ελληνικά	1.8/-	4.4M	↓
Français	1.8/-	4.1M	↓
Italiano	1.8/-	4.0M	↓

**Figure 159: Voice Prompt Package List**

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



LANGUAGE	OPTIONS
<input type="radio"/> English : en	⊗
<input type="radio"/> 中文 : zh	⊗
<input checked="" type="radio"/> Español(Español) : es_es	⊗

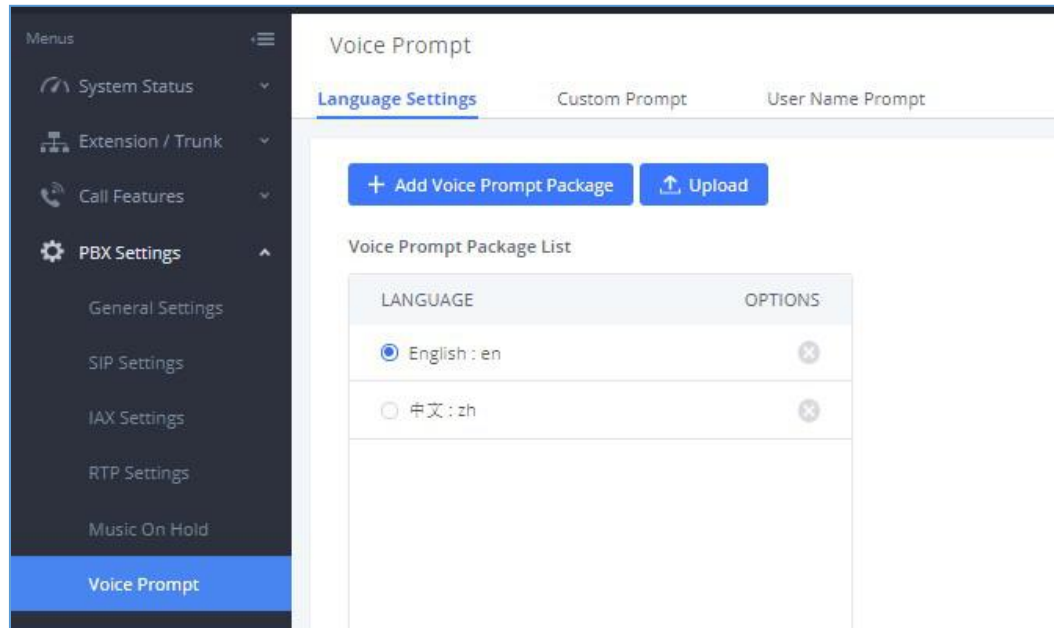
**Figure 160: New Voice Prompt Language Added**

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



## Upload Language Package

On the UCM6200, if the user needs to replace some specific customized prompt, the user can upload a single specific customized prompt from Web GUI→**PBX Settings**→**Voice Prompt**→**Language Settings** instead of the entire language pack.



**Figure 161: Upload Voice Prompts Package**

The package file should follow below requirements:

- Each file uploaded must be under 50MB.
- Package structure:

```
[Package]
├ [voice prompt dir]
│ └ [... dir]
│   └ [... files]
└ info.txt (containing the language name for display, in UTF8)
```

- Language dir name format:
- Custom dir name format: language\_XXX;  
 For example: If there is a Chinese custom directory named zh\_XXX, the custom voice prompt in zh\_XXX would be used first, then the Chinese voice prompt zh, then use the default language prompt (en); If not named the format as above, then the custom prompt will be used first, then use the default language prompt (en).

For more details, please refer to:

[http://www.grandstream.com/sites/default/files/Resources/ucm\\_voiceprompt\\_customization\\_guide.zip](http://www.grandstream.com/sites/default/files/Resources/ucm_voiceprompt_customization_guide.zip)



## Custom Prompt

### Record New Custom Prompt

In the UCM6200 Web GUI→**PBX Settings**→**Voice Prompt**→**Custom Prompt** page, click on "Record New IVR Prompt" and follow the steps below to record new IVR prompt.

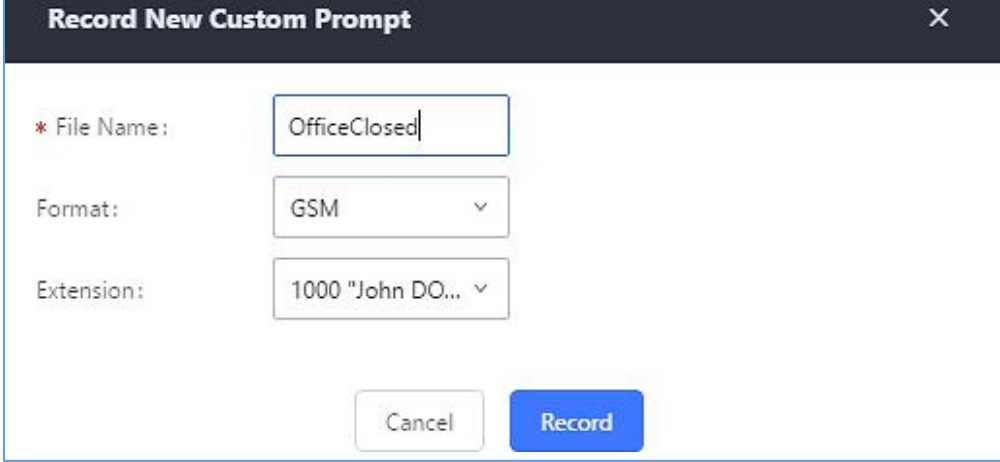


Figure 162: Record New IVR Prompt

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

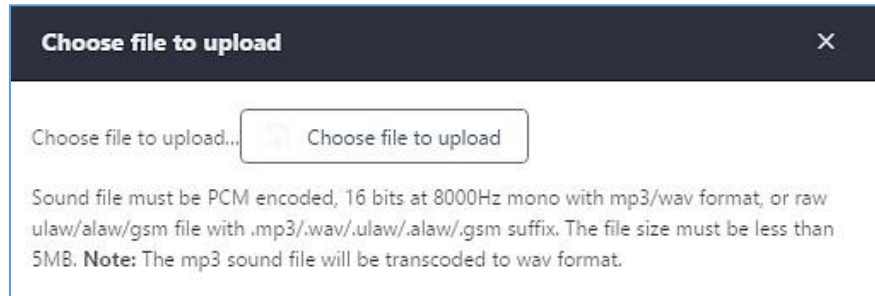
### Upload Custom Prompt

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

- PCM encoded.
- 16 bits.
- 8000Hz mono.
- In .mp3 or .wav format; or raw/ulaw/alaw/gsm file with .ulaw or .alaw suffix.
- File size under 5M.
- Filename should not exceed 100 characters.






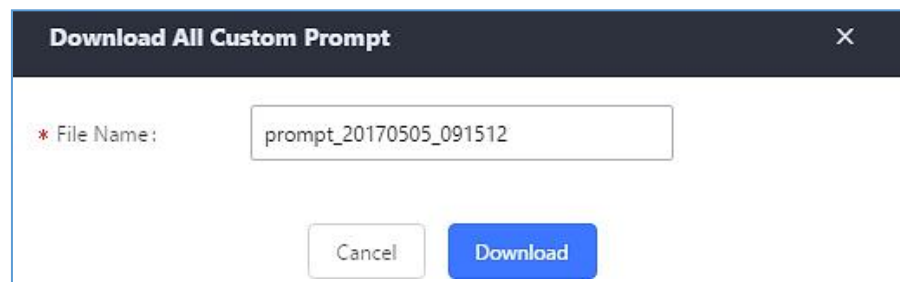


**Figure 163: Upload IVR Prompt**

Click on “choose file to upload” to select audio file from local PC and to start uploading. Once uploaded, the file will appear in the IVR Prompt web page.

### Download All Custom Prompt

On the UCM6200, the users can download all custom prompts from UCM Web GUI to local PC. To download all custom prompt, log in UCM Web GUI and navigate to **PBX Settings**→**Voice Prompt**→**Custom Prompt** and click on . The following window will pop up in order to set a name for the downloaded file.



**Figure 164: Download All Custom Prompt**

**Note:** The downloaded file will have a .tar extension.






### Username Prompt Customization

There are two ways to customize/set new username prompt:

#### Upload Username Prompt File from Web GUI

1. First, Users should have a pre-recorded file respecting the following format:
  - PCM encoded / 16 bits / 8000Hz mono.
  - In “.GSM” or “.WAV” format.
  - File size under 5M.



- Filename must be set as the extension number with 18 characters max. For example, the recorded file name 1000.wav will be used for extension 1000.
2. Go under web GUI **PBX Settings** → **Voice Prompt** → **Username Prompt** and click on  button.
  3. Select the recorded file to upload it and press Save and Apply Settings.
    - Click on  to record again the username prompt.
    - Click on  to play recorded username prompt.
    - Select username prompts and press  to delete specific file or select multiple files for deletion using the button .

### Record Username via Voicemail Menu

The second option to record username is using voicemail menu, please follow below steps:

- Dial \*98 to access the voicemail
- After entering the desired extension and voicemail password, dial “0” to enter the recordings menu and then “3” to record a name.

Another option is that each user can record their own name by following below steps:

- The user dials \*97 to access his/her voicemail
- After entering the voicemail password, the user can press “0” to enter the recordings menu and then “3” to record his name.



## VOICEMAIL

### Configure Voicemail

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

* Max Greeting (s):	<input type="text" value="60"/>
Dial "0" for Operator:	<input type="checkbox"/>
Operator Extension:	<input type="text" value="None"/>
* Max Messages Per Folder:	<input type="text" value="50"/>
Max Message Time:	<input type="text" value="15 minutes"/>
Min Effective Message Time:	<input type="text" value="3 seconds"/>
Announce Message Caller- ID:	<input type="checkbox"/>
Announce Message Duration:	<input type="checkbox"/>
Play Envelope:	<input checked="" type="checkbox"/>
Play from Last:	<input type="checkbox"/>
Allow User Review:	<input type="checkbox"/>
Voicemail Remote Access:	<input type="checkbox"/>
Forward Voicemail to Peered UCMs:	<input type="checkbox"/>
Voicemail Password:	<input type="text"/>

Figure 165: Voicemail Settings



**Table 70: Voicemail Settings**

<b>Max Greeting</b>	Configure the maximum number of seconds for the voicemail greeting. The default setting is 60 seconds.
<b>Dial '0' For Operator</b>	If enabled, the caller can press 0 to exit the voicemail application and connect to the configured operator's extension.
<b>Operator Extension</b>	Select the operator extension, which will be dialed when users press 0 to exit voicemail application. The operator extension can also be used in IVR.
<b>Max Messages Per Folder</b>	Configure the maximum number of messages per folder in users' voicemail. The valid range 10 to 1000. The default setting is 50.
<b>Max Message Time</b>	<p>Select the maximum duration of the voicemail message. The message will not be recorded if the duration exceeds the max message time. The default setting is 15 minutes. The available options are:</p> <ul style="list-style-type: none"> <li>• 1 minute</li> <li>• 2 minutes</li> <li>• 5 minutes</li> <li>• 15 minutes</li> <li>• 30 minutes</li> <li>• Unlimited</li> </ul>
<b>Min Effective Message Time</b>	<p>Configure the minimum duration (in seconds) of a voicemail message. Messages will be automatically deleted if the duration is shorter than the Min Message Time. The default setting is 3 seconds. The available options are:</p> <ul style="list-style-type: none"> <li>• No minimum</li> <li>• 1 second</li> <li>• 2 seconds</li> <li>• 3 seconds</li> <li>• 4 seconds</li> <li>• 5 seconds</li> </ul> <p><b>Note:</b> Silence and noise duration are not counted in message time.</p>
<b>Announce Message Caller-ID</b>	If enabled, the caller ID of the user who has left the message will be announced at the beginning of the voicemail message. The default setting is "No".
<b>Announce Message Duration</b>	If enabled, the message duration will be announced at the beginning of the voicemail message. The default setting is "No".
<b>Play Envelope</b>	If enabled, a brief introduction (received time, received from, and etc.) of each message will be played when accessed from the voicemail application. The default setting is "Yes".
<b>Play from Last</b>	If enabled, UCM will play from the voice message left most recently; if disabled, UCM will play from the earliest left voice message
<b>Allow User Review</b>	If enabled, users can review the message following the IVR before sending.



<b>Voicemail Remote Access</b>	<p>If enabled, external callers routed by DID and reaching VM will be prompted by the UCM with 2 options:</p> <ul style="list-style-type: none"> <li>• <b>Press 1 to leave a message.</b> To leave a message for the extension reached by DID.</li> <li>• <b>Press 2 to access voicemail management system.</b> This will allow caller to access any extension VM after entering extension number and its VM password.</li> </ul> <p><b>Note:</b> This option applies to inbound call routed by DID only. The default setting is “Disabled”.</p>
<b>Forward Voicemail to Peered UCMs</b>	<p>Enables the forwarding of voicemail to remote extensions on peered SIP trunks. The default setting is “Disabled”.</p>
<b>Voicemail Password</b>	<p>Configures the default voicemail password that will be used when an extension is reset.</p>

**Note:** Resetting an extension will reset Voicemail Password, Send Voicemail to Email, and Keep Voicemail after Emailing values to default. Previous custom voicemail prompts and messages will be deleted.

## Access Voicemail

If the voicemail is enabled for UCM6200 extensions, the users can dial the voicemail access number (by default \*97) to access their extension’s voicemail. The users will be prompted to enter the voicemail password and then can enter digits from the phone keypad to navigate in the IVR menu for different options.

Otherwise the user can dial the voicemail access code (by default \*98) followed by the extension number and password in order to access to that specific extension’s voicemail.

**Table 71: Voicemail IVR Menu**

Main Menu	Sub Menu 1	Sub Menu 2
<b>1 – New messages</b>	3 - Advanced options	1 - Send a reply 2 - Call the person who sent this message 3 - Hear the message envelop 4 - Leave a message * - Return to the main menu
	5 - Repeat the current message	
	7 - Delete this message	
	8 - Forward the message to another user	
	9 - Save	



<b>2 – Change folders</b>	* - Help	
	# - Exit	
	0 - New messages	
	1 - Old messages	
	2 - Work messages	
	3 - Family messages	
	4 - Friend messages	
	# - Cancel	
<b>3 – Advanced options</b>	1 - Send a reply	
	2 - Call the person who sent this message	
	3 - Hear the message envelop	
	4 - Leave a message	
	* - Return to the main menu	
<b>0 – Mailbox options</b>	1 - Record your unavailable message	1 - Accept this recording
		2 - Listen to it
		3 - Re-record your message
	2 - Record your busy message	1 - Accept this recording
		2 - Listen to it
		3 - Re-record your message
	3 - Record your name	1 - Accept this recording
		2 - Listen to it
		3 - Re-record your message
	4 - Record temporary greeting	1 - Accept this recording
		2 - Listen to it
		3 - Re-record your message
	5 - Change your password	
* - Return to the main menu		

## Leaving Voicemail

If an extension has voicemail enabled under basic settings “**Extension/Trunk → Extensions → Basic Settings**” and after a ring timeout or user not available, the caller will be automatically redirected to the voicemail in order to leave a message on which case they can press # in order to submit the message.

In case if the caller is calling from an internal extension, they will be directly forwarded to the extension’s voicemail box.

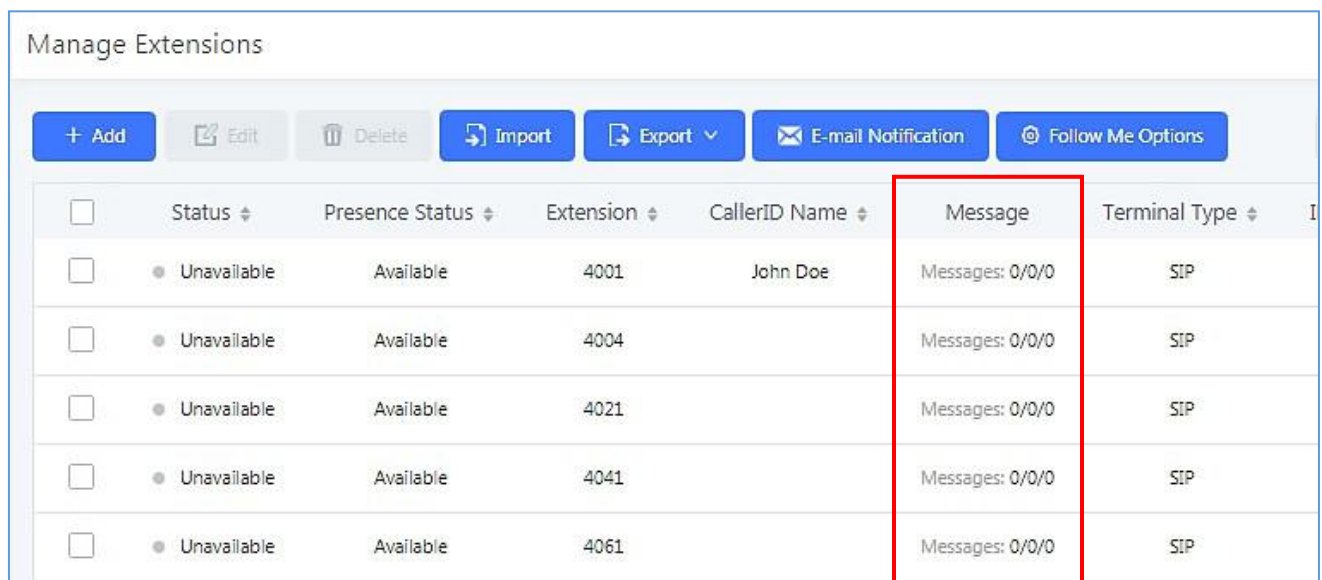


But if the caller is calling from outside the system and the incoming call is routed by DID to the destination extension, then the caller will be prompted with the choice to either press 1 to access voicemail management or press 2 to leave a message for the called extension. This feature could be useful for remote voicemail administration.

## Extension Voicemail Count

The UCM62xx provides an easy way to check the number of voicemail messages for each extension directly from UCM web GUI → Extension/Trunk → Extensions overview page.

Voicemail count (“Message” column) is displayed in the format **Urgent / Unread / Read**.



<input type="checkbox"/>	Status	Presence Status	Extension	CallerID Name	Message	Terminal Type
<input type="checkbox"/>	Unavailable	Available	4001	John Doe	Messages: 0/0/0	SIP
<input type="checkbox"/>	Unavailable	Available	4004		Messages: 0/0/0	SIP
<input type="checkbox"/>	Unavailable	Available	4021		Messages: 0/0/0	SIP
<input type="checkbox"/>	Unavailable	Available	4041		Messages: 0/0/0	SIP
<input type="checkbox"/>	Unavailable	Available	4061		Messages: 0/0/0	SIP

**Figure 166: Voicemail Count**

## Voicemail Email Settings

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

**Table 72: Voicemail Email Settings**

<b>Attach Recordings to E-Mail</b>	If enabled, voicemails will be sent to user's Email address. The default setting is "Yes".
<b>Keep Recordings</b>	If enabled, voicemail will be stored in the UCM6200 after the email is sent. The default setting is "Yes".

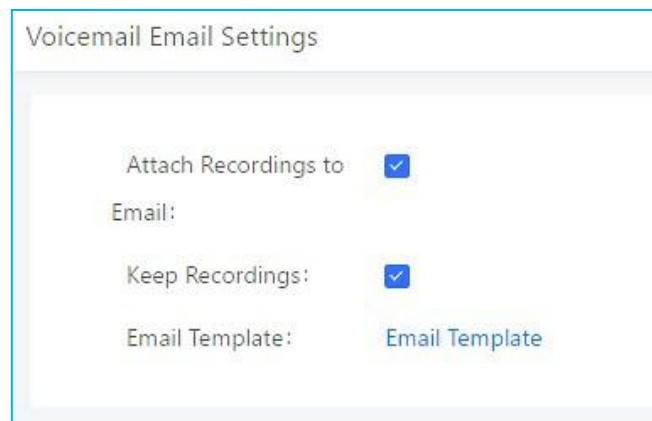


### Template for Voicemail Emails

Fill in the "Subject:" and "Message:" content, to be used in the Email when sending to the user.

The template variables are:

- \t: TAB
- \${VM\_NAME}: Recipient's first name and last name
- \${VM\_DUR}: The duration of the voicemail message
- \${VM\_MAILBOX}: The recipient's extension
- \${VM\_CALLERID}: The caller ID of the person who has left the message
- \${VM\_MSGNUM}: The number of messages in the mailbox
- \${VM\_DATE}: The date and time when the message is left



**Figure 167: Voicemail Email Settings**

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

### Configure Voicemail Group

The UCM6200 supports voicemail group and all the extensions added in the group will receive the voicemail to the group extension. The voicemail group can be configured under Web GUI → **Call Features** → **Voicemail** → **Voicemail Group**. Click on "Create New Voicemail Group" to configure the group.





### Create New Voicemail Group

\* Extension:

\* Name:

Voicemail Password:

Email Address:

Email:

2 Available Mailboxes

Search

1002

1005 "Marcel LAST"

2 Voicemail Group Mailboxes

Search

1000 "John DOE"

1001 None

<

>

**Figure 168: Voicemail Group**

**Table 73: Voicemail Group Settings**

<b>Voicemail Group Extension</b>	Enter the Voicemail Group Extension. The voicemail messages left to this extension will be forwarded to all the voicemail group members.
<b>Name</b>	Configure the Name to identify the voicemail group. Letters, digits, _ and - are allowed.
<b>Voicemail Password</b>	The Voicemail password for the user to check Voicemail messages.
<b>Email Address</b>	The Email address of current user.
<b>Member</b>	Select available extensions from the left list and add them to the right list. The extensions need to have voicemail enabled to be listed in available mailboxes list. <b>Note:</b> Members selected cannot exceed 27 extensions.

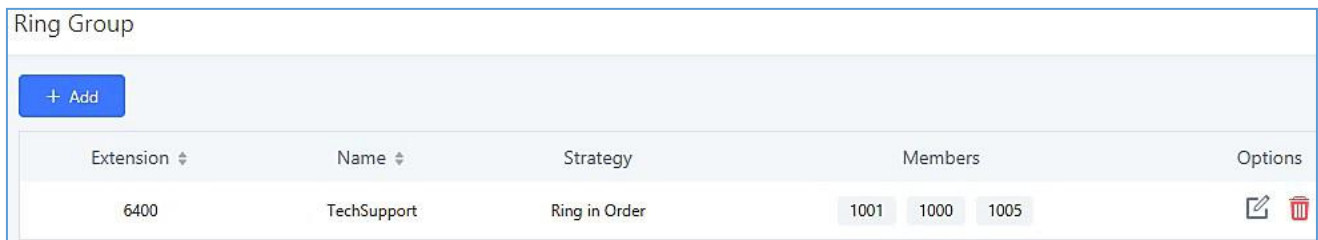


## RING GROUP

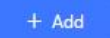


The UCM6200 supports ring group feature with different ring strategies applied to the ring group members. This section describes the ring group configuration on the UCM6200.

### Configure Ring Group





Ring group settings can be accessed via Web GUI → **Call Features** → **Ring Group**.



**Figure 169: Ring Group**

- Click on  to add ring group.
- Click on  to edit the ring group. The following table shows the ring group configuration parameters.
- Click on  to delete the ring group.

**Table 74: Ring Group Parameters**

<b>Ring Group Name</b>	Configure ring group name to identify the ring group. Letters, digits, _ and – are allowed.
<b>Extension</b>	Configure the ring group extension.
<b>Members</b>	Select available users from the left side to the ring group member list on the right side. Click on   to arrange the order.
<b>LDAP Phonebook</b>	Select available remote users from the left side to the ring group member list on the right side. Click on   to arrange the order. Note: LDAP Sync must be enabled first.
<b>Ring Strategy</b>	Select the ring strategy. The default setting is “Ring in order”. <ul style="list-style-type: none"> <li>• <b>Ring simultaneously.</b> Ring all the members at the same time when there is incoming call to the ring group extension. If any of the member answers the call, it will stop ringing.</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>Ring in order.</b> Ring the members with the order configured in ring group list. If the first member does not answer the call, it will stop ringing the first member and start ringing the second member.</li> </ul>
<b>Music On Hold</b>	Select the “Music On Hold” Class of this Ring Group, “Music On Hold” can be managed from the “Music On Hold” panel on the left.
<b>Custom Prompt</b>	<p>This option is to set a custom prompt for a ring group to announce to caller. Click on ‘Prompt’, it will direct the users to upload the customized voice prompts.</p> <p><b>Note:</b> Users can also refer to the page <b>PBX Settings→Voice Prompt→Custom Prompt</b>, where they could record new prompt or upload prompt files.</p>
<b>Ring Timeout on Each Member</b>	<p>Configure the number of seconds to ring each member. If set to 0, it will keep ringing. The default setting is 30 seconds.</p> <p><b>Note:</b> The actual ring timeout might be overridden by users if the phone has ring timeout settings as well.</p>
<b>Auto Record</b>	If enabled, calls on this ring group will be automatically recorded. The default setting is No. The recording files can be accessed from Web GUI→CDR→Recording Files.
<b>Endpoint Call Forwarding Support</b>	<p>This allows the UCM to work with endpoint-configured call forwarding settings to redirect calls to ring group. For example, if a member wants to receive calls to the ring group on his mobile phone, he would have to set his endpoint’s call forwarding settings to his mobile number. By default, it is disabled.</p> <p>However, this feature has the following limitations:</p> <ul style="list-style-type: none"> <li>• This feature will work only when call forwarding is configured on endpoints, not on the UCM.</li> <li>• If the forwarded call goes through an analog trunk, and polarity reversal is disabled, the other ring group members will no longer receive the call after it is forwarded.</li> <li>• If the forwarded call goes through a VoIP trunk, and the outbound route for it is PIN-protected and requires authentication, the other ring group members will no longer receive the call after it is forwarded.</li> <li>• If the forwarded call hits voicemail, the other ring group members will no longer receive the call.</li> </ul>



<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with the Ring Group name the caller know whether the call is incoming from a direct extension or a Ring Group.
<b>Skip Busy Agent</b>	If enabled calls to busy agents will be skipped and sent to the following available ones. Default is enabled.
<b>Enable Destination</b>	If enabled, users could select extension, voicemail, ring group, IVR, call queue, voicemail group as the destination if the call to the ring group has no answer. <b>Note:</b> Voicemail Password and Email address (limited by 128 characters) are required if voicemail is selected as the destination. Voicemail system will mention if a voicemail is from a ring group.

Save

Edit Ring Group: 6400

---

\* Ring Group Name:

\* Extension:

Members:

1 Available Extensions

Search

1002

Selected Extensions

3 Selected Extensions

Search

1001

1000 "John DOE@one"

1005 "Marcel LAST"

<
>

LDAP Phonebook:

1 Available LDAP

Search

1002(ou=GSEMEA,dc=pbx,dc=com)

None

Selected LDAP

0 Selected LDAP

Search

None

<
>

**Ring Group Options**

Ring Strategy:

Music On Hold:

Custom Prompt:  [Prompt](#)

\* Ring Timeout on E...:


Auto Record:

**Figure 170: Ring Group Configuration**



## Remote Extension in Ring Group

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

1. Creating SIP Peer Trunk between both UCM6200\_A and UCM6200\_B. **SIP Trunk** can be found under Web GUI→**Extension/Trunk**→**VoIP Trunks**. Also, please configure their Inbound/Outbound routes accordingly.
2. Click edit button in the menu , and check if **Sync LDAP Enable** is selected, this option will allow UCM6200\_A update remote LDAP server automatically from peer UCM6200\_B. In addition, **Sync LDAP Password** must match for UCM6200\_A and UCM6200\_B to synchronize LDAP contact automatically. Port number can be anything between 0~65535, and use the outbound rule created in step 1 for the **LDAP Outbound Rule** option.



### Edit SIP Trunk: test

Basic Settings      **Advanced Settings**

---

Codec Preference:

Available	Selected
<input type="checkbox"/> 11 items <input type="checkbox"/> G.722 <input type="checkbox"/> AAL2-G.726-32 <input type="checkbox"/> ADPCM <input type="checkbox"/> G.723	<input type="checkbox"/> 6 items <input type="checkbox"/> PCMU <input type="checkbox"/> PCMA <input type="checkbox"/> GSM <input type="checkbox"/> G.726

Send PPI Header:

Send PAI Header:

Passthrough PAI Header:

DID Mode: Request-line

DTMF Mode: Info

Enable Heartbeat Detection:

\* The Maximum Number of Call Lines: 0

Fax Mode: None

SRTP: Disabled

IPVT Mode:

Sync LDAP Enable:

\* Sync LDAP Password: .....

\* Sync LDAP Port: 4444

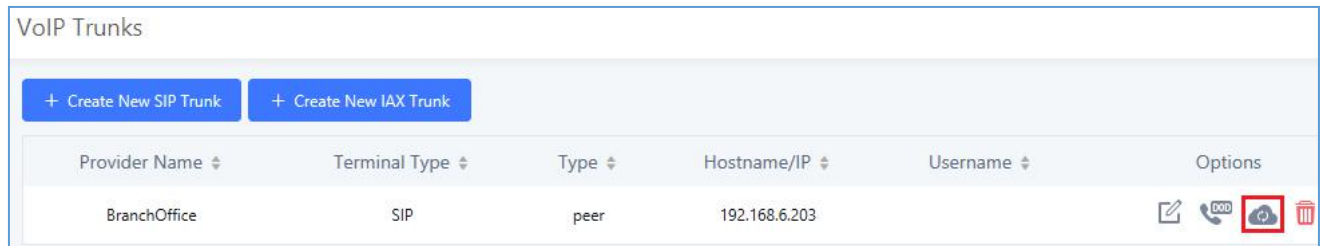
LDAP Outbound Rule: Self-defined

\* LDAP Dialed Prefix: 9

**Figure 171: Sync LDAP Server option**



- In case if LDAP server does not synchronize automatically, user can manually synchronize LDAP server. Under **VoIP Trunks** page, click sync button shown in the following figure to manually synchronize LDAP contacts from peer UCM6200.



**Figure 172: Manually Sync LDAP Server**

- Under **Ring Groups** setting page, click "Add". **Ring Groups** can be found under Web GUI→**Call Features**→**Ring Groups**.
- If LDAP server is synchronized correctly, **Available LDAP Numbers** box will display available remote extensions that can be included in the current ring group. Please also make sure the extensions in the peer UCM6200 can be included into that UCM6200's LDAP contact.



Create New Ring Group
Save

\* Ring Group Name:

\* Extension:

Members:

Available Extensions

106

Search

1000

1001

1002

1003

0 Selected Extensions

Search

None

LDAP Phonebook:

Available LDAP

15

Search

5000(ou=ucm6510,dc=pbx,dc=com)

5001(ou=ucm6510,dc=pbx,dc=com)

5002(ou=ucm6510,dc=pbx,dc=com)

5003(ou=ucm6510,dc=pbx,dc=com)

0 Selected LDAP

Search

None

**Ring Group Options**

Ring Strategy:

Music On Hold:

Custom Prompt:  Prompt

\* Ring Timeout on:

**Figure 173: Ring Group Remote Extension**







## PAGING AND INTERCOM GROUP

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

### Configure Paging/Intercom Group

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

### Configure Multicast Paging

Create New Paging/Intercom Groups

<b>*Name:</b>	<input type="text" value="Name"/>
<b>*Type:</b>	<input style="border-bottom: 1px solid #ccc;" type="text" value="Multicast Paging"/> ▾
<b>*Extension:</b>	<input type="text" value="Extension"/>
<b>*Maximum Call Duration.:</b>	<input type="text" value="0"/>
Custom Prompt:	<input style="border-bottom: 1px solid #ccc;" type="text" value="None"/> ▾ <span style="color: #0070c0; font-weight: bold;">Prompt</span>
<b>*Multicast IP Address:</b>	<input type="text" value="Configure multicast IP address"/>
<b>*Port:</b>	<input type="text" value="Configure the port number"/>

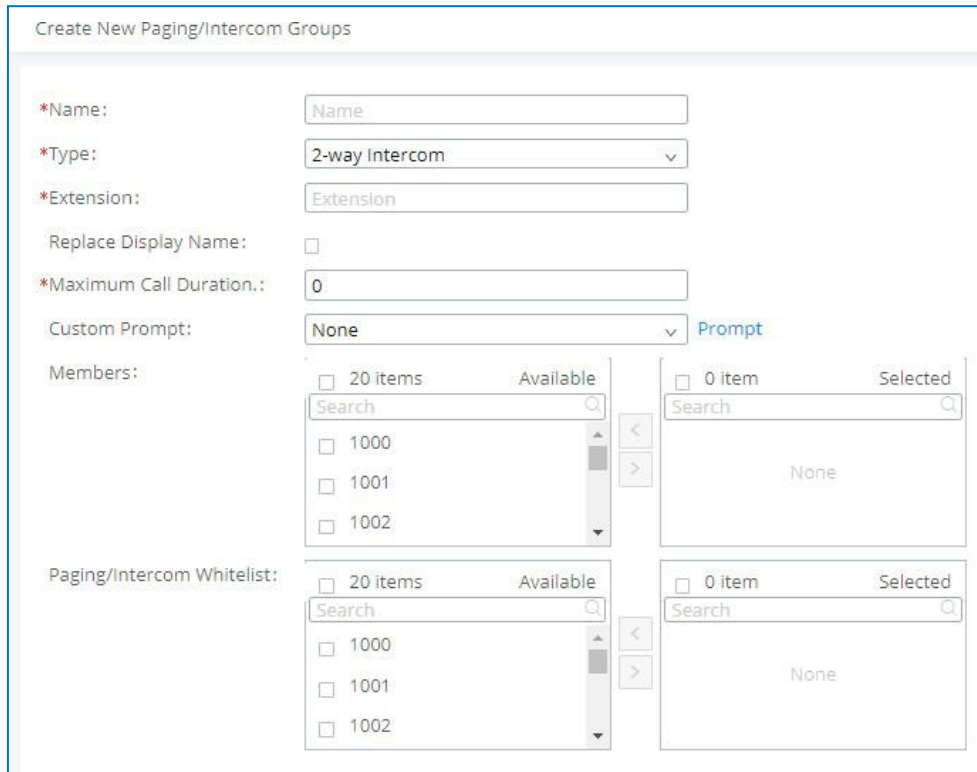
**Figure 174: Multicast Paging**



**Table 75: Multicast Paging Configuration Parameters**

<b>Name</b>	Configure paging/intercom group name.
<b>Type</b>	Select “Multicast Paging”.
<b>Extension</b>	Configure the paging/intercom group extension.
<b>Multicast IP Address</b>	The allowed multicast IP address range is 224.0.1.0 - 238.255.255.255. <b>Note:</b> This field appears only when “Type” is set to “Multicast Paging”.
<b>Maximum Call Duration</b>	Specify the maximum call duration in seconds. The default value 0 means no limit.
<b>Custom Prompt</b>	This option is to set a custom prompt for a paging/intercom group to announce to caller. Click on ‘Prompt’, it will direct the users to upload the customized voice prompts. <b>Note:</b> Users can also refer to the page PBX Settings→Voice Prompt→Custom Prompt, where they could record new prompt or upload prompt files.
<b>Multicast IP Address</b>	Select which extensions are allowed to use the paging/intercom feature for this paging group.
<b>Port</b>	Specify port for multicast paging. <b>Note:</b> This field appears only when “Type” is set to “Multicast Paging”.

## Configure 2-way Intercom



The screenshot shows the configuration interface for a 2-way Intercom group. The fields are as follows:

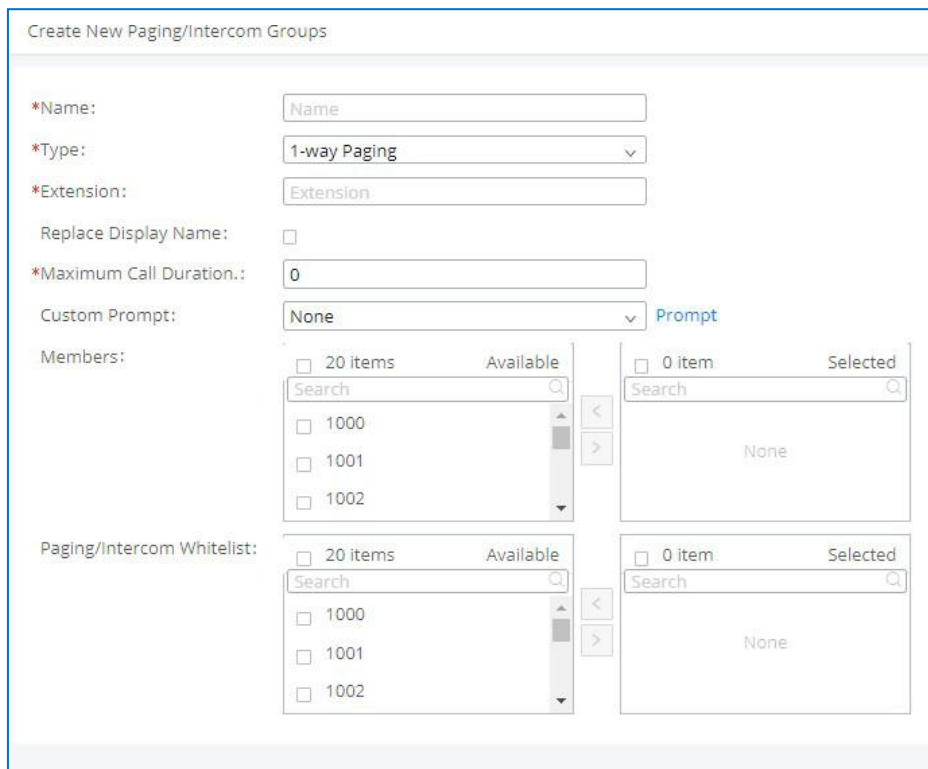
- \*Name:** Text input field with the value "Name".
- \*Type:** Dropdown menu set to "2-way Intercom".
- \*Extension:** Text input field with the value "Extension".
- Replace Display Name:** Unchecked checkbox.
- \*Maximum Call Duration.:** Text input field with the value "0".
- Custom Prompt:** Dropdown menu set to "None" with a "Prompt" button next to it.
- Members:** A list management interface showing 20 items available (1000, 1001, 1002) and 0 items selected.
- Paging/Intercom Whitelist:** A list management interface showing 20 items available (1000, 1001, 1002) and 0 items selected.

**Figure 175: 2-way Intercom**


**Table 76: 2-way Intercom Configuration Parameters**

<b>Name</b>	Configure paging/intercom group name.
<b>Type</b>	Select "2-way Intercom".
<b>Extension</b>	Configure the paging/intercom group extension.
<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with Paging/Intercom name.
<b>Maximum Call Duration</b>	Specify the maximum call duration in seconds. The default value 0 means no limit.
<b>Custom Prompt</b>	This option is to set a custom prompt for a paging/intercom group to announce to caller. Click on 'Prompt', it will direct the users to upload the customized voice prompts. <b>Note:</b> Users can also refer to the page <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Custom Prompt</b> , where they could record new prompt or upload prompt files.
<b>Members</b>	Select available users from the left side to the paging/intercom group member list on the right.
<b>Paging/Intercom Whitelist</b>	Select which extensions are allowed to use the paging/intercom feature for this paging group.

### Configure 1-way Paging



The screenshot shows the 'Create New Paging/Intercom Groups' configuration interface. It includes the following fields and sections:

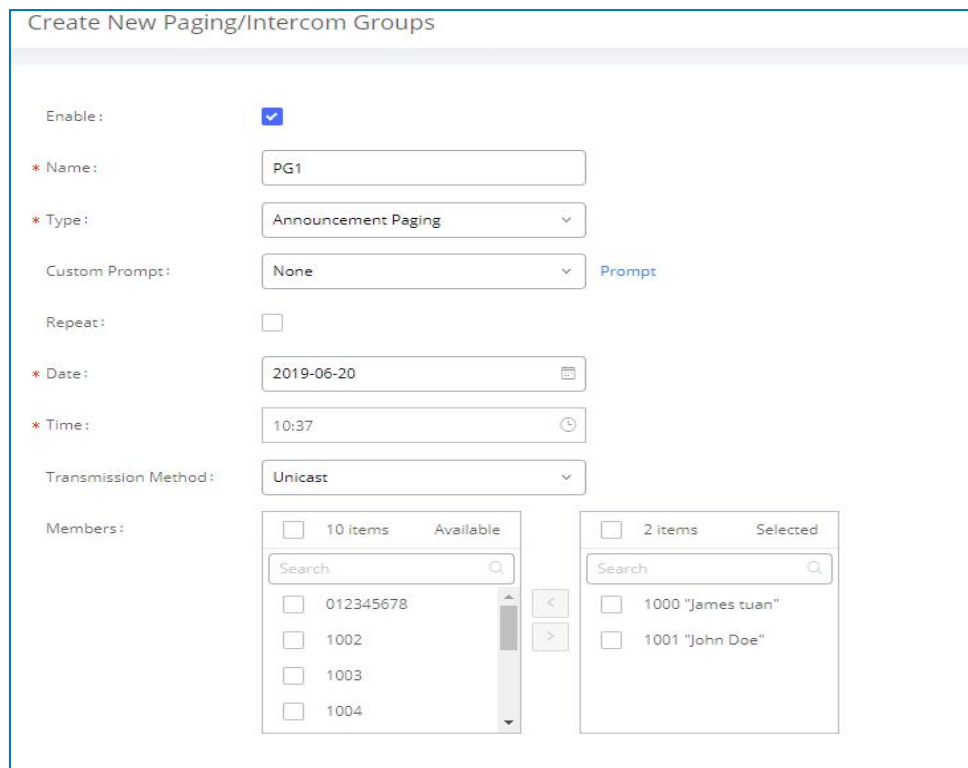
- Name:** A text input field labeled 'Name'.
- Type:** A dropdown menu set to '1-way Paging'.
- Extension:** A text input field labeled 'Extension'.
- Replace Display Name:** An unchecked checkbox.
- Maximum Call Duration.:** A text input field containing '0'.
- Custom Prompt:** A dropdown menu set to 'None' with a 'Prompt' button next to it.
- Members:** A list management interface with an 'Available' list (20 items, showing 1000, 1001, 1002) and a 'Selected' list (0 item, showing 'None').
- Paging/Intercom Whitelist:** A similar list management interface with an 'Available' list (20 items, showing 1000, 1001, 1002) and a 'Selected' list (0 item, showing 'None').

**Figure 176: 1-way Paging**


**Table 77: 1-way Paging Configuration Parameters**

<b>Name</b>	Configure paging/intercom group name.
<b>Type</b>	Select "1-way Paging".
<b>Extension</b>	Configure the paging/intercom group extension.
<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with Paging/Intercom name.
<b>Maximum Call Duration</b>	Specify the maximum call duration in seconds. The default value 0 means no limit.
<b>Custom Prompt</b>	This option is to set a custom prompt for a paging/intercom group to announce to caller. Click on 'Prompt', it will direct the users to upload the customized voice prompts. <b>Note:</b> Users can also refer to the page <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Custom Prompt</b> , where they could record new prompt or upload prompt files.
<b>Members</b>	Select available users from the left side to the paging/intercom group member list on the right.
<b>Paging/Intercom Whitelist</b>	Select which extensions are allowed to use the paging/intercom feature for this paging group.

## Configure Announcement Paging



Create New Paging/Intercom Groups

Enable:

\* Name:

\* Type:

Custom Prompt:  [Prompt](#)

Repeat:

\* Date:

\* Time:

Transmission Method:

Members:

Available	Selected
<input type="checkbox"/> 10 items <input type="checkbox"/> 012345678 <input type="checkbox"/> 1002 <input type="checkbox"/> 1003 <input type="checkbox"/> 1004	<input type="checkbox"/> 2 items <input type="checkbox"/> 1000 "James tuan" <input type="checkbox"/> 1001 "John Doe"

**Figure 177: Announcement Paging**


**Table 78: Announcement Paging Configuration Parameters**

<b>Enable</b>	Enable/Disable Announcement Paging.
<b>Name</b>	Configure paging/intercom group name.
<b>Type</b>	Select "Announcement Paging"
<b>Custom Prompt</b>	<p>This option is to set a custom prompt for a paging/intercom group to announce to caller. Click on 'Prompt', it will direct the users to upload the customized voice prompts.</p> <p><b>Note:</b> Users can also refer to the page <b>PBX Settings</b>→<b>Voice Prompt</b>→<b>Custom Prompt</b>, where they could record new prompt or upload prompt files.</p>
<b>Repeat</b>	If enabled, the announcement page will be repeated for the selected weekdays.
<b>Date</b>	Configure Announcement Paging Date.
<b>Time</b>	Configure Announcement Paging Time.
<b>Transmission Method</b>	<p>Configure Announcement Paging transmission method.</p> <p><b>Unicast:</b> Depending on members selection</p> <p><b>Multicast:</b> Depending on Multicast IP address and Port</p>
<b>Members</b>	Select available users from the left side to the paging/intercom group member list on the right.

### Configure Private Intercom

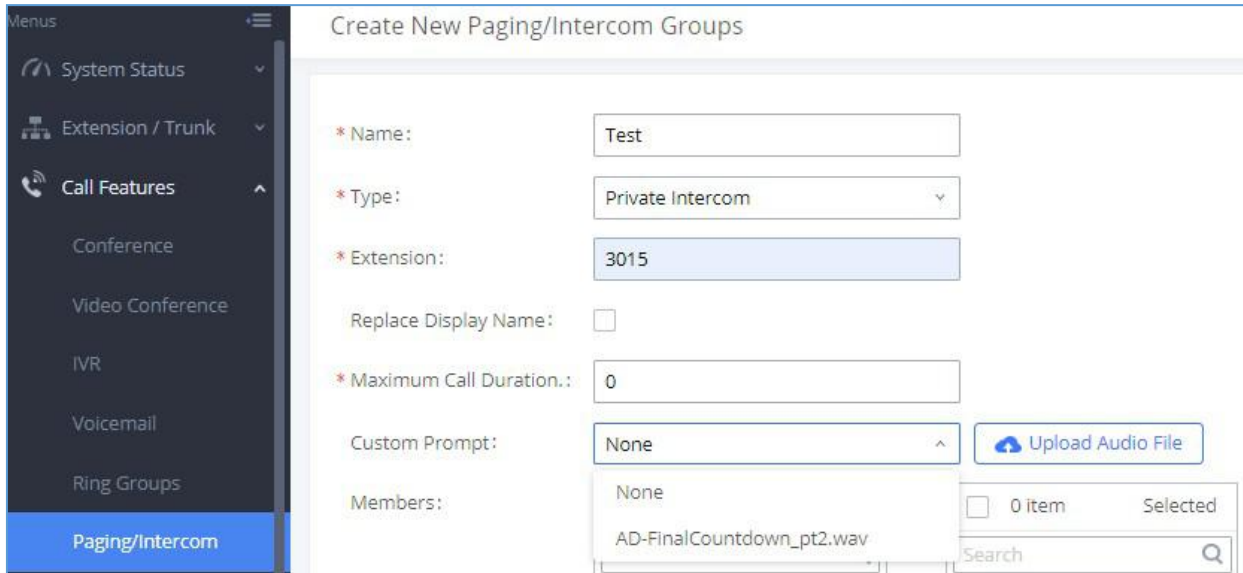
Private Intercom is a new paging type that is meant to be used with Grandstream GSC3510.

<http://www.grandstream.com/products/facility-management/intercoms-paging/product/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





The screenshot shows the 'Create New Paging/Intercom Groups' configuration page. On the left is a dark sidebar menu with 'Paging/Intercom' selected. The main content area has the following fields:

- Name:** Text input field containing 'Test'.
- Type:** Dropdown menu set to 'Private Intercom'.
- Extension:** Text input field containing '3015'.
- Replace Display Name:** A checkbox that is currently unchecked.
- Maximum Call Duration.:** Text input field containing '0'.
- Custom Prompt:** Dropdown menu set to 'None', with an 'Upload Audio File' button to its right.
- Members:** A list box showing 'None' and 'AD-FinalCountdown\_pt2.wav'. Below it is a search bar with '0 item Selected' and a magnifying glass icon.

**Figure 178: Private Intercom**

**Table 79: Private Intercom Configuration Parameters**

<b>Name</b>	Configure paging/intercom group name.
<b>Type</b>	Select "Private Intercom".
<b>Extension</b>	Configure the paging/intercom group extension.
<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with Paging/Intercom name.
<b>Maximum Call Duration</b>	Specify the maximum call duration in seconds. The default value 0 means no limit.
<b>Custom Prompt</b>	This option is to set a custom prompt for a paging/intercom group to announce to caller. Click on 'Prompt', it will direct the users to upload the customized voice prompts. <b>Note:</b> Users can also refer to the page <b>PBX Settings</b> → <b>Voice Prompt</b> → <b>Custom Prompt</b> , where they could record new prompt or upload prompt files.
<b>Members</b>	Select available users from the left side to the paging/intercom group member list on the right.
<b>Paging/Intercom Whitelist</b>	Select which extensions are allowed to use the paging/intercom feature for this paging group.



## Paging/Intercom Group Settings

### Paging & Intercom Settings

\* Alert-info Head...

### Paging/Intercom Feature Code Settings

Custom Prompt:  Prompt

Please go to [Feature Codes](#) Configure Paging/Intercom Feature Code.

**Figure 179: Page/Intercom Group Settings**

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

## Configure a Scheduled Paging/Intercom

Users can schedule paging/intercom calls by using the Schedule Paging/Intercom page. To schedule, click the Add button on the new page and configure the caller, the group to use, and the time to call out.

### Paging/Intercom Groups

Paging/Intercom Groups [Schedule Paging/Intercom](#)

+ Add

	Caller ↕	Paging/Intercom Group ↕
<input type="checkbox"/>	3000	5000
<input type="checkbox"/>	3002	5001

Total: 2 < 1 >

**Figure 180: Schedule Paging/Intercom page**



**Table 80: Schedule Paging / Intercom Settings**

<b>Caller</b>	Configure the caller ID for the paging / intercom group.
<b>Paging/Intercom Group</b>	Select the paging / intercom group from the list of the available groups.
<b>Start Time</b>	Configure the start time of the scheduled paging / intercom call.
<b>Type</b>	Select the type for the scheduled paging / intercom call. The available types are: <b>Single</b> time or <b>Daily</b> basis. Default is “Single”.
<b>Include Holidays</b>	If enabled Paging/Intercom will run during holidays.
<b>Action Status</b>	Display the action status of the scheduled paging / intercom call.

Create New Scheduled Paging/Intercom

---

\* Caller:

\* Paging/Intercom Group:

Type:

Include Holidays:

\* Start Time:

**Figure 181: Creating a scheduled paging/intercom call**

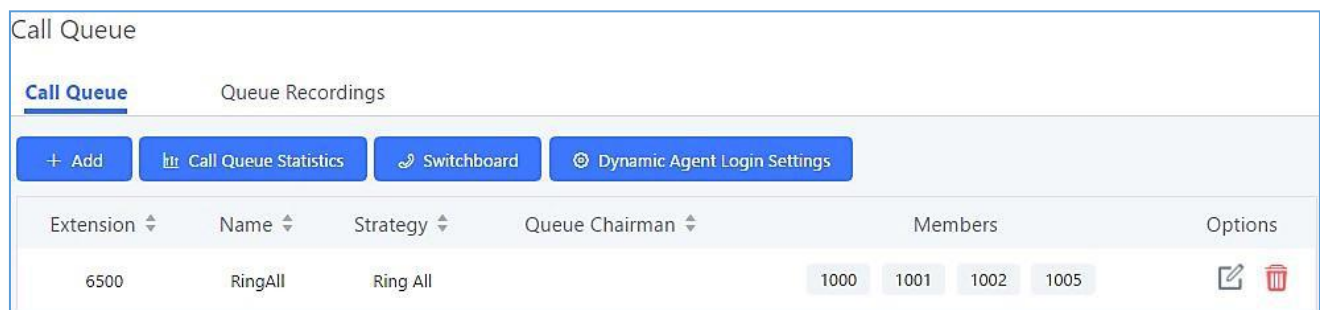



## CALL QUEUE

The UCM6200 supports call queue by using static agents or dynamic agents. Call Queue system can accept more calls than the available agents. Incoming calls will be held until next representative is available in the system. This section describes the configuration of call queue under Web GUI→**Call Features**→**Call Queue**.

### Configure Call Queue



Call queue settings can be accessed via Web GUI→**Call Features**→**Call Queue**.



**Figure 182: Call Queue**

UCM6200 supports custom prompt feature in call queue. This custom prompt will active after the caller waits for a period of time in the Queue. Then caller could choose to leave a message/ transfer to default extension or keep waiting in the queue.

To configure this feature, please go to UCM Web GUI→**Call Features**→**Call Queue**→Create New Queue/Edit Queue→Queue Options→set Enable Destination to Enter Destination with Voice Prompt. Users could configure the wait time with Voice Prompt Cycle.

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

**Table 81: Call Queue Configuration Parameters**

Basic Settings	
<b>Extension</b>	Configure the call queue extension number.
<b>Name</b>	Configure the call queue name to identify the call queue.
<b>Strategy</b>	Select the strategy for the call queue. <ul style="list-style-type: none"> <li>• <b>Ring All</b> Ring all available Agents simultaneously until one answer.</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>Linear</b> Ring agents in the specified order.</li> <li>• <b>Least Recent</b> Ring the agent who has been called the least recently.</li> <li>• <b>Fewest Calls</b> Ring the agent with the fewest completed calls.</li> <li>• <b>Random</b> Ring a random agent.</li> <li>• <b>Round Robin</b> Ring the agents in Round Robin scheduling with memory. The default setting is "Ring All".</li> </ul>
<b>Music On Hold</b>	<p>Select the Music On Hold class for the call queue.</p> <p><b>Note:</b> Music On Hold classes can be managed from Web GUI→<b>PBX Settings</b>→<b>Music On Hold</b>.</p>
<b>Max Queue Length</b>	<p>Configure the maximum number of calls to be queued at once. This number does not include calls that have been connected with agents. It only includes calls not connected yet. The default setting is 0, which means unlimited. When the maximum value is reached, the caller will be treated with busy tone followed by the next calling rule after attempting to enter the queue.</p>
<b>Wrapup Time</b>	<p>Configure the number of seconds before a new call can ring the queue after the last call on the agent is completed. If set to 0, there will be no delay between calls to the queue. The default setting is 10 seconds.</p>
<b>Retry Time</b>	<p>Configure the number of seconds to wait before ringing the next agent.</p>
<b>Ring Time</b>	<p>Configure the number of seconds an agent will ring before the call goes to the next agent. The default setting is 30 seconds.</p>
<b>Auto Record</b>	<p>If enabled, the calls on the call queue will be automatically recorded. The recording files can be accessed in Queue Recordings under Web GUI→<b>Call Features</b>→<b>Call Queue</b>.</p>
<b>Welcome Prompt</b>	<p>If enabled, users can upload an audio file that will be played as an initial tone when dialing the queue number.</p>
<b>Max Wait Time</b>	<p>Configure the timeout after which users will be disconnected from the call queue. The default setting is "60". 0 means unlimited.</p> <p><b>Note:</b> It is recommended to configure "Wait Time" longer than the "Wrapup Time".</p>
<b>Destination</b>	<p>Once Max Wait Time has been configured, select to which destination send the calls that have timed out. The default is to "Hang up" the call.</p>
<b>Reset Agent Call Counter - Enable</b>	<p>If this option is checked the agent call counter will be reset according to the Repeat time settings.</p>



<b>Repeat</b>	<p><b>Note:</b> This option is available only when the Strategy is set to “Fewest Calls”</p> <p>Specifies the frequency at which the Agent call counter will be reset.</p>
<b>Destination Prompt Cycle</b>	<p>Configure the voice prompt cycle (in seconds) of the call queue. Once all agents are busy and the voice prompt will be played, and you can press the appropriate key to transfer to failover destination.</p>
<b>Custom Prompt</b>	<p>When playing a custom prompt, press 1 to transfer to failover destination.</p>
<b>Destination</b>	<p>Select failover destination to send callers after pressing 1 upon hearing the custom prompt.</p>
<b>Advanced Settings</b>	
<ul style="list-style-type: none"> <li>- <b>Virtual Queue</b></li> <li>- <b>Position Announcement</b></li> <li>- <b>Queue Chairman</b></li> </ul>	<p>Refer to <b>Call Center Settings and Enhancements</b> section for detailed information about these features.</p>
<b>Enable Agent Login</b>	<p>Enables agent login/logout feature for static agents (supported only on GXP21XX phones with fw higher than 1.0.9.18).</p>
<b>Leave When Empty</b>	<p>Configure whether the callers will be disconnected from the queue or not if the queue has no agent anymore. The default setting is "Strict".</p> <ul style="list-style-type: none"> <li>• <b>Yes</b> Callers will be disconnected from the queue if all agents are paused or invalid.</li> <li>• <b>No</b> Never disconnect the callers from the queue when the queue is empty.</li> <li>• <b>Strict</b> Callers will be disconnected from the queue if all agents are paused, invalid or unavailable.</li> </ul>
<b>Dial in Empty Queue</b>	<p>Configure whether the callers can dial into a call queue if the queue has no agent. The default setting is "No".</p> <ul style="list-style-type: none"> <li>• <b>Yes</b> Callers can always dial into a call queue.</li> <li>• <b>No</b> Callers cannot dial into a queue if all agents are paused or invalid.</li> <li>• <b>Strict</b> Callers cannot dial into a queue if the agents are paused, invalid or unavailable.</li> </ul>
<b>Failover Destination</b>	<p>Choose the destination where the call will be directed when the queue is empty or when all the agents are not logged in, here are the destinations that can be configured:</p> <ul style="list-style-type: none"> <li>• Play Sound.</li> </ul>

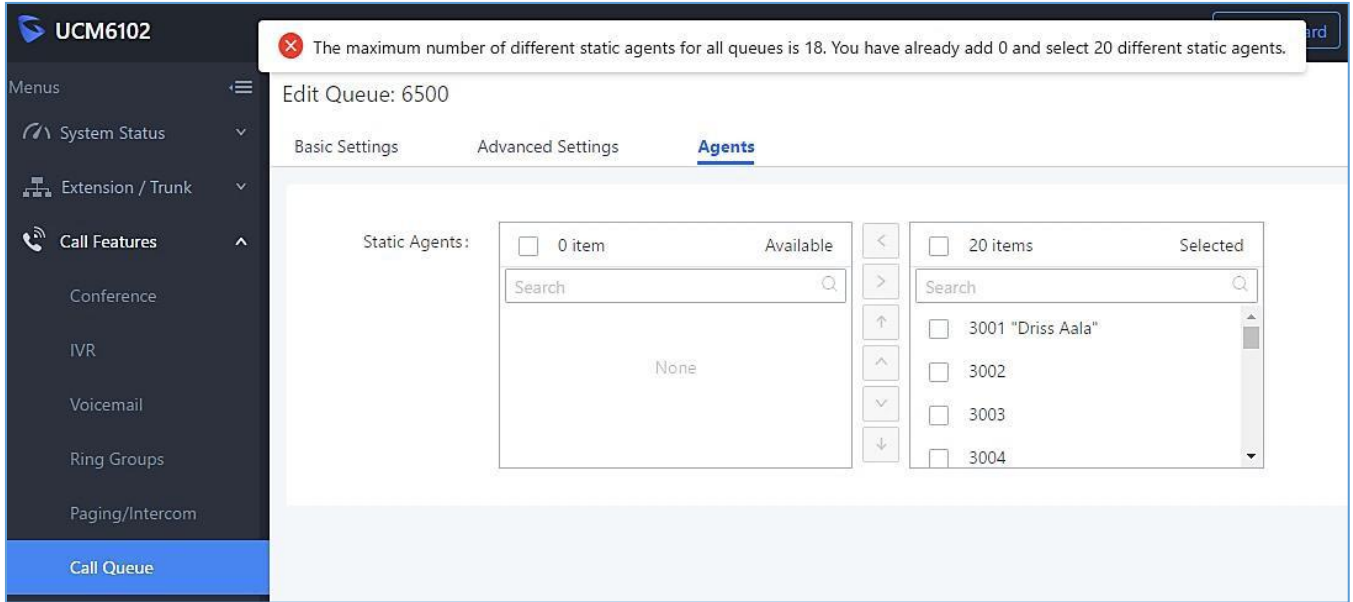


	<ul style="list-style-type: none"> <li>• Extension.</li> <li>• Voicemail.</li> <li>• Queues.</li> <li>• Ring Group.</li> <li>• Voicemail Group.</li> <li>• IVR.</li> <li>• External Number.</li> </ul>
<b>Report Hold Time</b>	If enabled, the UCM6200 will report (to the agent) the duration of time of the call before the caller is connected to the agent. The default setting is "No".
<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with the Call Queue name so that the caller knows the call is incoming from a Call Queue.
<b>Enable Feature Codes</b>	Enable feature codes option for call queue. For example, *83 is used for "Agent Pause". <b>Note:</b> Callers can no longer use feature codes in established callbacks.
<b>Dynamic Login Password</b>	If enabled, the configured PIN number is required for dynamic agent to log in. The default setting is disabled.
<b>Alert-Info</b>	Configure the call destination for the call to be routed to if no agent in this call queue answers the call.
<b>Agents</b>	
<b>Agents</b>	Go to "Agents" Tab and Select the available users to be the static agents in the call queue. Choose from the available users on the left to the static agents list on the right. Click on <input type="checkbox"/> <input type="checkbox"/> to choose. And use UP and Down arrow to select the order of the agent within the call queue.

### Static Agents limitation:

To guarantee a high level of audio quality with the call queue feature, UCMs will limit the number of static agents allowed to be assigned depending on the UCM model used. If the user attempts to configure the number of static agents to be more than the maximum allowed number, a warning message will appear.





**Figure 183: Static Agents limit**

The following table lists the maximum number of static agents for each UCM model:

**Table 82: Static Agent Limitation**

UCM Model	Max Static Agents in Call Queue
UCM6202	23
UCM6204	34
UCM6208	75
UCM6510	15
	0

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



### Dynamic Agent Login Settings

---

Agent Login Extension P...



Agent Logout Extension ...

Example: If Queue Extension is 6500,  
 Agent Login Extension Postfix is \*,  
 Agent Logout Extension Postfix is \*\*,  
 Dial **6500\*** to log in, dial **6500\*\*** to log out.

**Note:** Remove postfix will lead the agent that has  
 not log out yet cannot logout.


**Figure 184: Agent Login Settings**

For example, if the call queue extension is 6500, Agent Login Extension Postfix is \* and Agent Logout Extension Postfix is \*\*, users could dial 6500\* to login to the call queue as dynamic agent and dial 6500\*\* to logout from the call queue. Dynamic agent does not need to be listed as static agent and can log in/log out at any time.

- Call queue feature code "Agent Pause" and "Agent Unpause" can be configured under Web GUI→**Call Features**→**Feature Codes**. The default feature code is \*83 for "Agent Pause" and \*84 for "Agent Unpause".
- Queue recordings are shown on the Call Queue page under "Queue Recordings" Tab. Click on  to download the recording file in .wav format; click on  to delete the recording file. To delete multiple recording files by one click, select several recording files to be deleted and click on "Delete Selected Recording Files" or click on "Delete All Recording Files" to delete all recording files.

## Call Center Settings and Enhancements

UCM supports light weight call center features including virtual queue and position announcement, allowing the callers to know their position on the call queue and giving them the option to either stay on the line waiting for their turn or activate a callback which will be initiated by the UCM one an agent is free.

To configure call center features, press on  on an existing call queue and go under the advanced settings tab. Following parameters are available:



**Table 83: Call Center Parameters**

<b>Enable Virtual Queue</b>	Enable virtual queue to activate call center features.
<b>Virtual Queue Period</b>	Configure the time in (s) after which the virtual queue will take effect and the menu will be presented to the caller to choose an option. Default is 20s.
<b>Virtual Queue Mode</b>	<p><b>Offered to caller after timeout:</b> After the virtual queue period passes, the caller will enter the virtual call queue and be presented with a menu to choose an option, the choices are summarized below:</p> <ul style="list-style-type: none"> <li>• Press * to set current number as callback number.</li> <li>• Press 0 to set a callback number different than current caller number.</li> <li>• Press # to keep waiting on the call queue.</li> </ul> <p><b>Triggered on user request:</b> In this mode, the callers can activate the virtual queue by pressing 2, then they will be presented with the menu to choose an option as below:</p> <ul style="list-style-type: none"> <li>• Press * to set current number as callback number.</li> <li>• Press 0 to set a callback number different than current caller number.</li> <li>• Press # to keep waiting on the call queue.</li> </ul>
<b>Virtual Queue Outbound Prefix</b>	System will add this prefix to dialed numbers when calling back users.
<b>Enable Virtual Queue Timeout</b>	When this option is enabled and after a caller registers a call back request on the virtual queue. While all the agents are busy, the UCM will call an agent once he/she is idle again, this timeout is used for how long the UCM continues calling the agent and if the agent doesn't answer the call then the callback request will timeout and expire.
<b>Write Timeout</b>	Configure the virtual queue callback timeout period in seconds.
<b>Enable Position Announcement</b>	Enable the announcement of the caller's position periodically. <b>Note:</b> Queue position will now be announced to the caller upon entering the queue.
<b>Position Announcement Interval</b>	Configure the period of time in (s) during which the UCM will announce the caller's position in the call queue.
<b>Enable Hold time Announcement</b>	Enable the announcement of the estimated hold time to the caller periodically. <b>Note:</b> Hold time will not be announced if less than one minute.
<b>Queue Chairman</b>	Select the extension to act as chairman of the queue (monitoring). <b>Note:</b> One queue can have only 3 chairmen to manage it.
<b>Enable Agent Login</b>	When enabled, statics agents can conveniently log in and out of a queue by configuring a programmable key on their phones as a shortcut. <b>Notes:</b> <ul style="list-style-type: none"> <li>✓ This feature is currently available only for GXP21xx phones on firmware 1.0.9.18 or greater.</li> </ul>



	<ul style="list-style-type: none"> <li>✓ After enabling the feature, users need to set the option on GXP21XX phone under “<b>Account→SIP Settings→Advanced Features→Special Feature</b>” to “<b>UCM Call Center</b>”. A softkey labeled “UCM-CC” will appear on the bottom of the phone’s screen.</li> <li>✓ When this option is enabled, dynamic agent login will be no longer supported.</li> <li>✓ In case of concurrent registrations, changing agent status on one phone (login/logout) will be reflected on all phones.</li> </ul>
<b>Autofill</b>	Enable or Disable the autofill feature.

### Queue Auto fill enhancement:

In previous UCM firmware, the call queue has a serial type behavior in that the queue will make all waiting callers wait in the queue even if there is more than one available member ready to take calls until the head caller is connected with the member they were trying to get to.

The next waiting caller in line then becomes the head caller, and they are then connected with the next available member and all available members and waiting callers waits while this happens.

Starting from 1.0.14.x, the waiting callers are connecting with available members in a parallel fashion until there are no more available members or no more waiting callers.

For example, in a call queue with linear method, if there are two available agents, when two callers call in the queue at the same time, UCM will assign the two callers to each of the two available agents at the same time, rather than assigning the second caller to second available agent after the first agent answers the call from the first caller.

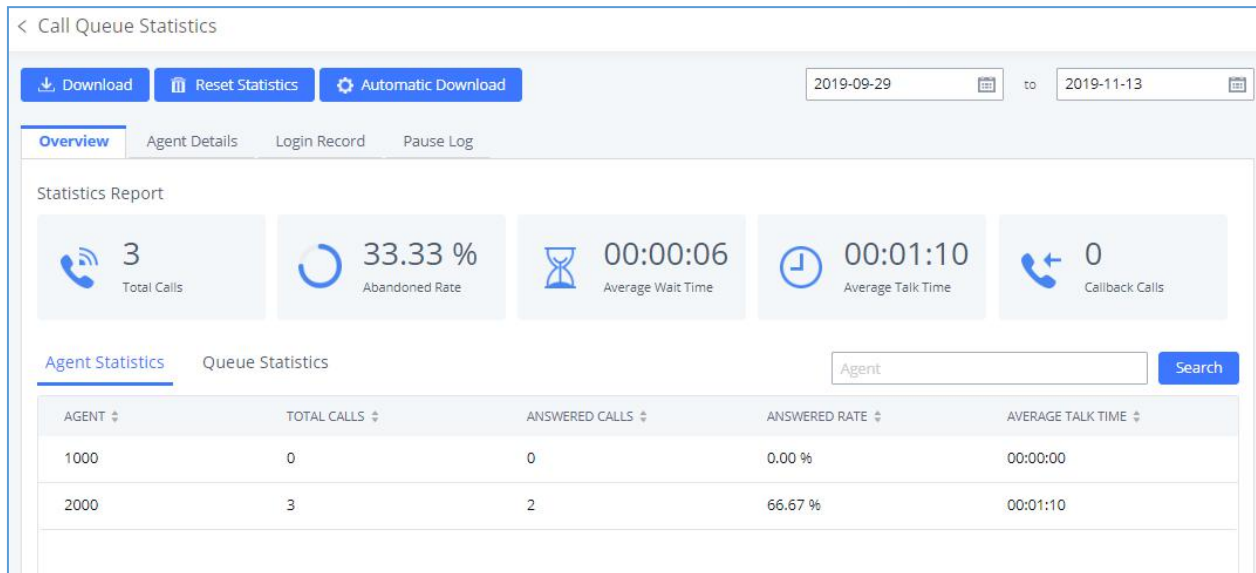
### Queue Statistics

Along with call center features, users can also gather detailed call queue statistics allowing them to make better changes/decision to manage better the call distribution and handling based on time, agent and queue.

To access call queue statistics, go to Web GUI→**Call Features→Call Queue** and click on “Call Queue Statistics”, the following page will be displayed:

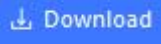






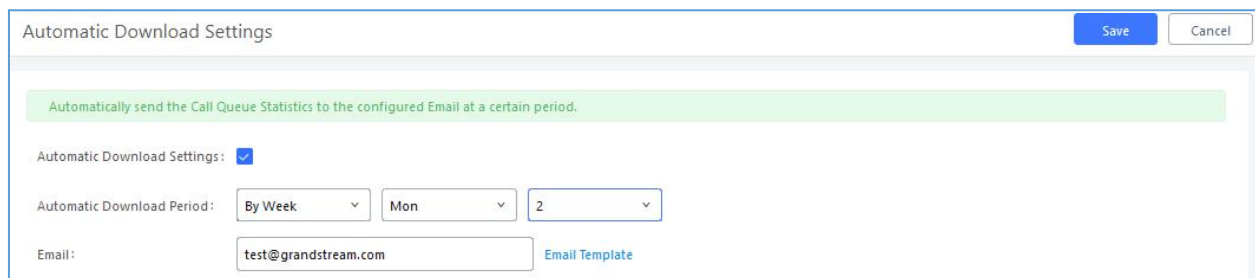


**Figure 185: Call Queue Statistics**

Select the time interval along with the queue(s) and agent(s) to get detailed statistics.

User can download statistics on CSV format by clicking on the download button , also the statistics can be cleared using “Clear Statistics” button .

The statistics can be automatically sent to a specific email address on a preconfigured Period, this can be done by clicking on “Automatic Download Settings” button , and user will be directed to below page where he can configure the download period (Day/Week/Month) and the Email where the statistics will be sent (Email settings should be configured correctly):



**Figure 186 : Automatic Download Settings - Queue Statistics**

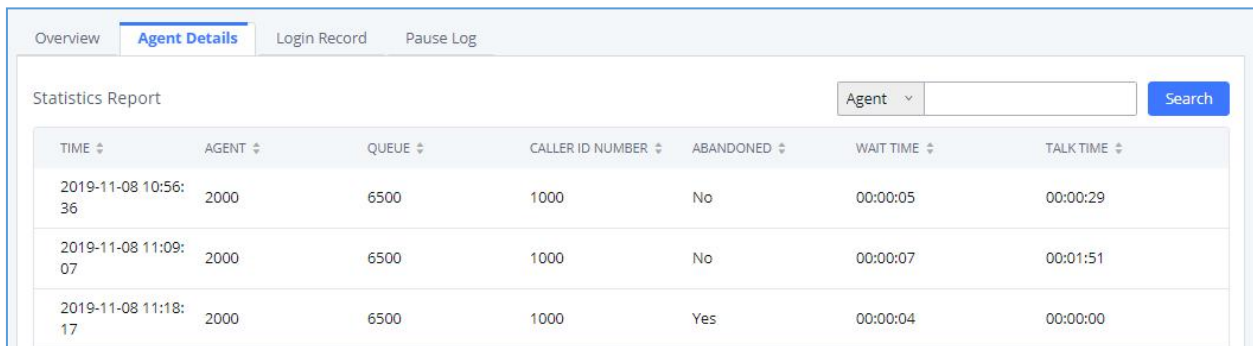
Significantly more information is now available UCM's queue statistics page. In addition to the information presented in previous firmware, users can now view a call log that displays calls to all agents and queues, a dynamic agent login/logout record, and a pause log. Statistics reports for these new 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.



## 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.



Overview **Agent Details** Login Record Pause Log

Statistics Report Agent  Search

TIME ↕	AGENT ↕	QUEUE ↕	CALLER ID NUMBER ↕	ABANDONED ↕	WAIT TIME ↕	TALK TIME ↕
2019-11-08 10:56:36	2000	6500	1000	No	00:00:05	00:00:29
2019-11-08 11:09:07	2000	6500	1000	No	00:00:07	00:01:51
2019-11-08 11:18:17	2000	6500	1000	Yes	00:00:04	00:00:00

**Figure 187: Agent details**

## 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.



Overview Agent Details <b>Login Record</b> Pause Log					
Statistics Report					
				Agent <input type="text"/>	Search
AGENT	QUEUE	LOGIN TIME	LOGOUT TIME	LOGIN DURATION	
2000	6500	2019-11-08 09:48:53	2019-11-08 09:53:00	00:04:07	
2000	6500	2019-11-08 09:53:10	2019-11-08 09:55:22	00:02:12	

**Figure 188: Login Record**

## 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.

Overview Agent Details Login Record <b>Pause Log</b>					
Statistics Report					
				Agent <input type="text"/>	Search
AGENT	QUEUE	PAUSE TIME	RESUME TIME	PAUSE DURATION	
2000	6500	2019-11-08 11:32:00	2019-11-08 11:33:33	00:01:33	
2000	6500	2019-11-08 11:32:00	2019-11-08 11:33:33	00:01:33	

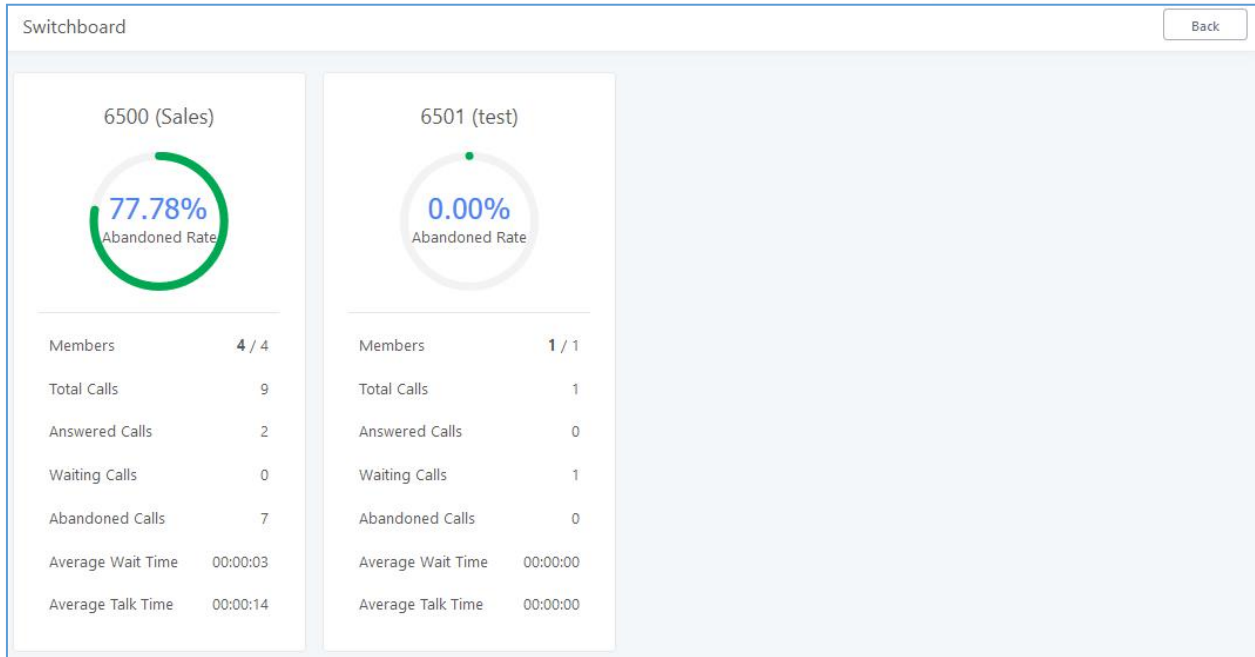
**Figure 189: Pause Log**

## Switchboard

Switchboard is a Web GUI tool for call queue monitoring and management, admin can access to it from the menu **Call Features**→**Call Queue** then press « Switchboard ».

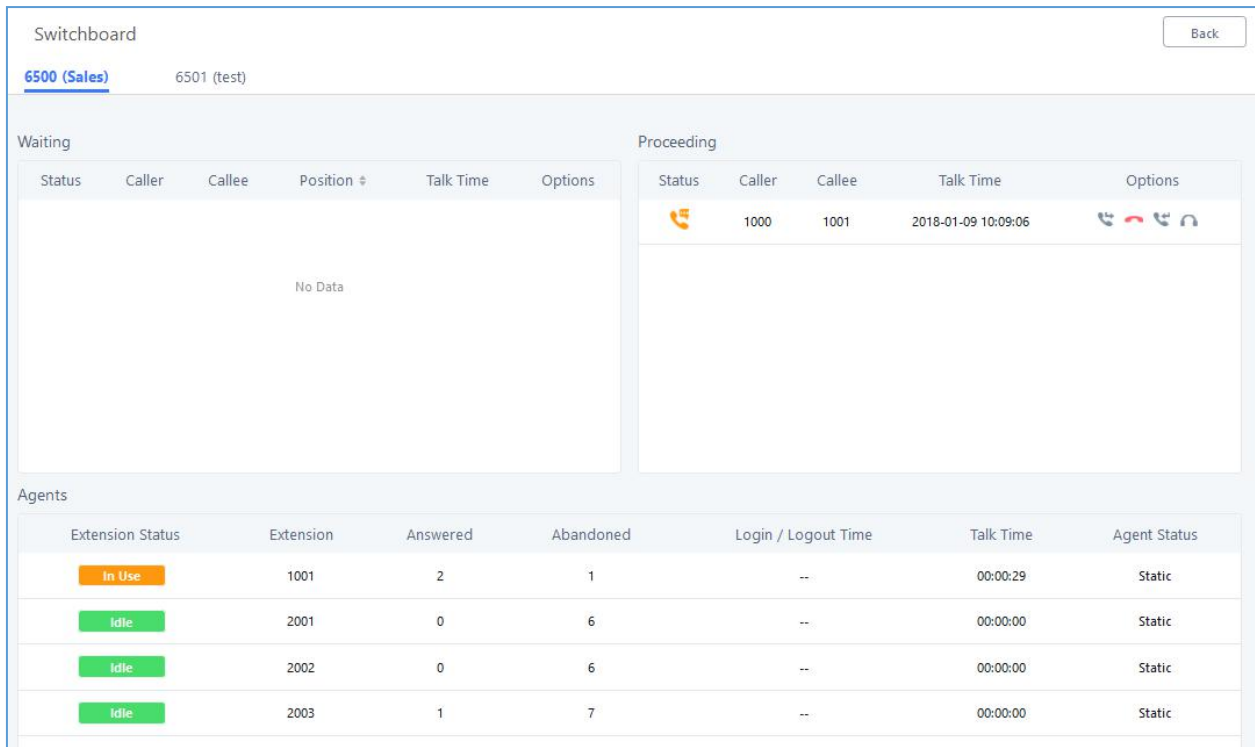
Following page will be displayed:





**Figure 190: Switchboard Summary**

Page above summarizes the available queues statistics and if one of the queues is clicked the user will be directed to page below:




**Figure 191: Call Queue Switchboard**



The table below gives a brief description for the main menus:

**Table 84: Switchboard Parameters**

<b>Waiting</b>	This menu shows the current waiting calls along with the caller id and the option to hang-up call by pressing on the  button.
<b>Proceeding</b>	Shows the current established calls along with the caller id and the callee (agent) as well as the option to hang-up, transfer, add conference or barge-in the call.
<b>Agents</b>	Displays the list of agents in the queue and the extension status (idle, ringing, in use or unavailable) along with some basic call statistics and agent's mode (static or dynamic).  <b>Note:</b> the dashboard will show the number of calls (answered and abandoned) of each agent. For dynamic agents, it will count the number of calls starting from the last login time.

There are three different privilege levels for Call Queue management from the switchboard: Super Admin, Queue Chairman, and Queue Agent.

- **Super Admin** - Default admin of the UCM. Call queue privileges include being able to view and edit all queue agents, monitor and execute actions for incoming and ongoing calls for each extension in Switchboard, and generate Call Queue reports to track performance.
- **Queue Chairman** - User appointed by Super Admin to monitor and manage an assigned queue extension via Switchboard. The Queue Chairman can log into the UCM user portal with his extension number and assigned user password. To access the Switchboard, click on “*Value-added Features*” in the side menu and click on “*Call Queue*”. In the image below, User 1012 is the Queue Chairman appointed to manage Queue Extension 6500 and can see all the agents of the queue in the Switchboard.  
Note: Super Admin can assign 3 Chairmen
- **Queue Agent** - User appointed by Super Admin to be a member of a queue extension. A queue agent can log into the UCM user portal with his extension number and assigned user password. To access the Switchboard, click on “*Value-added Features*” in the side menu and click on “*Call Queue*”. However, a queue agent can view and manage only his own calls and statistics, but not other agents' in the queue extension. In the image below, User 1000 is a queue agent and can see only his own information in the Switchboard.



Call Queue

6500 (test)

Waiting						Proceeding				
Status	Caller	Callee	Position	Wait Time	Options	Status	Caller	Callee	Talk Time	Options
	1003	6500	1	00:00:11		No Data				

Agents	Extension Status	Extension	Answered	Abandoned	Login / Logout Time	Talk Time	Agent Status
<b>Ringling</b>		1000	3	3	--	00:00:08	Static

**Figure 192: Queue Agent**

## Global Queue Settings

As explained before, under this section users can configure the feature codes for Dynamic agent login and logout, and also can now customize the keys for virtual queue options like shown below.

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 193: Global Queue Settings**



**Table 85: Global Queue Settings**

<b>Dynamic Agent Login Settings</b>	
<b>Agent Login Code Suffix</b>	Configure the code to dial after the queue extension to log into the queue (i.e. queue extension + suffix). If no suffix is configured, dynamic agents will not be able to log in
<b>Agent Logout Code Suffix</b>	Configure the code to dial after the queue extension to log out of the queue (i.e. queue extension + suffix). If no suffix is configured, dynamic agents will not be able to log out.
<b>Virtual Queue Callback Key Settings</b>	
<b>Call Back Current Number</b>	Press the feature key configured to set your current number as callback number.
<b>Custom Callback Number</b>	Press the feature key configured to set a custom callback number.
<b>Continue Waiting</b>	Press the feature key configured to continue waiting.






## PICKUP GROUPS

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

### Configure Pickup Groups

Pickup groups can be configured via Web GUI → **Call Features** → **Pickup Groups**.

- Click on  to create a new pickup group.
- Click on  to edit the pickup group.
- Click on  to delete the pickup group.

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

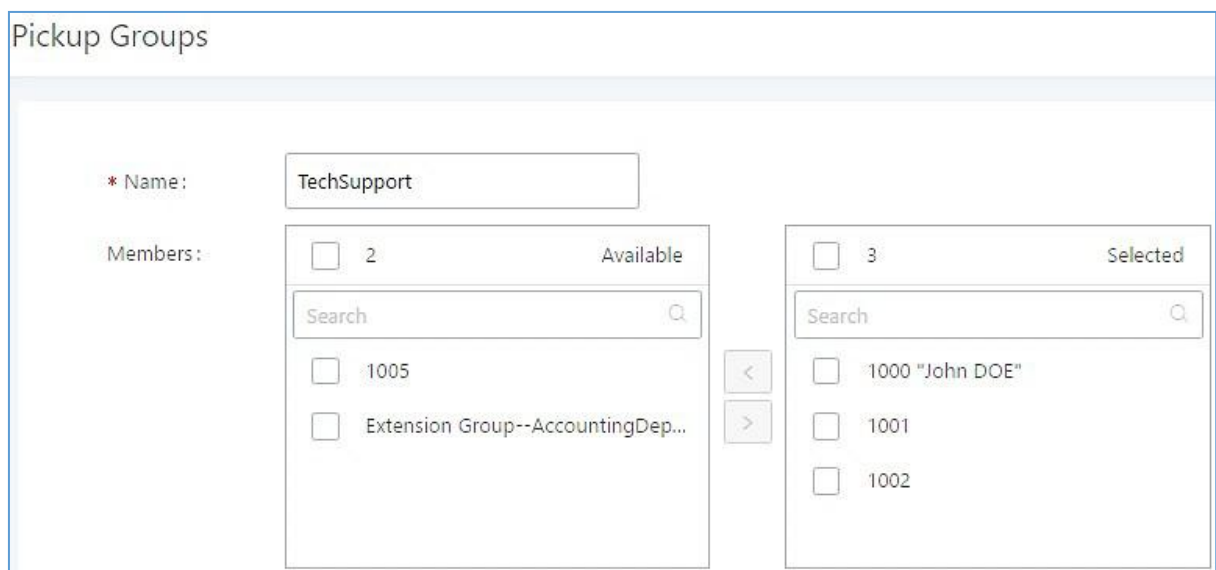


Figure 194: Edit Pickup Group

### Configure Pickup Feature Code

When picking up the call for the pickup group member, the user only needs to dial the pickup feature code. It is not necessary to add the extension number after the pickup feature code. The pickup feature code is configurable under Web GUI → **Call Features** → **Feature Codes**.

The default feature code for call pickup extension is \*8, otherwise if the person intending to pick up the call knows the ringing extension they can use \*\* followed by the extension number in order to perform the call pickup operation.





The following figure shows where you can customize these features codes

The screenshot displays the 'Feature Codes' configuration page in the Grandstream interface. The left sidebar shows a menu with 'Feature Codes' selected. The main content area has tabs for 'Feature Maps', 'DND/Call Forward', 'Feature Misc', and 'Feature Codes'. Below the tabs are 'Reset All' and 'Default All' buttons. The main area contains a list of feature codes, each with a text input field for the code and a checkbox. The '\* Pickup Extension...' feature code is highlighted with a green box, showing a code of '\*8' and a checked checkbox. Other feature codes include '\* Voicemail Access...', '\* Agent Pause:', '\* Paging Prefix:', '\* Blacklist Add:', '\* Call Pickup on R...', '\* Direct Dial Mob...', '\* Call Completion...', '\* Listen Spy:', '\* Barge Spy:', '\* PMS Wakeup Se...', '\* Presence Status...', '\* My Voicemail:', '\* Agent Unpause...', '\* Intercom Prefix:', '\* Blacklist Remov...', '\* Pickup In-call:', '\* Direct Dial Voice...', '\* Call Completion...', 'Enable Spy:', '\* Whisper Spy:', '\* Wakeup Service...', and '\* Update PMS Ro...'.

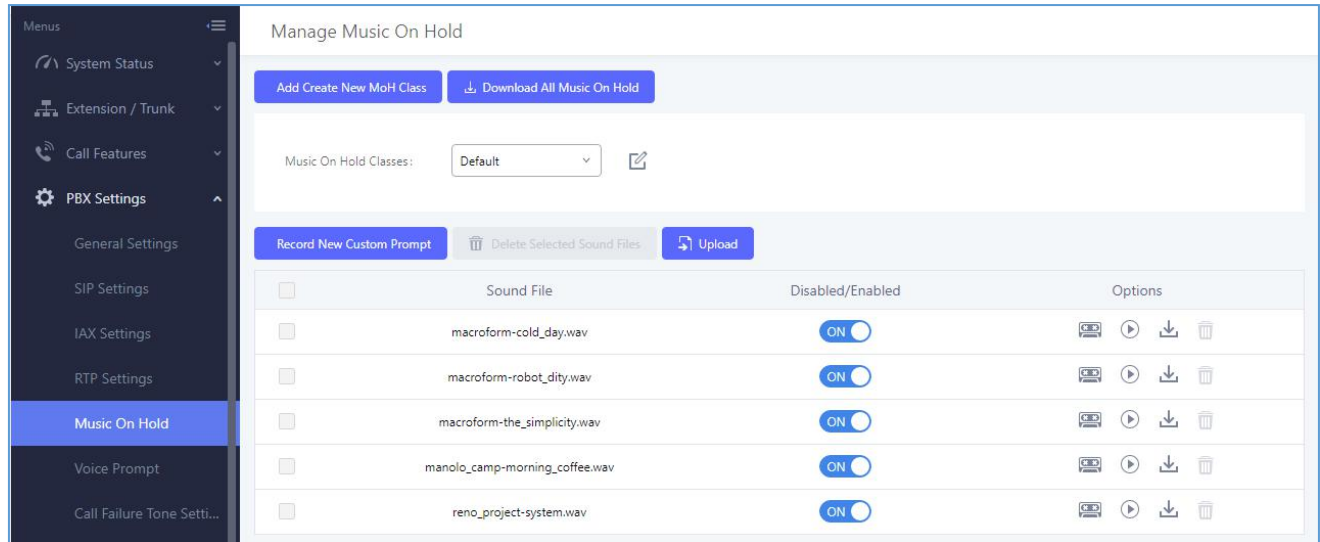
Feature Name	Code	Enabled
* Voicemail Acces...	*98	<input checked="" type="checkbox"/>
* Agent Pause:	*83	<input checked="" type="checkbox"/>
* Paging Prefix:	*81	<input checked="" type="checkbox"/>
* Blacklist Add:	*40	<input checked="" type="checkbox"/>
* Call Pickup on R...	**	<input checked="" type="checkbox"/>
* Pickup Extension...	*8	<input checked="" type="checkbox"/>
* Direct Dial Mob...	*88	<input checked="" type="checkbox"/>
* Call Completion...	*12	<input checked="" type="checkbox"/>
* Listen Spy:	*54	<input type="checkbox"/>
* Barge Spy:	*56	<input type="checkbox"/>
* PMS Wakeup Se...	*35	<input checked="" type="checkbox"/>
* Presence Status...	*48	<input checked="" type="checkbox"/>
* My Voicemail:	*97	<input checked="" type="checkbox"/>
* Agent Unpause...	*84	<input checked="" type="checkbox"/>
* Intercom Prefix:	*80	<input checked="" type="checkbox"/>
* Blacklist Remov...	*41	<input checked="" type="checkbox"/>
* Pickup In-call:	*45	<input type="checkbox"/>
* Direct Dial Voice...	*	<input checked="" type="checkbox"/>
* Call Completion...	*11	<input checked="" type="checkbox"/>
Enable Spy:		<input type="checkbox"/>
* Whisper Spy:	*55	<input type="checkbox"/>
* Wakeup Service...	*36	<input checked="" type="checkbox"/>
* Update PMS Ro...	*23	<input checked="" type="checkbox"/>

**Figure 195: Edit Pickup Feature Code**









## MUSIC ON HOLD


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



**Figure 196: Music On Hold Default Class**

- Click on "Create New MOH Class" to add a new Music On Hold class.
- Click on  to configure the MOH class sort method to be "Alpha" or "Random" for the sound files.
- Click on  next to the selected Music On Hold class to delete this Music On Hold class.
- Click on  to start uploading. Users can upload:
  - Single files with 8KHz Mono Music file, or
  - Music on hold files in a compressed package with .tar, .tar.gz and .tgz as the suffix. The file name can only be letters, digits or special characters -\_
  - the size for the uploaded file should be less than 30M, the compressed file will be applied to the entire MoH.
- Users could also download all the music on hold files from UCM. In the Music On Hold page, click on  and the file will be downloaded to your local PC.
- Click on  next to the sound file to disable it from the selected Music On Hold Class.
- Click on  next to the sound file to enable it from the selected Music On Hold Class.





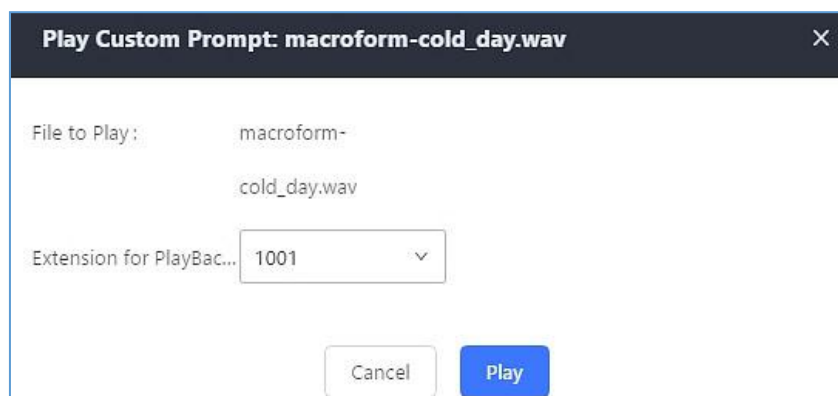
- Select the sound files and click on  Delete Selected Sound Files to delete all selected music on hold files.

**Note:** the size for the uploaded file should be less than 30M, the compressed file will be applied to the entire MoH.

The UCM6200 allows Users to select the Music on Hold file from WebGUI to play it. The UCM6200 will initiate a call to the selected extension and play this Music on Hold file once the call is answered.

Steps to play the music on hold file:


1. Click on the  button for the Music on Hold file.
2. In the prompted window, select the extension to playback and click .

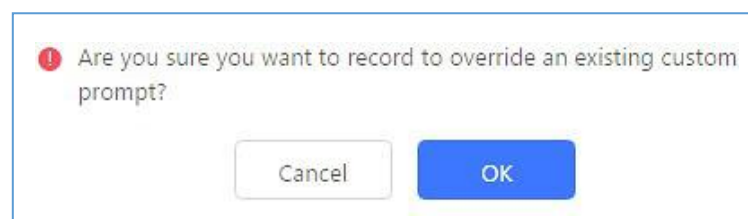


**Figure 197: Play Custom Prompt**



3. The selected extension will ring.
4. Answer the call to listen to the music playback.

Users could also record their own Music on hold to override an existing custom prompt, this can be done by following those steps:

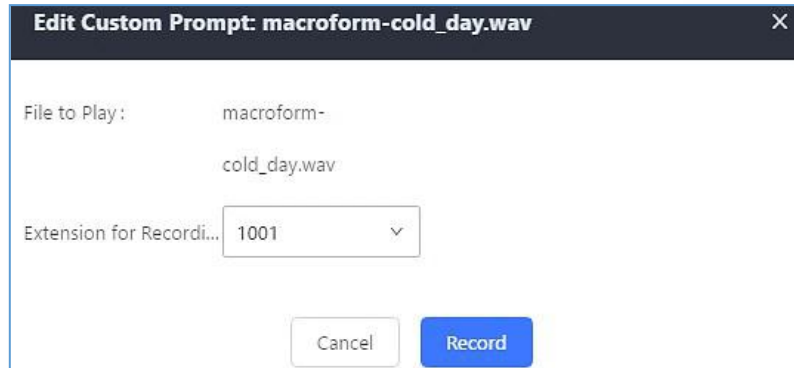
1. Click on .
2. A prompt of confirmation will pop up, as shown below.



**Figure 198: Information Prompt**

3. Click .
4. In the prompted window, select the extension to playback and click .





**Figure 199: Record Custom Prompt**

5. Answer the call and start to record your new music on hold.
6. Hangup the call and refresh Music On Hold page then you can listen to the new recorded file.

---

 **Notes:**

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



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



## FAX SERVER

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

### Configure Fax/T.38

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

Fax Settings

---

\*Enable Error Correction Mode:

\*Maximum Transfer Rate:

\*Minimum Transfer Rate:

\*Max Concurrent Sending Fax:

\*Fax Queue Length:

Fax Header Information:

Default Email Address:  [Email Template](#)

Enable Fax Resend:

Max Resend Attempts:

Figure 200: Fax Settings

Table 86: FAX/T.38 Settings

<b>Enable Error Correction Mode</b>	Configure to enable Error Correction Mode (ECM) for the Fax. The default setting is "Yes".
<b>Maximum Transfer Rate</b>	Configure the maximum transfer rate during the Fax rate negotiation. The possible values are 2400, 4800, 7200, 9600, 12000 and 14400. The default setting is 14400.



<b>Minimum Transfer Rate</b>	Configure the minimum transfer rate during the Fax rate negotiation. The possible values are 2400, 4800, 7200, 9600, 12000 and 14000. The default setting is 2400.
<b>Max Concurrent Sending Fax</b>	<p>Configure the concurrent fax that can be sent by UCM6200. Two modes “Only” and “More” are supported.</p> <ul style="list-style-type: none"> <li>• <b>Only</b> Under this mode, the UCM6200 allows only single user to send fax at a time.</li> <li>• <b>More</b> Under this mode, the UCM6200 supports multiple concurrent fax sending by the users.</li> </ul> <p>By default, this option is set to “only”.</p>
<b>Fax Queue Length</b>	Configure the maximum length of Fax Queue from 6 to 10. The default setting is 6.
<b>Fax Header Information</b>	Adds fax header into the fax file.
<b>Default Email Address</b>	<p>Configure the Email address to send the received Fax to if user's Email address cannot be found.</p> <p><b>Note:</b> The extension's Email address or the Fax's default Email address needs to be configured in order to receive Fax from Email. If neither of them is configured, Fax will not be received from Email.</p>
<b>Template Variables</b>	<p>Fill in the "Subject:" and "Message:" content, to be used in the Email when sending the Fax to the users.</p> <p>The template variables are:</p> <ul style="list-style-type: none"> <li>• <code>#{CALLERIDNUM}</code> : Caller ID Number</li> <li>• <code>#{CALLERIDNAME}</code> : Caller ID Name</li> <li>• <code>#{RECEIVEEXTEN}</code> : The extension to receive the Fax</li> <li>• <code>#{FAXPAGES}</code> : Number of pages in the Fax</li> <li>• <code>#{VM_DATE}</code> : The date and time when the Fax is received</li> </ul>
<b>Enable Fax Resend</b>	Enables the fax resend option which allow the UCM to keep attempting to send faxes up to a specified amount of times. Additionally, if a fax still fails to send, a <i>Resend</i> button will appear in the File Send Progress list in <i>Value-Added Features</i> → <i>Fax Sending</i> to allow manual resending.
<b>Max Resend Attempts</b>	Configures the maximum attempts number to resend the FAX.



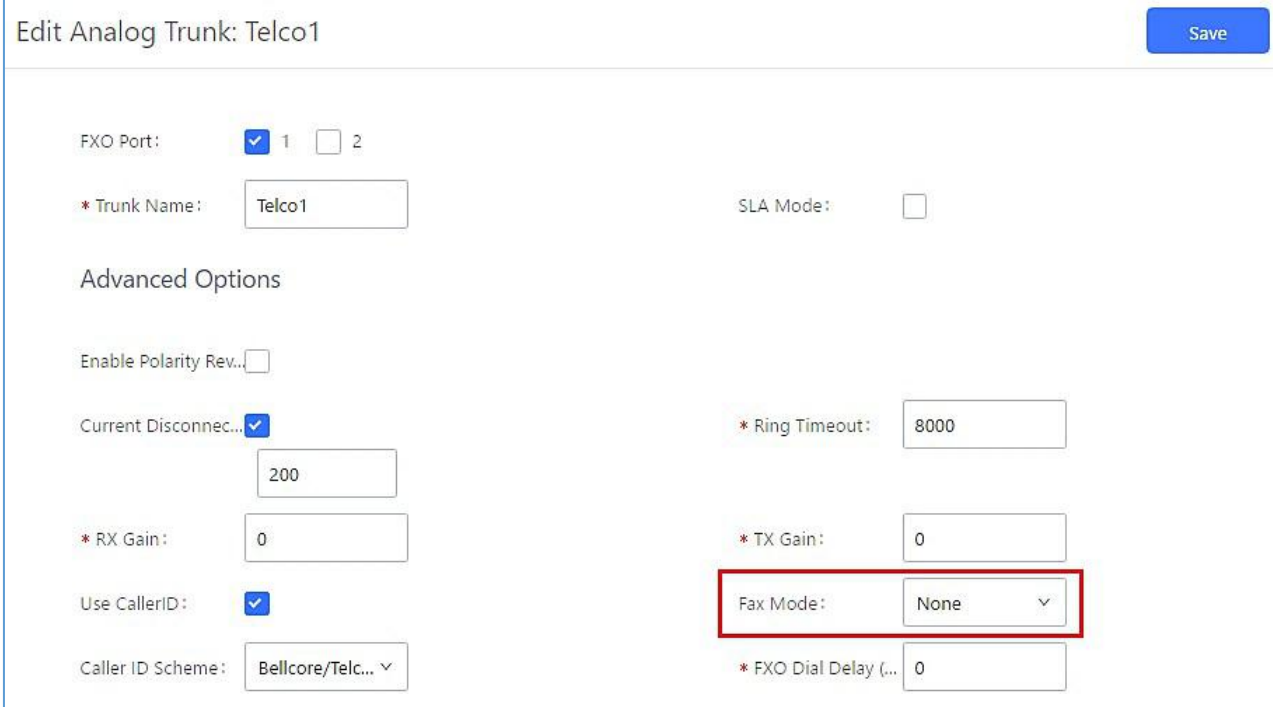
## Receiving Fax

### Sample Configuration to Receive Fax from PSTN Line

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

1. Connect Fax machine to the UCM6200 FXS port.
2. Connect PSTN line to the UCM6200 FXO port.
3. Go to Web GUI→**Extension/Trunk** page.
4. Create or edit the analog trunk for Fax as below.

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



The screenshot shows the 'Edit Analog Trunk: Telco1' configuration page. The 'Fax Mode' dropdown menu is highlighted with a red box and is set to 'None'. Other visible settings include:

- FXO Port:  1  2
- \* Trunk Name: Telco1
- SLA Mode:
- Advanced Options
- Enable Polarity Rev.:
- Current Disconnect...:  200
- \* Ring Timeout: 8000
- \* RX Gain: 0
- \* TX Gain: 0
- Use CallerID:
- Fax Mode: None (highlighted)
- Caller ID Scheme: Bellcore/Telc... ▼
- \* FXO Dial Delay (...): 0

Figure 201: Configure Analog Trunk without Fax Detection

5. Go to UCM6200 Web GUI→**Extension/Trunk**→**Extensions** page.
6. Create or edit the extension for FXS port.
  - **Analog Station:** Select FXS port to be assigned to the extension. By default, it is set to "None".
  - Once selected, analog related settings for this extension will show up in "**Analog Settings**" section.



Create New Extension Save

**Basic Settings**    Media    Features    Specific Time    Follow Me

\* Select Extension Type:  ▼

Select Add Method:  ▼

---

General

\* Extension:

CallerID Number:

Enable Voicemail:

Skip Voicemail Passwo...

Analog Station:  ▼

\* Permission:  ▼

\* Voicemail Password:

Disable This Extension...

**Figure 202: Configure Extension for Fax Machine: FXS Extension**

Create New Extension Save

Basic Settings    **Media**    Features    Specific Time    Follow Me

Analog Settings

Call Waiting:

\* RX Gain:

\* MIN RX Flash:

Enable Polarity Reversal:

3-way Calling:

\* Fax Mode:  ▼

Use "# as SEND:

\* TX Gain:

\* MAX RX Flash:

\* Echo Cancellation:  ▼

\* Send CallerID After:  ▼

None

Fax Detection

**Fax Gateway**

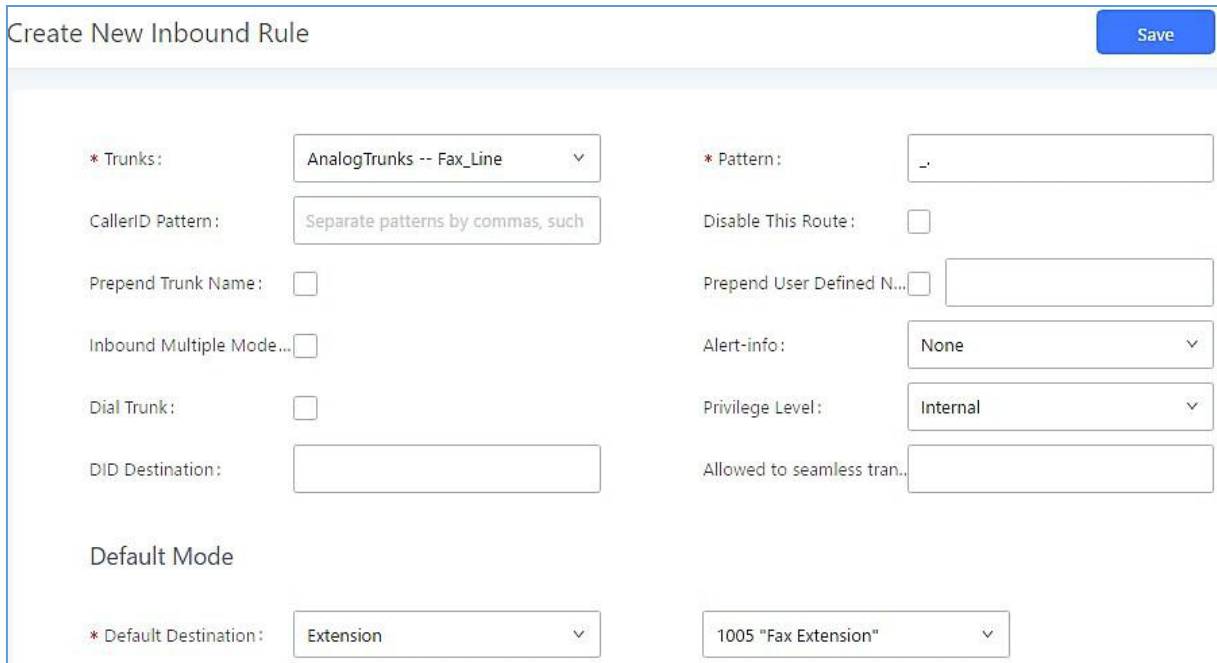
**Figure 203: Configure Extension for Fax Machine: Analog Settings**

7. Go to Web GUI→**Extension/Trunk**→**Inbound Routes** page.





8. Create an inbound route to use the Fax analog trunk. Select the created extension for Fax machine in step 4 as the default destination.



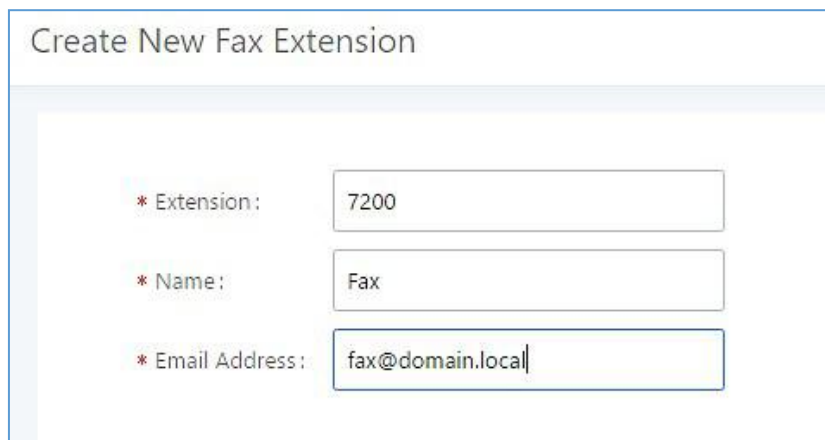
**Figure 204: Configure Inbound Rule for Fax**

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

### Sample Configuration for Fax-To-Email

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

1. Connect PSTN line to the UCM6200 FXO port.
2. Go to UCM6200 Web GUI → **Call Features** → **Fax/T.38** page. Create a new Fax extension.



**Figure 205: Create Fax Extension**



3. Go to UCM6200 Web GUI→**Extension/Trunk**→**Analog Trunks** page. Create a new analog trunk. Please make sure "Fax Detection" is set to "No".
4. Go to UCM6200 Web GUI→**Extension/Trunk**→**Inbound Routes** page. Create a new inbound route and set the default destination to the Fax extension.

Create New Inbound Rule
Save

<p>* Trunks: <input type="text" value="AnalogTrunks -- Fax_Line"/></p> <p>CallerID Pattern: <input type="text" value="Separate patterns by commas, such"/></p> <p>Prepend Trunk Name: <input type="checkbox"/></p> <p>Inbound Multiple Mode... <input type="checkbox"/></p> <p>Dial Trunk: <input type="checkbox"/></p> <p>DID Destination: <input type="text"/></p> <p>Default Mode</p> <p>* Default Destination: <input type="text" value="Fax"/></p>	<p>* Pattern: <input type="text" value="."/></p> <p>Disable This Route: <input type="checkbox"/></p> <p>Prepend User Defined N... <input type="checkbox"/></p> <p>Alert-info: <input type="text" value="None"/></p> <p>Privilege Level: <input type="text" value="Internal"/></p> <p>Allowed to seamless tran... <input type="text"/></p>
---	---

**Figure 206: Inbound Route to Fax Extension**

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

List of Fax Files

Delete Selected Fax Files
 Delete All

 Search

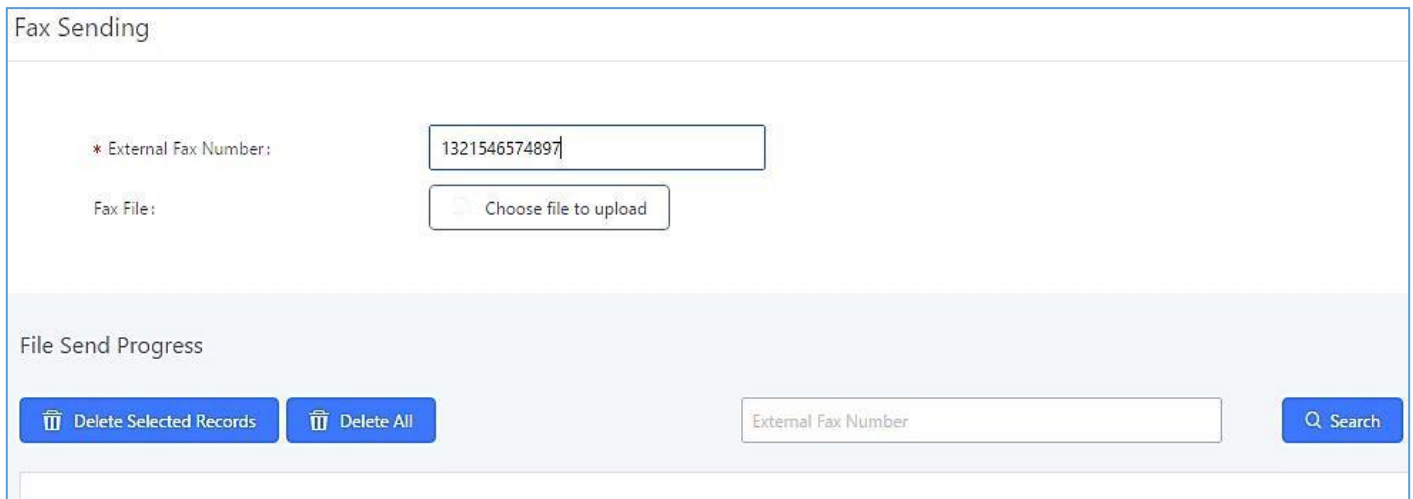
	Name	Date	Size	Options
<input type="checkbox"/>	VFAX-7200-20170511-101636-1494497796.5.pdf	2017-05-11 10:17:01 UTC+00:00	12834	

**Figure 207: List of Fax Files**

## FAX Sending

Besides the support of Fax machines, The UCM6200 supports also sending Fax via Web GUI access. This feature can be found on Web GUI→**Value-added Features**→**Fax Sending** page. To send fax, pre-setup for analog trunk and outbound route is required. Please refer to **[ANALOG TRUNKS]**, **[VOIP TRUNKS]** and **[Outbound Routes]** sections for configuring analog trunk and outbound route.

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



Fax Sending

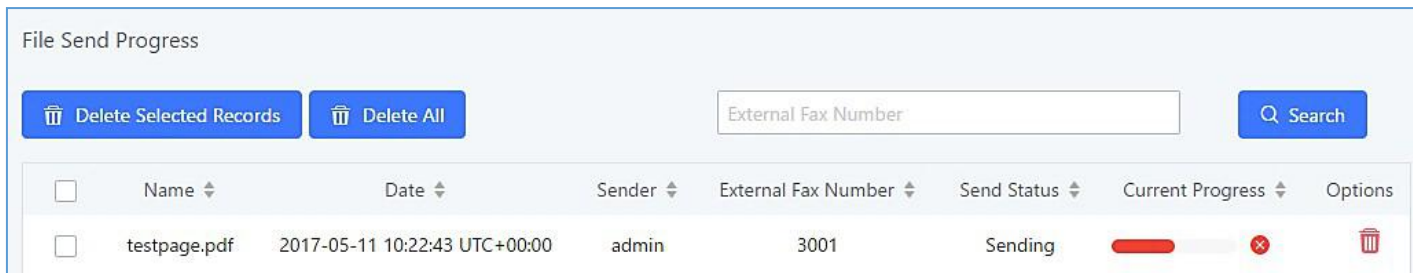
\* External Fax Number:

Fax File:

File Send Progress

**Figure 208: Fax Sending in Web GUI**

After that you can see the ongoing sending operation on the progress bar.



File Send Progress

<input type="checkbox"/>	Name ↕	Date ↕	Sender ↕	External Fax Number ↕	Send Status ↕	Current Progress ↕	Options
<input type="checkbox"/>	testpage.pdf	2017-05-11 10:22:43 UTC+00:00	admin	3001	Sending	<div style="width: 50%; background-color: red; height: 10px;"></div>	<input type="button" value="Delete"/>

**Figure 209: Fax Send Progress**

**Note:** Only A3, A4 and B4 paper sizes are supported for the Fax Sending.



## BUSY CAMP-ON

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

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

[http://www.grandstream.com/sites/default/files/Resources/ucm6xxx\\_busy\\_camp\\_on\\_guide.pdf](http://www.grandstream.com/sites/default/files/Resources/ucm6xxx_busy_camp_on_guide.pdf)



## PRESENCE

UCM does support SIP presence feature which allows users to advertise their current availability status and willingness to receive calls, this way other users can use their phones in order to monitor the presence status of each user and decide whether to call them or not based on their advertised availability.

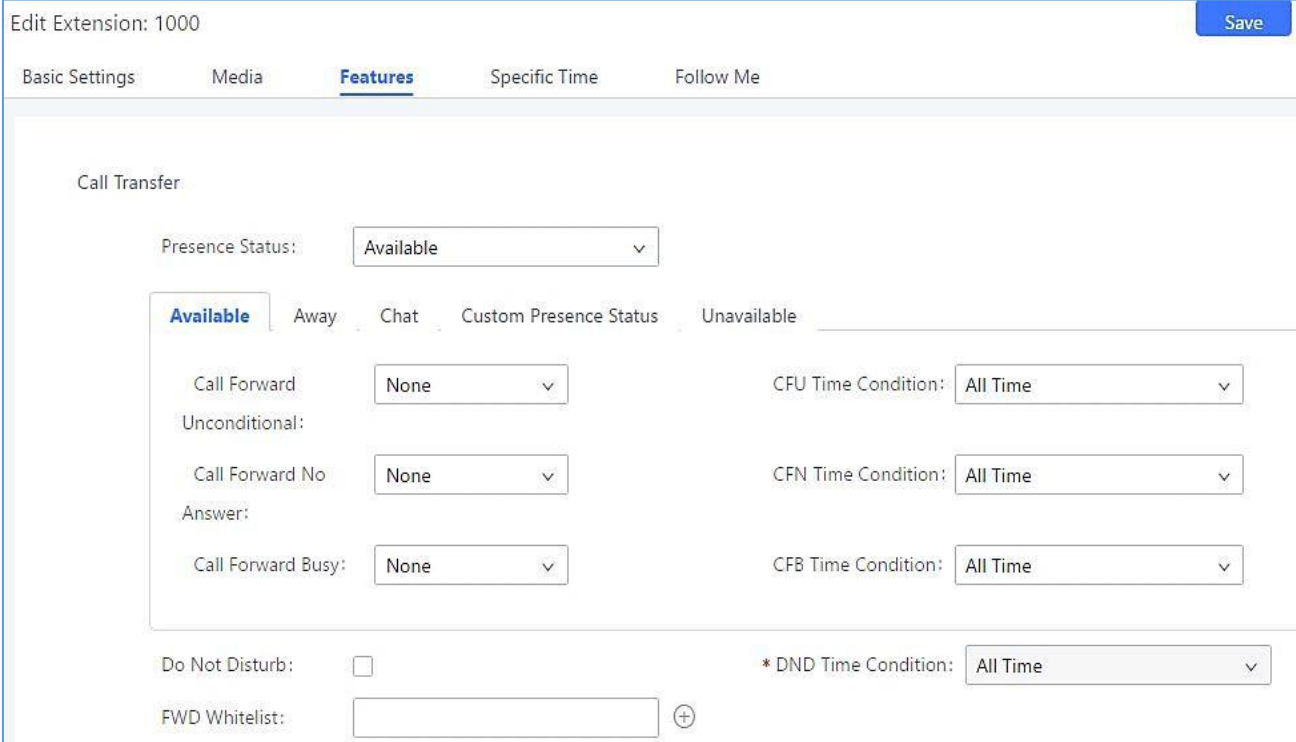
This feature is different than BLF which is mainly used to monitor the dialog status for each extension (Ringing, Idle or Busy). Instead the SIP presence module gives more options for users to choose which state they want to put themselves in.

In order to configure the presence status of an extension from the web GUI, users can access the menu of configuration using one of the two following methods:

- From admin account, go under the menu **Extension/Trunk**→**Extensions** and choose the desired extension to edit then navigate to the “Features” tab.

OR

- From the User Portal, go under the menu Basic **Information**→**Extensions** and navigate to the Features tab to have the following options.



Edit Extension: 1000 Save

Basic Settings   Media   **Features**   Specific Time   Follow Me

Call Transfer

Presence Status: Available ▾

**Available**   Away   Chat   Custom Presence Status   Unavailable

Call Forward: None ▾   CFU Time Condition: All Time ▾

Unconditional:   Call Forward No Answer: None ▾   CFN Time Condition: All Time ▾

Answer:   Call Forward Busy: None ▾   CFB Time Condition: All Time ▾

Do Not Disturb:    \* DND Time Condition: All Time ▾

FWD Whitelist:  ⊕

**Figure 210: SIP Presence Configuration**



Select which status to set from the presence status selection drop list, six options are available and below is a brief description of these states:

**Table 87: SIP Presence Status**

<b>Available</b>	The contact is online and can participate in conversations/phone calls.
<b>Away</b>	The contact is currently away (ex: for lunch break).
<b>Chat</b>	The contact has limited conversation flexibility and can only be reached via chat.
<b>Do Not Disturb</b>	The Contact is on DND (Do Not Disturb) mode.
<b>Custom Presence Status</b>	Please enter the presence status for this mode on the Web GUI. Up to 64 characters.
<b>Unavailable</b>	The contact is unreachable for the moment, please try to contact later.

Another option to set the presence status and which is more practical is using the feature code from the user's phone, one the user dials the feature code (default is \*48), a prompt will be played to select which status they want to put themselves in, by pressing the corresponding key.

The feature code can be enabled and customized from the Web GUI→**Call Features**→**Feature Codes**.

* Voicemail Access Code:	<input type="text" value="*98"/>	<input checked="" type="checkbox"/>	* My Voicemail:	<input type="text" value="*97"/>	<input checked="" type="checkbox"/>
* Agent Pause:	<input type="text" value="*83"/>	<input checked="" type="checkbox"/>	* Agent Unpause:	<input type="text" value="*84"/>	<input checked="" type="checkbox"/>
* Paging Prefix:	<input type="text" value="*81"/>	<input checked="" type="checkbox"/>	* Intercom Prefix:	<input type="text" value="*80"/>	<input checked="" type="checkbox"/>
* Blacklist Add:	<input type="text" value="*40"/>	<input checked="" type="checkbox"/>	* Blacklist Remove:	<input type="text" value="*41"/>	<input checked="" type="checkbox"/>
* Call Pickup on Ringing:	<input type="text" value="**"/>	<input checked="" type="checkbox"/>	* Pickup In-call:	<input type="text" value="*45"/>	<input type="checkbox"/>
* Pickup Extension:	<input type="text" value="*8"/>	<input checked="" type="checkbox"/>	* Direct Dial Voicemail Prefix:	<input type="text" value="*"/>	<input checked="" type="checkbox"/>
* Direct Dial Mobile Phone Prefix:	<input type="text" value="*88"/>	<input checked="" type="checkbox"/>	* Call Completion Request:	<input type="text" value="*11"/>	<input checked="" type="checkbox"/>
* Call Completion Cancel:	<input type="text" value="*12"/>	<input checked="" type="checkbox"/>	Enable Spy:	<input type="checkbox"/>	
* Listen Spy:	<input type="text" value="*54"/>	<input type="checkbox"/>	* Whisper Spy:	<input type="text" value="*55"/>	<input type="checkbox"/>
* Barge Spy:	<input type="text" value="*56"/>	<input type="checkbox"/>	* Wakeup Service:	<input type="text" value="*36"/>	<input checked="" type="checkbox"/>
* PMS Wakeup Service:	<input type="text" value="*35"/>	<input checked="" type="checkbox"/>	* Update PMS Room Status:	<input type="text" value="*23"/>	<input checked="" type="checkbox"/>
* Presence Status:	<input type="text" value="*48"/>	<input checked="" type="checkbox"/>	* Dynamic Agent Logout:	<input type="text" value="*85"/>	<input checked="" type="checkbox"/>

**Figure 211: SIP Presence Feature Code**

When a user does change his/her SIP presence status by making a call using presence feature code, the UCM will create a corresponding CDR entry showing the call as **Action type = PRSENCE\_STATUS**.



CDR Filter

By default, this page displays the CDR entries from the current month. Use the "Filter" button to specify a time range.

Delete All
Delete Search Result (s)
Download All Records
Download Search Result (s)
Automatic Download

Status	Call from	Call to	Action Type	Start Time	Call Time	Talk Time	Account Code	Recording File Options
+	"4004" 4004	*48	PRESENCE_STATUS	2018-01-30 05:53:02	0:00:30	0:00:30		-
+	"4002" 4002	4001	DIAL	2018-01-29 06:18:43	0:00:03	0:00:00		-
+	0622667315 [Trunk: GXP2160]	4001	DIAL	2018-01-29 06:13:01	0:00:14	0:00:03		-
+	"4001" 4001	5475 [Trunk: GXP2160]	DIAL	2018-01-29 04:47:13	0:00:05	0:00:04		-
+	"4001" 4001	65465476 [Trunk: GXP2160]	DIAL	2018-01-29 04:46:25	0:00:01	0:00:00		-
+	"4001" 4001	9	DIAL	2018-01-29 04:46:21	0:00:02	0:00:00		-

Total: 6 < 1 > 10 / page Goto 1


**Figure 212: Presence Status CDR**

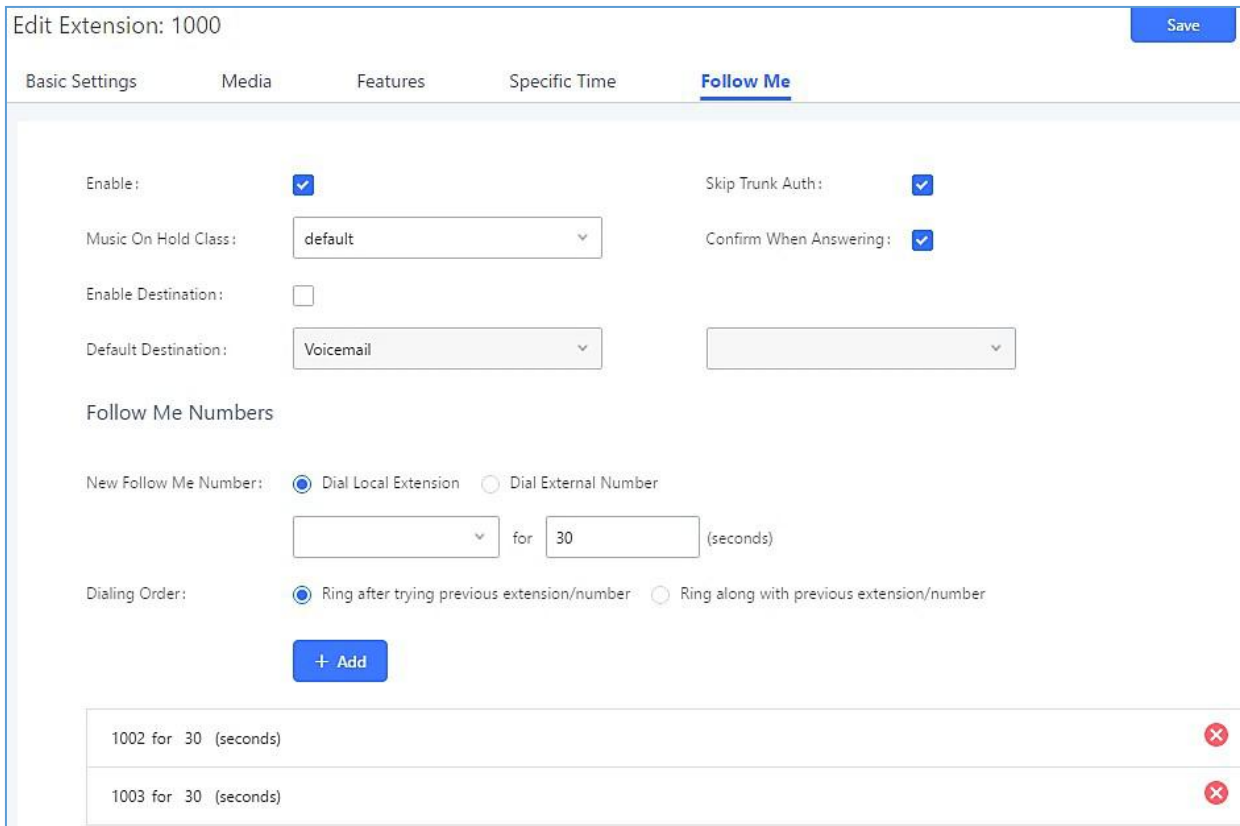


## FOLLOW ME

Follow Me is a feature on the UCM6200 that allows users to direct calls to other phone numbers and have them ring all at once or one after the other. Calls can be directed to users' home phone, office phone, mobile and etc. The calls will get to the user no matter where they are. Follow Me option can be found under extension settings page Web GUI → **Extension/Trunk** → **Extensions**.

To configure follow me:

1. Choose the extension and click on .
2. Go to the Follow me tab to add destination numbers and enable the feature.



**Edit Extension: 1000** Save

Basic Settings   Media   Features   Specific Time   **Follow Me**

Enable:    Skip Trunk Auth:

Music On Hold Class: default   Confirm When Answering:

Enable Destination:

Default Destination: Voicemail

**Follow Me Numbers**

New Follow Me Number:  Dial Local Extension    Dial External Number

for 30 (seconds)

Dialing Order:  Ring after trying previous extension/number    Ring along with previous extension/number

+ Add

1002 for 30 (seconds)	<span style="color: red; font-weight: bold;">✕</span>
1003 for 30 (seconds)	<span style="color: red; font-weight: bold;">✕</span>

**Figure 213: Edit Follow Me**





3. Click on + Add to add local extensions or external numbers to be called after ringing the extension selected in the first step.
4. Once created, it will be displayed on the follow me list. And you can click on ✕ to delete the Follow Me.

The following table shows the Follow Me configuration parameters:





**Table 88: Follow Me Settings**

<b>Enable</b>	Configure to enable or disable Follow Me for this user.
<b>Skip Trunk Auth</b>	If external number is added in the Follow Me, please make sure this option is enabled or the “Skip Trunk Auth” option of the extension is enabled, otherwise the external Follow Me number cannot be reached.
<b>Music On Hold Class</b>	Configure the Music On Hold class that the caller would hear while tracking users
<b>Confirm When Answering</b>	By default, it is enabled, and user will be asked to press 1 to accept the call or to press 2 to reject the call after answering a Follow Me call. If it is disabled, the Follow Me call will be established once after the user answers.
<b>Enable Destination</b>	When enabled, the call will be routed to the default destination if no one in the Follow Me extensions answers the call.
<b>Default Destination</b>	Configure the destination if no one in the Follow Me extensions answers the call. The available options are: <ul style="list-style-type: none"> <li>• Extension</li> <li>• Voicemail</li> <li>• Queues</li> <li>• Ring Group</li> <li>• Voicemail Group</li> <li>• IVR</li> <li>• External Number</li> </ul>
<b>Follow Me Numbers</b>	The added numbers are listed here. Click on   to arrange the order. Click on  to delete the number. Click on  to add new numbers.
<b>New Follow Me Number</b>	Add a new Follow Me number which could be a ‘Local Extension’ or ‘External Number’. The selected dial plan should have permissions to dial the defined external number.
<b>Dialing Order</b>	Select the order in which the Follow Me destinations will be dialed to reach the user: ring all at once or ring one after the other.

Click on “Follow Me Options” under Web GUI→**Extension/Trunk**→**Extension** page to enable or disable the options listed in the following table.

**Table 89: Follow Me Options**

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



## SPEED DIAL

The UCM6200 supports Speed Dial feature that allows users to call a certain destination by pressing one or four digits on the keypad. This creates a system-wide speed dial access for all the extensions on the UCM6200.

To enable Speed Dial, on the UCM6200 Web GUI, go to page Web GUI → **Call Features** → **Speed Dial**.

User should first click on + Add. Then decide from one digit up to four digits combination used for Speed Dial and select a dial destination from “Default Destination”. The supported destinations include extension, voicemail, conference room, voicemail group, IVR, ring group, call queue, page group, fax, DISA, Dial byName and external number.

### Create New Speed Dial

Enable

Destination:

\* Speed Dial

Extension:

Default Extension ▾ 1007 ▾

Destination:

**Figure 214: Speed Dial Destinations**

Extension ↕	Speed Dial	Default Destination	Default Destination	Options
1	Enable	Extension	1000	
2	Enable	Extension	1001	
3	Enable	External Number	0016175669300	
4	Enable	Ring Group	6400	
5	Enable	Queues	6500	

Total: 5 < 1 >
10 / page ▾ Goto




**Figure 215: List of Speed Dial**



## DISA

In many situations, the user will find the need to access his own IP PBX resources, but he is not physically near one of his extensions. However, he does have access to his own cell phone. In this case, we can use what is commonly known as DISA (Direct Inward System Access). Under this scenario, the user will be able to call from the outside, whether it is using his cell phone, pay phone, regular PSTN, etc. After calling into UCM6200, the user can then dial out via the SIP trunk or PSTN trunk connected to UCM6200 as it is an internal extension.

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

- Click on  to add a new DISA.
- Click on  to edit the DISA configuration.
- Click on  to delete the DISA.

### Create New DISA

* Name :	<input style="width: 90%;" type="text" value="Name"/>
* Password:	<input style="width: 90%;" type="password"/>
Permission :	<input style="border-bottom: 1px solid #ccc;" type="text" value="Internal"/>
* Response Timeout...	<input style="width: 90%;" type="text" value="10"/>
* Digit Timeout:	<input style="width: 90%;" type="text" value="5"/>
Allow Hang-up :	<input type="checkbox"/>
Replace Display Nam..	<input type="checkbox"/>

**Figure 216: Create New DISA**

The following table details the parameters to set and configure DISA feature on UCM6200 PBX.



**Table 90: DISA Settings**

<b>Name</b>	Configure DISA name to identify the DISA.
<b>Password</b>	Configure the password (digit only) required for the user to enter before using DISA to dial out.  <b>Note:</b> The password must be at least 4 digits.
<b>Permission</b>	Configure the permission level for DISA. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". If the user tries to dial outbound calls after dialing into the DISA, the UCM6200 will compare the DISA's permission level with the outbound route's privilege level. If the DISA's permission level is higher than (or equal to) the outbound route's privilege level, the call will be allowed to go through.
<b>Response Timeout</b>	Configure the maximum amount of time the UCM6200 will wait before hanging up if the user dials an incomplete or invalid number. The default setting is 10 seconds.
<b>Digit Timeout</b>	Configure the maximum amount of time permitted between digits when the user is typing the extension. The default setting is 5 seconds.
<b>Allow Hangup</b>	If enabled, during an active call, users can enter the UCM6200 Hangup feature code (by default it is *0) to disconnect the call or hang up directly. A new dial tone will be heard shortly for the user to make a new call. The default setting is "No".
<b>Replace Display Name</b>	If enabled, the UCM will replace the caller display name with the DISA name.

Once successfully created, users can configure the inbound route destination as "DISA" or IVR key event as "DISA". When dialing into DISA, users will be prompted with password first. After entering the correct password, a second dial tone will be heard for the users to dial out.



## EMERGENCY

UCM supports configuration and management of numbers to be called in emergency situation, thus bypassing the regular outbound call routing process and allowing users in critical situation to dial out for emergency help with the possibility to have redundant trunks as point of exit in case one of the lines is down.

UCM6xxx series are also now in full compliance with Kari's Law and Ray Baum's Act, for more information, please refer to the following links:


<https://www.fcc.gov/mlts-911-requirements>

[http://www.grandstream.com/sites/default/files/Resources/UCM\\_Emergency\\_Calls\\_Guide.pdf](http://www.grandstream.com/sites/default/files/Resources/UCM_Emergency_Calls_Guide.pdf)

In addition, Emergency calls can be automatically recorded by toggling on the new Auto Record and recordings can be viewed in the new Emergency Recordings tab on the same page. Additionally, users can have these recordings be sent to the configured email address(es).

Email alerts are also supported after enabling the notification for the event under “**Maintenance → System Events**”

To configure emergency numbers, users need to follow below steps:

1. Navigate on the web GUI under “**Call Features → Emergency Calls**”
2. Click on  to add a new emergency number.
3. Configure the required fields “Name, Emergency Number and Trunk(s) to be used to reach the number”.
4. Save and apply the configuration.



### Create New Emergency Call

\* Name:

\* Emergency Number:

Emergency Level:

Disable Hunt on Busy:

Custom Prompt:  [Prompt](#)

\* Use Trunks:

\* Members Notified:

11 items Available		1 item Selected	
<input type="checkbox"/>	1001 "John Doe"	<input type="checkbox"/>	1000 "James tuan"
<input type="checkbox"/>	1002		
<input type="checkbox"/>	1003		
<input type="checkbox"/>	1004		

Strip:

Prepend:

Auto Record:

Send Recording File:

Email Address:  [+](#)

**Figure 217: Emergency Number Configuration**

The table below gives more description of the configuration Parameters when creating emergency numbers.



**Table 91: Emergency Numbers Parameters**

<b>Name</b>	Configure the name of the emergency call. For example, "emergency911", "emergency211" and etc.
<b>Emergency Number</b>	Config the emergency service number. For example, "911", "211" and etc.
<b>Emergency Level</b>	Select the emergency level of the number. Level "3" means the most urgent.
<b>Disable Hunt on Busy</b>	If this option is not enabled, when the lines of trunks which the coming emergency call routes by are completely occupied, the line-grabbing function will automatically cut off a line from all busy lines so that the coming emergency call can seize it for dialing out. This option is not enabled by default.
<b>Custom Prompt</b>	This option sets a custom prompt to be used as an announcement to the person receiving an emergency call. The file can be uploaded from the page "Custom Prompt". Click "Prompt" to add additional record.
<b>Use Trunks</b>	Select the trunks for the emergency call. Select one trunk at least and select five trunks at most.
<b>Members Notified</b>	Select the members who will be notified when an emergency call occurs.
<b>Strip</b>	Specify the number of digits that will be Stripped from the beginning of the dialed number before the call is placed via the selected trunk. <b>Note:</b> Users can now strip the same amount of numbers as the emergency number length itself.
<b>Prepend</b>	Specify the digits to be Prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
<b>Auto Record</b>	When enabled, emergency call will be automatically recorded.
<b>Send Recording File</b>	When enabled recording files will be sent to the configured email address.
<b>Email Address</b>	The email address to where the recording files will be sent.



Emergency Calls				
+ Add				
Name ↕	Emergency Number ↕	Emergency Level ↕	Disable Hunt on Busy ↕	Options
911	911	1	No	 

**Figure 218: 911 Emergency Sample**





## CALLBACK

Callback is mainly designed for users who often use their mobile phones to make long distance or international calls which may have high service charges. The callback feature provides an economic solution for reduce the cost from this.

The callback feature works as follows:

1. Configure a new callback on the UCM6200.
2. On the UCM6200, configure destination of the inbound route for analog trunk to callback.
3. Save and apply the settings.
4. The user calls the PSTN number of the UCM6200 using the mobile phone, which goes to callback destination as specified in the inbound route.
5. Once the user hears the ringback tone from the mobile phone, hang up the call on the mobile phone.
6. The UCM6200 will call back the user.
7. The user answers the call.
8. The call will be sent to DISA or IVR which directs the user to dial the destination number.
9. The user will be connected to the destination number.

In this way, the calls are placed and connected through trunks on the UCM6200 instead of to the mobile phone directly. Therefore, the user will not be charged on mobile phone services for long distance or international calls.

To configure callback on the UCM6200, go to Web GUI→**Call Features**→**Callback** page and click on

[+ Create New Callback](#)

. Configuration parameters are listed in the following table.

**Table 92: Callback Configuration Parameters**

<b>Name</b>	Configure a name to identify the Callback. (Enter at least two characters)
<b>CallerID Pattern</b>	Configure the pattern of the callers allowed to use this callback. The caller who places the inbound call needs to have the CallerID match this pattern so that the caller can get callback after hanging up the call. <b>Note:</b> If leaving as blank, all numbers are allowed to use this callback.
<b>Outbound Prepend</b>	Configure the prepend digits to be added at before dialing the outside number. The number with prepended digits will be used to match the outbound route. '-' is the connection character which will be ignored.
<b>Delay Before Callback</b>	Configure the number of seconds to be delayed before calling back the user.
<b>Destination</b>	Configure the destination which the callback will direct the caller to. Two destinations are available: <ul style="list-style-type: none"> <li>• IVR</li> <li>• DISA</li> </ul> The caller can then enter the desired number to dial out via UCM6200 trunk.



## BLF AND EVENT LIST

### BLF

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

---



 **Note:**

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

---

### Event List

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

- Click on "Create New Event List" to add a new event list.
- Sort selected extensions manually in the Eventlist
- Click on  to edit the event list configuration.
- Click on  to delete the eventlist.



### Create New Event List

**\* URI:**

**Event Type:**

**Local Extensions:**

4/5 items Available

Search here

- 1000 "John DOE"
- 1001
- 1002
- 1003
- 1004

0 item Selected

Search here

None

**Remote Extensions:**

0 item Available

Search here

None

0 item Selected

Search here

None

**Special Extensions:**

**Figure 219: Create New Event List**

**Table 93: Event List Settings**

<b>URI</b>	Configure the name of this event list (for example, office_event_list). Please note the URI name cannot be the same as the extension name on the UCM6200. The valid characters are letters, digits, _ and -.
<b>Local Extensions</b>	Select the available extensions/Extension Groups listed on the local UCM6200 to be monitored in the event list.
<b>Remote Extensions</b>	If LDAP sync is enabled between the UCM6200 and the peer UCM6200, the remote extensions will be listed under "Available Extensions". If not, manually enter the remote extensions under "Special Extensions" field.



**Special Extensions**

Manually enter the remote extensions in the peer/register trunk to be monitored in the event list. Valid format: 5000,5001,9000

Remote extension monitoring works on the UCM6200 via event list BLF, among Peer SIP trunks or Register SIP trunks (register to each other). Therefore, please properly configure SIP trunks on the UCM6200 first before using remote BLF feature. Please note the SIP end points need support event list BLF in order to monitor remote extensions.

When an event list is created on the UCM6200 and remote extensions are added to the list, the UCM6200 will send out SIP SUBSCRIBE to the remote UCM6200 to obtain the remote extension status. When the SIP end points register and subscribe to the local UCM6200 event list, it can obtain the remote extension status from this event list. Once successfully configured, the event list page will show the status of total extension and subscribers for each event list. Users can also select the event URI to check the monitored extension's status and the subscribers' details.

**Notes:**

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

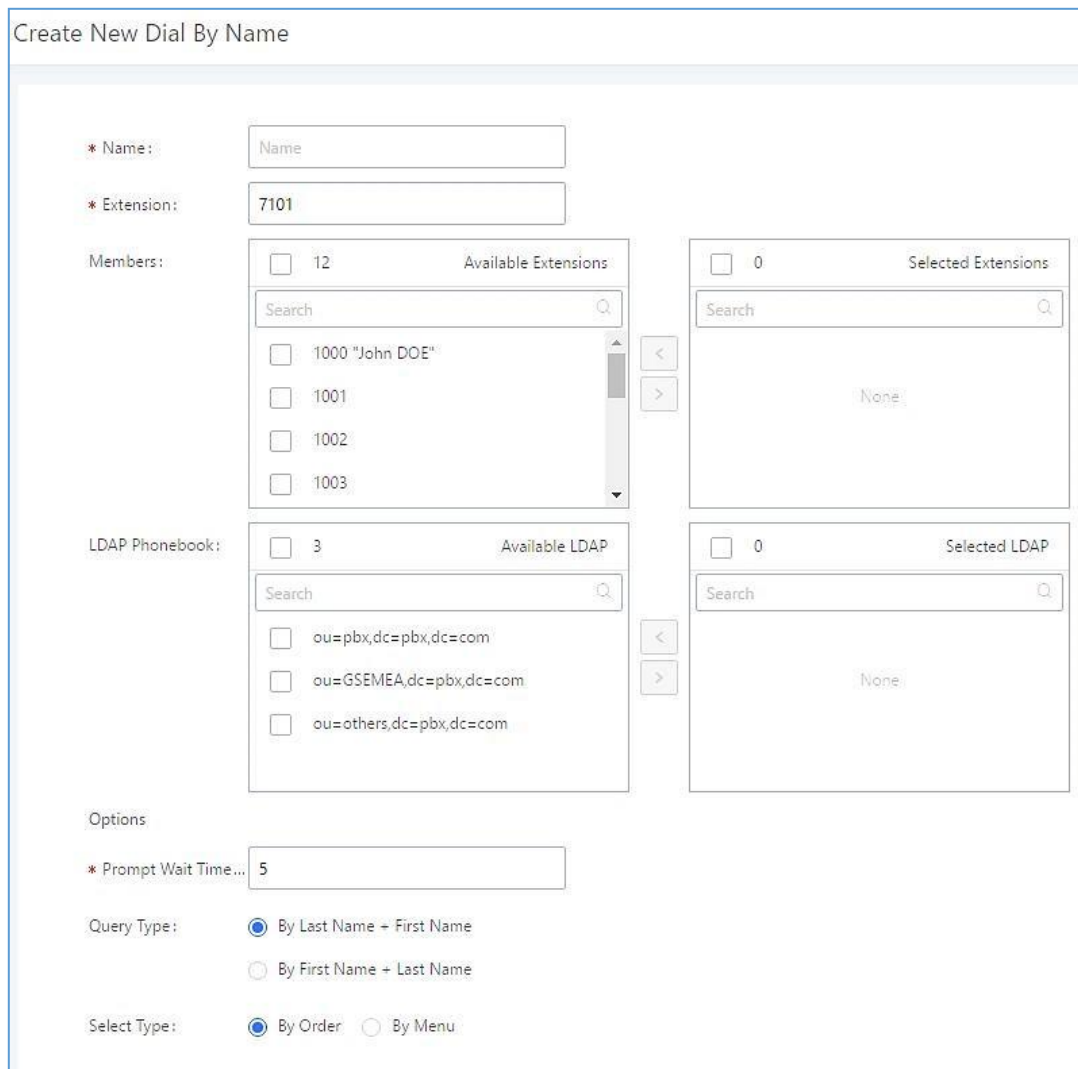


## DIAL BY NAME

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

### Dial by Name Configuration

The administrators can create the dial by name group under Web GUI→**Call Features**→**Dial By Name**.

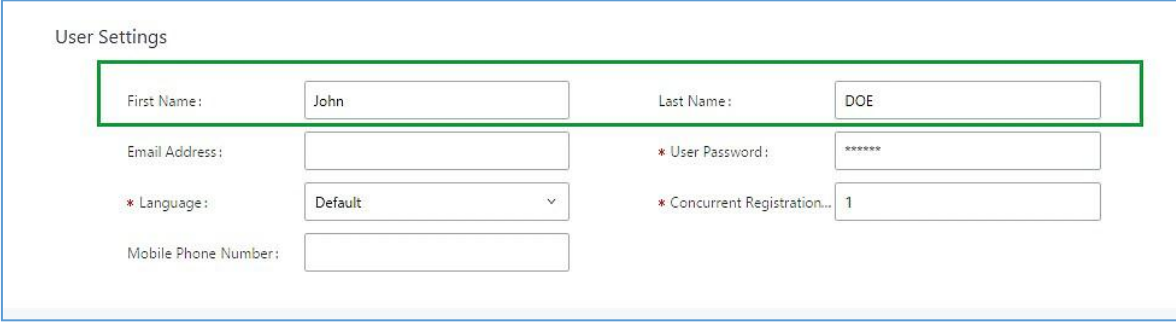


The screenshot shows the 'Create New Dial By Name' configuration page. It includes the following fields and sections:

- \* Name:** Text input field containing 'Name'.
- \* Extension:** Text input field containing '7101'.
- Members:** A list of available extensions with checkboxes. The list shows 12 available extensions, with a search bar and a list containing '1000 "John DOE"', '1001', '1002', and '1003'. There are left and right arrow buttons between the available and selected lists.
- Selected Extensions:** A list of selected extensions with checkboxes, currently showing 0 selected extensions and a search bar.
- LDAP Phonebook:** A list of available LDAP entries with checkboxes. The list shows 3 available LDAP entries, with a search bar and a list containing 'ou=pbx,dc=pbx,dc=com', 'ou=GSEMEA,dc=pbx,dc=com', and 'ou=others,dc=pbx,dc=com'. There are left and right arrow buttons between the available and selected lists.
- Selected LDAP:** A list of selected LDAP entries with checkboxes, currently showing 0 selected LDAP entries and a search bar.
- Options:**
  - \* Prompt Wait Time...:** Text input field containing '5'.
  - Query Type:** Radio buttons for 'By Last Name + First Name' (selected), 'By First Name + Last Name'.
  - Select Type:** Radio buttons for 'By Order' (selected), 'By Menu'.

**Figure 220: Create Dial by Name Group**





User Settings

First Name:	<input type="text" value="John"/>	Last Name:	<input type="text" value="DOE"/>
Email Address:	<input type="text"/>	* User Password:	<input type="password" value="*****"/>
* Language:	<input type="text" value="Default"/>	* Concurrent Registration...	<input type="text" value="1"/>
Mobile Phone Number:	<input type="text"/>		

**Figure 221: Configure Extension First Name and Last Name**

### 1. Name

Enter a Name to identify the Dial By Name group.

### 2. Extension

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

### 3. Custom Prompt

This option sets a custom prompt for directory to announce to a caller. The file can be uploaded from the page "Custom Prompt". Click "Upload Audio File" to add additional record.

### 4. Available Extensions/Selected Extensions

Select available extensions from the left side to the right side as the directory for the Dial By Name group. Only the selected extensions here can be reached by the Dial By Name IVR when dialing into this group. The extensions here must have a valid first name and last name configured under Web GUI→**Extension/Trunk**→**Extensions** in order to be searchable in Dial By Name directory through IVR. By specifying the extensions here, the administrators can make sure unscreened calls will not reach the company employee if he/she does not want to receive them directly.

### 5. Prompt Wait Time

Configure "Prompt Wait Time" for Dial By Name feature. During Dial By Name call, the caller will need to input the first letters of First/Last name before this wait time is reached. Otherwise, timeout will occur, and the call might hang up. The timeout range is between 3 and 60 seconds.

### 6. Query Type

Specify the query type. This defines how the caller will need to enter to search the directory.

By First Name: enter the first 3 digits of the first name to search the directory.

By Last Name: enter the first 3 digits of the last name to search the directory.

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



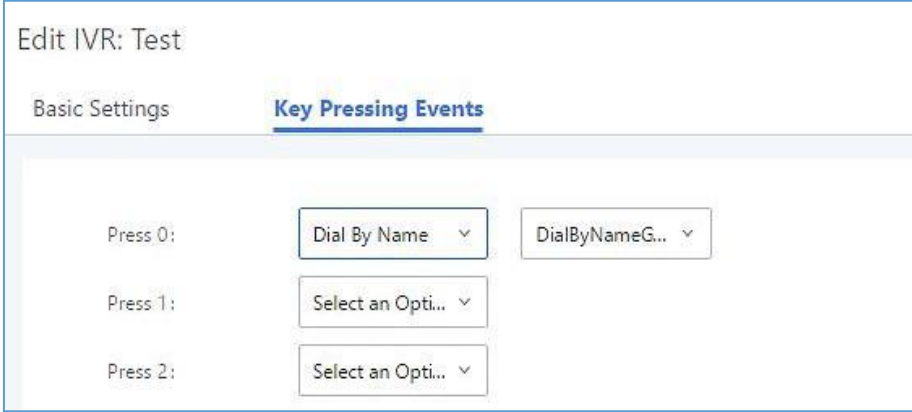
## 7. Select Type

Specify the select type on the searching result. The IVR will confirm the name/number for the party the caller would like to reach before dialing out.

**By Order:** After the caller enters the digits, the IVR will announce the first matching party's name and number. The caller can confirm and dial out if it is the destination party, or press \* to listen to the next matching result if it is not the desired party to call.

**By Menu:** After the caller enters the digits, the IVR will announce 8 matching results. The caller can press number 1 to 8 to select and call or press 9 for results in next page.

The Dial by Name group can be used as the destination for inbound route and key pressing event for IVR. The group name defined here will show up in the destination list when configuring IVR and inbound route. If Dial by Name is set as a key pressing event for IVR, user could use "\*" to exit from Dial by Name, then re-enter IVR and start a new event. The following example shows how to use this option.



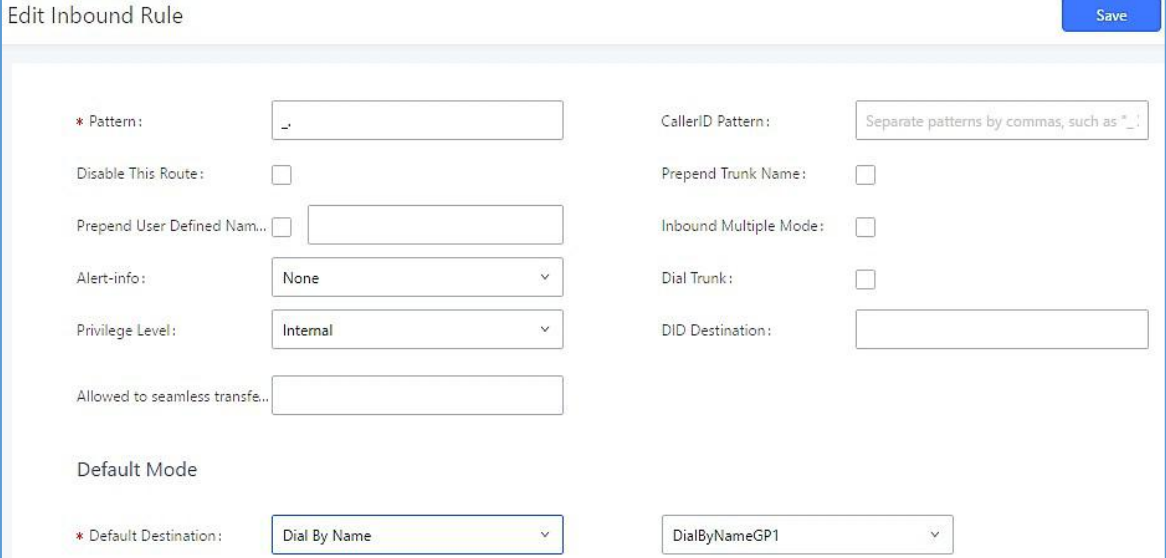
Edit IVR: Test

Basic Settings      **Key Pressing Events**

Press 0:	Dial By Name ▾	DialByNameG... ▾
Press 1:	Select an Opti... ▾	
Press 2:	Select an Opti... ▾	

**Figure 222: Dial By Name Group In IVR Key Pressing Events**





**Edit Inbound Rule** Save

\* Pattern:

CallerID Pattern:

Disable This Route:

Prepend Trunk Name:

Prepend User Defined Nam...

Inbound Multiple Mode:

Alert-info:

Dial Trunk:

Privilege Level:

DID Destination:

Allowed to seamless transfe...

Default Mode

\* Default Destination:



**Figure 223: Dial By Name Group In Inbound Rule**

## Username Prompt Customization

Starting from fw 1.0.15.x, the Dial By Name feature can use the recorded name prompt of a user to announce his/her name assigned to the dialed extension. If no name prompt greeting exists, the name will be spelt out like in previous versions.



There are two ways to customize/set new username prompt for an extension:

### Upload Username Prompt File from Web GUI

1. First, Users should have a pre-recorded file respecting the following format:
  - PCM encoded / 16 bits / 8000Hz mono.
  - In .GSM or .WAV format.
  - File size under 5M.
  - Filename must be set as the extension number. For example, the recorded file name 1000.wav will be used for extension 1000.
2. Go under web GUI **PBX Settings** → **Voice Prompt** → **Username Prompt** and click on  button.
3. Select the recorded file to upload it and press Save and Apply Settings.
  - Click on  to record again the username prompt.
  - Click on to play recorded username prompt.





- Select username prompts and press  to delete specific file or select multiple files for deletion using the button .

### **Record Username via Voicemail Menu**

The second option to record username is using voicemail menu, please follow below steps:

- Dial \*98 to access the voicemail
- After entering the desired extension and voicemail password, dial “0” to enter the recordings menu and then “3” to record a name.

Another option is that each user can record their own name by following below steps:

- The user dials \*97 to access his/her voicemail
- After entering the voicemail password, the user can press “0” to enter the recordings menu and then “3” to record his name.

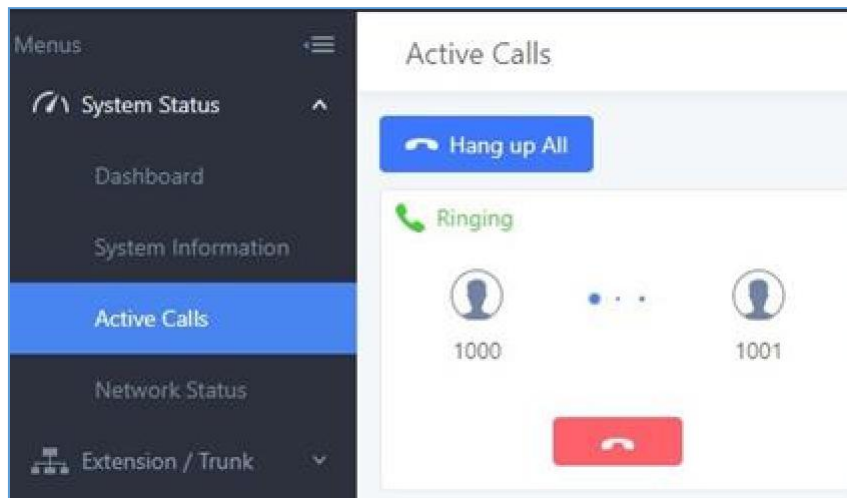


## ACTIVE CALLS AND MONITOR

The active calls on the UCM6200 are displayed in Web GUI→**System Status**→**Active Calls** page. Users can monitor the status, hang up the call as well as barge in the active calls in real time manner.

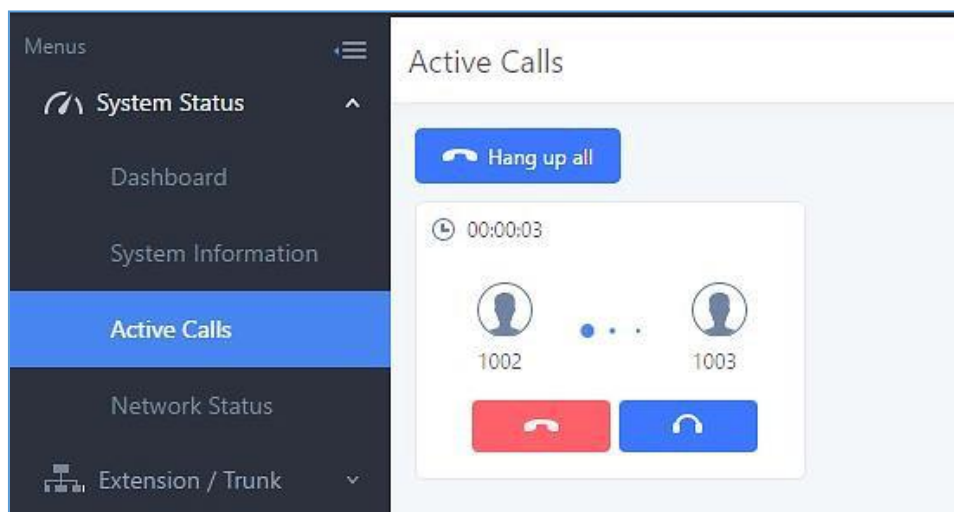
### Active Calls Status

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



**Figure 224: Status→PBX Status→Active Calls - Ringing**

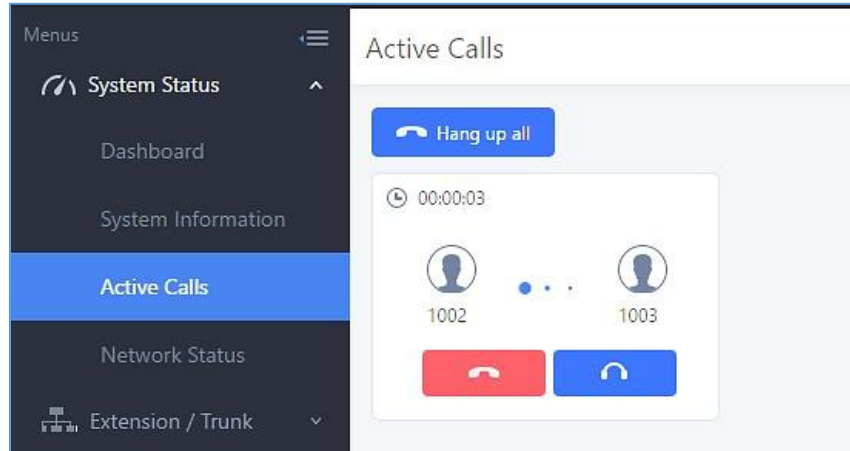
The following figure shows the call between 1002 and 1003 is established.



**Figure 225: Status→PBX Status→Active Calls – Call Established**

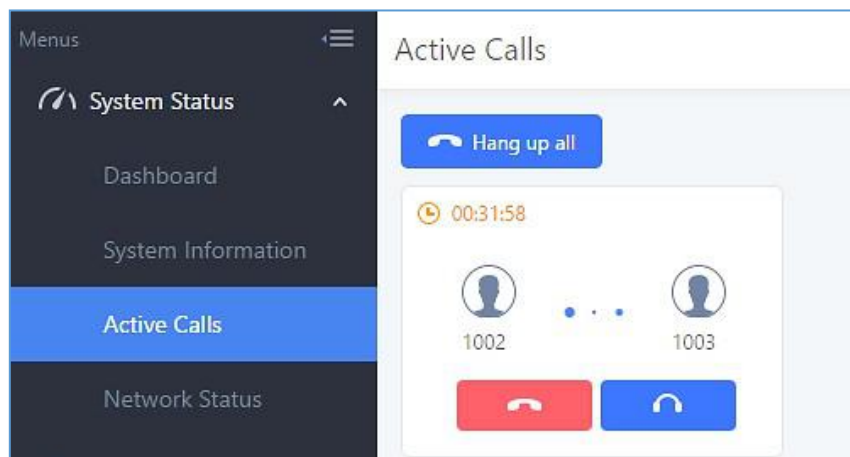


The green color of the active call means the connection of call time is less than half an hour. It means this call is normal.



**Figure 226: Call Connection less than half hour**

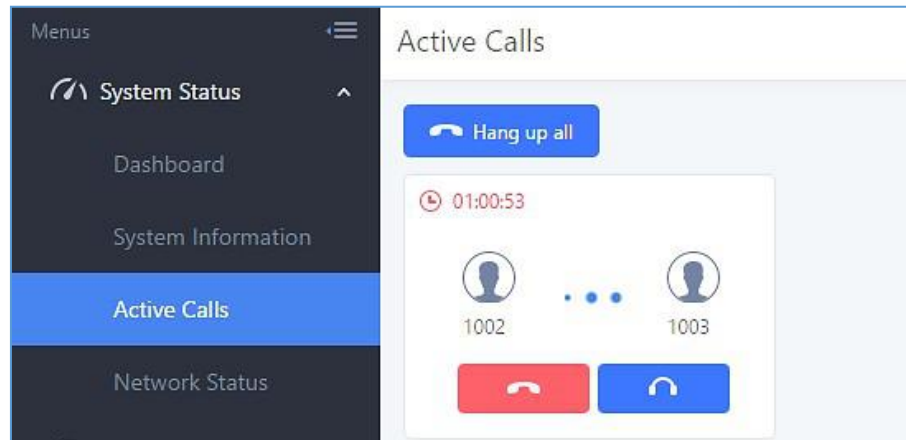
The yellow color of the active call means the connection of call time is greater than half an hour but less than one hour. It means this call is a bit long.



**Figure 227: Call Connection between half an hour and one hour**



The red color of the active call means the connection of call time is more than one hour. It means this call could be abnormal.





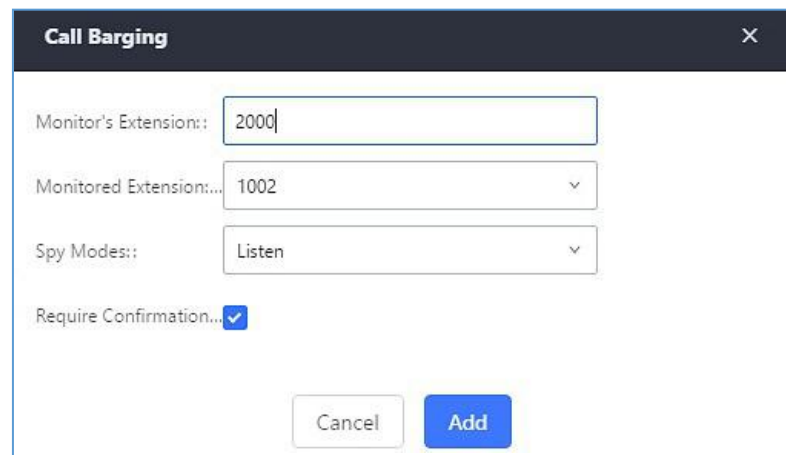
**Figure 228: Call Connection more than one hour**

## Hang Up Active Calls

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

## Call Monitor

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



**Figure 229: Configure to Monitor an Active Call**

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

1. Enter an available extension for “Monitor’s Extension” which will be used to monitor the active call.
2. “Monitored Extension” must be one of the parties in the active call to be monitored.
3. Select spy mode. There are three options in “Spy Mode”.



- **Listen**

In “Listen” mode, the extension monitoring the call can hear both parties in the active call but the audio of the user on this extension will not be heard by either party in the monitored active call.

- **Whisper**

In “Whisper” mode, the extension monitoring the call can hear both parties in the active call. The user on this extension can only talk to the selected monitored extension and he/she will not be heard by the other party in the active call. This can be usually used to supervise calls.

- **Barge**

In “Barge” mode, the extension monitoring the call can talk to both parties in the active call. The call will be established similar to three-way conference.

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

Another way to monitor active calls is to dial the corresponding feature codes from an extension. Please refer to **[Table 94: UCM6200 Feature Codes]** and **[Call Recording]** section for instructions.



## CALL FEATURES

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

### Feature Codes

**Table 94: UCM6200 Feature Codes**

Feature Maps	
<b>Blind Transfer</b>	<ul style="list-style-type: none"> <li>• Default code: <b>#1</b></li> <li>• Enter the code during active call. After hearing "Transfer", you will hear dial tone. Enter the number to transfer to. Then the user will be disconnected, and transfer is completed.</li> <li>• Options:               <ul style="list-style-type: none"> <li><b>Disable</b></li> <li><b>Allow Caller:</b> Enable the feature code on caller side only.</li> <li><b>Allow Callee:</b> Enable the feature code on callee side only.</li> <li><b>Allow Both:</b> Enable the feature code on both caller and callee.</li> </ul> </li> </ul>
<b>Attended Transfer</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*2</b></li> <li>• Enter the code during active call. After hearing "Transfer", you will hear the dial tone. Enter the number to transfer to and the user will be connected to this number. Hang up the call to complete the attended transfer. In case of the called party does not answer, users could press *0 to cancel the call and retrieve the first call leg.</li> <li>• Options:               <ul style="list-style-type: none"> <li><b>Disable</b></li> <li><b>Allow Caller:</b> Enable the feature code on caller side only.</li> <li><b>Allow Callee:</b> Enable the feature code on callee side only.</li> <li><b>Allow Both:</b> Enable the feature code on both caller and callee.</li> </ul> </li> </ul>
<b>Seamless Transfer</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*44</b> (Disabled by default).</li> <li>• Seamless Transfer allows user to perform blind transfer using UCM feature code without having music on hold presented during the transfer process, it minimizes the interruption during transfer, making the process smooth and simple.</li> <li>• During an active call use the feature code (*44 by default) followed by the number you want to transfer to in order to perform the seamless transfer.</li> </ul>



<b>Disconnect</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*0</b></li> <li>• Enter the code during active call. It will disconnect the call.</li> <li>• Options:  <b>Disable</b>  <b>Allow Caller:</b> Enable the feature code on caller side only.  <b>Allow Callee:</b> Enable the feature code on callee side only.  <b>Allow Both:</b> Enable the feature code on both caller and callee.</li> </ul>
<b>Call Park</b>	<ul style="list-style-type: none"> <li>• Default code: <b>#72</b></li> <li>• Enter the code during active call to park the call.</li> <li>• Options:  <b>Disable</b>  <b>Allow Caller:</b> Enable the feature code on caller side only.  <b>Allow Callee:</b> Enable the feature code on callee side only.  <b>Allow Both:</b> Enable the feature code on both caller and callee.</li> </ul>
<b>Audio Mix Record</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*3</b></li> <li>• Enter the code followed by # or SEND to start recording the audio call and the UCM6200 will mix the streams natively on the fly as the call is in progress.</li> <li>• Options:  <b>Disable</b>  <b>Allow Caller:</b> Enable the feature code on caller side only.  <b>Allow Callee:</b> Enable the feature code on callee side only.  <b>Allow Both:</b> Enable the feature code on both caller and callee.</li> </ul>
<b>Feature Code Digits Timeout</b>	Set the maximum interval (ms) between digits for feature code activation
<b>DND/Call Forward</b>	
<b>Do Not Disturb (DND) Activate</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*77</b></li> </ul>
<b>Do Not Disturb (DND) Deactivate</b>	<ul style="list-style-type: none"> <li>• Default code: <b>*78</b></li> </ul>
<b>Call Forward Busy Activate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*90</b></li> <li>• Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.</li> </ul>
<b>Call Forward Busy Deactivate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*91</b></li> </ul>
<b>Call Forward No Answer Activate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*92</b></li> <li>• Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.</li> </ul>



<b>Call Forward No Answer Deactivate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*93</b></li> </ul>
<b>Call Forward Unconditional Activate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*72</b></li> <li>• Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.</li> </ul>
<b>Call Forward Unconditional Deactivate</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*73</b></li> </ul>
<b>Remote Call Forward Enable</b>	<ul style="list-style-type: none"> <li>• If enabled, this option will allow specific extensions to dial the remote call forwarding feature codes to set call forwarding for any extension.</li> </ul>
<b>Remote Call Forward Busy Enable</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*65</b></li> <li>• Enter the code and follow the voice prompt to set the remote extension number where you want to enable Call Forward Busy and the target destination.</li> </ul>
<b>Remote Call Forward Busy Disable</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*651</b></li> <li>• Enter the code and follow the voice prompt to set the remote extension number where you want to disable the Call Forward Busy.</li> </ul>
<b>Remote Call Forward No Answer Enable</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*66</b></li> <li>• Enter the code and follow the voice prompt to set the remote extension number where you want to enable Call Forward No Answer and the target destination.</li> </ul>
<b>Remote Call Forward No Answer Disable</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*661</b></li> <li>• Enter the code and follow the voice prompt to set the remote extension number where you want to disable the Call Forward No Answer.</li> </ul>
<b>Remote Call Forward Unconditional Enable</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*67</b></li> <li>• Enter the code and follow the voice prompt to set the remote extension number where you want to enable Call Forward Unconditional and the target destination.</li> </ul>
<b>Remote Call Forward Unconditional Disable</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*671</b></li> <li>• Enter the code and follow the voice prompt to set the remote extension number where you want to disable the Call Forward Unconditional.</li> </ul>
<b>Remote Call Forward Whitelist</b>	<p>Only the Extensions selected in this whitelists can configure call forwarding for any extension via feature codes.</p>





Feature Codes	
<b>Voicemail Access Code</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*98</b></li> <li>• Enter *98 and follow the voice prompt. Or dial *98 followed by the extension and # to access the entered extension's voicemail box.</li> </ul>
<b>My Voicemail</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*97</b></li> <li>• Press *97 to access the voicemail box.</li> </ul>
<b>Agent Pause</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*83</b></li> <li>• Pause the agent in all call queues.</li> </ul>
<b>Agent Unpause</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*84</b></li> <li>• Unpause the agent in all call queues.</li> </ul>
<b>Paging Prefix</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*81</b></li> <li>• To page an extension, enter the code followed by the extension number.</li> </ul>
<b>Intercom Prefix</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*80</b></li> <li>• To intercom an extension, enter the code followed by the extension number.</li> </ul>
<b>Blacklist Add</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*40</b></li> <li>• To add a number to blacklist for inbound route, dial *40 and follow the voice prompt to enter the number.</li> </ul>
<b>Blacklist Remove</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*41</b></li> <li>• To remove a number from current blacklist for inbound route, dial *41 and follow the voice prompt to remove the number.</li> </ul>
<b>Call Pickup on Ringing</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>**</b></li> <li>• To pick up a call for any extension xxxx, enter the code followed by the extension number xxxx.</li> </ul>
<b>Pickup In-call</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*45</b> (Disabled by default).</li> <li>• If “Pickup In-call” feature is enabled, only the extensions added in “Allowed to seamless transfer” in the extension’s Seamless Transfer Privilege Control List” can pick up the call.</li> </ul>
<b>Pickup Extension</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*8</b></li> <li>• This code is for the pickup group, which can be assigned for each extension on the extension configuration page.</li> <li>• If there is an incoming call to an extension, the other extensions within the same pickup group can dial *8 directly to pick up the call.</li> </ul>

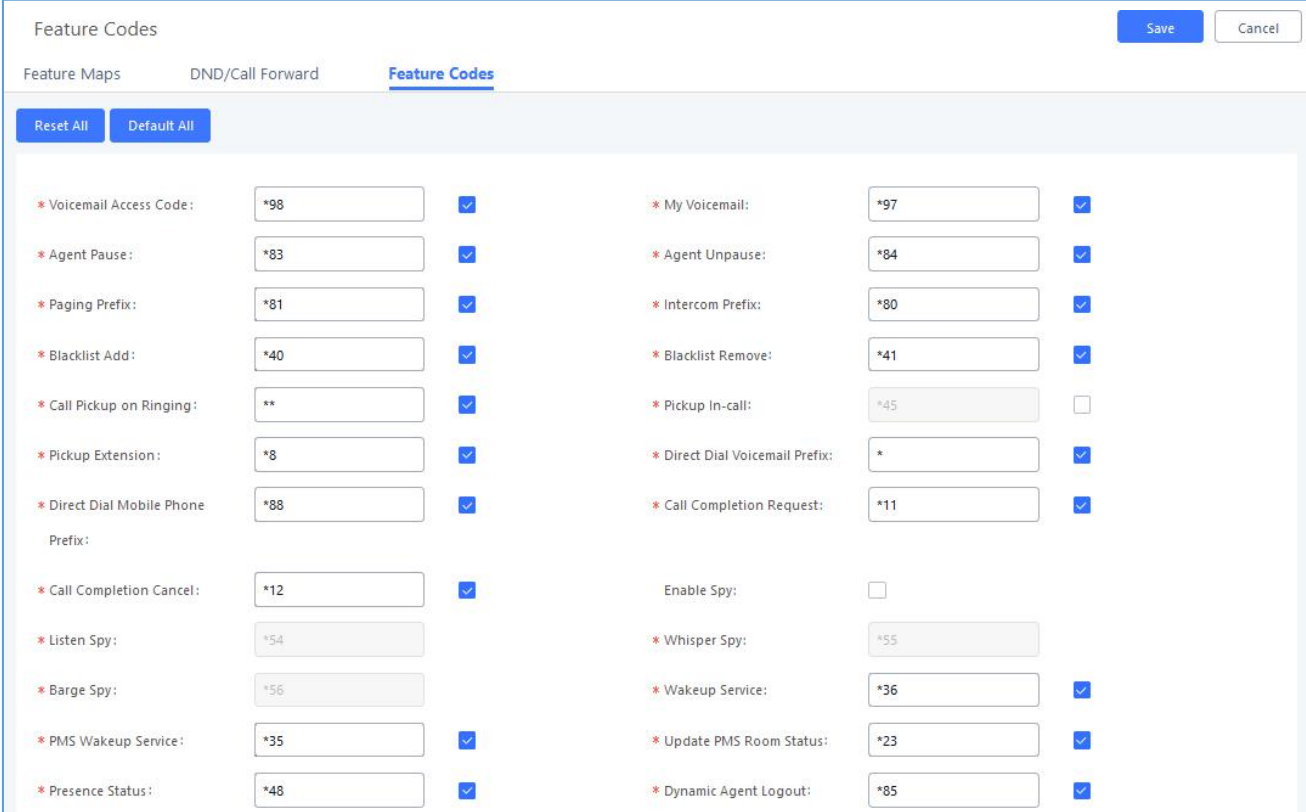


<b>Direct Dial Voicemail Prefix</b>	<ul style="list-style-type: none"> <li>• Default Code: *</li> <li>• This code is for the user to directly dial or transfer to an extension's voicemail.</li> <li>• For example, directly dial *5000 will have to call go into the extension 5000's voicemail. If the user would like to transfer the call to the extension 5000's voicemail, enter *5000 as the transfer target number.</li> </ul>
<b>Direct Dial Mobile Phone Prefix</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*88</b></li> <li>• If you have the permission to call mobile phone number, use this prefix plus the extension number can dial the mobile phone number of this extension directly.</li> </ul>
<b>Call Completion Request</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*11</b></li> <li>• This code is for the user who wants to use Call Completion to complete a call.</li> </ul>
<b>Call Completion Cancel</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*12</b></li> <li>• This code is for the user who wants to cancel Call Completion request.</li> </ul>
<b>Enable Spy</b>	<p>Check this box to enable spy feature codes.          Disabled by default.</p>
<b>Listen Spy</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*54</b> ("Enable Spy" needs to be checked)</li> <li>• This is the feature code to listen in on a call to monitor performance. Monitor's line will be muted, and neither party will hear from the monitor's extension.</li> </ul>
<b>Whisper Spy</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*55</b> ("Enable Spy" needs to be checked)</li> <li>• This is the feature code to speak to one side of the call (for example, whisper to employees to help them handle a call). Only one side will be able to hear from the monitor's extension.</li> </ul>
<b>Barge Spy</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*56</b> ("Enable Spy" needs to be checked)</li> <li>• This is the feature code to join in on the call to assist both parties.</li> </ul>
<b>Wakeup Service</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*36</b></li> <li>• Dial this code to access UCM wakeup service, you can add, update, activate or deactivate wakeup service.</li> </ul>
<b>PMS Wakeup Service</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*35</b></li> <li>• Dial this code to access UCM PMS wakeup service, you can add, update, activate or deactivate PMS wakeup service.</li> </ul>



<b>Update PMS Room Status</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*23</b></li> <li>• Use this code with maid code to update PMS room status. Choose the status to set after hearing the prompt, for example: for maid 001 dial *23001 and then 1 after hearing the prompt.</li> </ul>
<b>Presence Status</b>	<ul style="list-style-type: none"> <li>• Dial this code to set the presence status of the extension.</li> <li>• Possible options are:           <ul style="list-style-type: none"> <li>1:"unavailable"</li> <li>2:"available"</li> <li>3:"away"</li> <li>4:"chat"</li> <li>5:"dnd"</li> <li>6:"userdef"</li> </ul> </li> </ul>
<b>Dynamic Agent Logout</b>	<ul style="list-style-type: none"> <li>• Default Code: <b>*85</b></li> <li>• Use this code to logout the dynamic agent from all queues.</li> </ul>

The UCM6200 also allows user to one click enable / disable specific feature code as shown below:



The screenshot shows the 'Feature Codes' configuration page. It includes a 'Save' button and a 'Cancel' button. The page is divided into sections for 'Feature Maps', 'DND/Call Forward', and 'Feature Codes'. The 'Feature Codes' section contains a list of feature codes with their respective values and checkboxes for enabling or disabling them.

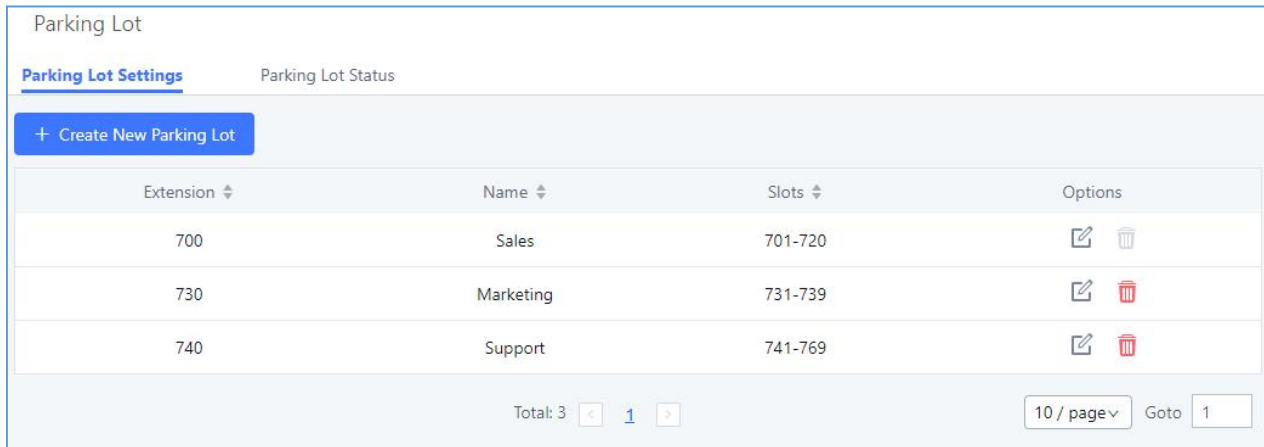
Feature Code	Value	Enabled
* Voicemail Access Code:	*98	<input checked="" type="checkbox"/>
* Agent Pause:	*83	<input checked="" type="checkbox"/>
* Paging Prefix:	*81	<input checked="" type="checkbox"/>
* Blacklist Add:	*40	<input checked="" type="checkbox"/>
* Call Pickup on Ringing:	**	<input checked="" type="checkbox"/>
* Pickup Extension:	*8	<input checked="" type="checkbox"/>
* Direct Dial Mobile Phone Prefix:	*88	<input checked="" type="checkbox"/>
* Call Completion Cancel:	*12	<input checked="" type="checkbox"/>
* Listen Spy:	*54	<input type="checkbox"/>
* Barge Spy:	*56	<input type="checkbox"/>
* PMS Wakeup Service:	*35	<input checked="" type="checkbox"/>
* Presence Status:	*48	<input checked="" type="checkbox"/>
* My Voicemail:	*97	<input checked="" type="checkbox"/>
* Agent Unpause:	*84	<input checked="" type="checkbox"/>
* Intercom Prefix:	*80	<input checked="" type="checkbox"/>
* Blacklist Remove:	*41	<input checked="" type="checkbox"/>
* Pickup In-call:	*45	<input type="checkbox"/>
* Direct Dial Voicemail Prefix:	*	<input checked="" type="checkbox"/>
* Call Completion Request:	*11	<input checked="" type="checkbox"/>
Enable Spy:		<input type="checkbox"/>
* Whisper Spy:	*55	<input type="checkbox"/>
* Wakeup Service:	*36	<input checked="" type="checkbox"/>
* Update PMS Room Status:	*23	<input checked="" type="checkbox"/>
* Dynamic Agent Logout:	*85	<input checked="" type="checkbox"/>

Figure 230: Enable/Disable Feature codes



## Parking Lot

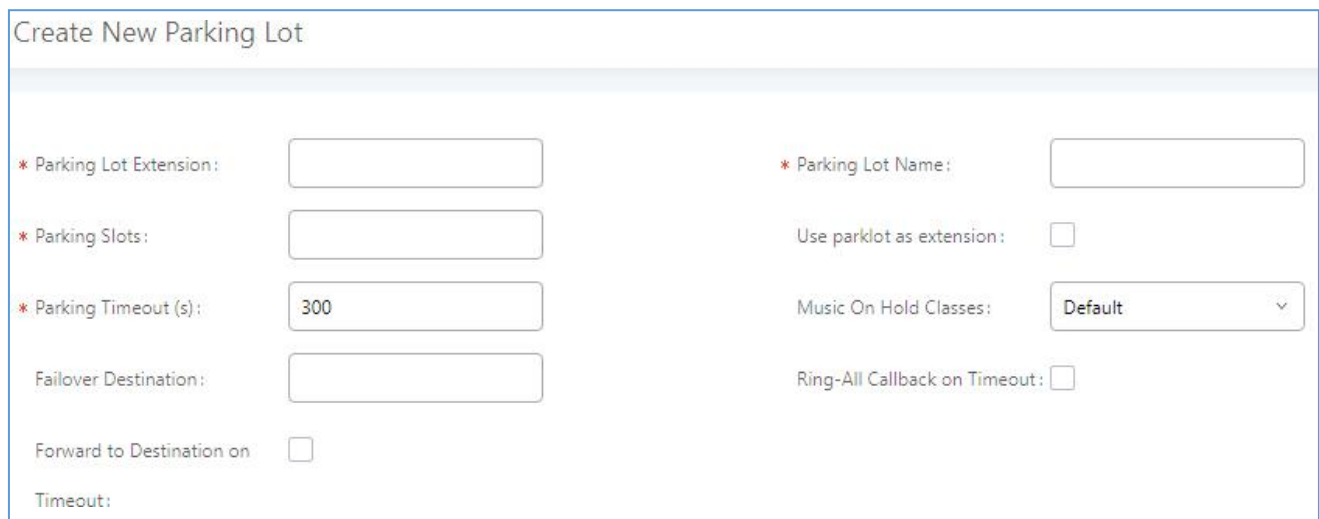
User can create parking lots and their related slots under Web GUI → **Call Features** → **Parking Lot**. In the Parking Lot page, users can create lots of their own. This allows different groups within an organization to have their own parking lots instead of sharing one large parking lot with others. While creating a new parking lot, users can assign it a range that they think is appropriate for the group that will use the parking lot.



Parking Lot			
Parking Lot Settings		Parking Lot Status	
+ Create New Parking Lot			
Extension	Name	Slots	Options
700	Sales	701-720	
730	Marketing	731-739	
740	Support	741-769	
Total: 3			10 / page Goto 1

Figure 231: Parking Lot

User can create a new Parking lot by clicking on button “Create New Parking Lot” .



Create New Parking Lot

* Parking Lot Extension:	<input type="text"/>	* Parking Lot Name:	<input type="text"/>
* Parking Slots:	<input type="text"/>	Use parklot as extension:	<input type="checkbox"/>
* Parking Timeout (s):	<input type="text" value="300"/>	Music On Hold Classes:	<input type="text" value="Default"/>
Failover Destination:	<input type="text"/>	Ring-All Callback on Timeout:	<input type="checkbox"/>
Forward to Destination on Timeout:	<input type="checkbox"/>		

Figure 232: New Parking Lot

Table 95 : Parking Lot

<b>Parking Lot Extension</b>	<ul style="list-style-type: none"> <li>• Default Extension: <b>700</b></li> <li>• During an active call, initiate blind transfer and then enter this code to park the call.</li> </ul>
<b>Parking Lot Name</b>	<ul style="list-style-type: none"> <li>• Set a name to the parking lot</li> </ul>



<b>Parked Slots</b>	<ul style="list-style-type: none"> <li>• Default Extension: <b>701-720</b></li> <li>• These are the extensions where the calls will be parked, i.e., parking lots that the parked calls can be retrieved.</li> </ul>
<b>Use Parklot as Extension</b>	<ul style="list-style-type: none"> <li>• If checked, the parking lot number can be used as extension. The user can transfer the call to the parking lot number to park the call. Please note this parking lot number range might conflict with extension range.</li> </ul>
<b>Parking Timeout (s)</b>	<ul style="list-style-type: none"> <li>• Default setting is <b>300</b> seconds and the maximum limit is <b>99.999</b> seconds.</li> <li>• This is the timeout allowed for a call to be parked. After the timeout, if the call is not picked up, the extension who parks the call will be called back.</li> </ul>
<b>Music On Hold Classes</b>	Select the Music on Hold Class.
<b>Failover Destination</b>	<p>Configures a callback failover destination when the extension that is called back is busy. The call will be routed to the destination number and this reduces the chance of dropping parked calls.</p> <p><b>Note:</b> This field cannot exceed 32 characters.</p>
<b>Ring All Callback on Timeout</b>	<p>If enabled, all registered endpoints of the extension will ring when callback occurs. Otherwise, only the original endpoint will be called back.</p> <p><b>Note:</b> This option will not be available if Forward to Destination on Timeout is enabled.</p>
<b>Forward to destination on timeout</b>	If enabled, the call will be routed to the configured destination upon timeout. Otherwise, the call will be routed back to the original caller.
<b>Timeout Destination</b>	This option appears once Forward to Destination on Timeout is enabled. Upon park timeout, the call will be routed to the configured destination.

## Call Park

The UCM6200 provides call park and call pickup features via feature code.

### Park a Call

There are two feature codes that can be used to park the call.

- **Feature Maps→Call Park (Default code #72)**  
 During an active call, press #72 and the call will be parked. Parking lot number (default range 701 to 720) will be announced after parking the call.



- **Feature Misc→Call Park (Default code 700)**

During an active call, initiate blind transfer (default code #1) and then dial 700 to park the call. Parking lot number (default range 701 to 720) will be announced after parking the call.

### Retrieve Parked Call

To retrieve the parked call, simply dial the parking lot number and the call will be established. If a parked call is not retrieved after the timeout, the original extension who parks the call will be called back.

### Monitor Call Park CID Name Information (GXP21xx Phones Only)

Users can see the CID name information of parked calls. VPK/MPKs must be configured as “Monitored Call Park” with the desired parking lot extension. The display will alternate between displaying the parking lot extension and the call’s CID name. There is no need to configure anything on the UCM.



**Note:** This feature requires Grandstream GXP21xx new firmware support. This new firmware is not available in Grandstream website yet. Please check GXP21xx upcoming firmware release information for availability.



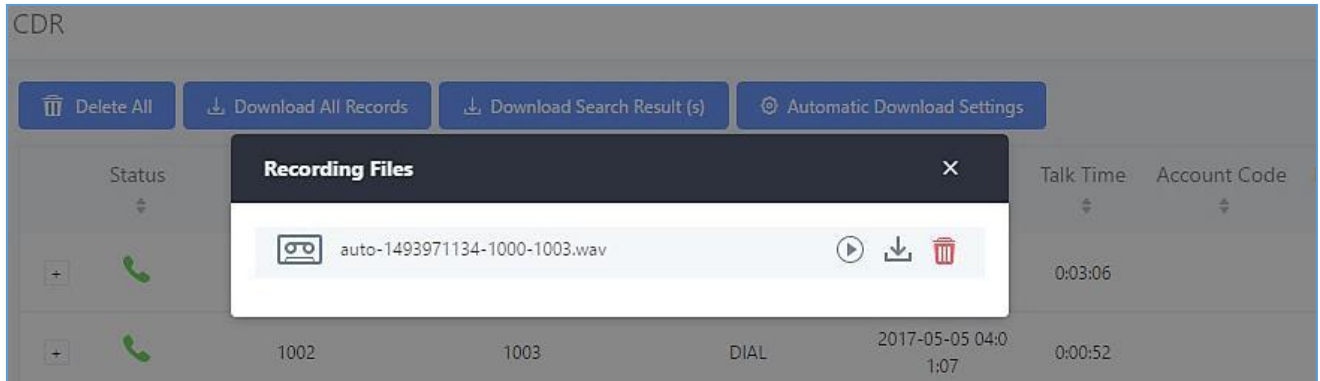
Figure 233: Monitored call park CID name

### Call Recording

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

1. Make sure the feature code for "Audio Mix Record" is configured and enabled.
2. After establishing the call, enter the "Audio Mix Record" feature code (by default it is \*3) followed by # or SEND to start recording.
3. To stop the recording, enter the "Audio Mix Record" feature code (by default it is \*3) followed by # or SEND again. Or the recording will be stopped once the call hangs up.
4. The recording file can be retrieved under Web GUI→**CDR**. Click on  to show and play the recording or click on  to download the recording file.

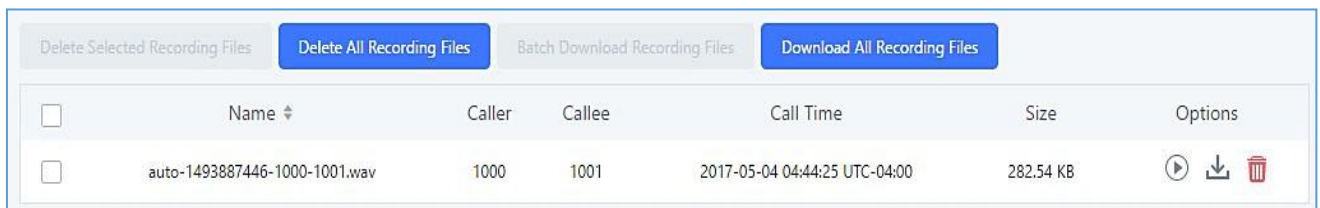




**Figure 234: Download Recording File from CDR Page**

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

**Note:** Starting firmware 1.0.20.17, Music on Hold will be also included in the recording.



**Figure 235: Download Recording File from Recording Files Page**

## Enable Spy

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

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

### **Caution:**

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

## Shared Call Appearance (SCA)

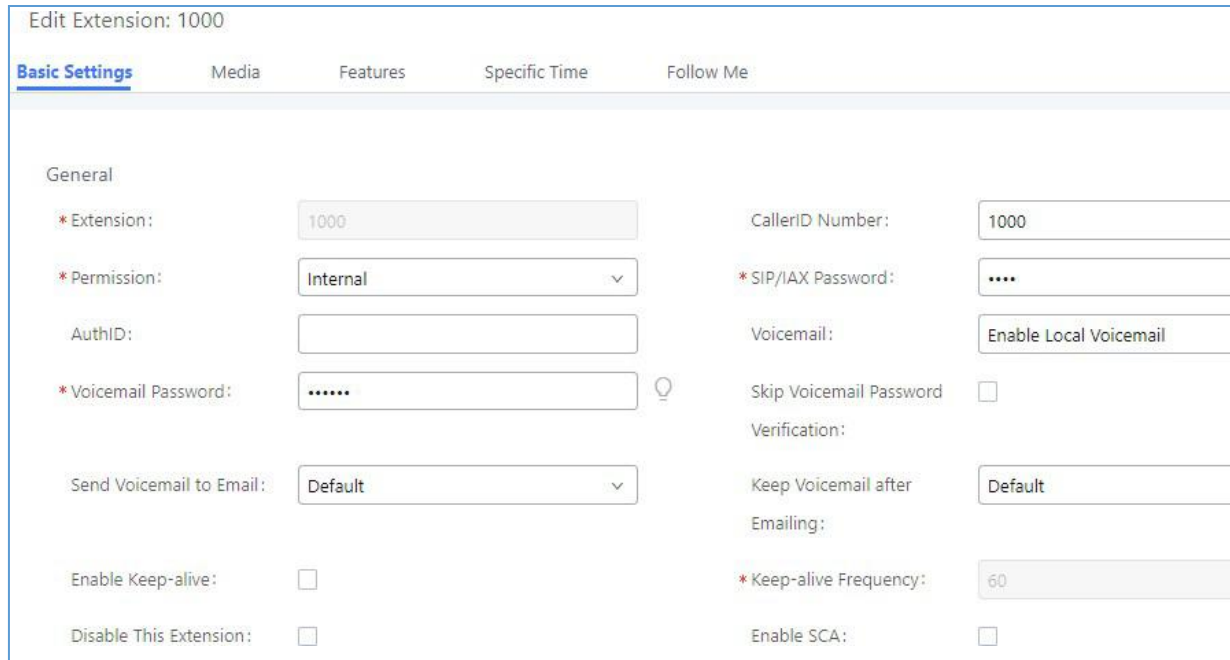
Shared Call Appearance (SCA) functionality has been added to the UCM. With SCA, users can assign multiple devices to one extension, configure endpoints to monitor that extension, make actions on behalf of that extension such as viewing call status and placing and receiving calls, and even barging into existing calls. To configure the



SCA functionality, please follow the steps below:

1. Users can enable SCA by navigating to the Extensions page, editing the desired extension, and enabling the option SCA.

**Note:** With SCA enabled, the Concurrent Registrations field can only have a value of 1.



Edit Extension: 1000

**Basic Settings**   Media   Features   Specific Time   Follow Me

General

\* Extension: 1000      CallerID Number: 1000

\* Permission: Internal      \* SIP/IAX Password: \*\*\*\*

AuthID:      Voicemail: Enable Local Voicemail

\* Voicemail Password: \*\*\*\*\*      Skip Voicemail Password Verification:

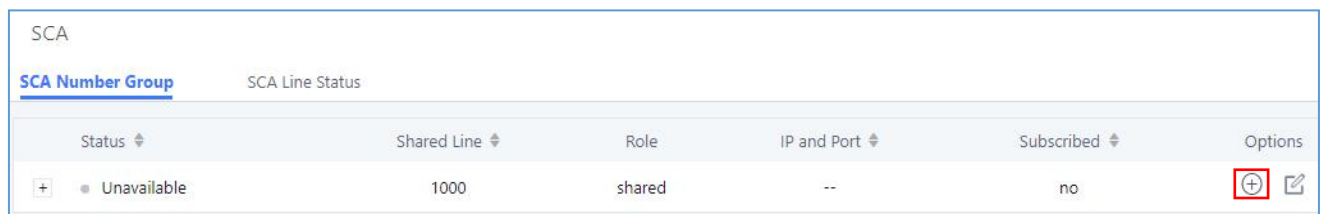
Send Voicemail to Email: Default      Keep Voicemail after Emailing: Default

Enable Keep-alive:       \* Keep-alive Frequency: 60

Disable This Extension:       Enable SCA:


**Figure 236: Enabling SCA option under Extension's Settings**

2. After enabling the option, navigate to *Call Features* → *SCA*. The newly enabled SCA extension will be listed. Click the "+" button under the Options column to add a number that will share the main extension's call appearance, which will be called private numbers.



SCA

**SCA Number Group**      SCA Line Status

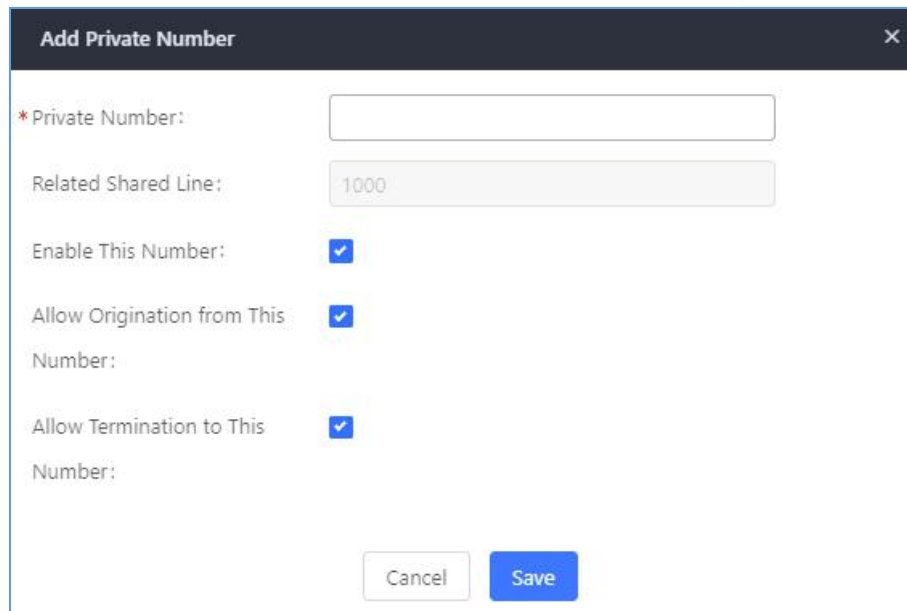
Status	Shared Line	Role	IP and Port	Subscribed	Options
+ Unavailable	1000	shared	--	no	<input checked="" type="checkbox"/> 

**Figure 237: SCA Number Configuration**

3. Configure the private number as desired.







**Add Private Number** [X]

\* Private Number:

Related Shared Line: 1000

Enable This Number:

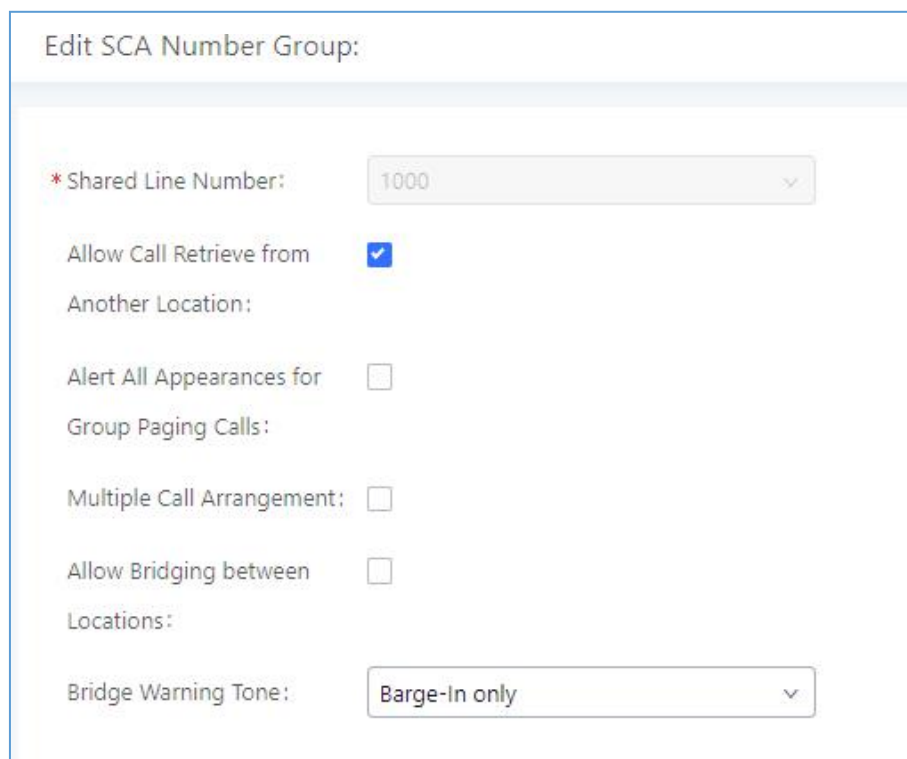
Allow Origination from This Number:

Allow Termination to This Number:

Cancel Save

**Figure 238: SCA Private Number Configuration**

- Once the private number has been created, users must now register a device to it. To properly register a device to the private number, use the configured private number as the SIP User ID. Auth ID and Password will be the same as the main extension's. Once registration is complete, SCAs are now configured.



**Edit SCA Number Group:**

\* Shared Line Number: 1000

Allow Call Retrieve from Another Location:

Alert All Appearances for Group Paging Calls:

Multiple Call Arrangement:

Allow Bridging between Locations:

Bridge Warning Tone: Barge-In only

**Figure 239: SCA Options**



SCA has various options to change its behavior:

- **Allow Call Retrieve from Another Location** – Allows users to retrieve held calls using any device associated with the SCA extension.
- **Alert All Appearances for Group Paging Calls** – Alerts all devices associated with the SCA extension.
- **Multiple Call Arrangement** – Allows all devices associated with the SCA extension to make different calls at the same time.
- **Allow Bridging between Locations** – Allows devices associated with the SCA extension to barge into existing calls of the same SCA group.
- **Bridge Warning Tone** – Notification sound that will play when a party barges into the call. Three options are available:
  - **None** – No notification sound will play
  - **Barge-In Only** – The notification sound will play once when a party barges in.
  - **Barge-In and Repeat** – The notification sound will play when a party barges in and will play again after every 30 seconds.

5. Next, configure the VPK or MPK to Shared for both the main extension and the private number. SCA is now configured for both endpoint devices.

The following table describe the SCA Number configuration setting:

**Table 96: Add SCA Private Number**

<b>Private Number</b>	Configures the private number for the SCA.
<b>Related Shared Line</b>	Display the related shared line.
<b>Enable This Number</b>	Whether enable this private number. If not enabled, this private number is only record in DB, it will not affect other system feature.
<b>Allow Origination from This Number</b>	Enable this option will allow calling from this private number. By default, it is enabled.
<b>Allow Termination to This Number</b>	Enable this option will allows calls to this private number. By default, it is enabled.

The following table describes the options available when editing the SCA number:

**Table 97: Editing the SCA Number**

<b>Shared Line Number</b>	While SCA is enabled, this number will be the same as the extension number.
---------------------------	---

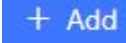


<b>Allow Call Retrieve from Another Location</b>	Allows remote call retrieval. Must be enabled in public hold. By default, it is enabled.
<b>Alert All Appearances for Group Paging Calls</b>	Allows all SCA group members to ring when the SCA shared number is paged. If disabled, only the SCA shared number will ring when paged. By default, it is disabled.
<b>Multiple Call Arrangement</b>	Allows simultaneous calls in an SCA group. By default, it is disabled.
<b>Allow Bridging between Locations</b>	Allows location bridging for SCA group. Must be enabled when using the Barge-In feature. By default, it is disabled.
<b>Bridge Warning Tone</b>	<p>Configures the notification in the bridge when another party join.</p> <ul style="list-style-type: none"> <li>• <b>None:</b> No notification sound.</li> <li>• <b>Barge-In only:</b> Notification sound will play when another party join.</li> <li>• <b>Barge-In and Repeat:</b> Notification sound will play when another party joins and repeat every 30 seconds.</li> </ul> <p>By default, it is set to “Barge-In Only”.</p>

## 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] or in [IVR].

To configure the Announcement, users need to follow below steps:

1. Navigate on the web GUI under “**Call Features → Announcement**”
2. Click on  to add a new Announcement.
3. Configure the required fields Name, Prompt, Default Destination to be used for the announcement.
4. Save and apply the configuration.



### Create New Announcement

---

\* Name:

Prompt:  Upload Audio File

Default Destination:

**Figure 240: Announcement settings**

The table below gives more description of the configuration parameters when creating Announcement.



**Table 98: Announcement Parameters**

<b>Name</b>	<p>Configure the name of the Announcement.</p> <p><b>Note:</b> Please use letters, digits, _ or – only and no more than 64 characters.</p>
<b>Prompt</b>	<p>Audio file that will be played before ringing the configured default destination.</p> <p><b>Note:</b> Sound file must be PCM encoded, 16 bits at 8000Hz mono in mp3/wav format or raw ulaw/alaw/gsm file with .mp3/.wav/.ulaw/.alaw/.gsm suffix. The file size must be less than 5MB. If uploading a compressed file, the file must have .tar/.tgz/.tar.gz suffix. the file name must contain only letters, numbers or special characters - _ . The file size must be less than 30MB. Filename should not exceed 100 characters.</p>
<b>Default Destination</b>	<p>Select the destination where to send the call after playing the announcement.</p> <p>The available default destinations are:</p> <ul style="list-style-type: none"> <li>• Extension</li> <li>• Conference Rooms</li> <li>• Video Conference</li> <li>• Voicemail</li> <li>• Voicemail Group</li> <li>• IVR</li> </ul>





- Ring Group
- Queues
- Fax
- DISA
- Dial By Name
- External Number
- Hang-up

Created Announcements will be listed as shown below:

Announcement				
+ Add				
NAME ↕	PROMPT ↕	DEFAULT DESTINATION	DEFAULT DESTINATION	OPTIONS
Announcement1	Alarm01	Extension	1000	 
		<input type="text" value="1"/>		Total: 1 <input type="text" value="30 / page"/> Goto <input type="text" value="1"/>

**Figure 241: Announcement**

- Press  to edit the announcement.
- Press  to delete the announcement.



## PBX SETTINGS

This section describes internal options that have not been mentioned in previous sections yet. The settings in this section can be applied globally to the UCM6200, including general configurations, jitter buffer, RTP settings, ports config and STUN monitor. The options can be accessed via Web GUI→**PBX Settings**→**General Settings**.

### PBX Settings/General Settings

Table 99: Internal Options/General

General Preferences	
<b>Global Outbound CID</b>	Configure the global CallerID used for all outbound calls when no other CallerID is defined with higher priority. If no CallerID is defined for extension or trunk, the global outbound CID will be used as CallerID.
<b>Global Outbound CID Name</b>	Configure the global CallerID Name used for all outbound calls. If configured, all outbound calls will have the CallerID Name set to this name. If not, the extension's CallerID Name will be used.
<b>Ring Timeout</b>	Configure the number of seconds to ring an extension before the call goes to the user's voicemail box. The default setting is 60. <b>Note:</b> This is the global value used for each extension if "Ring Timeout" field is left empty on the extension configuration page.
<b>Call Duration Limit</b>	Configure the maximum duration of call-blocking.
<b>Record Prompt</b>	If enabled, users will hear voice prompt before recording is started or stopped. For example, before recording, the UCM6200 will play voice prompt "The call will be recorded". The default setting is "No".
<b>Enable 486 to Failover Trunk</b>	Reroutes failed outbound calls that receive a 486 response through the failover trunk to retry the call. If disabled, calls that receive a 486 response will be terminated.
<b>Device Name</b>	Enter a name to identify the UCM. The name will be displayed on UCM web interface.
<b>International Call Prefix</b>	Configure the International prefix. Default is 00. If empty, international call prefix can be empty or +. This parameter helps the UCM to identify the international call prefix for the country (00, 011, 810...) to avoid any conflict when using blacklist for specific countries. When outbound blacklist is enabled, UCM will apply the following rule "International Call Prefix+Country Code+Destination number". When making outbound call, UCM will check "International Call Prefix" then "Country Code", if the combination exists in the blacklist, then the call will fail, otherwise, the call will be authorized.





If "Enable Pick Extension" under zero config settings is selected, the extension list defined in "Pick Extension Segment" will be sent out to the device after receiving the device's request. This "Pick Extension Segment" should be a subset of the "Pick Extensions" range here. This feature is for the GXP series phones that support selecting extension to be provisioned via phone's LCD.

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

## PBX Settings/RTP Settings

### RTP Settings

**Table 100: Internal Options/RTP Settings**

<b>RTP Start</b>	Configure the RTP port starting number. The default setting is 10000.
<b>RTP End</b>	Configure the RTP port ending address. The default setting is 20000.
<b>Strict RTP</b>	Configure to enable or disable strict RTP protection. If enabled, RTP packets that do not come from the source of the RTP stream will be dropped. The default setting is "Disable".
<b>RTP Checksums</b>	Configure to enable or disable RTP Checksums on RTP traffic. The default setting is "Disable".
<b>ICE Support</b>	Configure whether to support ICE. The default setting is enabled. ICE is the integrated use of STUN and TURN structure to provide reliable VoIP or video calls and media transmission, via a SIP request/ response model or multiple candidate endpoints exchanging IP addresses and ports, such as private addresses and TURN server address.





<b>STUN Server</b>	<p>Configure STUN server address. STUN protocol is a Client/Server and also a Request/Response protocol. It is used to check the connectivity between the two terminals, such as maintaining a NAT binding entries keep-alive agreement. The default STUN Server is stun.ipvideotalk.com.</p> <p>Valid format:          [(hostname   IP-address) [:' port]          The default port number is 3478 if not specified.</p>
--------------------	---

## Payload

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

**Table 101: Internal Options/Payload**

<b>AAL2-G.726</b>	Configure payload type for ADPCM (G.726, 32kbps, AAL2 codeword packing). The default setting is 112.
<b>DTMF</b>	Configured payload type for DTMF. The default setting is 101.
<b>G.721 Compatible</b>	Configure to enable/disable G.721 compatible. The default setting is Yes.
<b>G.726</b>	Configure the payload type for G.726 if "G.721 Compatible" is disabled. The default setting is 111.
<b>iLBC</b>	Configure the payload type for iLBC. The default setting is 97.
<b>H.264</b>	Configure the payload type for H.264. The default setting is 99.
<b>H.265</b>	Configure the payload type for H.264. The default setting is 114.
<b>H.263P</b>	Configure the payload type for H.263+. The default setting is 100 103.
<b>VP8</b>	Configure the payload type for VP8. The default setting is 108.

## PBX Settings/NAS

The UCM supports adding and backing up recordings to a network-attached storage (NAS) server. Following table describes NAS settings:

**Table 102: NAS Settings**

<b>Enable</b>	Enabled / Disable the NAS recording functionality.
<b>Host</b>	Configure the Domain or IP address of the NAS server. <b>Note:</b> Currently, only IP addresses are supported in the Host/IP field.
<b>Share Name</b>	Specify the name of the shared folder.
<b>Username</b>	Specify the account username to access the NAS server.



<b>Password</b>	Configure the account password to access the NAS server.
<b>Status</b>	If configured correctly, the Status field will show “Mounted”, and the newly added NAS server will be shown on the Mounted Netdisk List. Additionally, the NAS will appear as a selectable storage option in the PBX Settings->Recording Storage page and CDR→Recording Files page.

## PBX Settings/Voice Prompt Customization

### Record New Custom Prompt

In the UCM6200 Web GUI→**PBX Settings**→**Voice Prompt**→**Custom Prompt** page, click on “Record New Custom Prompt” and follow the steps below to record new IVR prompt.

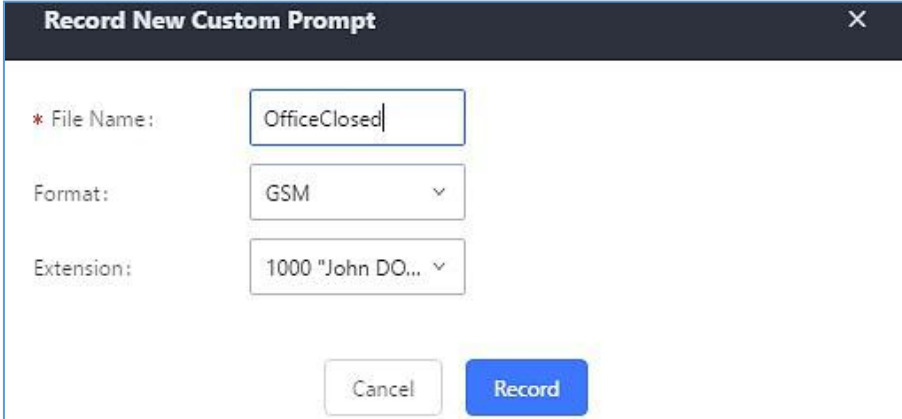


Figure 242: Record New Custom Prompt

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



## Upload Custom Prompt

If the user has a pre-recorded IVR prompt file, click on “Upload Custom Prompt” in Web GUI→PBX Settings→Voice Prompt→Custom Prompt page to upload the file to the UCM6200. The following are required for the IVR prompt file to be successfully uploaded and used by the UCM6200:

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

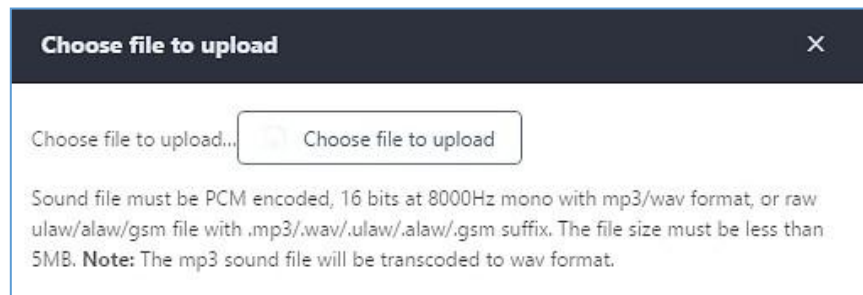


Figure 243: Upload Custom Prompt

Click on “choose file to upload” to start uploading. Once uploaded, the file will appear in the Custom Prompt web page.

## Download All Custom Prompt

On the UCM62XX, the users can download all custom prompts from UCM Web GUI to local PC. To download all custom prompt, log in UCM Web GUI and navigate to **PBX Settings→Voice Prompt→Custom Prompt** and click on [Download All Custom Prompt](#). The following window will pop up in order to set a name for the downloaded file.

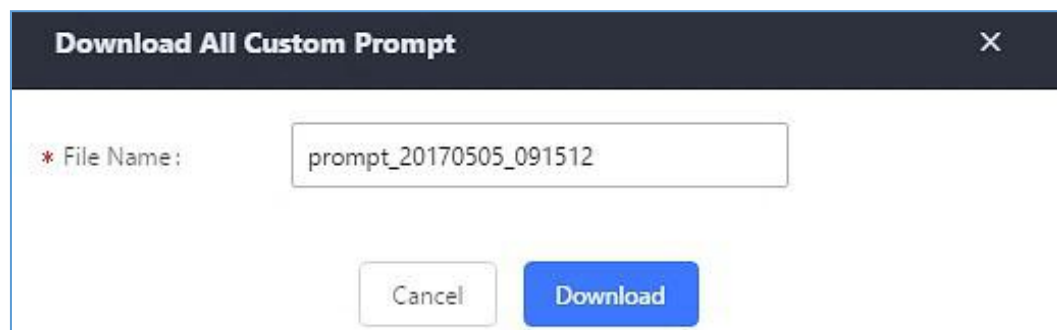


Figure 244: Download All Custom Prompt

**Note:** The downloaded file will have a .tar extension.



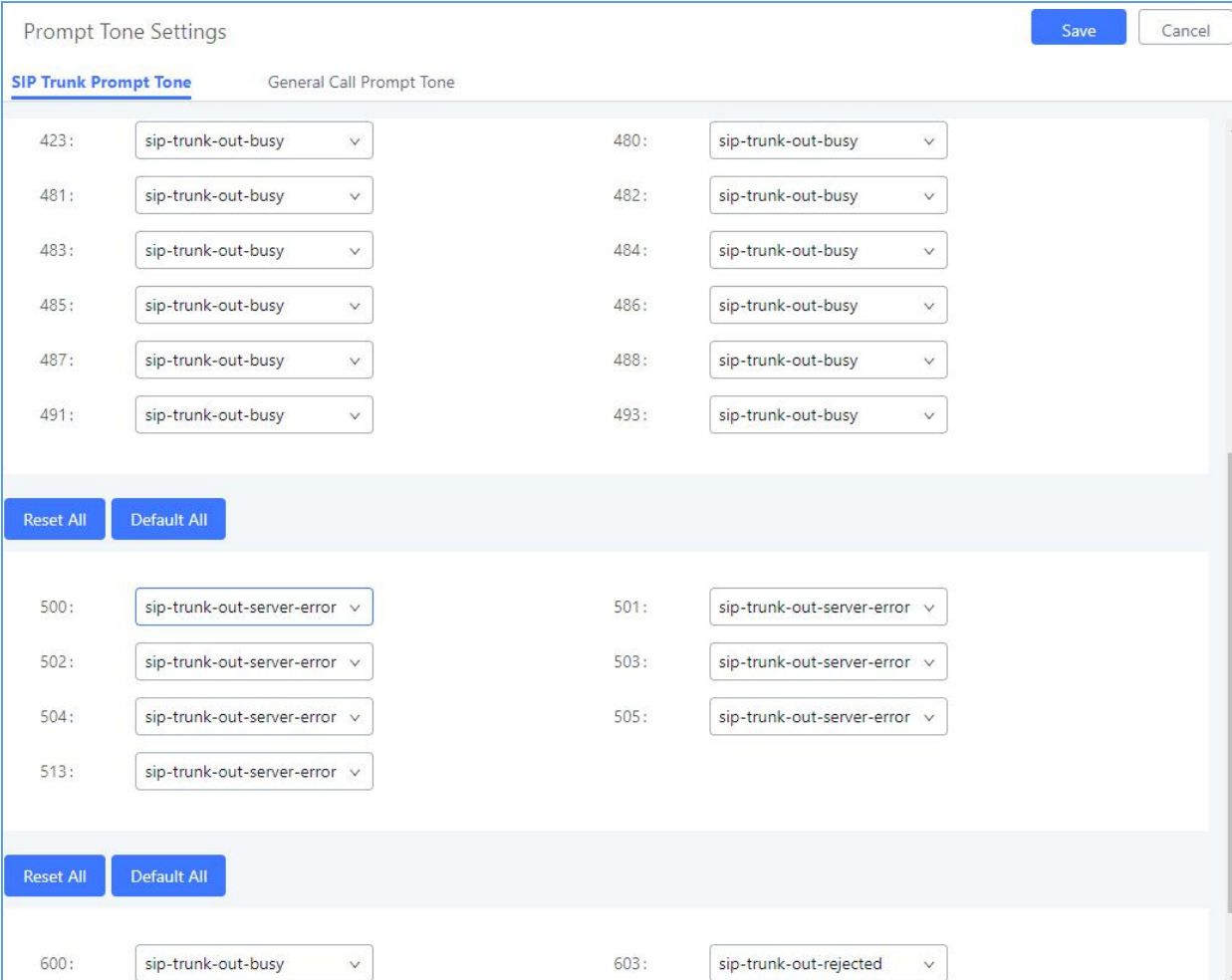
## PBX Settings/ Call Failure Tone Settings

### SIP Trunk Prompt Tone

**Prompt Tone Settings** tab has been added to the UCM to help users choose which prompt will be played by the UCM during call failure, the following voice message responses have been added and can be set to be played for 4XX, 5XX, and 6XX call failures:

- Default for 404 and 604 status codes: *“Your call can’t be completed as dialed. Please check the number and dial again.”*
- Default for 5xx status codes: *“Server error. Please check your device.”*
- Default for 403 and 603 status codes: *“The call was rejected by the server. Please try again later.”*
- Default for all other status codes: *“All circuits are busy now. Please try again later.”*

Additionally, custom voice messages recorded and uploaded in **PBX Settings**→**Voice Prompt**→**Custom Prompt** can be used for these failure responses instead of the default messages.



Prompt Tone Settings Save Cancel

**SIP Trunk Prompt Tone**      General Call Prompt Tone

423:	<input type="text" value="sip-trunk-out-busy"/>	480:	<input type="text" value="sip-trunk-out-busy"/>
481:	<input type="text" value="sip-trunk-out-busy"/>	482:	<input type="text" value="sip-trunk-out-busy"/>
483:	<input type="text" value="sip-trunk-out-busy"/>	484:	<input type="text" value="sip-trunk-out-busy"/>
485:	<input type="text" value="sip-trunk-out-busy"/>	486:	<input type="text" value="sip-trunk-out-busy"/>
487:	<input type="text" value="sip-trunk-out-busy"/>	488:	<input type="text" value="sip-trunk-out-busy"/>
491:	<input type="text" value="sip-trunk-out-busy"/>	493:	<input type="text" value="sip-trunk-out-busy"/>

500:	<input type="text" value="sip-trunk-out-server-error"/>	501:	<input type="text" value="sip-trunk-out-server-error"/>
502:	<input type="text" value="sip-trunk-out-server-error"/>	503:	<input type="text" value="sip-trunk-out-server-error"/>
504:	<input type="text" value="sip-trunk-out-server-error"/>	505:	<input type="text" value="sip-trunk-out-server-error"/>
513:	<input type="text" value="sip-trunk-out-server-error"/>		

600:	<input type="text" value="sip-trunk-out-busy"/>	603:	<input type="text" value="sip-trunk-out-rejected"/>
------	---	------	---

**Figure 245: SIP Trunk Prompt Tone**



## General Call Failure Tone

Moreover, users also have the possibility to customize the prompt for typical call failure reasons like (no permission to allow outbound calls, busy lines, incorrect number dialed ...Etc.).

To customize these prompts user could record and upload their own files under “**PBX Settings → Voice Prompt → Custom Prompts**” then select each one for specific call failure case under “**PBX Settings -> Prompt Tone Settings → General Call Prompt Tone**” page as shown on the following figure:

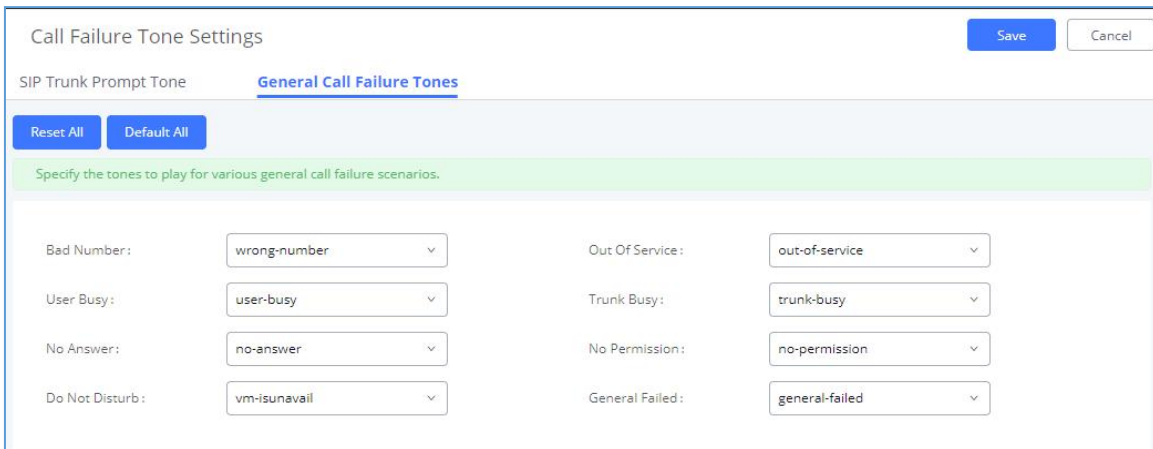


Figure 246: General call Failure Prompts

## PBX Settings/Jitter Buffer

Table 103: Internal Options/Jitter Buffer

SIP Jitter Buffer	
<b>Enable Jitter Buffer</b>	Select to enable jitter buffer on the sending side of the SIP channel. The default setting is "No".
<b>Jitter Buffer Size</b>	Configure the time (in ms) to buffer. This is the jitter buffer size used in "Fixed" jitter buffer or used as the initial time for "adaptive" jitter buffer. The default setting is 100.
<b>Max Jitter Buffer</b>	Configure the maximum time (in ms) to buffer for "Adaptive" jitter buffer implementation or used as the jitter buffer size for "Fixed" jitter buffer implementation. The default setting is 200.
<b>Implementation</b>	Configure the jitter buffer implementation on the sending side of a SIP channel. The default setting is "Fixed". <ul style="list-style-type: none"> <li>• <b>Fixed</b> The size is always equal to the value of "Max Jitter Buffer".</li> </ul>



- **Adaptive**  
The size is adjusted automatically and the maximum value equals to the value of "Max Jitter Buffer".

## PBX Settings/Recordings Storage

The UCM6200 supports call recordings automatically or manually and the recording files can be saved in external storage plugged in the UCM6200 or on the UCM6200 locally. To manage the recording storage, users can go to UCM6200 Web GUI→**PBX Settings**→**Recordings Storage** page and select whether to store the recording files in USB Disk, SD card or locally on the UCM6200.

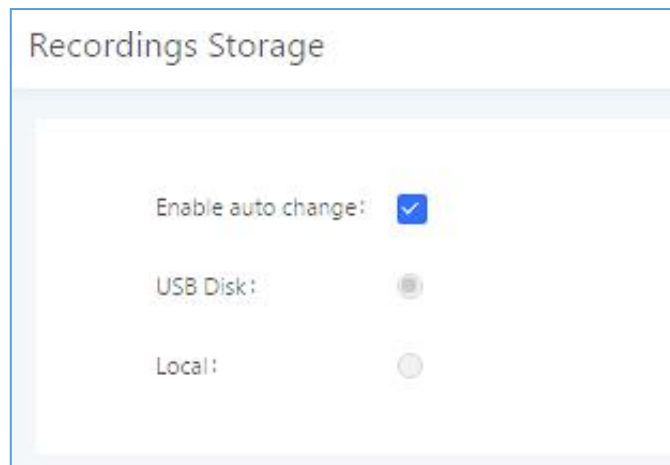
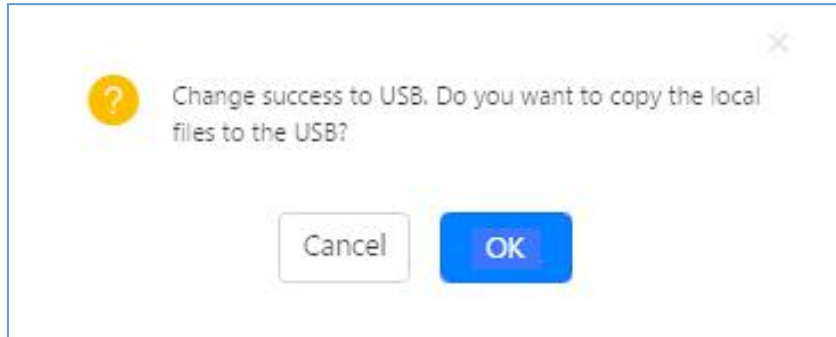


Figure 247: Settings→Recordings Storage

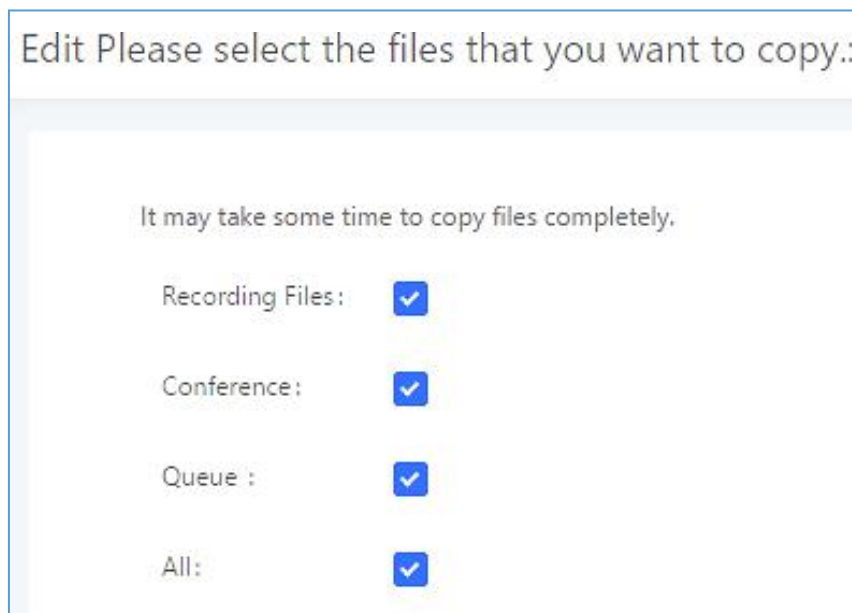
- If **“Enable Auto Change”** is selected, the recording files will be automatically saved in the available USB Disk or SD card plugged into the UCM6200. If both USB Disk and SD card are plugged in, the recording files will be always saved in the USB Disk.
- If **“Local”** is selected, the recordings will be stored in UCM6200 internal storage.
- If **“USB Disk”** or **“SD Card”** is selected, the recordings will be stored in the corresponding plugged in external storage device. Please note the options **“USB Disk”** and **“SD Card”** will be displayed only if they are plugged into the UCM6200.  
Once **“USB Disk”** or **“SD Card”** is selected, click on **“OK”**. The user will be prompted to confirm to copy the local files to the external storage device.





**Figure 248: Recordings Storage Prompt Information**

Click on “OK” to continue. The users will be prompted a new dialog to select the categories for the files to be copied over.



**Figure 249: Recording Storage Category**

On the UCM6200, recording files are generated and exist in 3 categories: normal call recording files, conference recording files, and call queue recording files. Therefore, users have the following options when select the categories to copy the files to the external device:

- **Recording Files:** Copy the normal recording files to the external device.
- **Conference:** Copy the conference recording files to the external device.
- **Queue:** Copy the call queue recording files to the external device.
- **All:** Copy all recording files to the external device.



## PBX Settings/NAS

The UCM supports adding and backing up recordings to a network-attached storage (NAS) server. Following table describes NAS settings:

**Table 104: NAS Settings**

<b>Enable</b>	Enabled / Disable the NAS recording functionality.
<b>Host</b>	Configure the Domain or IP address of the NAS server. <b>Note:</b> Currently, only IP addresses are supported in the Host/IP field.
<b>Share Name</b>	Specify the name of the shared folder.
<b>Username</b>	Specify the account username to access the NAS server.
<b>Password</b>	Configure the account password to access the NAS server.
<b>Status</b>	If configured correctly, the Status field will show "Mounted", and the newly added NAS server will be shown on the Mounted Netdisk List. Additionally, the NAS will appear as a selectable storage option in the PBX Settings→Recording Storage page and CDR→Recording Files page.





## SIP SETTINGS

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

### SIP Settings/General

**Table 105: SIP Settings/General**

<b>Realm For Digest Authentication</b>	Configure the host name or domain name for the UCM6200. Realms MUST be globally unique according to RFC3261. The default setting is Grandstream.
<b>Bind UDP Port</b>	Configure the UDP port used for SIP. The default setting is 5060.
<b>Bind IPv4 Address</b>	Configure the IPv4 address to bind to. The default setting is 0.0.0.0, which means binding to all addresses.
<b>Bind IPv6 Address</b>	Configure the IPv6 address to bind to. The default is : "[::]" and it means to bind to all IP addresses.
<b>Allow Guest Calls</b>	<p>If enabled, the UCM6200 allows unauthorized INVITE coming into the PBX and the call can be made. The default setting is "No".</p> <p><b>Warning:</b> Please be aware of the potential security risk when enabling "Allow Guest Calls" as this will allow any user with the UCM6200 address to dial into the UCM6200.</p>
<b>Allow Transfer</b>	If set to "No", all transfers initiated by the endpoint in the UCM6200 will be disabled (unless enabled in peers or users). The default setting is "Yes".
<b>MWI From</b>	When sending MWI NOTIFY requests, this value will be used in the "From:" header as the "name" field. If no "From User" is configured, the "user" field of the URI in the "From:" header will be filled with this value.
<b>Enable Diversion Header</b>	<p>If disabled, the UCM will not forward the diversion header.</p> <p><b>Note:</b> Diversion header will be included for Forward and transfer when the option is enabled.</p>
<b>Block Collect Calls</b>	<p>If enabled, collect calls will be blocked.</p> <p><b>Note:</b> Collect calls are indicated by the header "P-Asserted-Service-Info: service-code=Backward Collect Call, P-Asserted-Service-Info: service-code=Collect Call".</p>



## SIP Settings/MISC

Table 106: SIP Settings/Misc

Outbound SIP Registrations	
<b>Register Timeout</b>	Configure the register retry timeout (in seconds). The default setting is 20.
<b>Register Attempts</b>	Configure the number of registration attempts before the UCM6200 gives up. The default setting is 0, which means the UCM6200 will keep trying until the server side accepts the registration request.
Video	
<b>Max Bit Rate (kb/s)</b>	Configure the maximum bit rate (in kb/s) for video calls. The default setting is 384.
<b>Support SIP Video</b>	Select to enable video support in SIP calls. The default setting is "Yes".
<b>Reject Non-Matching INVITE</b>	If enabled, when rejecting an incoming INVITE or REGISTER request, the UCM6200 will always reject with "401 Unauthorized" instead of notifying the requester whether there is a matching user or peer for the request. This reduces the ability of an attacker to scan for valid SIP usernames. The default setting is "No".
SDP Attribute Passthrough	
<b>Enable Attribute Passthrough</b>	If enable, and if the service does not know the attribute of FEC/FECC/BFCP, then the attribute will be passthrough.
Early Media	
<b>Enable Use Final SDP</b>	If enabled, call negotiation will use final response SDP.
Blind Transfer	
<b>Allow callback when blind transfer fails</b>	If enabled, the UCM will call back to the transferrer when blind transfer fails (reason of failure includes busy and no answer). <b>Note:</b> This feature takes effect only on internal calls.
<b>Blind transfer timeout</b>	Configure the timeout in (s) for the transferrer waiting for the destination to answer. Default is 60s.
DNS	
<b>DNS mode</b>	This option affects the DNS query only during Calls. When you choose A&AAAA, UCM will do both A and AAAA type DNS query; when you chose A, UCM will only do A type DNS query; and when you chose AAAA, UCM will only do AAAA type DNS query.
Hold	
<b>Forward HOLD Requests</b>	Configure the UCM to forward HOLD requests instead of processing holds internally. This serves to meet the standards set by some providers that require HOLD requests to be passed along from endpoint to endpoint. This option is disabled by default. <b>Note:</b> Enabling this option may cause hold retrieval issues and MOH to not be heard.



## SIP Settings/Session Timer

**Table 107: SIP Settings/Session Timer**

<b>Force Timer</b>	If checked, always request and run session timer.
<b>Timer</b>	If checked, run session timer only when requested by other UA.
<b>Session Expire</b>	Configure the maximum session refresh interval (in seconds). The default setting is 1800.
<b>Min SE</b>	Configure the minimum session refresh interval (in seconds). The default setting is 90.

## SIP Settings/TCP and TLS

**Table 108: SIP Settings/TCP and TLS**

<b>TCP Enable</b>	Configure to allow incoming TCP connections with the UCM6200. The default setting is "No".
<b>TCP Bind Address</b>	Configure the IP address for TCP server to bind to. 0.0.0.0 means binding to all interfaces. The port number is optional. If not specified, 5060 will be used.
<b>TLS Enable</b>	Configure to allow incoming TLS connections with the UCM6200. The default setting is "No".
<b>TLS Bind Address</b>	Configure the IP address for TLS server to bind to. 0.0.0.0 means binding to all interfaces. The port number is optional. If not specified, 5061 will be used. <b>Note:</b> The IP address must match the common name (hostname) in the certificate. Please do not bind a TLS socket to multiple IP addresses. For details on how to construct a certificate for SIP, please refer to the following document: <a href="http://tools.ietf.org/html/draft-ietf-sip-domain-certs">http://tools.ietf.org/html/draft-ietf-sip-domain-certs</a>
<b>TLS Client Protocol</b>	Select the TLS protocol for outbound client connections. The default setting is TLSv1.
<b>TLS Do Not Verify</b>	If enabled, the TLS server's certificate will not be verified when acting as a client. The default setting is "Yes".
<b>TLS Self-Signed CA</b>	This is the CA certificate if the TLS server being connected to requires self-signed certificate, including server's public key. This file will be renamed as "TLS.ca" automatically. <b>Note:</b> The size of the uploaded ca file must be under 2MB..



<b>TLS Cert</b>	<p>This is the Certificate file (*.pem format only) used for TLS connections. It contains private key for client and signed certificate for the server. This file will be renamed as "TLS.pem" automatically.</p> <p><b>Note:</b> The size of the uploaded certificate file must be under 2MB.</p>
<b>TLS CA Cert</b>	<p>This file must be named with the CA subject name hash value. It contains CA's (Certificate Authority) public key, which is used to verify the accessed servers.</p> <p><b>Note:</b> The size of the uploaded CA certificate file must be under 2MB.</p>
<b>TLS CA List</b>	Display a list of files under the CA Cert directory.

## SIP Settings/NAT

**Table 109: SIP Settings/NAT**

<b>External Host</b>	Configure a static IP address and port (optional) used in outbound SIP messages if the UCM6200 is behind NAT. If it is a host name, it will only be looked up once.
<b>Use IP address in SDP</b>	If enabled, the SDP connection will use the IP address resolved from the external host.
<b>External TCP Port</b>	Configure the externally mapped TCP port when the UCM6200 is behind a static NAT or PAT.
<b>External TLS Port</b>	Configures the externally mapped TLS port when UCM6200 is behind a static NAT or PAT.
<b>Local Network Address</b>	<p>Specify a list of network addresses that are considered inside of the NAT network. Multiple entries are allowed. If not configured, the external IP address will not be set correctly.</p> <p>A sample configuration could be as follows: 192.168.0.0/16</p>

## SIP Settings/TOS

**Table 110: SIP Settings/ToS**

<b>ToS For SIP</b>	Configure the Type of Service for SIP packets. The default setting is None.
<b>ToS For RTP Audio</b>	Configure the Type of Service for RTP audio packets. The default setting is None.
<b>ToS For RTP Video</b>	Configure the Type of Service for RTP video packets. The default setting is None.
<b>Default Incoming/Outgoing Registration Time</b>	Configure the default duration (in seconds) of incoming/outgoing registration. The default setting is 120.



<b>Max Registration/Subscription Time</b>	Configure the maximum duration (in seconds) of incoming registration and subscription allowed by the UCM6200. The default setting is 3600.
<b>Min Registration/Subscription Time</b>	Configure the minimum duration (in seconds) of incoming registration and subscription allowed by the UCM6200. The default setting is 60.
<b>Enable Relaxed DTMF</b>	Select to enable relaxed DTMF handling. The default setting is "No".
<b>DTMF Mode</b>	Select DTMF mode to send DTMF. The default setting is RFC4733. If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit codec PCMU and PCMA are required. When "Auto" is selected, "RFC4733" will be used if offered, otherwise "Inband" will be used. The default setting is "RFC4733".
<b>RTP Timeout</b>	During an active call, if there is no RTP activity within the timeout (in seconds), the call will be terminated. The default setting is no timeout.  <b>Note:</b> This setting does not apply to calls on hold.
<b>RTP Hold Timeout</b>	When the call is on hold, if there is no RTP activity within the timeout (in seconds), the call will be terminated. This value of RTP Hold Timeout should be larger than RTP Timeout. The default setting is no timeout.
<b>RTP Keep-alive</b>	This feature can be used to avoid abnormal call drop when the remote provider requires RTP traffic during proceeding. For example, when the call goes into voicemail and there is no RTP traffic sent out from UCM, configuring this option can avoid voicemail drop. When configured, RTP keep-alive packet will be sent to remote party at the configured interval. If set to 0, RTP keep-alive is disabled.
<b>100rel</b>	Configure the 100rel setting on UCM6200. The default setting is "Yes".
<b>Trust Remote Party ID</b>	Configure whether the Remote-Party-ID should be trusted. Default setting is "No".
<b>Send Remote Party ID</b>	Configure whether the Remote-Party-ID should be sent or not. The default setting is "No".
<b>Generate In-Band Ringing</b>	Configure whether the UCM6200 should generate inband ringing or not. The default setting is "Never". <ul style="list-style-type: none"> <li>• <b>Yes:</b> The UCM6200 will send 180 Ringing followed by 183 Session Progress and in-band audio.</li> <li>• <b>No:</b> The UCM6200 will send 180 Ringing if 183 Session Progress has not been sent yet. If audio path is established already with 183 then send in-band ringing.</li> <li>• <b>Never:</b> Whenever ringing occurs, the UCM6200 will send 180 Ringing as long as 200OK has not been set yet. Inband ringing will not be generated even the end point device is not working properly.</li> </ul>



**Server User Agent**

Configure the user agent string for the UCM6200.

**Send Compact SIP Headers**

If enabled, compact SIP headers will be sent. The default setting is "No".

## Transparent Call-Info header

UCM supports transparent call info header in order to integrate GDS door system with GXP21XX Color phones, the UCM will forward the call-info header to the phone in order to request the live view from GDS door system and give the option to open the door via softkey.

```

Session Initiation Protocol (INVITE)
  Request-Line: INVITE sip:3001@192.168.6.36:5064 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP 192.168.6.187:5060;rport;branch=z9hG4bKPj3217e67b-e74f-4f9f-b06f-afd3dcbbe29b
    From: "3002" <sip:3002@192.168.6.187>;tag=202dca4f-2b9d-4880-924c-d48cea7d0596
    To: <sip:3001@192.168.6.36>
    Contact: <sip:68aae6ea-f1d4-4e62-9987-446e718a2448@192.168.6.187:5060>
    Call-ID: 7f66bb20-0b9f-4828-a355-698853b8d9fb
    CSeq: 17559 INVITE
    Allow: OPTIONS, INFO, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, MESSAGE, REGISTER, REFER
    Supported: 100rel, timer, replaces, noferesub
    Session-Expires: 1800
    Min-SE: 90
    Call-Info: <https://192.168.6.186:443/capture/8001> ;purpose=GDS-view
    Max-Forwards: 70
    User-Agent: Grandstream UCM6202V1.5A 1.0.13.15
    Content-Type: application/sdp
    Content-Length: 547
  Message Body
  
```

Figure 250: Transparent Call-Info



## IAX SETTINGS

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

### IAX Settings/General

Table 111: IAX Settings/General

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

### IAX Settings/Registration

Table 112: IAX Settings/Registration

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



<b>Auto Kill</b>	If set to "yes", the connection will be terminated if ACK for the NEW message is not received within 2000ms. Users could also specify number (in milliseconds) in addition to "yes" and "no". The default setting is "yes".
<b>Authentication Debugging</b>	If enabled, authentication traffic in debugging will not show. The default is "No".
<b>Codec Priority</b>	Configure codec negotiation priority. The default setting is "Reqonly". <ul style="list-style-type: none"> <li>• <b>Caller</b> Consider the callers preferred order ahead of the host's.</li> <li>• <b>Host</b> Consider the host's preferred order ahead of the caller's.</li> <li>• <b>Disabled</b> Disable the consideration of codec preference all together.</li> <li>• <b>Reqonly</b> This is almost the same as "Disabled", except when the requested format is not available. The call will only be accepted if the requested format is available.</li> </ul>
<b>Type of Service</b>	Configure ToS bit for preferred IP routing.
<b>IAX Trunk Options</b>	
<b>Trunk Frequency</b>	Configure the frequency of trunk frames (in milliseconds). The default is 20.
<b>Trunk Time Stamps</b>	If enabled, time stamps will be attached to trunk frames. The default is "No".

## IAX Settings/Security

**Table 113: IAX Settings/Static Defense**

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



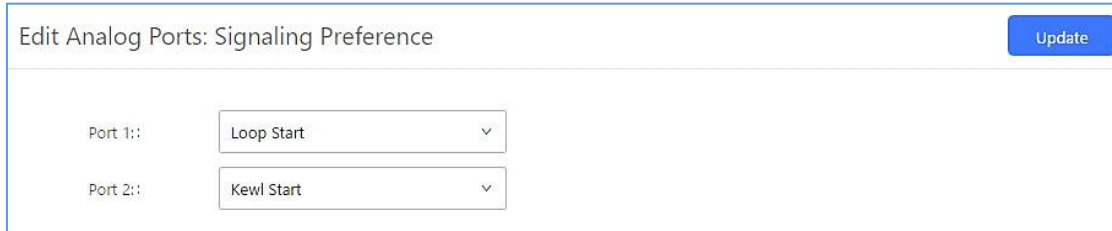


## INTERFACE SETTINGS

### Analog Hardware

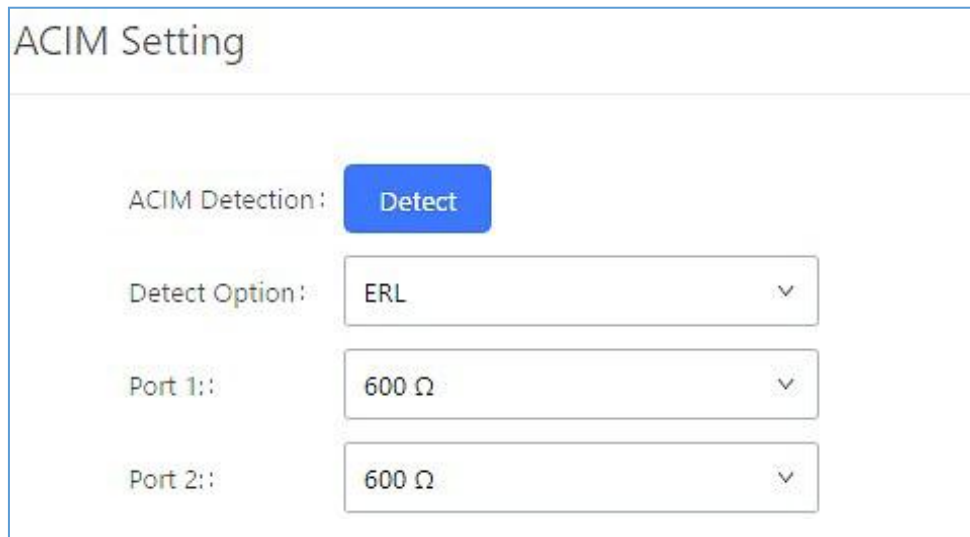
The analog hardware (FXS port and FXO port) on the UCM6200 will be listed in this page. Click on to edit signaling preference for FXS port or configure ACIM settings for FXO port.

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



**Figure 251: FXS Ports Signaling Preference**

For FXO port, users could manually enter the ACIM settings by selecting the value from dropdown list for each port. Or users could click on "Detect" and choose the detection algorithm, two algorithms exist (ERL, Pr) for the UCM6200 to automatically detect the ACIM value. The detecting value will be automatically filled into the settings.



**Figure 252: FXO Ports ACIM Settings**

**Table 114: PBX Interface Settings**



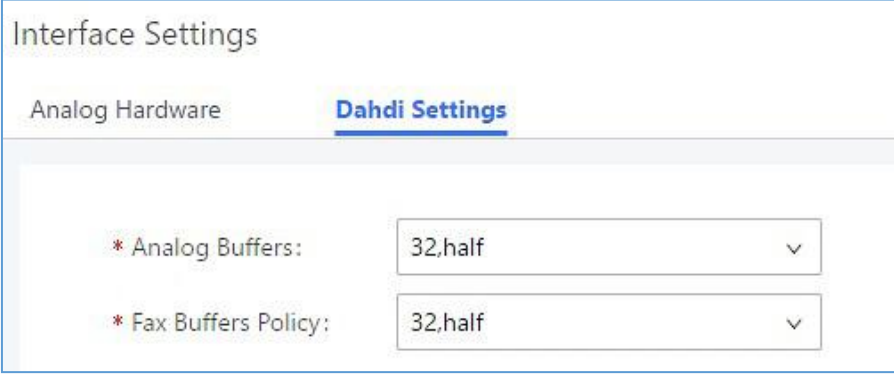
<b>Tone Region</b>	Select country to set the default tones for dial tone, busy tone, ring tone and etc. to be sent from the FXS port. The default setting is "United States of America (USA)".
<b>Advanced Settings</b>	
<b>FXO Opermode</b>	Select country to set the On-Hook Speed, Ringer Impedance, Ringer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



<b>FXS MWI Mode</b>	Configure the type of Message Waiting Indicator on FXS lines. The default setting is "FSK". <ul style="list-style-type: none"> <li>• <b>FSK</b>: Frequency Shift Key Indicator</li> <li>• <b>NEON</b>: Light Neon Bulb Indicator.</li> </ul>
<b>FXO Frequency Tolerance</b>	Allows users to adjust the tolerance of the FXO ringing frequency. 63Hz is considered the standard value and is selected by default.

## DAHDI Settings

When users encounter issues such as audio delay in outbound calls using the analog trunk, they can adjust DAHDI settings on the UCM to attempt to lessen or resolve the issues.



The screenshot shows the 'Interface Settings' page with the 'Dahdi Settings' tab selected. Under this tab, there are two configuration options, both currently set to '32,half':

- \* Analog Buffers: 32,half
- \* Fax Buffers Policy: 32,half

Figure 253: DAHDI Settings

For the value of the option such as "32, half":

The number in the option indicates the number of read/write buffers for TDM (DAHDI).

The "Half", "Immediate" or "Full" option indicates the strategy when reading/writing data from buffer.

- **"Half"**: Data will be read/written from buffer when half of the buffer is occupied with data.
- **"Immediate"**: Read/write from buffer whenever there is data occupying the buffer.
- **"Full"**: Data will be read/written from buffer when buffer is fully occupied with data.

Normally, DAHDI settings should be kept default and should be adjusted only when users encounter analog trunk/Fax-related issues.

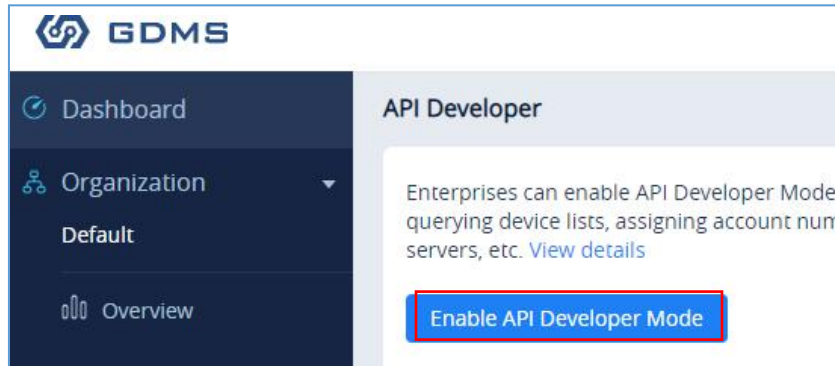


## GDMS SETTINGS

UCM can synchronize its SIP accounts to GDMS cloud management system.

<http://www.grandstream.com/products/device-management/gdms>

To 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.

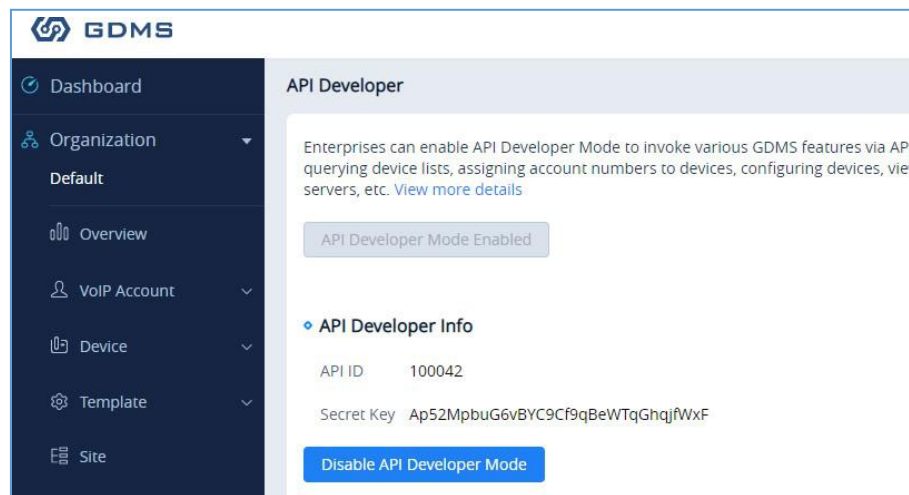


**Figure 254: GDMS Developer Mode Button**

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

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



**Figure 255: GDMS API Credentials**



### GDMS Settings

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

Enable:

\* Server Region:

\* Account Email/Username:

\* Password:

\* API ID:

\* Secret Key:

Connection Status:

Organization:

**Authenticate**

**Figure 256: GDMS Settings**

<b>Enable</b>	<p>Enable GDMS.</p> <p>Once toggled on all SIP extensions will automatically synchronize to the GDMS Cloud Platform.</p>
<b>Server Region</b>	<p>Select your GDMS server region:</p> <ul style="list-style-type: none"> <li>• <b>US region</b></li> <li>• <b>EU region</b></li> </ul>
<b>Account Email/Username</b>	The account email/username for GDMS Cloud Platform Authentication.
<b>Password</b>	The password for GDMS Cloud Platform Authentication.
<b>API ID</b>	<p>API ID from GDMS account.</p> <p>Refer to [Figure 255: GDMS API Credentials].</p>
<b>Secret Key</b>	<p>API secret key from GDMS account.</p> <p>Refer to [Figure 255: GDMS API Credentials].</p>
<b>Connection Status</b>	Provides the connection status once authenticated.
<b>Organization</b>	Name of the organization in GDMS that the UCM SIP server and its extensions will be under.
<b>Authenticate</b>	Authenticate button, in order to send all the information for authentication.



If the authentication is successful, you will now be able to select an organization on GDMS to synchronize 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 synchronized to GDMS.

SIP Server	
<input type="button" value="Delete"/>	
<input type="checkbox"/> Server Name	Server Address
<input type="checkbox"/> 192.168.42.63 <b>UCM</b>	192.168.42.63
Total 1	

**Figure 257: UCM on GDMS**

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

**Figure 258: UCM SIP Extensions on GDMS**

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

**Note:** UCM extensions’ names currently cannot be synchronized.



## API CONFIGURATION

The UCM6200 supports third party billing interface API for external billing software to access CDR and call recordings on the PBX. The API uses HTTPS to request the CDR data and call recording data matching given parameters as configured on the third-party application. More methods are also supported to provide better integration with 3<sup>rd</sup> party systems.

Before accessing the API, the administrators need enable API and configure the access/authentication information on the UCM6200 first under **Value-added Features**→**API Configuration**. The API configuration parameters are available for HTTPS API Settings (New), HTTPS API Settings (Old), CDR Real-time Output Settings & “Upload Prompts User Configuration”.

### HTTPS API (New)

Starting from firmware 1.0.20.17, UCM6200 supports new HTTPS API interface to query, edit PBX settings and implement multiple call functions on another server connected to it via API. PBX will actively send system reports and call reports to this other server. Additionally, legacy CDR API, REC API and PMS API are supported.

The table below lists configuration parameters for HTTPS API.

**Table 115: API Configuration Parameters**

HTTPS API Settings (New)	
<b>Enable</b>	Enable/Disable API. The default setting is disabled.
<b>Username</b>	Configure the username for API Authentication.
<b>Password</b>	Configure the password for API Authentication.
<b>Call Control</b>	If enabled, 3 <sup>rd</sup> party applications will be able to manage inbound calls via API actions. <b>acceptCall</b> will accept incoming calls while <b>refuseCall</b> will reject them. If no actions are done within 10 seconds, calls will automatically be accepted.

**Note:** HTTPS API uses web interface port (default is 8089).

The table below lists new HTTPS API supported methods.

**Table 116: New API Supported Queries**

getSystemStatus	addInboundRoute	listPaginggroup
getSystemGeneralStatus	getInboundRoute	addPaginggroup
listAccount	updateInboundRoute	getPaginggroup
getSIPAccount	deleteInboundRoute	updatePaginggroup



updateSIPAccount	playPromptByOrg	deletePaginggroup
listVoIPTrunk	listBridgedChannels	MulticastPaging
addSIPTrunk	listUnBridgedChannels	MulticastPagingHangup
getSIPTrunk	Hangup	listIVR
updateSIPTrunk	Callbarge	addIVR
deleteSIPTrunk	listQueue	getIVR
listOutboundRoute	getQueue	updateIVR
addOutboundRoute	updateQueue	deleteIVR
getOutboundRoute	addQueue	cdrazi
updateOutboundRoute	deleteQueue	recapi
deleteOutboundRoute	loginLogoffQueueAgent	pmsapi
listInboundRoute	pauseUnpauseQueueAgent	queueapi

For more details, please refer to online how-to guide available in our website.

## HTTPS API (Old)

Table 117: API Configuration Parameters (Old)

HTTPS API Settings (Old)	
Basic Settings	
<b>Enable</b>	Enable/Disable API. The default setting is disabled.
<b>TLS Bind Address</b>	Configure the IP address for TLS server to bind to. "0.0.0.0" means binding to all interfaces. The port number is optional, and the default port number is 8443. The IP address must match the common name (host name) in the certificate so that the TLS socket will not bind to multiple IP addresses. The default setting is 0.0.0.0:8443.
<b>Username</b>	Configure the username for TLS authentication.
<b>Password</b>	Configure the password for TLS authentication.
<b>Permitted IP(s)</b>	Specify a list of IP addresses permitted by API. This creates an API-specific access control list. Multiple entries are allowed. For example, "192.168.40.3/255.255.255.255" denies access from all IP addresses except 192.168.40.3. The default setting is blank, meaning all IP addresses will be denied.





Other Settings	
<b>TLS Private Key</b>	Upload TLS private key. The size of the key file must be under 2MB. This file will be renamed as 'private.pem' automatically.
<b>TLS Cert</b>	Upload TLS cert. The size of the certificate must be under 2MB. This is the certificate file (*.pem format only) for TLS connection. This file will be renamed as "certificate.pem" automatically. It contains private key for the client and signed certificate for the server.
API Module	
<b>CDR API</b>	Enable/disable CDR API module.
<b>REC API</b>	Enable/disable REC API module.
<b>PMS API</b>	Enable/disable PMS API module.

For more details on CDR API (Access to Call Detail Records), REC API (Access to Call Recording Files) and PMS API, please refer the document in the link here:

- [CDR API](#)
- [REC API](#)
- [PMS API](#)

## CDR Real-time Output Settings

CDR Real-time output feature allows to automatically send CDR records once available (after the call is terminated) to specified server/port.

Table 118: CDR Real-time Output Settings

CDR Real-time Output Settings	
<b>Enable</b>	Enables real-time CDR output module. This module connects to selected IP addresses and ports and posts CDR strings as soon as it is available.
<b>Server Address</b>	CDR server IP address
<b>Port</b>	CDR server IP port

For more details, refer to online how-to guide available at:

[http://www.grandstream.com/sites/default/files/Resources/CDR\\_Real-time\\_Output\\_Feature\\_Guide.pdf](http://www.grandstream.com/sites/default/files/Resources/CDR_Real-time_Output_Feature_Guide.pdf)



## Upload Voice Prompt via API

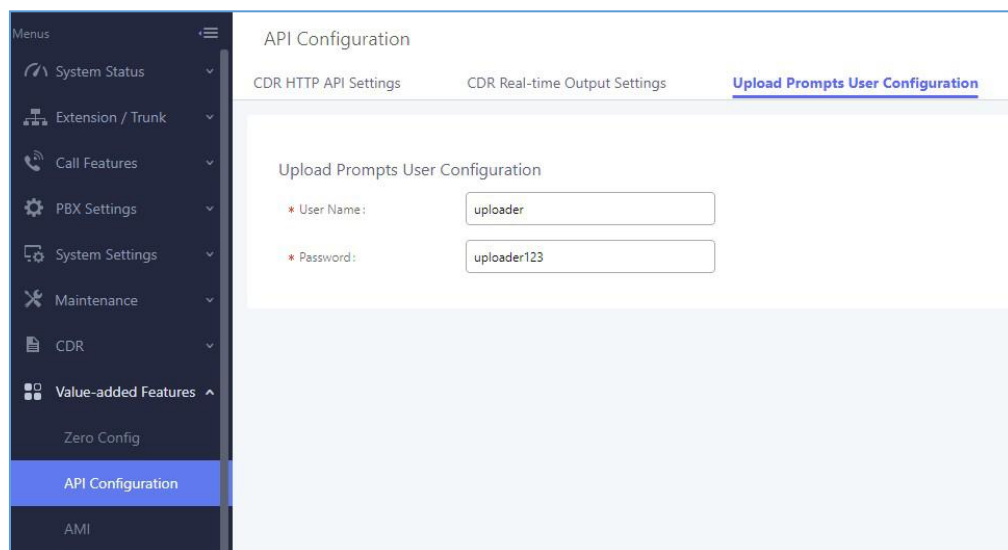
Customers now can use the “Upload Prompts User Configuration” to upload/replace voice prompt files as an alternative method to the manual upload method on UCM PBX Settings → Voice Prompt → Custom Prompt.

The workflow of the prompt file upload goes as:

An HTTP/HTTPS request is sent to the UCM to upload/replace a voice prompt file, the request should include authentication details to the UCM and the name of the file to be uploaded. Then the UCM will contact an FTP server that should be hosted on the same IP address of the HTTP/HTTPS requester and download the prompt file from the FTP server.

The steps and conditions to upload the voice prompt via API are listed below:

1. Configure the prompt User under **value-added Features** → **API Configuration** → **Upload Prompts User Configuration**. By default, the username and password for voice prompt user are “Username: uploader; Password: uploader123”.



**Figure 259: Upload Prompt User Configuration**

2. Hash the password of the user configured to an MD5 Encryption format.
3. Set the permission on the FTP server to Anonymous on the local computer hosting the FTP server and make sure that the default FTP port 21 is used.
4. Send an HTTP/HTTPS command to trigger the Prompt file upload on the UCM. If UCM's HTTP server is set to HTTPS, the example of the request sent to the UCM is:
   
<https://192.168.124.89:8089/cgi?action=uploadprompt&username=uploader&password=9191a6394c21b3aabd779213c7179462&filename=test.mp3>



If UCM's HTTP server is set to HTTP, the example of the request sent to the UCM is:

<http://192.168.124.89:8089/cgi?action=uploadprompt&username=uploader&password=9191a6394c21b3aabd779213c7179462&filename=test.mp3>

**Note:** If the File name on the HTTP/HTTPS request exists already on the UCM's Custom voice prompts list the existing file will be overwritten by the new file downloaded from the FTP server.

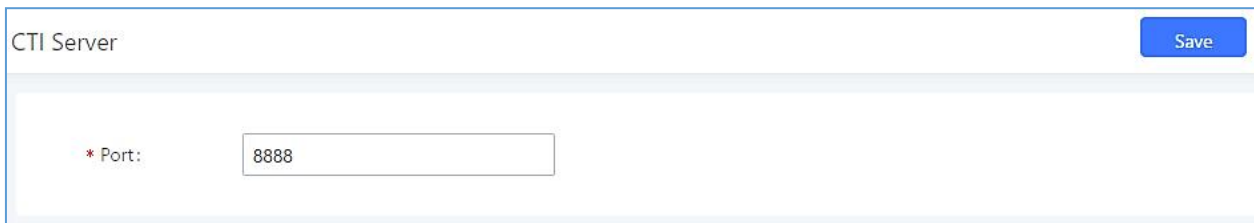


## CTI SERVER

UCM does support CTI server capabilities which are designed to be a part of the CTI solution suite provided by Grandstream, including GXP21XX and GXP17XX enterprise IP phones along with GS Affinity app.

Mainly the UCM will by default listening on port TCP 8888 for the connections from GS affinity application in order to interact, modify and serve data requests by the application which includes setting call features for the connected extension as call forward and DND.

Users can change the listening port under the menu page, Web GUI→**Value-added Features**→**CTI Server** as shown on below screenshot:



CTI Server Save

\* Port:

**Figure 260: CTI Server Listening port**

More information about GS affinity and CTI Support on Grandstream products series please refer to the following link: [http://www.grandstream.com/sites/default/files/Resources/GS\\_Affinity\\_Guide.pdf](http://www.grandstream.com/sites/default/files/Resources/GS_Affinity_Guide.pdf)



## ASTERISK MANAGER INTERFACE (RESTRICTED ACCESS)

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

User could configure AMI parameters on UCM6200 Web GUI→**Value-added Features**→**AMI**. For details on how to use AMI on UCM6200, please refer to the following AMI guide:

[http://www.grandstream.com/sites/default/files/Resources/UCM\\_series\\_AMI\\_guide.pdf](http://www.grandstream.com/sites/default/files/Resources/UCM_series_AMI_guide.pdf)

---

 **Warning:**

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

---



## CRM INTEGRATION

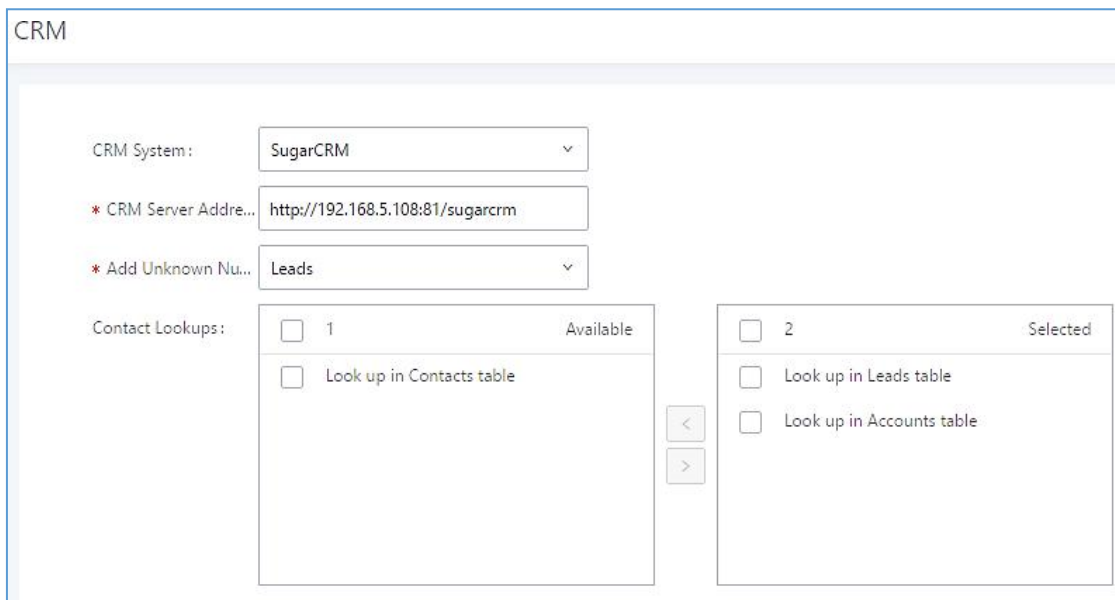
**Customer relationship management (CRM)** is a term that refers to practices, strategies and technologies that companies use to manage and analyze customer interactions and data throughout the customer lifecycle, with the goal of improving business relationships with customers.

The UCM6200 support the following CRM API: SugarCRM, vTigerCRM, ZohoCRM and Salesforce CRM, which allows users to look for contact information in the Contacts, Leads and / or Accounts tables, shows the contact record in CRM page, and saves the call information in the contact's history.

**Note:** Starting firmware 1.0.17.16, emergency calls will not be logged into CRM servers.

### SugarCRM

Configuration page of the SugarCRM can be accessed via admin login, on the UCM webGUI→**Value-added Features**→**CRM**.



**Figure 261: SugarCRM Basic Settings**



1. Select “SugarCRM” from the CRM System Dropdown in order to use SugarCRM.

**Table 119: SugarCRM Settings**



<b>CRM System</b>	Select a CRM system from the dropdown menu, four CRM systems are available: SugarCRM, vTigerCRM, ZohoCRM (legacy v1 API), ZohoCRM (v2 API), Salesforce or ACT! CRM.
<b>CRM Server Address</b>	Enter the IP address of the CRM server.
<b>Add Unknown Number</b>	Add the new number to this module if it cannot be found in the selected module.


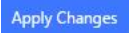


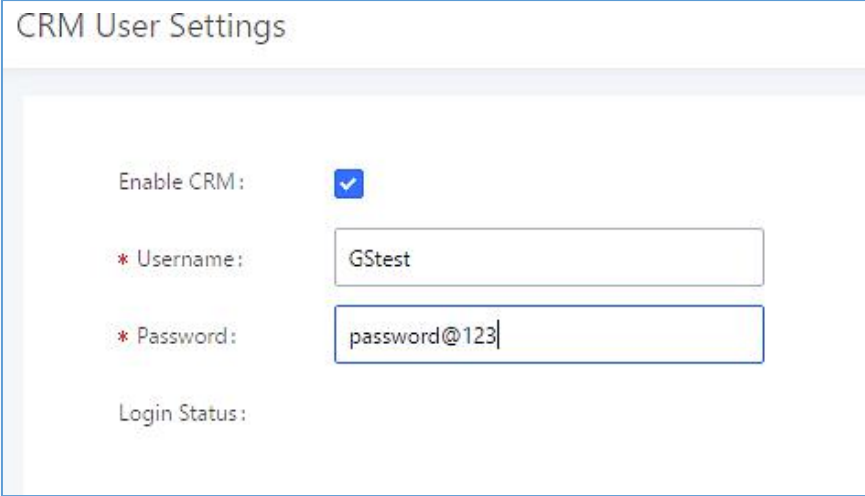
### Contact Lookups

Select from the **“Available”** list of lookups and press   to select where the UCM can perform the lookups on the CRM tables, Leads, Accounts, and Contacts.

Once settings on admin access are configured:

2. Click on  and .
3. Logout from admin access.
4. Login to the UCM as user and navigate under “User Portal→Value-added Feature→CRM User Settings”.

Click on **“Enable CRM”** and enter the username/password associated with the CRM account then click on  and . The status will change from “Logged Out” to “Logged In”. User can start then using SugarCRM features.



The screenshot shows the 'CRM User Settings' form. It includes the following fields:

- Enable CRM:** A checkbox that is checked.
- \* Username:** A text input field containing 'GStest'.
- \* Password:** A text input field containing 'password@123'.
- Login Status:** A label with no input field.

**Figure 262: CRM User Settings**

## vTigerCRM

Configuration page of the vTigerCRM can be accessed via admin login, on the UCM webGUI→**Value-added Features→CRM**.




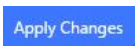
**Figure 263: vTigerCRM Basic Settings**



1. Select “vTigerCRM” from the CRM System Dropdown in order to use vTigerCRM.

**Table 120: vTigerCRM Settings**

<b>CRM System</b>	Select a CRM system from the dropdown menu, four CRM systems are available: SugarCRM, vTigerCRM, ZohoCRM (legacy v1 API), ZohoCRM (v2 API), Salesforce or ACT! CRM.
<b>CRM Server Address</b>	Enter the IP address of the CRM server.
<b>Add Unknown Number</b>	Add the new number to this module if it cannot be found in the selected module.
<b>Contact Lookups</b>	Select from the “ <b>Available</b> ” list of lookups and press to select where the UCM can perform the lookups on the CRM tables, Leads, Organizations, and Contacts.

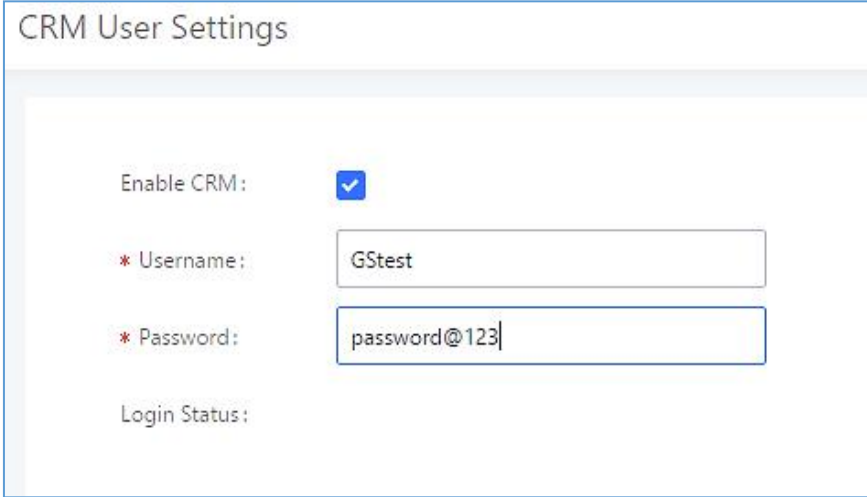
Once settings on admin access are configured:

2. Click on  and .
3. Logout from admin access.
4. Login to the UCM as user and navigate under “User Portal→Value-added Feature→CRM User Settings”.

Click on “**Enable CRM**” and enter the username/password associated with the CRM account then click on  and . The status will change from “Logged Out” to “Logged In”. User can start then using SugarCRM features.



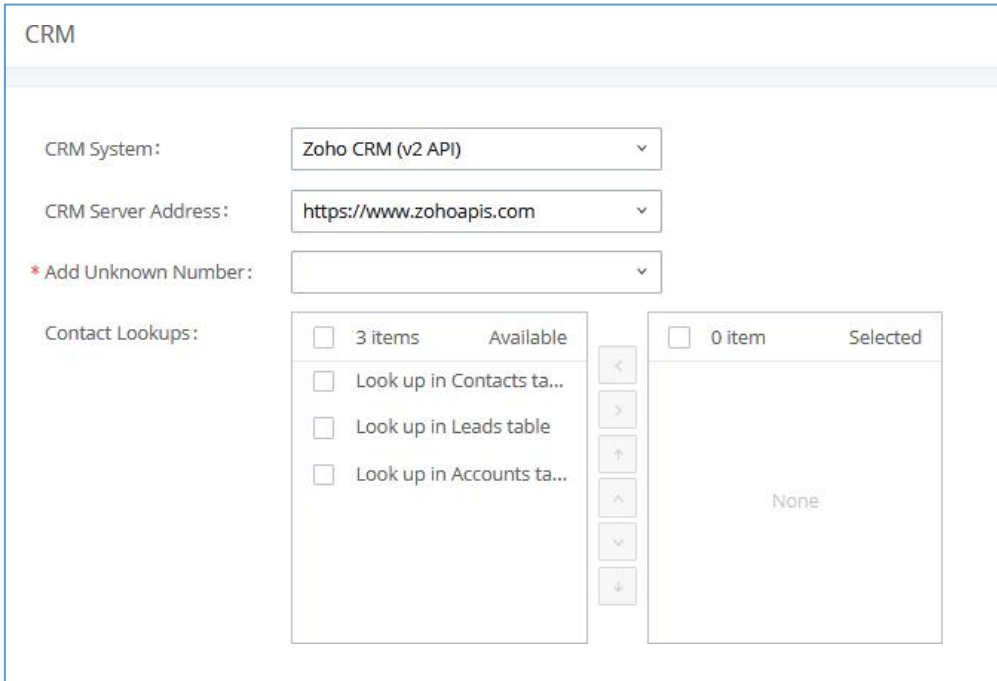




**Figure 264: CRM User Settings**

## ZohoCRM

Configuration page of the ZohoCRM v1 and ZohoCRM v2 can be both accessed via admin login, on the UCM Web GUI → **Value-added Features** → **CRM**.



**Figure 265: ZohoCRM Basic Settings**

1. Select “ZohoCRM (v2 API)” from the CRM System Dropdown in order to use ZohoCRM.

**Note:** Zoho CRM (legacy v1 API) will no longer be supported after 2019. Please use Zoho CRM (v2 API).



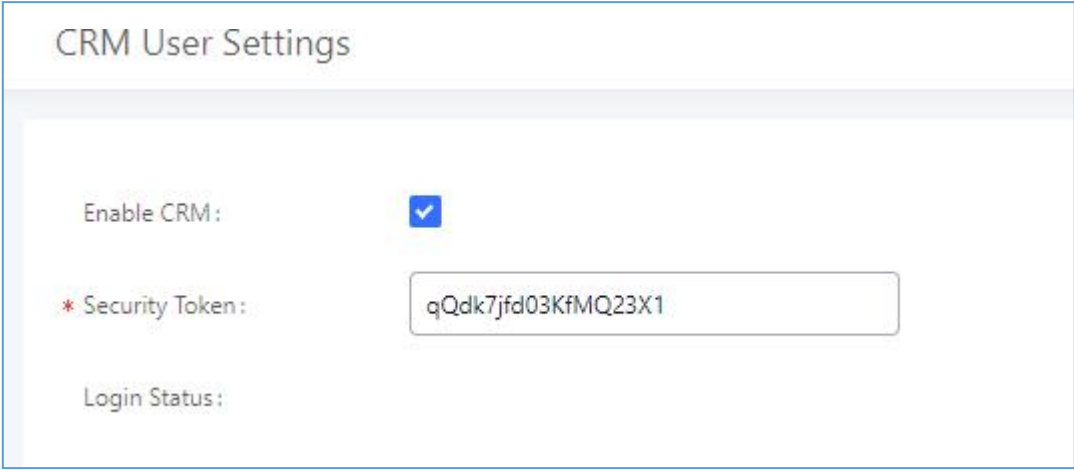
**Table 121: ZohoCRM Settings**

<b>CRM System</b>	Select CRM system from the dropdown menu, four CRM systems are available: SugarCRM, vTigerCRM, ZohoCRM (legacy v1 API), ZohoCRM (v2 API), Salesforce or ACT! CRM.
<b>CRM Server Address</b>	Select Zoho CRM URL from the list. Available options are: <ul style="list-style-type: none"> <li>• <a href="https://www.zohoapis.com">https://www.zohoapis.com</a></li> <li>• <a href="https://www.zohoapis.com.cn">https://www.zohoapis.com.cn</a></li> <li>• <a href="https://www.zohoapis.eu">https://www.zohoapis.eu</a></li> </ul>
<b>Add Unknown Number</b>	Add the new number to this module if it cannot be found in the selected module.
<b>Contact Lookups</b>	Select from the “ <b>Available</b> ” list of lookups and press <input type="button" value="OK"/> to select where the UCM can perform the lookups on the CRM tables, Leads, Accounts, and Contacts.

Once settings on admin access are configured:

2. Click on  and .
3. Logout from admin access.
4. Login to the UCM as user and navigate under “User Portal→Value-added Feature→CRM User Settings”.

Click on “**Enable CRM**” and enter the username/password associated with the CRM account then click on  and . The status will change from “Logged Out” to “Logged In”. User can start then using ZohoCRM features.



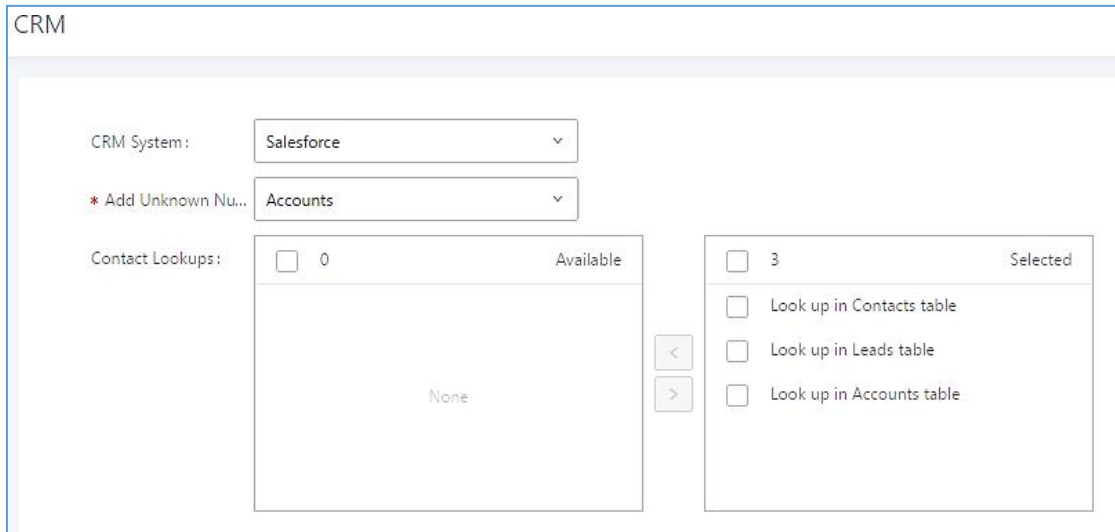
The screenshot shows the 'CRM User Settings' page. It includes the following elements:

- Enable CRM:** A checkbox that is checked with a blue checkmark.
- \* Security Token:** A text input field containing the value 'qQdk7jfd03KfMQ23X1'.
- Login Status:** A label for the login status, which is currently blank.

**Figure 266: CRM User Settings**


## Salesforce CRM

Configuration page of the Salesforce CRM can be accessed via admin login, on the UCM Web GUI → **Value-added Features** → **CRM**.



**Figure 267: Salesforce Basic Settings**

1. Select “Salesforce” from the CRM System Dropdown in order to use Salesforce CRM.

**Table 122: Salesforce Settings**

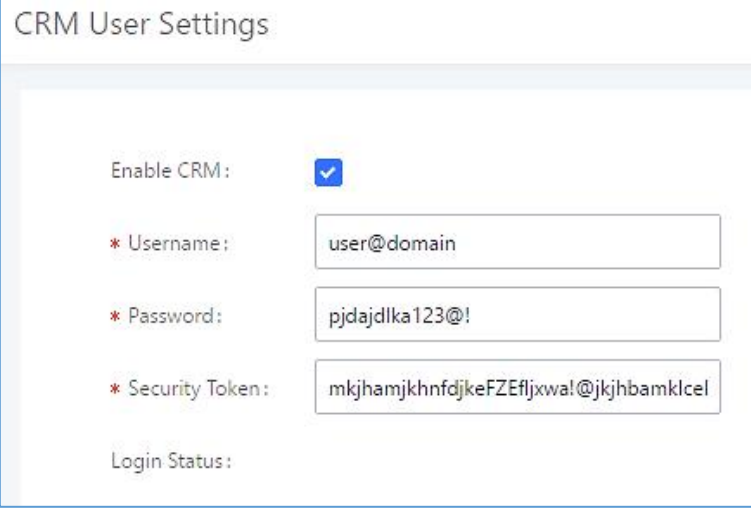
<b>CRM System</b>	Select a CRM system from the dropdown menu, four CRM systems are available: SugarCRM, vTigerCRM, ZohoCRM (legacy v1 API), ZohoCRM (v2 API), Salesforce or ACT! CRM.
<b>Add Unknown Number</b>	Add the new number to this module if it cannot be found in the selected module.
<b>Contact Lookups</b>	Select from the “ <b>Available</b> ” list of lookups and press to select where the UCM can perform the lookups on the CRM tables, Leads, Accounts, and Contacts.

Once settings on admin access are configured:

2. Click on **Save** and **Apply Changes**.
3. Logout from admin access.
4. Login to the UCM as user and navigate under “User Portal → Value-added Feature → CRM User Settings”.

Click on “**Enable CRM**” and enter the **username**, **password** and **Security Token** associated with the CRM account then click on **Save** and **Apply Changes**. The status will change from “Logged Out” to “Logged In”. User can start then using Salesforce CRM features.





CRM User Settings

Enable CRM:

\* Username:

\* Password:

\* Security Token:

Login Status:

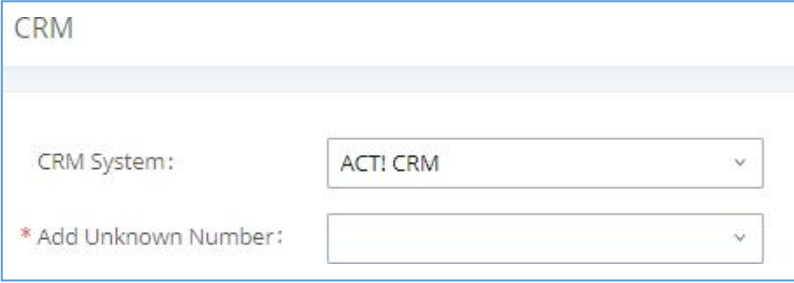
**Figure 268: Salesforce User Settings**

## ACT! CRM

Configuration page of the ACT! CRM can be accessed via admin login, on the UCM Web GUI → **Value-added Features** → **CRM**.

The configuration steps of the ACT! CRM are as follows:

1. Navigate to **Value-Added Features** → **CRM** and select the “ACT! CRM” option.



CRM

CRM System:

\* Add Unknown Number:

**Figure 269: Enabling ACT! CRM**

2. Log into the UCM as a regular user and navigate to **Value-Added Features** → **CRM User Settings** and check “Enable CRM” option and enter the username and password, which will be the ACT! CRM account’s **API Key** and **Developer Key**, respectively. To obtain these, please refer to the ACT! CRM API developer’s guide here: <https://mycloud.act.com/act/Help>



### CRM User Settings

---

Enable CRM:

Username:

Password:

Login Status:

**Figure 270: Enabling CRM on the User Portal**

**Note:** For more information on the ACT! CRM integration, please refer to the ACT! CRM documentation on our website.



## PMS INTEGRATION

UCM6200 supports Hotel Property Management System PMS, including check-in/check-out services, wakeup calls, room status, Do Not Disturb which provide an ease of management for hotel applications. This feature can be found on Web GUI→**Value-added Features**→**PMS**.

**Note:** The PMS integration on UCM is currently supported only with one of the three following solutions.

The PMS module built-in the UCM supports the following features based on each solution:

**Table 123: PMS Supported Features**

Feature	Mite l	HMobil e	HS C
Check-In	✓	✓	✗
Check-out	✓	✓	✗
Wake-up Call	✓	✓	✗
Name Change	✓	✗	✓
Update	✗	✓	✗
Set Credit	✓	✗	✗
Set Station Restriction	✓	✗	✓
Room Status	✗	✓	✗
Room Move	✗	✓	✗
Do Not Disturb	✗	✓	✓
Mini Bar	✗	✓	✗
MSG	✗	✓	✗
MWI	✗	✗	✓
Unconditional Call Forward	✗	✗	✓

### HMoblie PMS Connector

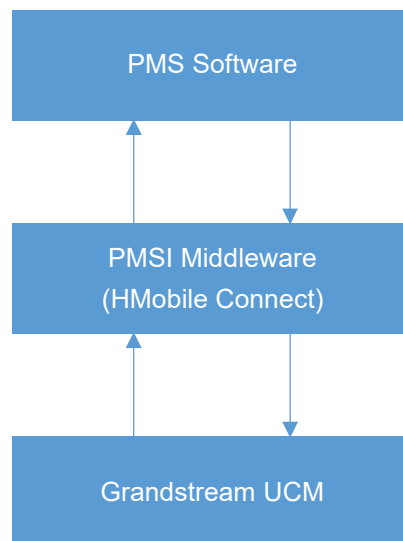
In this mode, the system can be divided into three parts:

- PMS (Property Management System)
- PMSI (Property Management System Interface)
- PBX

Grandstream UCM6XXX series have integrated HMobile Connect PMSI which supports a large variety of PMS software providing following hospitality features: Check-in, Check-out, set Room Status, Wake-up call and more.

The following figure illustrates the communication flow between the PBX (Grandstream UCM6xxx Series) and PMS software, which is done through a middleware system (HMobile Connect) acting as interface between both parties.





**Figure 271: UCM & PMS interaction**

## HSC PMS

In this mode, the system can be divided into two parts:

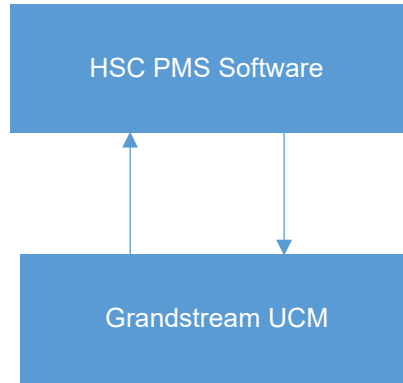
- PMS (Property Management System)
- PBX

Grandstream UCM6XXX series have integrated HSC PMS providing following features:

- Changing Display Name
- Set Station Restriction
- Call forwarding
- DND
- Name Change
- MWI

The following figure illustrates the communication flow between the PBX (Grandstream UCM6xxx Series) and PMS software (HSC). The communication between both parties is direct with no middleware.





**Figure 272: UCM & HSC PMS interaction**

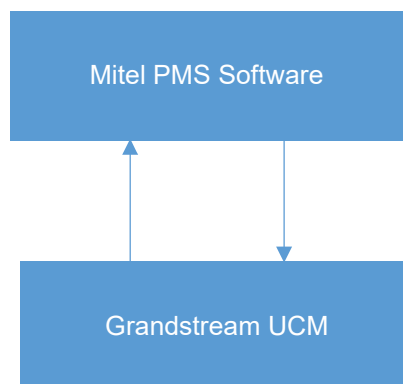
## Mitel PMS

In this mode, the system can be divided into two parts:

- PMS (Property Management System)
- PBX

Grandstream UCM6XXX series have integrated Mitel PMS providing following hospitality features: Check-in, Check-out, set Room Status, Wake-up call and more.

The following figure illustrates the communication flow between the PBX (Grandstream UCM6xxx Series) and PMS software (Mitel). The communication between both parties is direct with no middleware.



**Figure 273: UCM & Mitel PMS interaction**





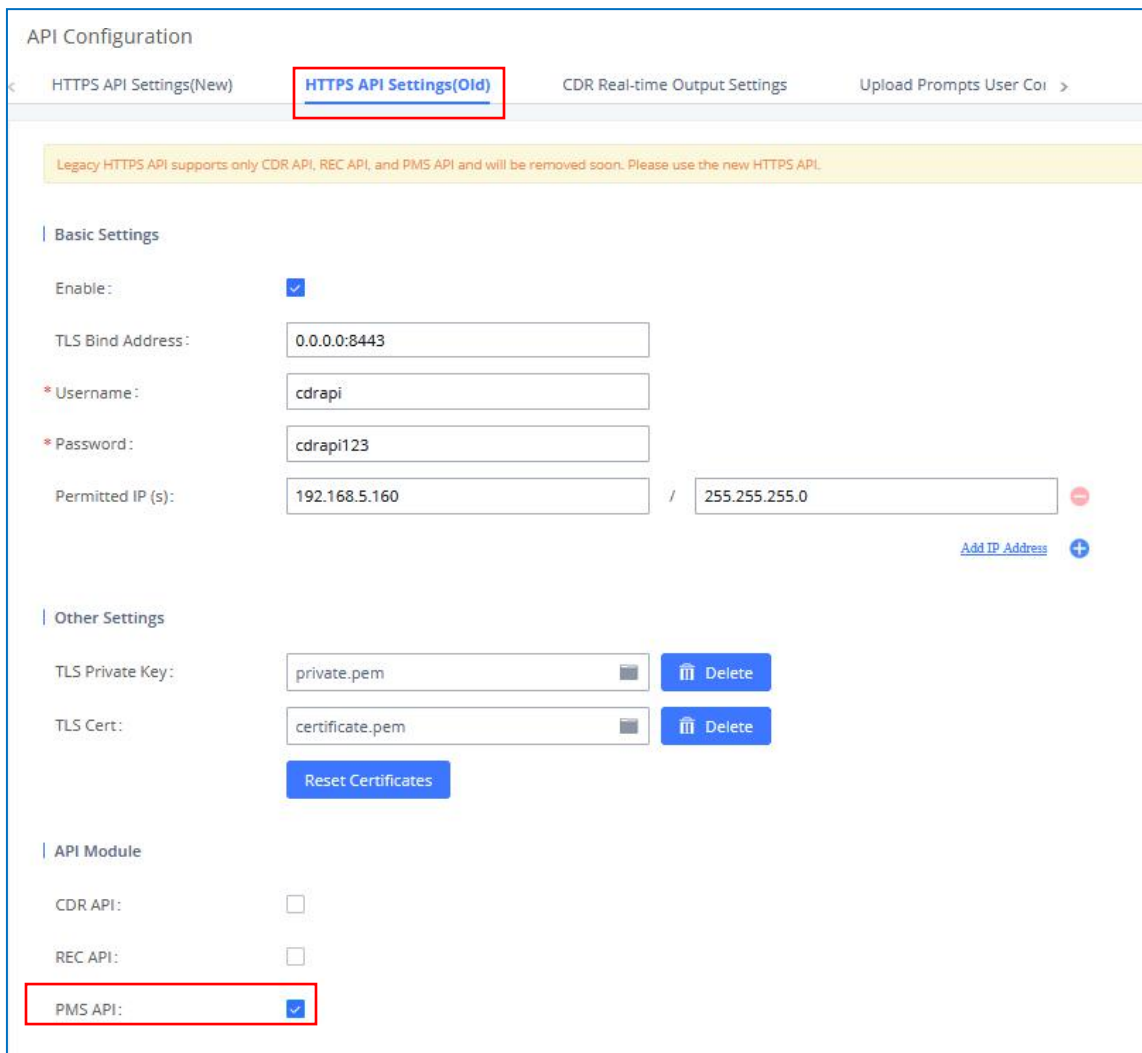
## PMS API

The PMS API allows users to use their own middleware to work with PMS systems instead of currently supported integrations.

Additionally, this API allows access to read and modify certain UCM parameters that current supported PMS integrations cannot. To use this, users must first enable and configure the HTTPS API settings.

On the HTTPS API settings, permitted IP addresses must be configured. Otherwise, the API will be inaccessible.

On the “Old” HTTPS API settings, permitted IP addresses must be configured. Otherwise, the API will be inaccessible. Make sure to check “Enable PMS API”.



API Configuration

[HTTPS API Settings\(New\)](#)
[HTTPS API Settings\(Old\)](#)
[CDR Real-time Output Settings](#)
[Upload Prompts User Coi >](#)

Legacy HTTPS API supports only CDR API, REC API, and PMS API and will be removed soon. Please use the new HTTPS API.

**Basic Settings**

Enable:

TLS Bind Address:

\* Username:

\* Password:

Permitted IP (s):  / 
[Add IP Address](#)

**Other Settings**

TLS Private Key:  [Delete](#)

TLS Cert:  [Delete](#)

[Reset Certificates](#)

**API Module**

CDR API:

REC API:

**PMS API:**

Figure 274: Enable PMSAPI

On the “New” HTTPS API, just need to enable API which include PMS API by default.

For more details, regarding Old/New HTTPS API, please refer to [API CONFIGURATION].

For more details, please refer to online [PMS API Guide](#).



## Connecting to PMS

On the UCM WebGUI→**Value-added Features**→**PMS**→**Basic Settings**” set the connection information for the PMS platform.

Table 124: PMS Basic Settings

Field	Description
<b>PMS Module</b>	Users can select the desired PMS module from the drop-down list. <ul style="list-style-type: none"> <li>• Hmobile.</li> <li>• Mitel.</li> <li>• HSC.</li> <li>• PMS API</li> </ul>
<b>Wake Up Prompt</b>	Prompt used when answering the wakeup calls it can be customized from <b>“PBX Settings→Voice Prompt→Custom Prompt.</b>
<b>PMS URL</b>	Enter the PMS system URL. If using “Hmobile” PMS Module only.
<b>UCM Port</b>	Enter the Port used by the PMS system. Default is 8081.
<b>Username</b>	Enter the Username to connect to the PMS system. If using “Hmobile” or “HSC” PMS Module only.
<b>Password</b>	Enter the password to connect to the PMS system. If using “Hmobile” or “HSC” PMS Module only.
<b>Site</b>	Enter the site to connect to the PMS system. If using “Hmobile” PMS Module only.
<b>Back Up Voicemail Recordings</b>	If enabled, this option allows backing up voicemail recordings to external storage after check-out. Voicemail can be backed up to SD card, USB disk, NAS, or an SFTP server
<b>Email address</b>	Configure the email address to send the backup to.

In order to use some PMS features please activate the feature code associated under **“Call Features→Feature Codes”**

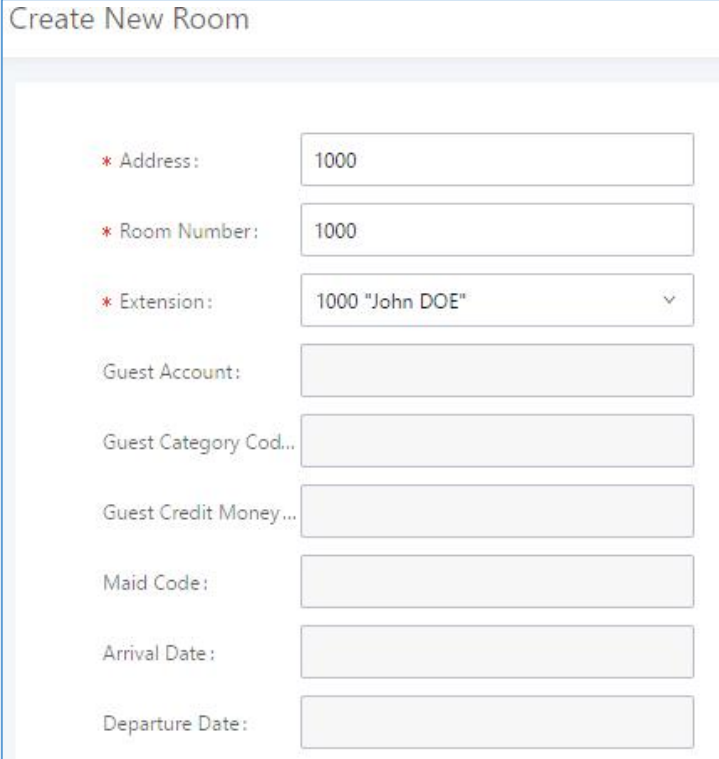
- Update PMS Room Status
- PMS Wake Up Service



## PMS Features

### Room Status

User can create Rooms by clicking on [+ Create New Room](#), the following Figure will be displayed then.



**Create New Room**

\* Address: 1000

\* Room Number: 1000

\* Extension: 1000 "John DOE"

Guest Account:

Guest Category Cod...:

Guest Credit Money...:

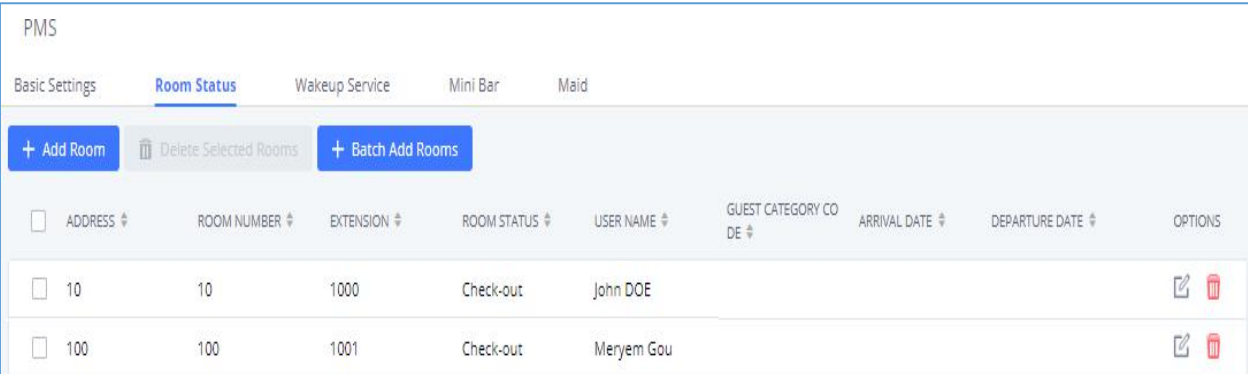
Maid Code:

Arrival Date:

Departure Date:

**Figure 275: Create New Room**

Click "Save" to create the new room, the fields above can be configured from the PMS platform, once set the following screen will be shown:



PMS

Basic Settings **Room Status** Wakeup Service Mini Bar Maid

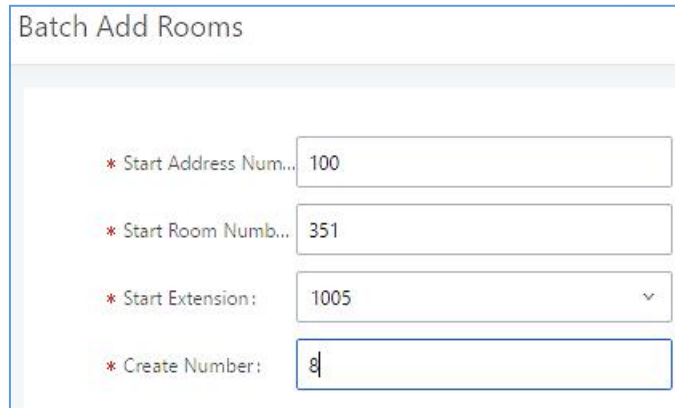
[+ Add Room](#) [Delete Selected Rooms](#) [+ Batch Add Rooms](#)

<input type="checkbox"/>	ADDRESS	ROOM NUMBER	EXTENSION	ROOM STATUS	USER NAME	GUEST CATEGORY CODE	ARRIVAL DATE	DEPARTURE DATE	OPTIONS
<input type="checkbox"/>	10	10	1000	Check-out	John DOE				<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	100	100	1001	Check-out	Meryem Gou				<a href="#">Edit</a> <a href="#">Delete</a>

**Figure 276: Room Status**

User can create a batch of rooms as well by clicking on [+ Batch Add Rooms](#), the following window will pop up:

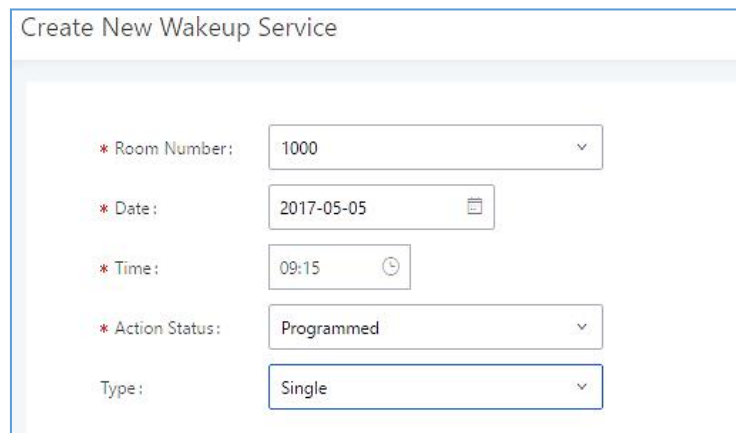




**Figure 277: Add batch rooms**

## Wake Up Service

To create a New Wake up service, user can click on [+ Create New Wakeup Service](#), the following window will pop up:





**Figure 278: Create New Wake Up Service**

**Table 125: PMS Wake up Service**

Field	Description
<b>Room Number</b>	Select the room number where to call with a limitation of 63 characters.
<b>Time</b>	Set the time of the wakeup call
<b>Action Status</b>	Show the status of the call: <ul style="list-style-type: none"> <li>• <u>Programmed</u>: the call is scheduled for the time set</li> <li>• <u>Cancelled</u>: the call is canceled</li> <li>• <u>Executed</u>: the wakeup call is made</li> </ul> <b>Note:</b> Editing an already executed wakeup service will automatically change the service's status to "Programmed".
<b>Type</b>	<ul style="list-style-type: none"> <li>• <u>Single</u>: The call will be made once on the specific time.</li> <li>• <u>Daily</u>: The call will be repeated every day on the specific time</li> </ul>




Once the call is made on the time specified, the following figure show the status of the wakeup call.

<input type="checkbox"/>	Name ↕	Extension ↕	Status	Action Status ↕	Answer Status ↕	Date	Time	Options
<input type="checkbox"/>	John	1000	Enabled	Executed	Answered	2017-05-04	05:18	 

**Figure 279: Wakeup Call executed**

This call has been executed but has been rejected, that why we can see the **“Busy”** status.

## Mini Bar

In order to create a new mini bar, click on  under UCM webGUI→**Value-added Features**→**PMS**→**Mini Bar**, the following window will pop up:

Create New Mini Bar

---

\* Code :

\* Name :

\* Prompt :  Prompt


Skip Maid and Passw...

Enable Continuous M...

**Figure 280: Create New Mini Bar**

**Table 126: Create New Mini Bar**

<b>Code</b>	Enter a non-existing extension number to be dialed when using the mini bar feature.
<b>Name</b>	Enter a name for the mini bar.
<b>Prompt</b>	Select the Prompt to play once connected to the mini bar.
<b>Skip Maid and Password Authentication</b>	If enabled, the default maid code will be 0000, no authentication is required. (Enter 0000 followed by # to access the consumer goods)
<b>Enable Continuous Multi Goods Billing</b>	If enabled, please separate the goods' codes by*.

In order to create a new maid, click on  under UCM webGUI→**Value-added Features**→**PMS**→**Mini Bar**, the following window will popup.



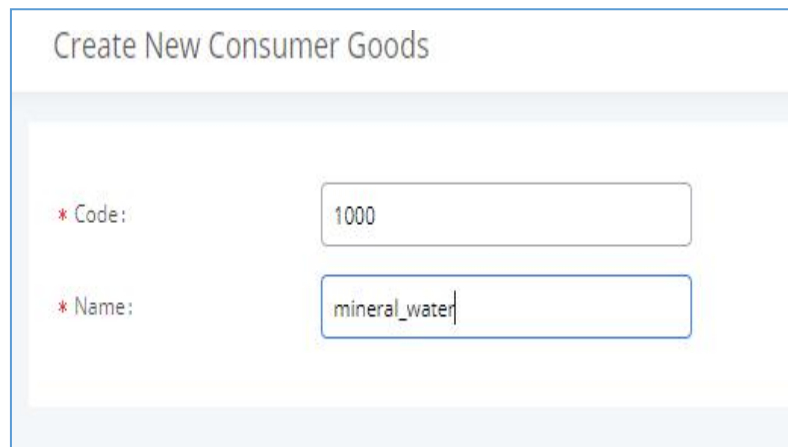


**Figure 281: Create New Maid**

**Table 127: Create New Maid**

<b>Maid Code</b>	Enter the Code to use when the maid wants to use the Mini Bar.
<b>Secret</b>	Enter the password associated with the maid.

In order to create a new consumer goods, click on [+ Create New Consumer Goods](#) under UCM webGUI→**Value-added Features**→**PMS**→**Mini Bar**, the following window will popup.



**Figure 282: Create New Consumer Goods**

<b>Code</b>	Enter the Goods Code.
<b>Name</b>	Enter the Name of the Goods



The Minibar page displays as:





PMS

Basic Settings    Room Status    Wakeup Service    Mini Bar

+ Create New Mini Bar



Code ↕	Name ↕	Options
4000	MiniBar	 

+ Create New Maid

Maid Code ↕	Secret	Options
1100	123456	 

Total: 1  1       10 条/页 ▾    跳至  页

+ Create New Consumer Goods

Code ↕	Name ↕	Options
7000	mineral_water	 

Total: 1  1       10 条/页 ▾    跳至  页

**Figure 283: Mini Bar**



## WAKEUP SERVICE

The Wakeup service can be used to schedule a reminder or wake up calls to any valid destination. This service is available on the UCM6200 as a separated module.

There are three ways to set up Wakeup Service:

- Using admin login
- Using user portal
- Using feature code

### Wakeup Service using Admin Login

1. Login to the UCM as admin.
2. Wakeup service can be found under Web GUI → **Value-added Features** → **Wakeup Service**, click on [+ Create New Wakeup Service](#) to create a new wakeup service. The following window will pop up.

The screenshot shows a web form titled "Create New Wakeup Service". The form contains the following fields and options:

- Enable Wakeup Service:**
- \* Name:**
- Prompt:**  Prompt
- Custom Date:**
- \* Date:**
- \* Time:**
- Members:**
  - Available (3 items):**  1002,  1003,  1004
  - Selected (2 items):**  1000,  1001

**Figure 284: Create New Wakeup Service**


3. Fill out the required fields and select the members to add to the wakeup group.





**Table 128: Wakeup Service**

<b>Enable Wakeup Service</b>	Enable Wakeup service.
<b>Name</b>	Enter a name (up to 64 characters) to identify the wakeup service.
<b>Prompt</b>	Select the prompt to play for that extension.
<b>Custom Date</b>	If disabled, users can select a specific date and time. If enabled users can select multiple days of the week to perform the wakeup.
<b>Date</b>	Select the date or dates when to performs the wakeup call.
<b>Time</b>	Select the time when to play the wakeup call.
<b>Members</b>	Select the members involved within the wakeup service group.

4. Click  and  to apply the changes.



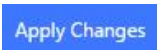
A wakeup service entry is created. The UCM will send a wakeup call to every extension in the member list at the scheduled date and time.

**Note:** the wakeup service has the following limitation on how many members can be added depending on UCM model.

**Table 129: Max Wakeup Members**

UCM Model	Max members in a Wakeup Service
UCM6202	50
UCM6204	50
UCM6208	100
UCM6510	100


## Wakeup Service from User Portal

1. Login to the user portal on the UCM6200.
2. Wakeup service can be found under “Value-added Features→Wakeup Service”, click on  to create a new wakeup service.
3. Configures the Name, Prompt, Date and Time for the user to make the wakeup to.
4. Click  and  to apply the changes.



## Wakeup Service using Feature Code

- Login to the UCM as admin.
- Enable “Wakeup Service” from the WebGUI under “**Call Features**→**Feature Codes**”.



* Listen Spy :	*54	
* Barge Spy :	*56	
* PMS Wakeup Servi...	*35	<input checked="" type="checkbox"/>
* Presence Status :	*48	<input checked="" type="checkbox"/>
* Whisper Spy :	*55	
* Wakeup Service :	*36	<input checked="" type="checkbox"/>
* Update PMS Room...	*23	<input checked="" type="checkbox"/>

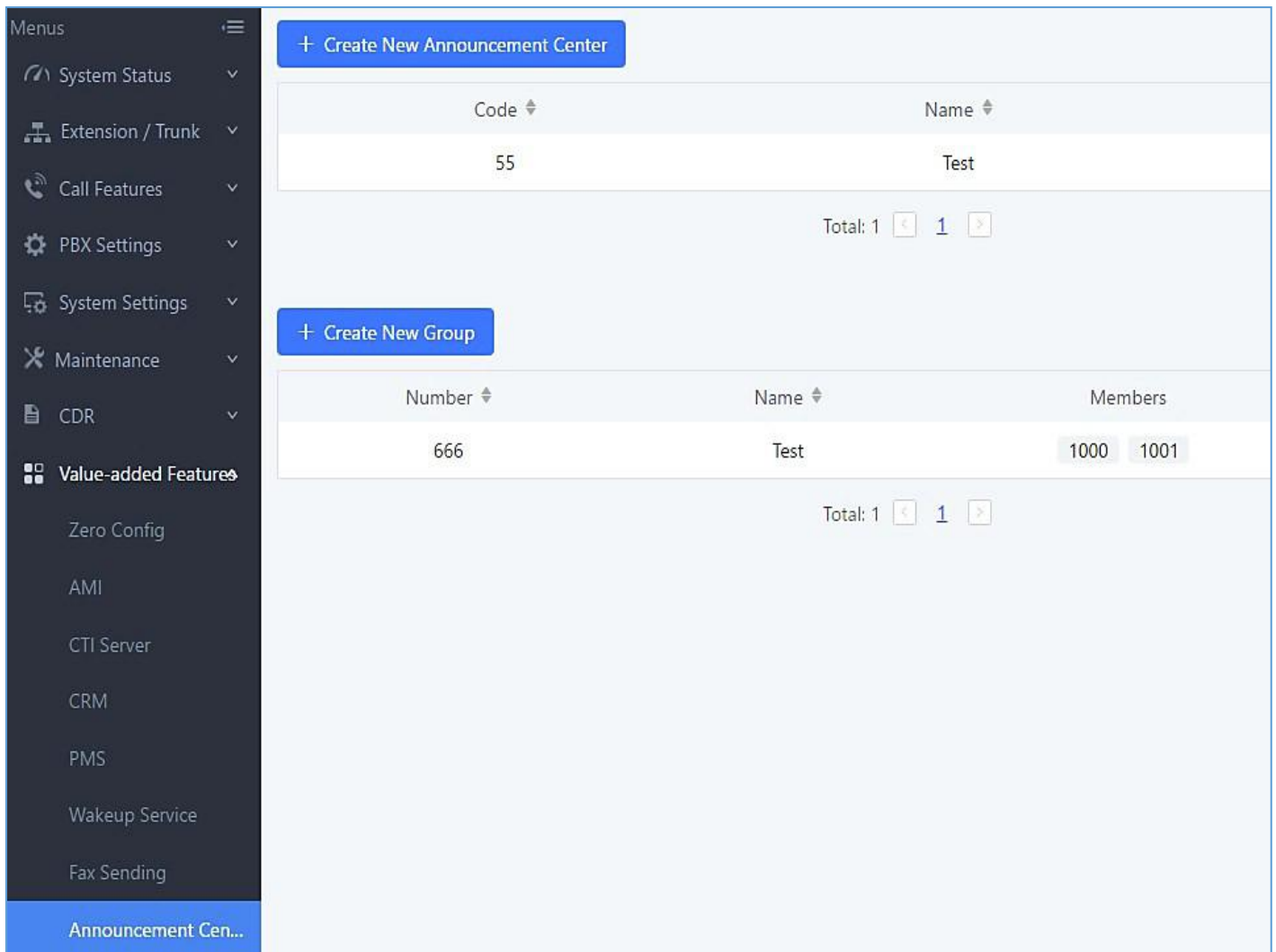
Figure 285: Wakeup Service Feature Code

- Click **Save** and **Apply Changes** to apply the changes.
- Dial “\*36” which is the feature code by default to access to the UCM wakeup service to add, update, activate or deactivate UCM wakeup service.
- Users have the choice to set a wakeup service for tomorrow or specify another day when adding a wakeup service.
- A wakeup service entry is created. The UCM will send a wakeup call at the scheduled date and time.



## ANNOUNCEMENTS CENTER

The UCM6200 supports Announcements Center feature which allows users to pre-record and store voice message into UCM6200 with a specified code. The users can also create group with specified extensions. When the code and the group number are dialed together in the combination of **code + group number**, the specified voice message is sent to all group members and only extensions in the group will hear the voice message.



The screenshot displays the 'Announcements Center' configuration page. On the left is a navigation menu with the following items: System Status, Extension / Trunk, Call Features, PBX Settings, System Settings, Maintenance, CDR, Value-added Features (Zero Config, AMI, CTI Server, CRM, PMS, Wakeup Service, Fax Sending), and Announcements Center (highlighted). The main content area features a '+ Create New Announcement Center' button. Below this is a table listing announcement centers:

Code	Name
55	Test

Below the table, the total count is shown as 'Total: 1' with a page number '1'. A '+ Create New Group' button is located below the table. Underneath is a table listing groups:

Number	Name	Members
666	Test	1000 1001

Below this table, the total count is shown as 'Total: 1' with a page number '1'.

Figure 286: Announcements Center



## Announcements Center Settings

**Table 130: Announcements Center Settings**


<b>Name</b>	Configure a name for the newly created Announcements Center to identify this announcement center.
<b>Code</b>	Enter a code number for the custom prompt. This code will be used in combination with the group number. For example, if the code is 55, and group number is 666. The user can dial 55666 to send prompt 55 to all members in group 666. <b>Note:</b> The combination number must not conflict with any number in the system such as extension number or conference number.
<b>Custom Prompt</b>	This option is to set a custom prompt as an announcement to notify group members. The file can be uploaded from page 'Custom Prompt'. Click 'Prompt' to add additional record.
<b>Ring Timeout</b>	Configure the ring timeout for the group members. The default value is 30 seconds.

## Group Settings

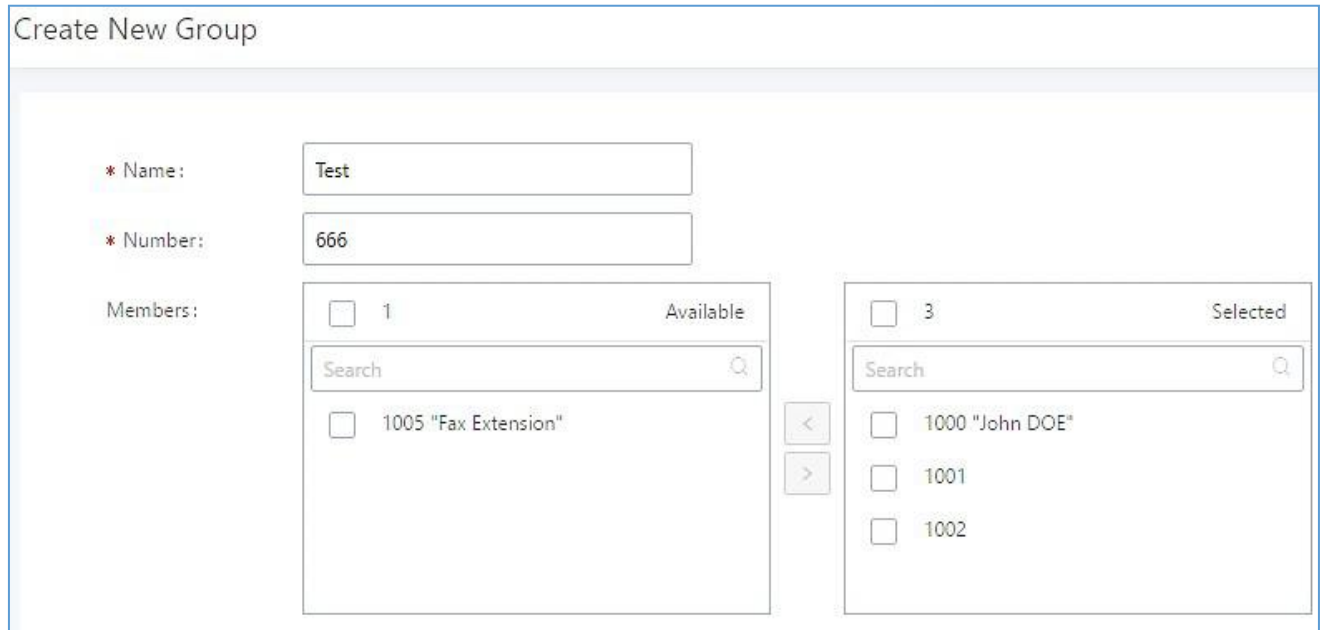
**Table 131: Group Settings**

<b>Name</b>	Configure a name for the newly created group to identify the group. <b>Note:</b> Name cannot exceed 64 characters
<b>Number</b>	Configure the group number. The group number is used in combination with the code. For example, if group number is 666, and code is 55. The user can dial 55666 to send prompt 55 to all members in group 666. <b>Note:</b> The combination number must not conflict with any number in the system such as extension number or conference number.

Announcements Center feature can be found under Web GUI→**Value-added Features**→**Announcements Center**. The following example demonstrates the usage of this feature.

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

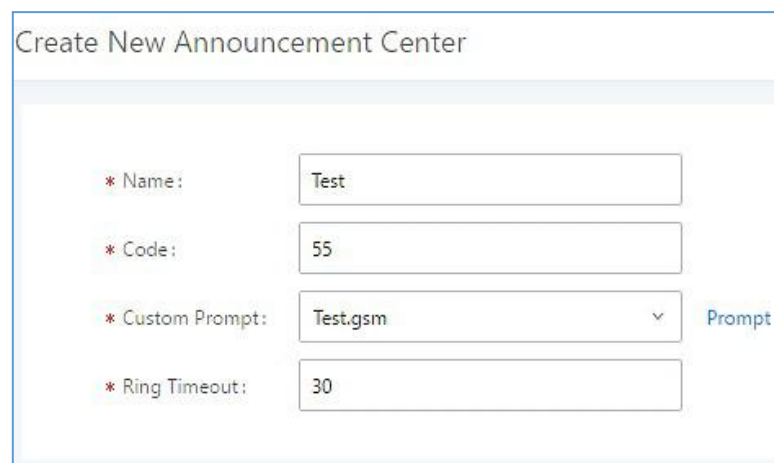




**Figure 287: Announcements Center Group Configuration**

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

1. Click [+ Create New Announcement Center](#) to create a new Announcement Center.
2. Give a name to the newly created Announcement Center.
3. Specify the code which will be used with group number to send the voice message to.
4. Select the message that will be used by the code from the Custom Prompt drop down menu. To create a new Prompt, please click “Prompt” link and follow the instructions in that page.



**Figure 288: Announcements Center Code Configuration**



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

+ Create New Announcement Center

Code ↕	Name ↕	Options
55	Test	<span style="font-size: 1.2em;">✎</span> <span style="font-size: 1.2em;">🗑️</span>

Total: 1 ◀ 1 ▶

10 条/页 跳至 1

+ Create New Group

Number ↕	Name ↕	Members	Options
666	Test	<span style="background-color: #e9ecef; padding: 2px 5px; border-radius: 3px;">1000</span> <span style="background-color: #e9ecef; padding: 2px 5px; border-radius: 3px; margin-left: 5px;">1001</span> <span style="background-color: #e9ecef; padding: 2px 5px; border-radius: 3px; margin-left: 5px;">1002</span>	<span style="font-size: 1.2em;">✎</span> <span style="font-size: 1.2em;">🗑️</span>

**Figure 289: Announcements Center Example**



## QUEUOMETRICS INTEGRATION

UCM6200 series have integrated QueueMetrics which is a highly scalable monitoring and reporting suite that addresses the needs of thousands of contact centers worldwide and offers a broad range of integrated benefits.

UCM6200 is currently supporting the following features with QueueMetrics: Agent login, Agent Logoff, Realtime Monitoring-Pausing, Realtime Monitoring-Barging, Realtime Monitoring-Transferring, Realtime Monitoring-End calls, Generating Performance Report-Quick and Agent today.

### API Configuration Parameters

Configuration page of the QueueMetrics can be accessed via admin login, on the UCM Web GUI → **Value-added Features** → **QueueMetrics**.

To Integrate QueueMetrics with the UCM, please follow the steps below:

1. Enable the option QueueMetrics Integration.
2. Enter the QueueMetrics URL provided.
3. Set the Username and the account code to the account name and password provided from QueueMetrics.
4. Click on Save and Apply Changes.

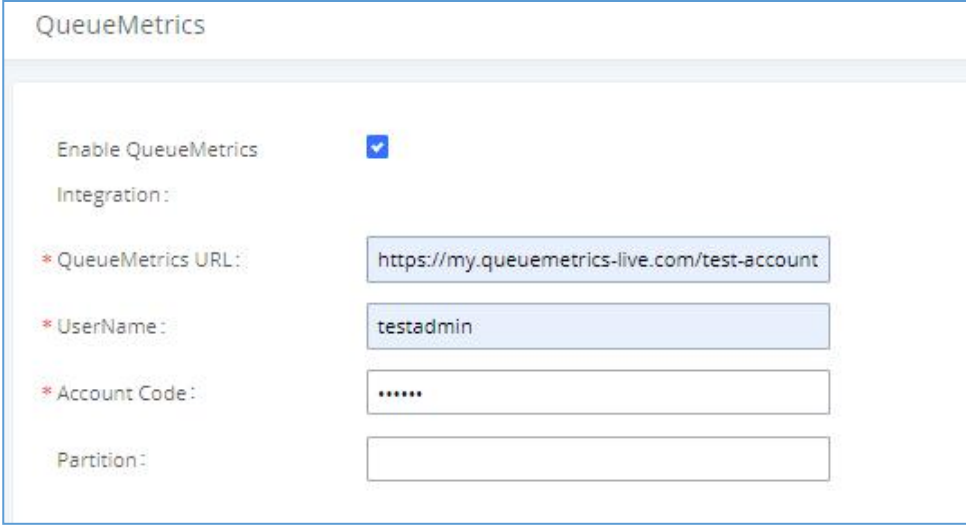


Figure 290: QueueMetrics configuration



**Table 132: QueueMetrics Configuration Parameters**

<b>QueueMetrics</b>	
<b>Enable QueueMetrics Integration</b>	Enable/Disable QueueMetrics Integration.
<b>QueueMetrics URL</b>	Configure the URL for QueueMetrics Integration.
<b>Username</b>	Configure the username for QueueMetrics Authentication.
<b>Account Code</b>	Configure the password for QueueMetrics Authentication.
<b>Partition</b>	Configure Data storage partition identifier.

For more details, please refer to online guide:

[http://www.grandstream.com/sites/default/files/Resources/QueueMetrics\\_integration.pdf](http://www.grandstream.com/sites/default/files/Resources/QueueMetrics_integration.pdf)





## STATUS AND REPORTING

### PBX Status

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

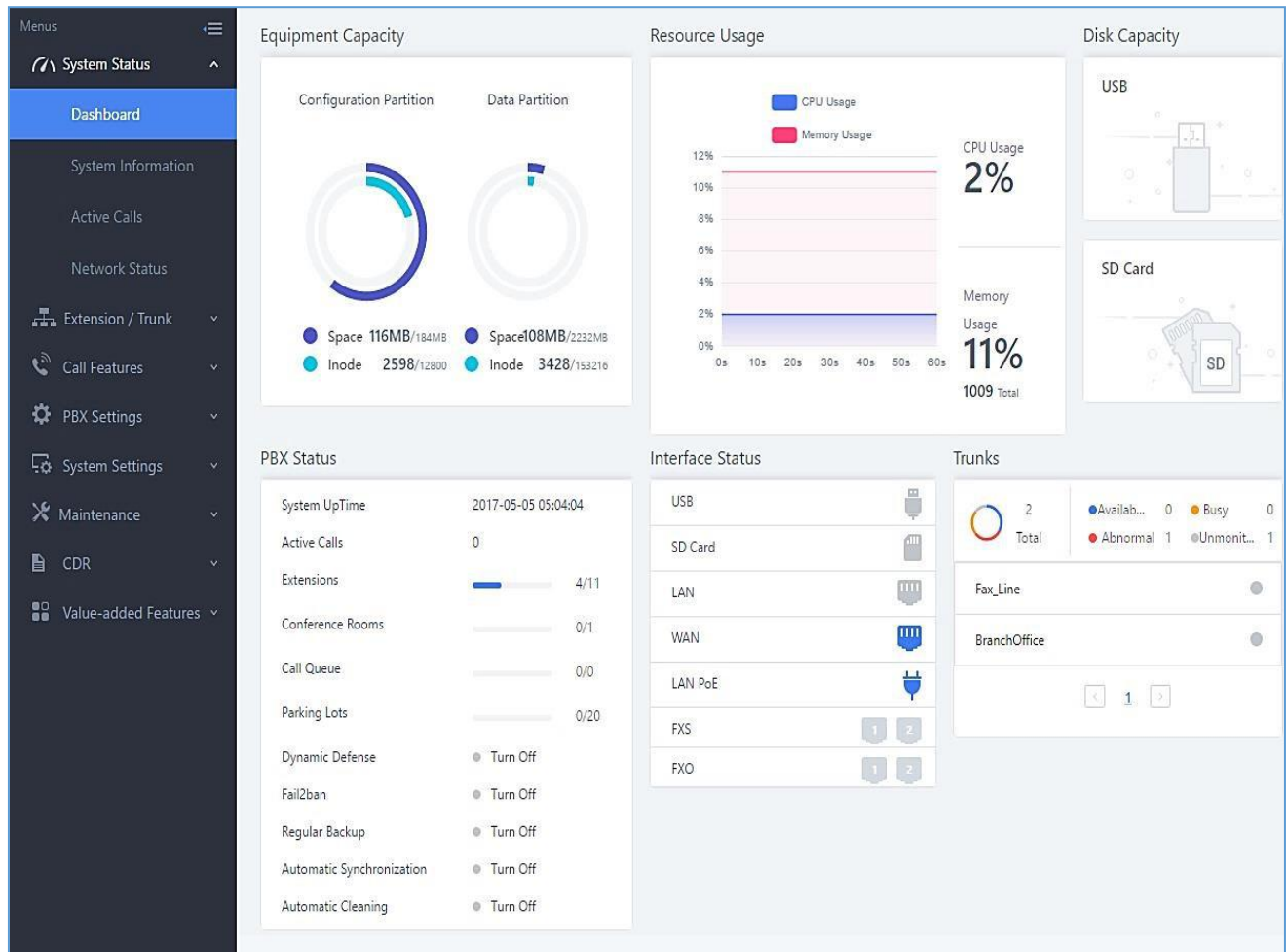
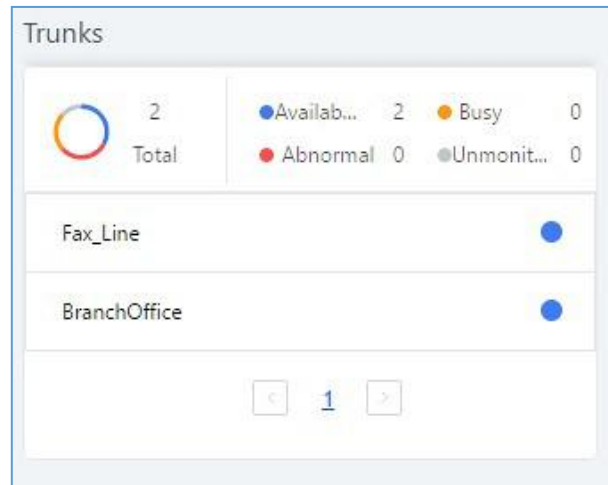


Figure 291: Status→PBX Status



## Trunks

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



**Figure 292: Trunk Status**

**Table 133: Trunk Status**

<p><b>Status</b></p>	<p>Display trunk status.</p> <ul style="list-style-type: none"> <li><u>Analog trunk status:</u> <ul style="list-style-type: none"> <li><b>Available</b></li> <li><b>Busy</b></li> <li><b>Unavailable</b></li> <li><b>Unknown Error</b></li> </ul> </li> <li><u>SIP Peer trunk status:</u> <ul style="list-style-type: none"> <li><b>Unreachable:</b> The hostname cannot be reached.</li> <li><b>Unmonitored:</b> Heartbeat feature is not turned on to be monitored.</li> <li><b>Reachable:</b> The hostname can be reached.</li> </ul> </li> <li><u>SIP Register trunk status:</u> <ul style="list-style-type: none"> <li><b>Registered</b></li> <li><b>Unrecognized Trunk</b></li> </ul> </li> </ul>
<p><b>Trunks</b></p>	<p>Display trunk name</p>
<p><b>Type</b></p>	<p>Display trunk Type:</p> <ul style="list-style-type: none"> <li>Analog</li> <li>SIP</li> <li>IAX</li> </ul>
<p><b>Username</b></p>	<p>Display username for this trunk.</p>
<p><b>Port/Hostname/IP</b></p>	<p>Display Port for analog trunk, or Hostname/IP for VoIP (SIP/IAX) trunk.</p>



## Extensions

Extensions Status can be seen from the same configuration page, users can go under Web GUI→**Extension/Trunk**→**Extensions** and following page will be displayed listing the extensions and their status information.

Follow Me Options		Search						
<input type="checkbox"/>	Status	Presence Status	Extension	CallerID Name	Terminal Type	IP and Port	Email Status	Options
<input type="checkbox"/>	In Use	Available	1000	John DOE	SIP	192.168.6.238:46365		
<input type="checkbox"/>	Unavailable	Available	1001		SIP	--		
<input type="checkbox"/>	In Use	Available	1002		SIP	192.168.6.238:46365		
<input type="checkbox"/>	Ringling	Available	1003		SIP	192.168.6.102:5060		
<input type="checkbox"/>	Idle	Available	1004		SIP	192.168.6.102:5062		

**Figure 293: Extension Status**

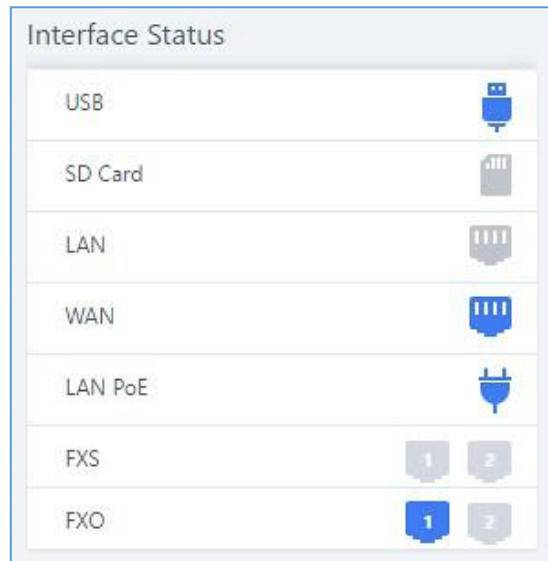
**Table 134: Extension Status**

<b>Status</b>	Display extension number (including feature code). The color indicator has the following definitions. <ul style="list-style-type: none"> <li>• Green: Free</li> <li>• Blue: Ringling</li> <li>• Yellow: In Use</li> <li>• Grey: Unavailable</li> </ul>
<b>Presence Status</b>	Display the presence status of the extension.
<b>Extension</b>	Display the extension number.
<b>CallerID Name</b>	First name and last name of the extension.
<b>IP and Port</b>	Display the IP and port number of the registered device.
<b>Email</b>	Display Email Notification status for the extension. When notification is waiting for be sent, shows  and once sent it will display
<b>Terminal Type</b>	Displays extension type. <ul style="list-style-type: none"> <li>• SIP User</li> <li>• IAX User</li> <li>• Analog User</li> <li>• Ring Groups</li> <li>• Voicemail Groups</li> </ul>






## Interfaces Status

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






**Figure 294: UCM6204 Interfaces Status**

**Table 135: Interface Status Indicators**

	USB disconnected.
	USB connected.
	SD Card connected.
	SD Card disconnected.
	LAN/WAN connected.
	LAN/WAN not configured.
	LAN/WAN disconnected.
	FXS/FXO connected.
	FXS/FXO waiting.



	FXS/FXO busy.
	FXS/FXO not configured.
	FXS/FXO disconnected.

## System Status

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

### General

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

Table 136: System Status→General

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

### Network

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



**Table 137: System Status→Network**

<b>System Status→System Status→Network</b>	
<b>MAC Address</b>	Global unique ID of device, in HEX format. The MAC address can be found on the label coming with original box and on the label located on the bottom of the device.
<b>IP Address</b>	IP address.
<b>Gateway</b>	Default gateway address.
<b>Subnet Mask</b>	Subnet mask address.
<b>DNS Server</b>	DNS Server address.
<b>Speed</b>	Network Port Speed
<b>Duplex Mode</b>	Shows the Duplex Mode (Full/Half Duplex)

## Storage Usage

Users could access the storage usage information from Web GUI→**System Status**→**Dashboard**→**Storage Usage**. It shows the available and used space for Space Usage and Inode Usage.

Space Usage includes:

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

Inode Usage includes:

- **Configuration partition**
- **Data partition**

### Note:

Inode is the pointer used for file reference in the system. The system usually has limited resources of pointers.



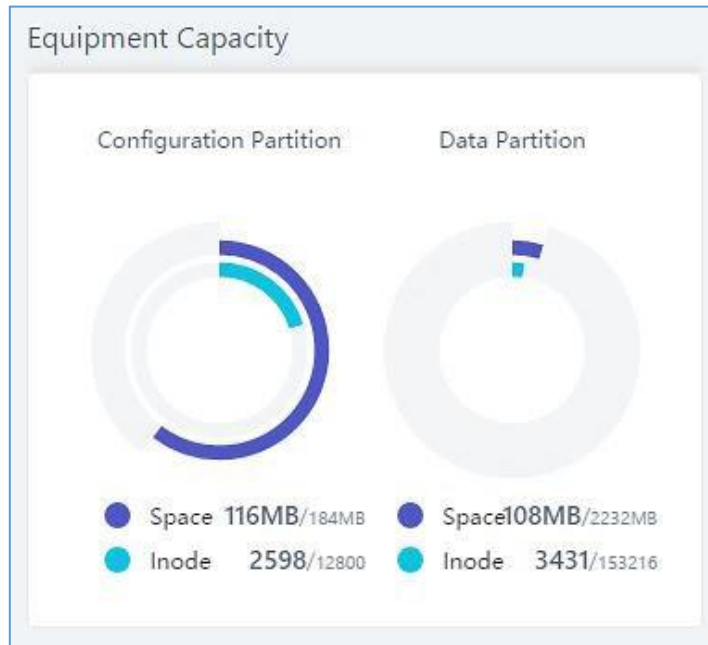


Figure 295: System Status→Storage Usage

## Resource Usage

When configuring and managing the UCM6200, users could access resource usage information to estimate the current usage and allocate the resources accordingly. Under Web GUI:

→**System Status**→**Dashboard**→**Resource Usage**, the current CPU usage and Memory usage are shown in the pie chart.

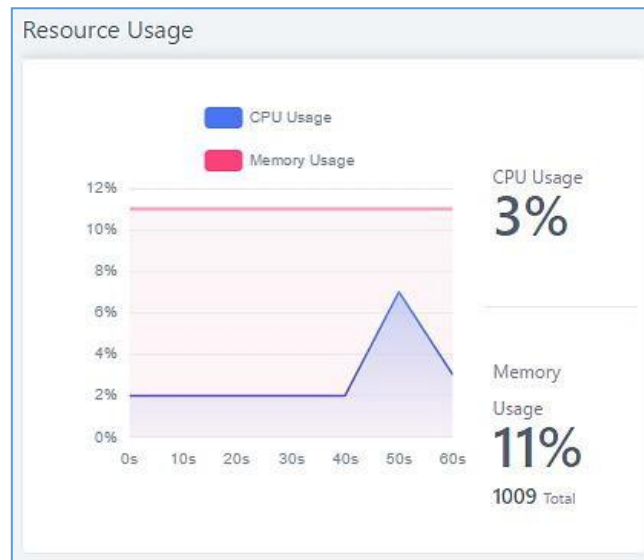


Figure 296: System Status→Resource Usage



## System Events

The UCM6200 can monitor important system events, log the alerts and send Email notifications to the system administrator.

### Alert Events List


The system alert events list can be found under Web GUI→**Maintenance**→**System Events**. The following event and their actions are currently supported on the UCM62xx which will have alert and/or Email generated if occurred:

Table 138: Alert Events

Action index	Alert Events
1	Disk Usage
2	Modify Super Admin Password
3	Memory Usage
4	System Reboot
5	System Update
6	System Crash
7	Register SIP failed
8	Register SIP trunk failed
9	Restore Config
10	User login success
11	User login failed
12	SIP Internal Call Failure
13	SIP Outgoing Call through Trunk Failure
14	Fail2ban Blocking
15	SIP Lost Registration
16	SIP Peer Trunk Status
17	User Login Banned
18	External Disk Usage
19	HA failure warning
20	Emergency Calls
21	The CDR database is corrupted
22	NAS





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

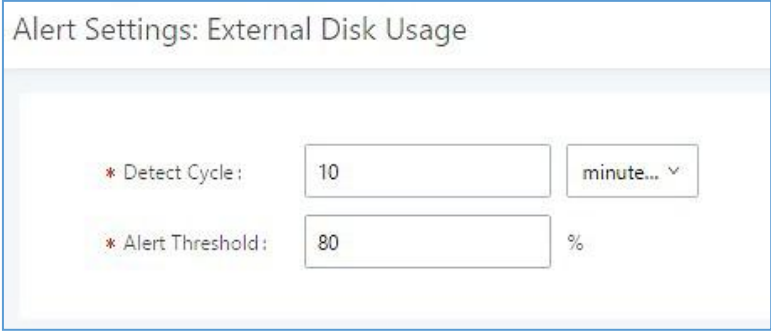
### 1. Disk Usage



**Figure 297: System Events→Alert Events Lists: Disk Usage**

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

### 2. External Disk Usage

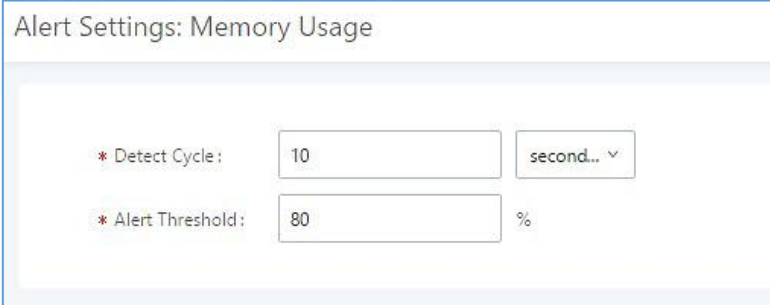


**Figure 298: System Events→Alert Events Lists: External Disk Usage**

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



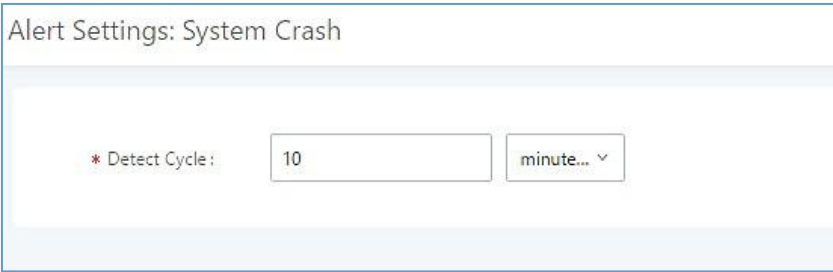
### 3. Memory Usage



**Figure 299: System Events→Alert Events Lists: Memory Usage**


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

### 4. System Crash



**Figure 300: System Events→Alert Events Lists: System Crash**

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

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

## Alert Log

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



System Events

**Alert Log**    Alert Events List    Alert Contact

---

Alert Log Filter

Delete Search Result (s)    Delete All

Time	Event Name	Type	Content
2017-05-04 04:33:20	User login success	Generate Alert	Logged in system successfully! The username is: admin!P:192.168.6.246
2017-05-04 04:33:15	User login failed	Generate Alert	Logged in system failed! The username is: admin!P:192.168.6.246

**Figure 301: System Events→Alert Log**

User could also filter alert logs by selecting a certain event category, type of alert log, and/or specifying a certain time period. The matching results will be displayed after clicking on Filter. Alert logs are classified into two types by the system:

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

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

**Alert Log**    Alert Events List    Alert Contact

---

Alert Log Filter    Reset

Event Name:

Type:

Start Time:

End Time:

Delete Search Result (s)    Delete All

Time	Event Name	Type	Content
2017-05-04 04:33:15	User login failed	Generate Alert	Logged in system failed! The username is: admin!P:192.168.6.246

**Figure 302: Filter for Alert Log**



## Alert Contact

This feature allows the administrator to be notified when one of the Alert events mentioned above happens. Users could add administrator's Email address under Web GUI→**Maintenance**→**System Events**→**Alert Contact** to send the alert notification to an email (Up to 10 Email addresses can be added) or also specify an HTTP server where to send this alert.

**Table 139: Alert Contact**

<b>Super Admin Email</b>	Configure the email addresses to send alert notifications to. Up to 10 email addresses can be added.
<b>Admin Email</b>	Configure the email addresses to send alert notifications to. Up to 10 email addresses can be added.
<b>Email Template</b>	Please refer to section <b>Email Templates</b>
<b>Protocol</b>	Protocol used to communicate with the server. HTTP or HTTPS. Default one is <b>HTTP</b> .
<b>HTTP Server</b>	The IP address or FQDN of the HTTP/HTTPS server.
<b>HTTP Server Port</b>	HTTP/HTTPS port
<b>Warning Template</b>	Customize the template used for system warnings. By default: <code>{"action":"\${ACTION}","mac":"\${MAC}","content":"\${WARNING_MSG}"}</code>
<b>Notification Template</b>	Customize the notification template to receive relevant alert information. By default: <code>{"action":"\${ACTION}","cpu":"\${CPU_USED}","memery":"\${MEM_USED}","disk":"\${DISK_USED}","external_disk":"\${EXTERNAL_DISK_USED}"}</code> <b>Note:</b> The notification message with <b>"action:0"</b> will be sent periodically if Notification Interval is set.
<b>Notification Interval</b>	Modifies the frequency at which notifications are sent in seconds. No notifications will be sent if the value is "0". Default value: <b>20</b>
<b>Template Variables</b>	<code>\${MAC}</code> : MAC Address <code>\${WARNING_MSG}</code> : Warning message <code>\${TIME}</code> : Current System Time <code>\${CPU_USED}</code> : CPU Usage <code>\${MEM_USED}</code> : Memory Usage <code>\${ACTION}</code> : Message Type. Refer to [Table 138: Alert Events] <code>\${DISK_USED}</code> : Disk Usage <code>\${EXTERNAL_DISK_USED}</code> : Disk Usage

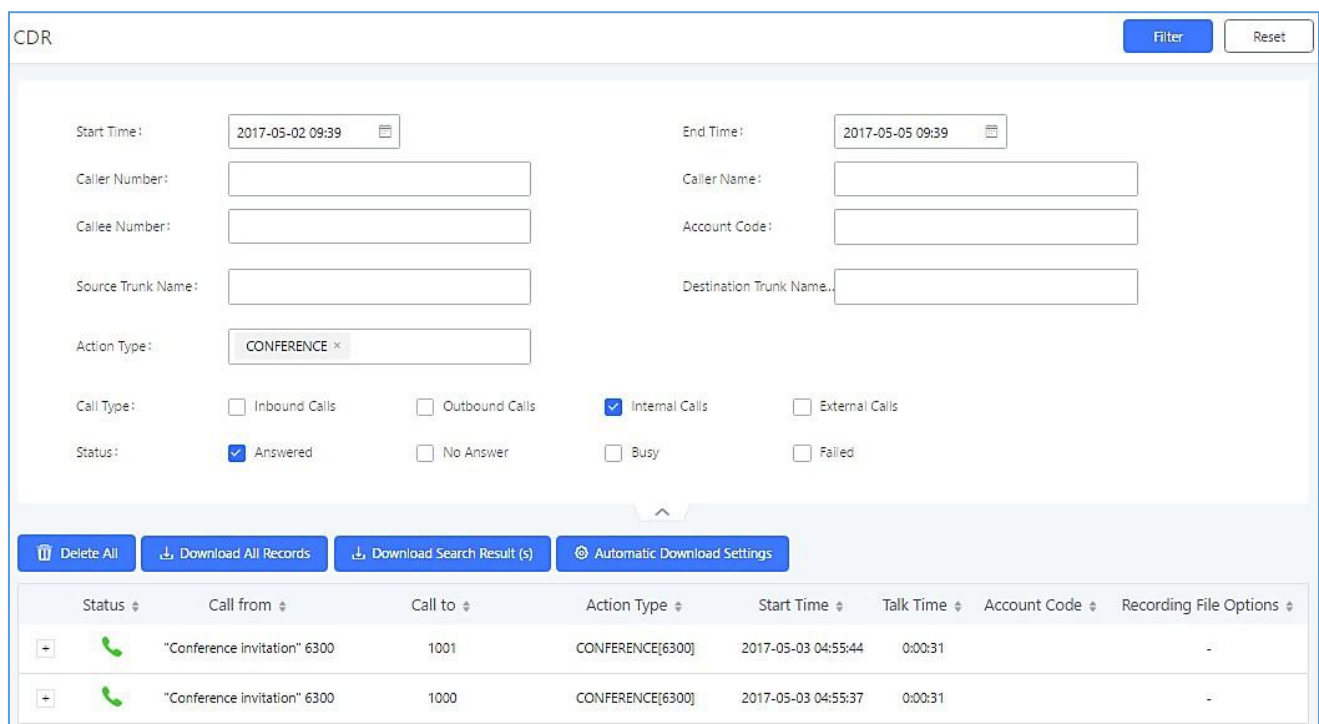


## CDR

CDR (Call Detail Record) is a data record generated by the PBX that contains attributes specific to a single instance of phone call handled by the PBX. It has several data fields to provide detailed description for the call, such as phone number of the calling party, phone number of the receiving party, start time, call duration, etc.

On the UCM6200, the CDR can be accessed under Web GUI→**CDR**→**CDR**. Users could filter the call report by specifying the date range and criteria, depending on how the users would like to include the logs to the report. Click on "Search" button to display the generated report.

**Important Note:** Starting from fw 1.0.15.16, the UCM has **CDR separation** and it will display only CDR for the current month. And in order to get older data, users need to apply filter by specifying the date range.



The screenshot shows the CDR Filter web interface. At the top right, there are 'Filter' and 'Reset' buttons. The main area contains various input fields for filtering: Start Time (2017-05-02 09:39), End Time (2017-05-05 09:39), Caller Number, Caller Name, Callee Number, Account Code, Source Trunk Name, and Destination Trunk Name. There is also an 'Action Type' dropdown menu set to 'CONFERENCE'. Below these are radio buttons for 'Call Type' (Inbound Calls, Outbound Calls, Internal Calls, External Calls) and 'Status' (Answered, No Answer, Busy, Failed). At the bottom, there are buttons for 'Delete All', 'Download All Records', 'Download Search Result (5)', and 'Automatic Download Settings'. Below the buttons is a table with the following data:

Status	Call from	Call to	Action Type	Start Time	Talk Time	Account Code	Recording File Options
<input checked="" type="checkbox"/>	"Conference Invitation" 6300	1001	CONFERENCE[6300]	2017-05-03 04:55:44	0:00:31		-
<input checked="" type="checkbox"/>	"Conference Invitation" 6300	1000	CONFERENCE[6300]	2017-05-03 04:55:37	0:00:31		-

**Figure 303: CDR Filter**

**Table 140: CDR Filter Criteria**

<b>Call Type</b>	Groups the following: <ul style="list-style-type: none"> <li>• <b>Inbound calls:</b> Inbound calls are calls originated from a non-internal source (like a VoIP trunk) and sent to an internal extension.</li> <li>• <b>Outbound calls:</b> Outbound calls are calls sent to a non-internal source (like a VoIP trunk) from an internal extension.</li> <li>• <b>Internal calls:</b> Internal calls are calls from one internal extension to another extension, which are not sent over a trunk.</li> </ul>
------------------	---




	<ul style="list-style-type: none"> <li>• <b>External calls:</b> External calls are calls sent from one trunk to another trunk, which are not sent to any internal extension.</li> </ul>
<b>Status</b>	Filter with the call status, the available statuses are the following: <ul style="list-style-type: none"> <li>• Answered</li> <li>• No Answer</li> <li>• Busy</li> <li>• Failed</li> </ul>
<b>Source Trunk Name</b>	Select source trunk(s) and the CDR of calls going through inbound the trunk(s) will be filtered out.
<b>Destination Trunk Name</b>	Select destination trunk(s) and the CDR of calls going outbound through the trunk(s) will be filtered out.
<b>Action Type</b>	Filter calls using the Action Type, the following actions are available: <ul style="list-style-type: none"> <li>• Dial</li> <li>• Announcements</li> <li>• Callback</li> <li>• Call Forward</li> <li>• Conference</li> <li>• DISA</li> <li>• Fax</li> <li>• Follow Me</li> <li>• IVR</li> <li>• Page</li> <li>• Parked Call</li> <li>• Queue</li> <li>• Ring Group</li> <li>• Transfer</li> <li>• VFax</li> <li>• VM</li> <li>• VMG</li> <li>• Wakeup</li> <li>• Emergency Call</li> <li>• Emergency Notify</li> <li>• SCA</li> </ul>
<b>Extension Group</b>	Specify the Extension Group name to filter with.
<b>Account Code</b>	Select the account Code to filter with. If pin group CDR is enabled, the call with pin group information will be displayed as part of the CDR under Account Code Field.
<b>Start Time</b>	Specify the start time to filter the CDR report. Click on the calendar icon on the right and the calendar will show for users to select the exact date and time.
<b>End Time</b>	Specify the end time to filter the CDR report. Click on the calendar icon on the right and the calendar will show for users to select the exact date and time.




<b>Caller Number</b>	<p>Enter the caller number to filter the CDR report. CDR with the matching caller number will be filtered out.</p> <p>User could specify a particular caller number or enter a pattern. '.' matches zero or more characters, only appears in the end. 'X' matches any digit from 0 to 9, case-insensitive, repeatable, only appears in the end.</p> <p><u>For example:</u></p> <p><b>3XXX:</b> It will filter out CDR that having caller number with leading digit 3 and of 4 digits' length.</p> <p><b>3.:</b> It will filter out CDR that having caller number with leading digit 3 and of any length.</p>
<b>Caller Name</b>	Enter the caller name to filter the CDR report. CDR with the matching caller name will be filtered out.
<b>Callee Number</b>	Enter the callee number to filter the CDR report. CDR with the matching callee number will be filtered out.

The call report will display as the following figure shows.

Status	Call from	Call to	Action Type	Start Time	Talk Time	Account Code	Recording File Option
	"John DOE" 1000	1001	DIAL	2017-05-04 04:41:49	0:00:05		-

Status	Call from	Call to	Action Type	Start Time	Talk Time	Account Code	Recording File Options
	"John DOE" 1000	1001	DIAL	2017-05-04 04:41:49	0:00:05		-

**Figure 304: Call Report**

The CDR report has the following data fields:

- Start Time**  
 Format: 2016-09-03 00:06:16
- Call Type**  
 Example:  
 IVR  
 DIAL  
 WAKEUP
- Call From**  
 Example format:  
 "John Doe" 2000
- Call To**  
 Example format: 2002



- **Call Time**  
Format: 0:00:02
- **Talk Time**  
Format: 0:00:00
- **Account Code**  
Example format:  
Grandstream/Test


- **Status**  
Answered, Busy, No answer or Failed.


Users could perform the following operations on the call report.

- **Sort by “Start Time”**  
Click on the header of the column to sort the report by "Start Time". Clicking on "Start Time" again will reverse the order.

- **Download Searched Results**  
Click on “Download Search Result(s)” to export the records filtered out to a .csv file.



- **Download All Records**  
Click on “Download All Records” to export all the records to a .csv file.

- **Delete All**  
Click on  **Delete All** button to remove all the call report information.

- **Delete Search Result**  
On the bottom of the page, click on  **Delete Search Result (s)** button to remove CDR records that appear on search results.

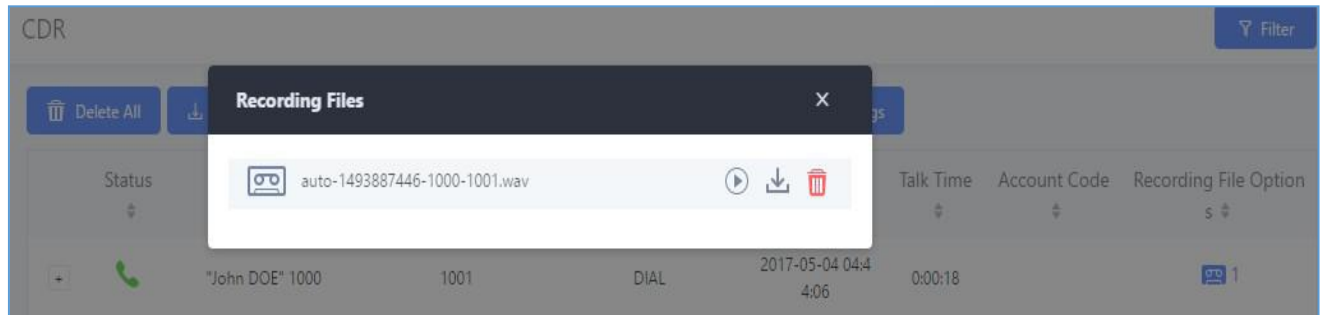
**Note:** When deleting CDR, a prompt will now appear asking whether to delete all recording files or not.

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

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



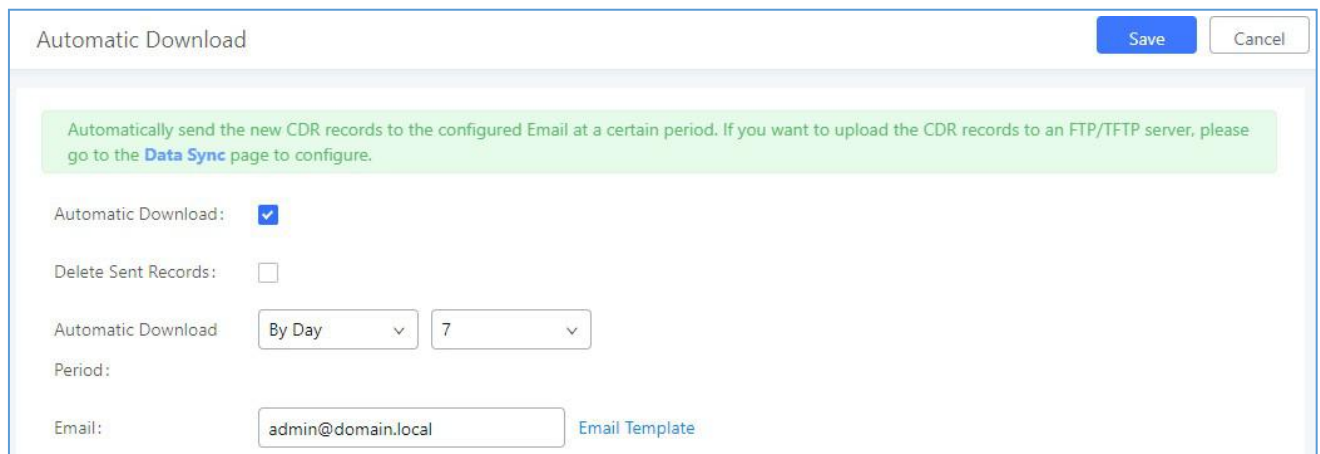




**Figure 305: Call Report Entry with Audio Recording File**

- **Automatic Download CDR Records**

User could configure the UCM6200 to automatically download the CDR records and send the records to multiple Email recipients in a specific hour. Click on “Automatic Download Settings” and configure the parameters in the dialog below.



**Figure 306: Automatic Download Settings**


To receive CDR record automatically from Email, check “Enable” and select a time period “By Day” “By Week” or “By Month”, select Hour of the day as well for the automatic download period. Make sure you have entered an Email or multiple email addresses where to receive the CDR records.

**Note:** users have the option to delete the sent records “Delete Sent Records”








### CDR Improvement

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



Status	Call from	Call to	Action Type	Start Time	Talk Time	Account Code	Recording File Option
	1002	1001	DIAL	2017-05-04 04:47:58	0:00:29		-

**Figure 307: CDR Report**

Status	Call from	Call to	Action Type	Start Time	Talk Time	Account Code	Recording File Option																								
	1002	1001	DIAL	2017-05-04 04:47:58	0:00:29		-																								
<table border="1"> <thead> <tr> <th>Status</th> <th>Call from</th> <th>Call to</th> <th>Action Type</th> <th>Start Time</th> <th>Talk Time</th> <th>Account Code</th> <th>Recording File Options</th> </tr> </thead> <tbody> <tr> <td></td> <td>1002</td> <td>1001</td> <td>DIAL</td> <td>2017-05-04 04:47:58</td> <td>0:00:14</td> <td></td> <td>-</td> </tr> <tr> <td></td> <td>1002</td> <td>1002</td> <td>TRANSFER</td> <td>2017-05-04 04:48:13</td> <td>0:00:15</td> <td></td> <td>-</td> </tr> </tbody> </table>								Status	Call from	Call to	Action Type	Start Time	Talk Time	Account Code	Recording File Options		1002	1001	DIAL	2017-05-04 04:47:58	0:00:14		-		1002	1002	TRANSFER	2017-05-04 04:48:13	0:00:15		-
Status	Call from	Call to	Action Type	Start Time	Talk Time	Account Code	Recording File Options																								
	1002	1001	DIAL	2017-05-04 04:47:58	0:00:14		-																								
	1002	1002	TRANSFER	2017-05-04 04:48:13	0:00:15		-																								

**Figure 308: Detailed CDR Information**

## Downloaded CDR File

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

- Caller number, Callee number**

"Caller number": the caller ID.

"Callee number": the callee ID.

If the "Source Channel" contains "DAHDI", this means the call is from FXO/PSTN line.

caller number	callee number	context	calerid	source channel	dest channel	lastapp
	2009	from-internal	"Wake Up Call" <WakeUp>	Local/2009@from-internal-00000001;2	PJSIP/2009-00000013	Dial
2007	31100	from-internal	"" <2007>	PJSIP/2007-00000014	DAHDI/1-1	Dial
2009	1100	from-internal	"John Doe" <2009>	PJSIP/2009-00000015	PJSIP/trunk_1-00000016	Dial
1100	2014	from-did-direct	"1100" <1100>	DAHDI/1-1	PJSIP/2014-00000017	Dial

**Figure 309: Downloaded CDR File Sample**

- Context**

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

**from-internal**: internal extension makes outbound calls.

**ext-did-XXXXX**: inbound calls. It starts with "ext-did", and "XXXXX" content varies case by case, which also relate to the order when the trunk is created.

**ext-local**: internal calls between local extensions.

- Source Channel, Dest Channel**

**Sample 1:**

caller number	callee number	context	calerid	source channel	dest channel	disposition
2007	31100	from-internal	"" <2007>	PJSIP/2007-00000014	DAHDI/1-1	ANSWERED

**Figure 310: Downloaded CDR File Sample - Source Channel and Dest Channel 1**



DAHDI means it is an analog call, FXO or FXS.

For UCM6202, DAHDI/(1-2) are FXO ports, and DAHDI(3-4) are FXS ports.

For UCM6204, DAHDI/(1-4) are FXO ports, and DAHDI(5-6) are FXS ports.

For UCM6208, DAHDI/(1-8) are FXO ports, and DAHDI(9-10) are FXS ports.

### Sample 2:

caller number	callee number	context	calerid	source channel	dest channel	lastapp
2009	1100	from-internal	"John Doe" <2009>	PJSIP/2009-00000015	PJSIP/trunk_1-00000016	Dial

Figure 311: Downloaded CDR File Sample - Source Channel and Dest Channel 2

"SIP" means it is a SIP call. There are three possible formats:

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

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

(c) **PJSIP/trunk\_X-XXXXXX**, where trunk\_X is the internal trunk name and it is an inbound call from this trunk. The last XXXXX is a random string and can be ignored.

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

**IAX2/NUM-XXXXXX**: it means this is an IAX call.

**Local/@from-internal-XXXXX**: it is used internally to do some special feature procedure. We can simply ignore it.

**Hangup**: the call is hung up from the dialplan. This indicates there are some errors, or it has run into abnormal cases.

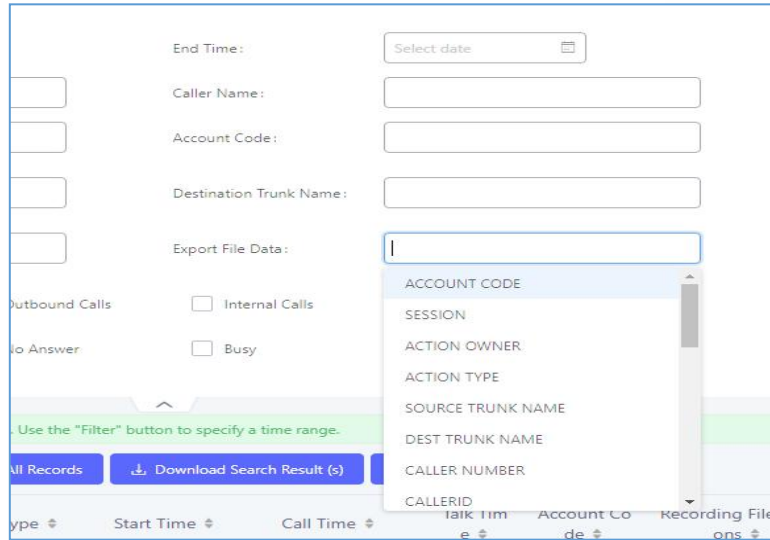
**Playback**: play some prompts to you, such as 183 response or run into an IVR.

**ReadExten**: collect numbers from user. It may occur when you input PIN codes or run into DISA

### CDR Export Customization

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



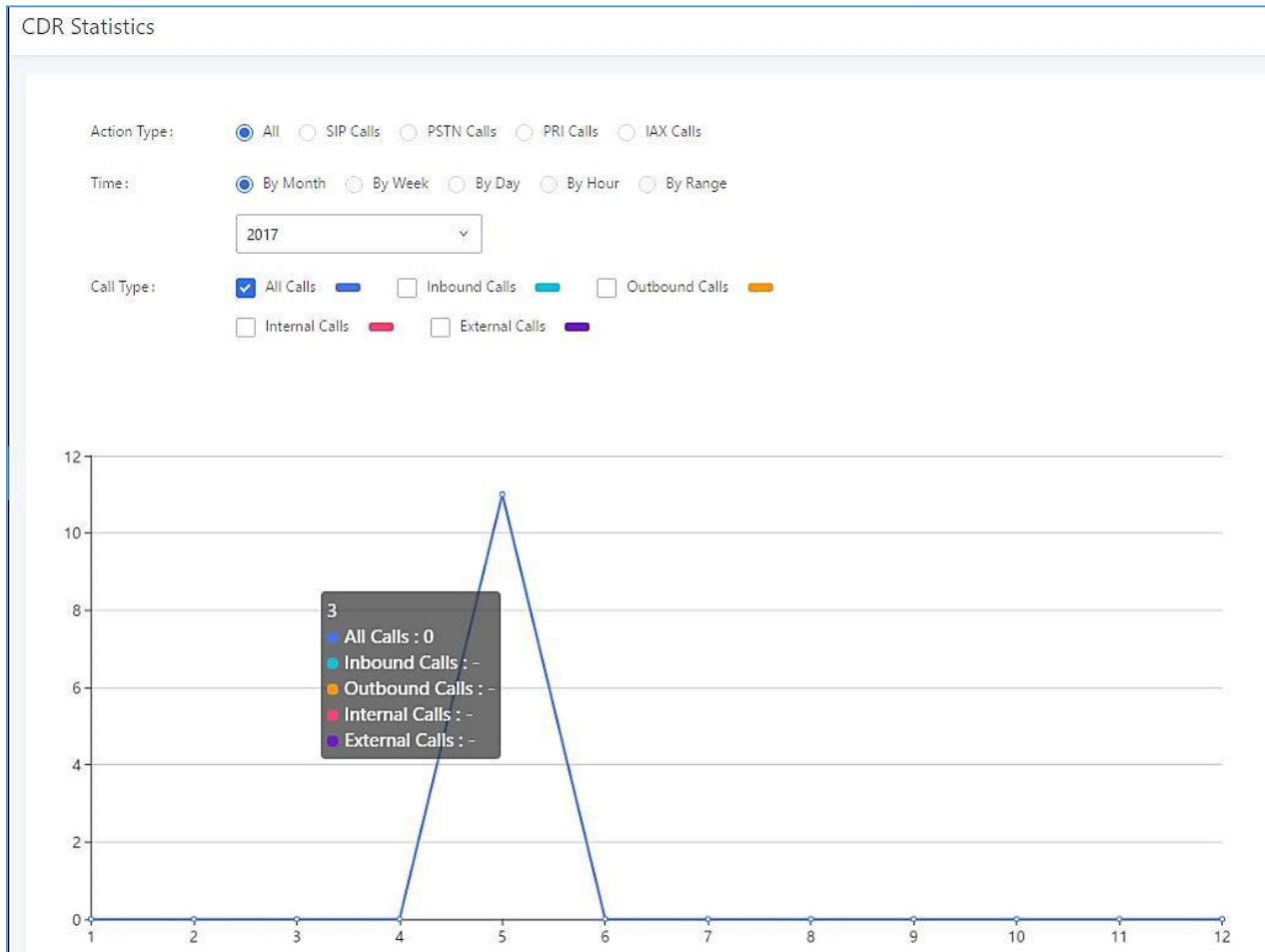


**Figure 312: CDR Export File data**

## Statistics

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





**Figure 313: CDR Statistics**

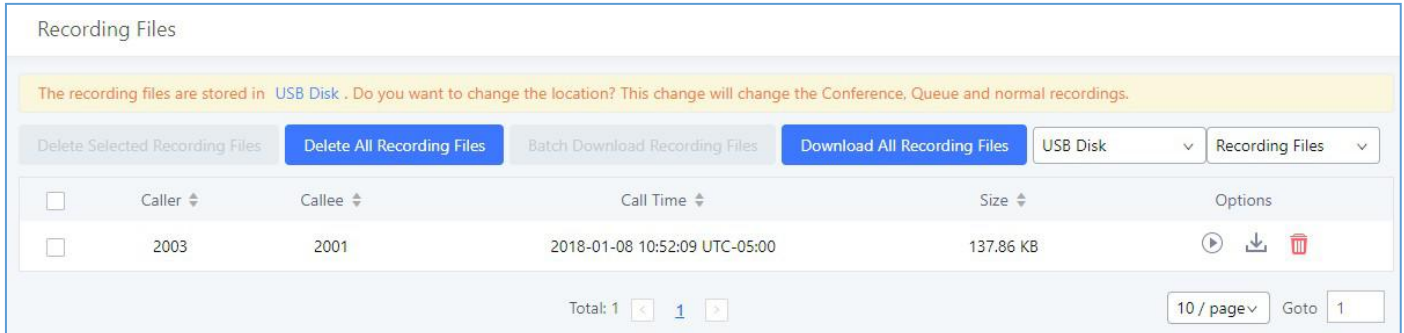
**Table 141: CDR Statistics Filter Criteria**

<b>Trunk Type</b>	Select one of the following trunk types. <ul style="list-style-type: none"> <li>All</li> <li>SIP Calls</li> <li>PSTN Calls</li> </ul>
<b>Call Type</b>	Select one or more in the following checkboxes. <ul style="list-style-type: none"> <li>Inbound calls</li> <li>Outbound calls</li> <li>Internal calls</li> <li>External calls</li> <li>All calls</li> </ul>
<b>Time Range</b>	<ul style="list-style-type: none"> <li>By month (of the selected year).</li> <li>By week (of the selected year).</li> <li>By day (of the specified month for the year).</li> <li>By hour (of the specified date).</li> <li>By range. For example, 2016-01 To 2016-03.</li> </ul>



## Recording Files

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



Recording Files

The recording files are stored in **USB Disk**. Do you want to change the location? This change will change the Conference, Queue and normal recordings.



USB Disk
Recording Files

<input type="checkbox"/>	Caller ↕	Callee ↕	Call Time ↕	Size ↕	Options
<input type="checkbox"/>	2003	2001	2018-01-08 10:52:09 UTC-05:00	137.86 KB	<input type="button" value="Play"/> <input type="button" value="Download"/> <input type="button" value="Delete"/>

Total: 1

10 / page

Figure 314: CDR→Recording Files

- Click on “**Delete Selected Recording Files**” to delete the recording files.
- Click on “**Delete All Recording Files**” to delete all recording files.
- Click on “**Batch Download Recording Files**” in order to download the selected recording files.
- Click on “**Download All Recording Files**” to download all recordings files.
- Select Either “**USB Disk**” or “**Local**” to show recording files stored on external or internal storage, depending on selected storage space.
- Select whether to show call recordings, queue recordings or conference recordings.
- Click on  to download the recording file in .wav format.
- Click on  to delete the recording file.
- To sort the recording file, click on the title "Caller", "Callee" or "Call Time" for the corresponding column. Click on the title again can switch the sorting mode between ascending order or descending order.


## API Configuration

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



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

**Table 142: API Configuration Files**

<b>Enable</b>	Enable/Disable API. The default setting is disabled.
<b>TLS Bind Address</b>	Configure the IP address for TLS server to bind to. "0.0.0.0" means binding to all interfaces. The port number is optional, and the default port number is 8443. The IP address must match the common name (host name) in the certificate so that the TLS socket will not bind to multiple IP addresses. The default setting is 0.0.0.0:8443.
<b>Username</b>	Configure the Username for API Authentication.
<b>Password</b>	Configure the Password for API Authentication.
<b>TLS Private Key</b>	Upload TLS private key. The size of the key file must be under 2MB. This file will be renamed as 'private.pem' automatically.
<b>TLS Cert</b>	Upload TLS cert. The size of the certificate must be under 2MB. This is the certificate file (*.pem format only) for TLS connection. This file will be renamed as "certificate.pem" automatically. It contains private key for the client and signed certificate for the server.
<b>Permitted IPs</b>	Specify a list of IP addresses permitted by API. This creates an AIP-specific access control list. Multiple entries are allowed. For example, "192.168.5.20/255.255.255.255" denies access from all IP addresses except 192.168.5.20. The default setting is blank, meaning all IPs will be denied. Users must set permitted IP address before connecting to the API.
<b>Reset Certificates</b>	Press  button to restore UCM's default certificates.

For more details on CDR API (Access to Call Detail Records) and REC API (Access to Call Recording Files), please refer the document in the link here:

[http://www.grandstream.com/sites/default/files/Resources/ucm6xxx\\_cdr\\_rec\\_api\\_guide.pdf](http://www.grandstream.com/sites/default/files/Resources/ucm6xxx_cdr_rec_api_guide.pdf)



## USER PORTAL

Users could log into their web GUI portal using the extension number and user password. When an extension is created in the UCM62XX, the corresponding user account for the extension is automatically created. The user portal allows access to a variety of features which include user information, extension configuration and CDR as well as settings and managing value-added features like Fax Sending, Call Queue, Wakeup Service and CRM.

Users also can access their personal data files (call recordings, Fax files, Voicemail Prompts ...).

The login credentials are configured by Super Admin. The following figure shows the dialog of editing the account information by Super Admin. The Username must be the extension number and it is not configurable, and the password is set on “User Password” field and it should not be confused with the SIP extension password.

**Edit User Information: 1000**

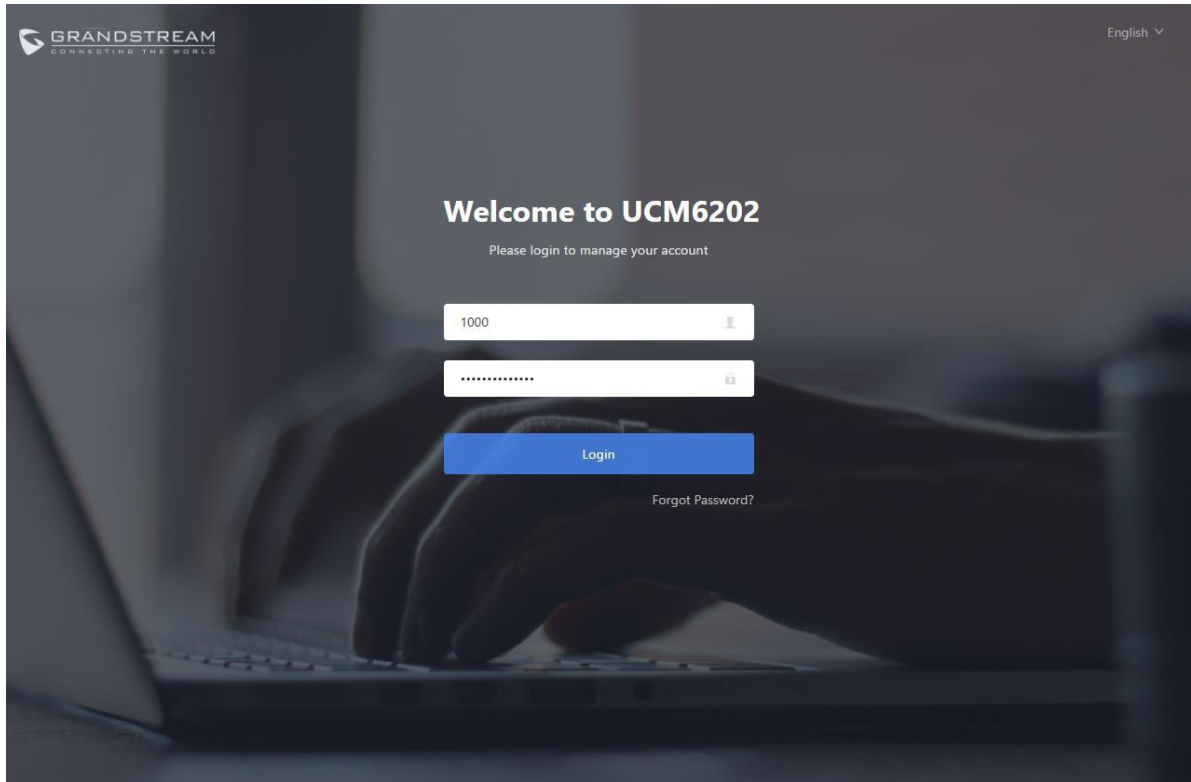
<p>* User Name: <input type="text" value="1000"/></p> <p>Privilege: <input style="border: none; border-bottom: 1px solid #ccc; text-decoration: none; background-color: #f0f0f0; padding: 2px 5px; width: 100%;" type="text" value="Consumer"/></p> <p>Fax: <input type="text"/></p> <p>First Name: <input style="background-color: #ffff00;" type="text" value="John"/></p> <p>Home Number: <input type="text"/></p>	<p>* User Password: <input type="text" value="mYpassWord!"/></p> <p>Department: <input type="text" value="Support"/></p> <p>Email Address: <input type="text" value="user1000@domain.local"/></p> <p>Last Name: <input type="text" value="DOE"/></p> <p>Mobile Phone Numb... <input type="text"/></p>
---	---

**Figure 315: Edit User Information by Super Admin**

The following screenshot shows an example of login page using extension number 1000 as the username.



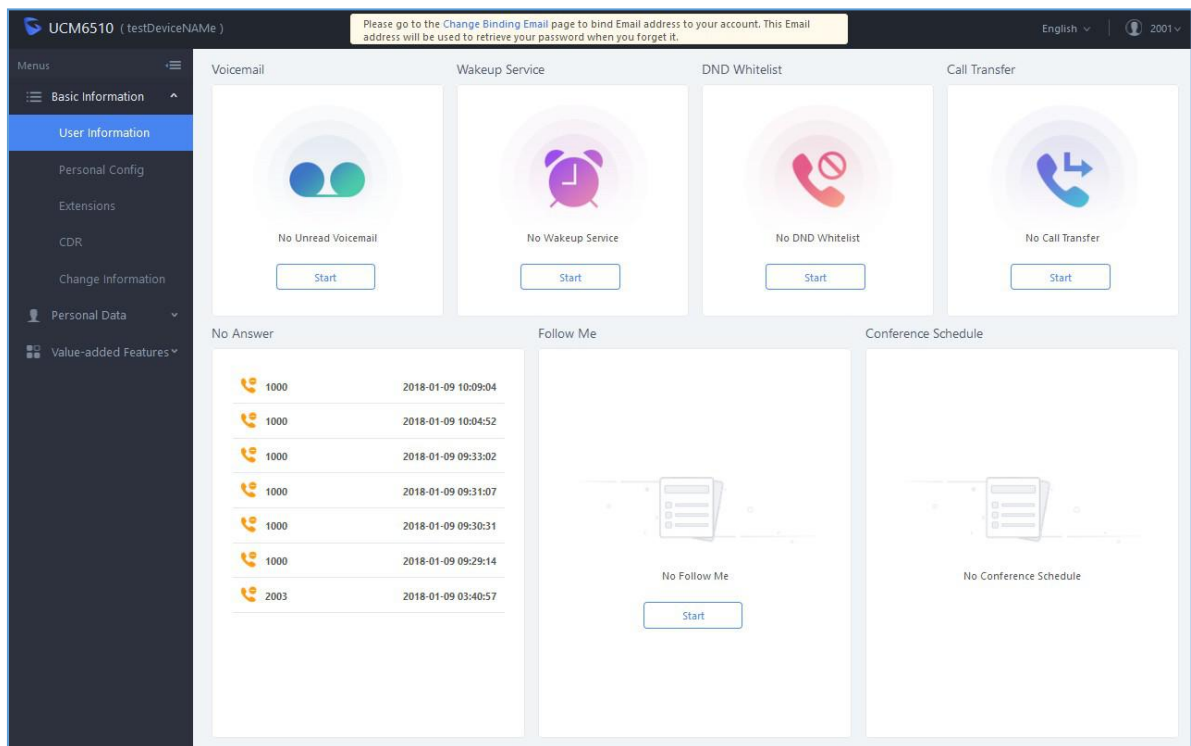




**Figure 316: User Portal Login**

After login, the Web GUI display is shown as below.

**Note:** Regular users will be logged out automatically if an admin resets their extensions.



**Figure 317: User Portal Layout**



After successful login, the user has the following three configuration tabs:

## Basic Information

Under this menu, the user can configure and change his/her personal information including (first name, last name, password, email address, department...). And they can also set and activate their extension features (presence status, call forward, DND ....) to be reflected on the UCM.

Also, the user can see from this menu the Call Details Records and search for specific ones along with the possibility to download the records on CSV format for later usage.

## Personal Data

Under this section, the user can access and manage their personal data files which includes (voicemail files, call recordings, and fax files) along with the possibility to set Follow me feature to without requesting the Super admin to set the feature from admin account.

## Value-added Features

On this section, the user has access to manage and use all rich value-added features which includes.

- + Sending Fax files using PDF or TIF/TIFF format.
- + If user is a member of call queue, they can check the queue's activity from the "Call Queue" section.
- + Create and enable Wakeup service.
- + Enable and configure CRM connection to either SugarCRM or Salesforce.

For the configuration parameter information in each page, please refer to **[Table 143: User Management→Create New User]** for options in **User Portal→Basic Information→User Information** page; please refer to **[EXTENSIONS]** for options in **User Portal→Basic Information→Extension** page; please refer to **[CDR]** for **User Portal→Basic Information→CDR** page.



# MAINTENANCE

## User Management

User management is on Web GUI→**Maintenance**→**User Management** page. User could create multiple accounts for different administrators to log in the UCM6200 Web GUI. Additionally, the system will automatically create user accounts along with creating new extensions for extension users to login to the Web GUI using their extension number and password. All existing user accounts for Web GUI login will be displayed on User Management page as shown in the following figure.

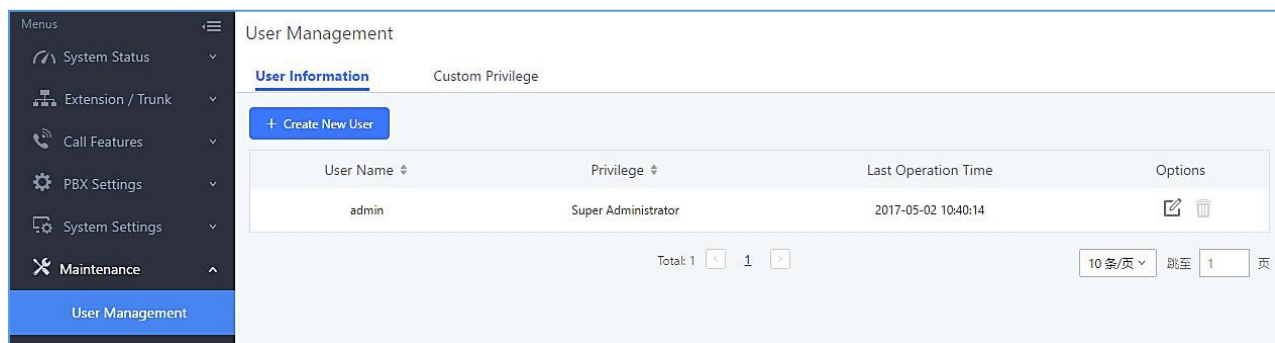



Figure 318: User Management Page Display

## User Information

When logged in as Super Admin, click on  to create a new account for Web GUI user. The following dialog will prompt. Configure the parameters as shown in below table.

### Create New User Information



<p>* User Name: <input type="text" value="John"/></p> <p>Privilege: <input type="text" value="Administrator"/></p> <p>Fax: <input type="text"/></p> <p>First Name: <input type="text" value="John"/></p> <p>Home Number: <input type="text"/></p>	<p>* User Password: <input type="text" value="admin123"/></p> <p>Department: <input type="text" value="IT"/></p> <p>Email Address: <input type="text" value="john@domain.local"/></p> <p>Last Name: <input type="text" value="DOE"/></p> <p>Mobile Phone Numb...: <input type="text" value="123456789"/></p>
---	--

Figure 319: Create New User



**Table 143: User Management→Create New User**





<b>Username</b>	Configure a username to identify the user which will be required in Web GUI login. Letters, digits and underscore are allowed in the username.
<b>User Password</b>	Configure a password for this user which will be required in Web GUI login. Letters, digits and underscore are allowed.
<b>Privilege</b>	This is the role of the Web GUI user. Currently only “Admin” is supported when Super Admin creates a new user.
<b>Department</b>	Enter the necessary information to keep a record for this user.
<b>Fax</b>	
<b>Email Address</b>	
<b>First Name</b>	
<b>Last Name</b>	
<b>Home Number</b>	
<b>Phone Number</b>	

Once created, the Super Admin can edit the users by clicking on  or delete the user by clicking on .

User Management

User Information    Custom Privilege

[+ Create New User](#)

User Name	Privilege	Last Operation Time	Options
admin	Super Administrator	2017-05-02 10:45:09	 
John	Administrator		 

**Figure 320: User Management – New Users**

## Custom Privilege

Four privilege levels are supported:

- **Super Administrator**


- This is the highest privilege. Super Admin can access all pages on UCM6200 Web GUI, change configuration for all options and execute all the operations.
- Super Admin can create, edit and delete one or more users with “Admin” privilege
- Super Admin can edit and delete one or more users with “Consumer” privilege
- Super Admin can view operation logs generated by all users.

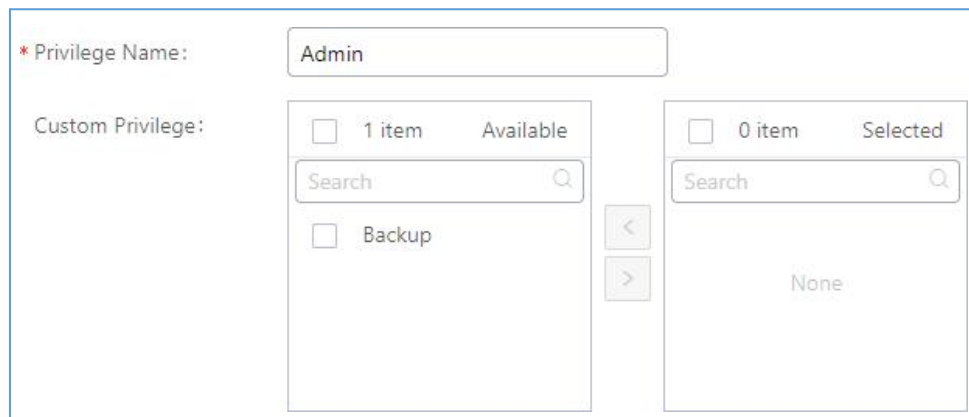


- By default, the user account “admin” is configured with “Super Admin” privilege and it is the only user with “Super Admin” privilege. The Username and Privilege level cannot be changed or deleted.
- Super Admin could change its own login password on Web GUI→**Maintenance**→**Change Information** page.
- Super Admin could view operations done by all the users in Web GUI→**Maintenance**→**User Management**→**Operation Log**

- **Administrator**

- Users with “Admin” privilege can only be created by “Super Admin” user.
- “Admin” privilege users are not allowed to access the following pages:  
**Maintenance**→**Upgrade**  
**Maintenance**→**Cleaner**  
**Maintenance**→**Reset/Reboot**  
**Settings**→**User Management**→**Operation Log**
- “Admin” privilege users cannot create new users for login.

**Note:** By default, administrator accounts are not allowed to access backup menu, but this can be assigned to them by editing the option “**Maintenance** → **User Management** → **Custom Privilege**” then press  to edit the “Admin” account and include backup operation permission for these types of users.




**Figure 321: Assign Backup permission to "Admin" users**

- **Consumer**

- A user account for Web GUI login is created automatically by the system when a new extension is created.
- The user could log in the Web GUI with the extension number and password to access user information, extension configuration, CDR of that extension, personal data and value-added features. For more details; please refer to [User Portal Guide](#).



- The SuperAdmin user can click on  on the “General\_User” in order to enable/disable the custom privilege from deleting their own recording files, changing SIP credentials and disabling voicemail service in their user portal account.

Edit Custom Privilege: General\_User

---

\* Privilege Name:

Enable Delete Recording

Files:

Allowed to change Auth ID   
and SIP password:

Allow to Enable Voicemail:

**Figure 322: General User**

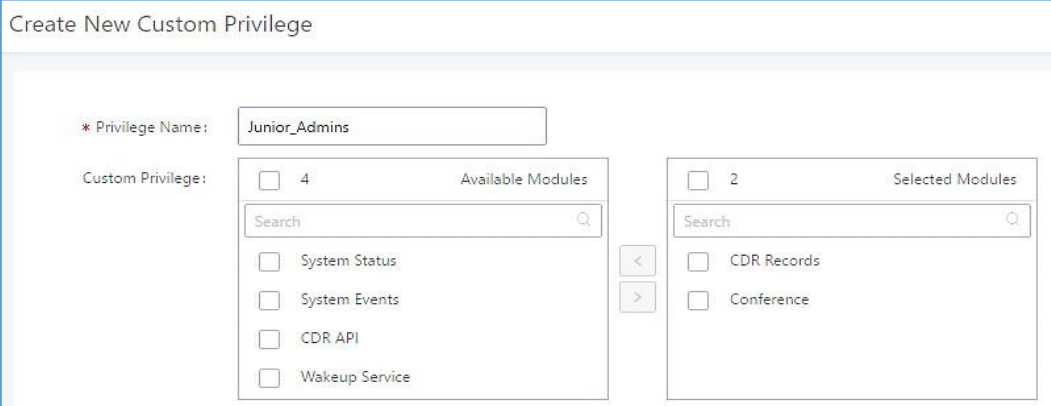
- **Custom Privilege**

The Super Admin user can create users with different privileges. following modules are available for privilege customization.

- System Status
- Conference
- System Events
- CDR Records
- CDR API
- Wakeup Service
- Extensions
- IVR
- Voicemail
- Ring Groups
- Paging/Intercom
- Call Queue
- Pickup Groups
- Speed Dial
- DISA
- Event List
- Call back
- Feature Codes
- Fax/T.38



- Parking Lot
- Backup
- Dial by Name
- Emergency Calls
- CDR Records
- CDR Statistics
- CDR Recordings
- Voice Prompt
- Inbound Routes
- Outbound Routes



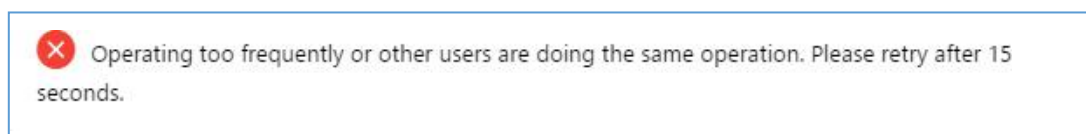
**Figure 323: Create New Custom Privilege**

Log in UCM6200 as super admin and go to **Maintenance**→**User Management**→**Custom Privilege**, create privilege with customized available modules.

To assign custom privilege to a sub-admin, navigate to UCM Web GUI→**Maintenance**→**User Management**→**User Information**→Create New User/Edit Users, select the custom privilege from “Privilege” option.

### Concurrent Multi-User Login

When there are multiple Web GUI users created, concurrent multi-user login is supported on the UCM6200. Multiple users could edit options and have configurations take effect simultaneously. However, if different users are editing the same option or making the same operation (by clicking on “Apply Changes”), a prompt will pop up as shown in the following figure.



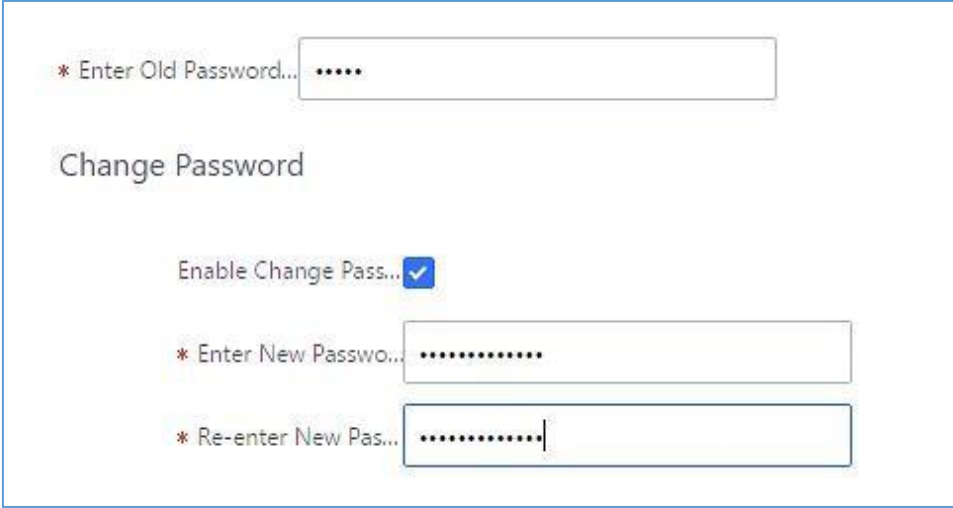
**Figure 324: Multiple User Operation Error Prompt**



## Change Password

After logging in the UCM6200 Web GUI for the first time, it is highly recommended for users to change the default password "admin" to a more complicated password for security purpose. Follow the steps below to change the Web GUI access password.

1. Go to Web GUI→**Maintenance**→**Change Information** page.
2. Enter the old password first.
3. Enter the new password and re-type the new password to confirm.  
**Note:** If PBX Settings→General Settings→Enable Strong Password is toggled on, the minimum password requirements are as follows:
  - Must contain at least one number.
  - Must contain at least one uppercase letter, lower case letter, OR special character.
4. Configure the Email Address that is used when login credential is lost.
5. Click on "Save" and the user will be automatically logged out.
6. Once the web page comes back to the login page again, enter the username "admin" and the new password to login.



**Figure 325: Change Password**

<b>Enter Old Password</b>	Enter the Old Password for UCM6200
<b>Enter New Password</b>	Enter the New Password for UCM6200
<b>Retype New Password</b>	Retype the New Password for UCM6200

## Change Username

UCM62xx allows users now to change Super Administrator username.







Figure 326: Change Username

## Change binding Email

UCM6200 allows user to configure binding email in case login password is lost. UCM6200 login credential will be sent to the designated email address. The feature can be found under Web GUI→**System Settings**→**User Management**→**Change Binding Email**.

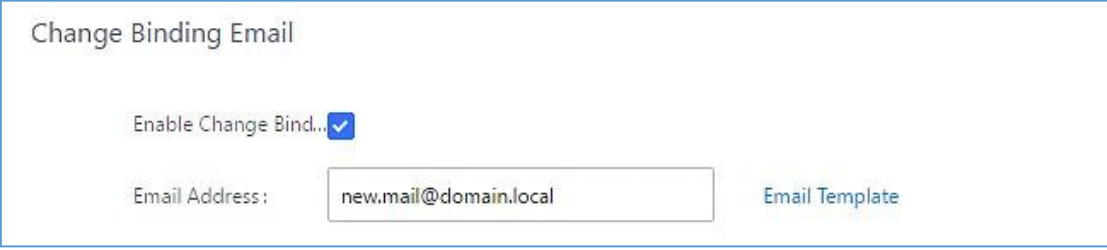


Figure 327: Change Binding Email

Table 144: Change Binding Email option

<b>Enter the password of the account</b>	Enter the current login user credential for UCM6200
<b>Email Address</b>	Email Address is used to retrieve password when password is lost

## Login Settings

After the user logs in the UCM6200 Web GUI, the user will be automatically logged out after certain timeout, or he/she can be banned for a specific period if the login timeout is exceeded. Those values can be specified under UCM6200 web GUI→**Maintenance**→**Change Information**→**Login Settings** page.



The **“User Login Timeout”** value is in minute and the default setting is 10 minutes. If the user does not make any operation on Web GUI within the timeout, the user will be logged out automatically. After that, the Web GUI will be redirected to the login page and the user will need to enter username and password to log in.

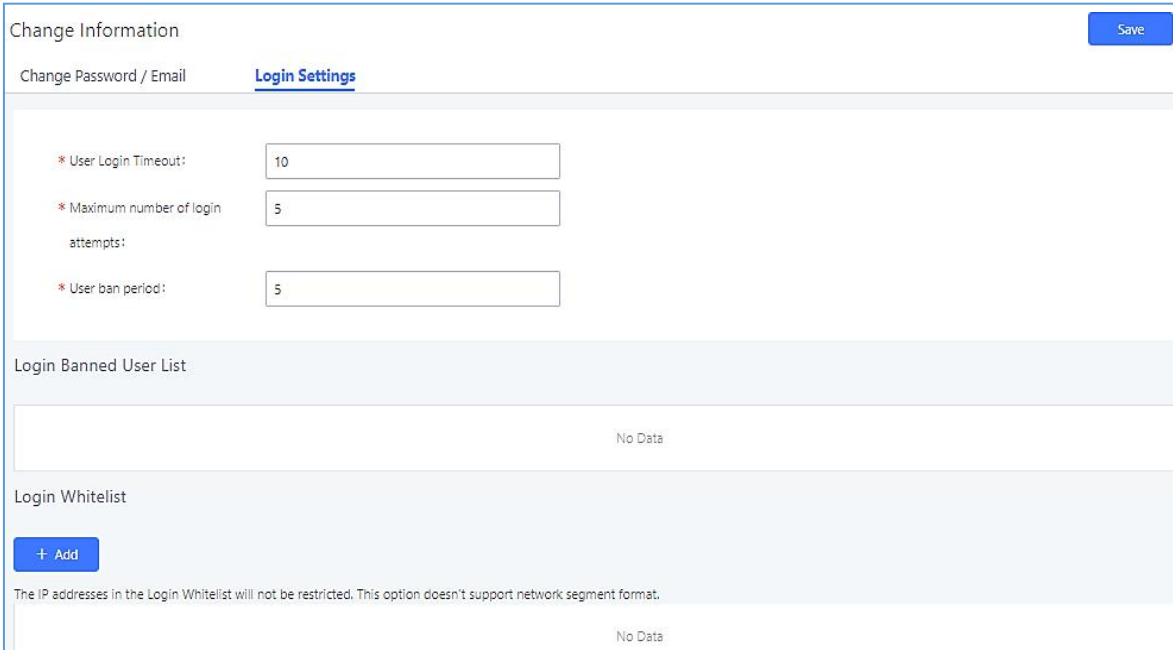
If set to 0, there is no timeout for the Web GUI login session and the user will not be automatically logged out.

**“Maximum number of login attempts”** can prevent the UCM6200 from brutal force decryption, if this number is exceeded user IP address will be banned from accessing the UCM for a period of time based on user configuration, the default value is 5.

**“User ban period”** specify the period of time in minutes an IP will be banned from accessing the UCM if the User max number of try login is exceeded, the default value is 5.

**“Login Banned User List”** show the list of IPs’ banned from the UCM.

**“Login White List”** User can add a list of IPs’ to avoid the above restriction, thus, they can exceed the User max number of try login.



The screenshot shows a web interface titled "Change Information" with a "Save" button in the top right. Below the title, there are links for "Change Password / Email" and "Login Settings" (which is highlighted). The "Login Settings" section contains three input fields:

- \* User Login Timeout: 10
- \* Maximum number of login attempts: 5
- \* User ban period: 5

Below these settings are two sections:

- Login Banned User List:** A table area showing "No Data".
- Login Whitelist:** A section with a "+ Add" button and a note: "The IP addresses in the Login Whitelist will not be restricted. This option doesn't support network segment format." Below this is another "No Data" table area.

**Figure 328: Login Timeout Settings**

## Operation Log

Super Admin has the authority to view operation logs on UCM6200 Web GUI→**Settings**→**User Management**→**Operation Log** page. Operation logs list operations done by all the Web GUI users, for example,



Web GUI login, creating trunk, creating outbound rule and etc. There are 7 columns to record the operation details “Date”, “Username”, “IP Address”, “Results”, “Page Operation”, “Specific Operation” and “Remark”.

Operation Log <span style="float: right;">Filter</span>						
Delete Search Result (5)		Delete All Logs				
Date	User Name	IP Address	Results	Page Operation	Specific Operation	Remark
2017-08-03 05:40:50	admin	192.168.6.223	Operation successful	Apply Changes		Click to modify notes.
2017-08-03 05:40:49	admin	192.168.6.223	Operation successful	Extensions: Create New SIP Extension	Extension: 1000,1001,1002,1003,1004. ⓘ	Click to modify notes.
2017-08-03 05:40:49	admin	192.168.6.223	Operation successful	Follow Me: Create New Follow Me	ⓘ	Click to modify notes.
2017-08-03 05:33:07	admin	192.168.6.223	Operation successful	Login: Login	User Name: admin. ⓘ	Click to modify notes.
2017-08-03 05:17:58	admin	192.168.6.223	Operation successful	Login: Login	User Name: admin. ⓘ	Click to modify notes.
2017-08-03 04:48:18	admin	192.168.6.223	Operation successful	Login: Login	User Name: admin. ⓘ	Click to modify notes.
2017-08-03 04:19:47	admin	192.168.6.223	Operation successful	Login: Login	User Name: admin. ⓘ	Click to modify notes.
2017-08-03 03:51:20	admin	192.168.6.223	Operation successful	Login: Login	User Name: admin. ⓘ	Click to modify notes.

Total: 8 1 10 / page Goto 1

**Figure 329: Operation Logs**


The operation log can be sorted and filtered for easy access. Click on the header of each column to sort. For example, clicking on "Date" will sort the logs according to operation date and time. Clicking on "Date" again will reverse the order.

**Table 145: Operation Log Column Header**

<b>Date</b>	The date and time when the operation is executed.
<b>Username</b>	The username of the user who performed the operation.
<b>IP Address</b>	The IP address from which the operation is made.
<b>Results</b>	The result of the operation.
<b>Page Operation</b>	The page where the operation is made. For example, login, logout, delete user, create trunk and etc.




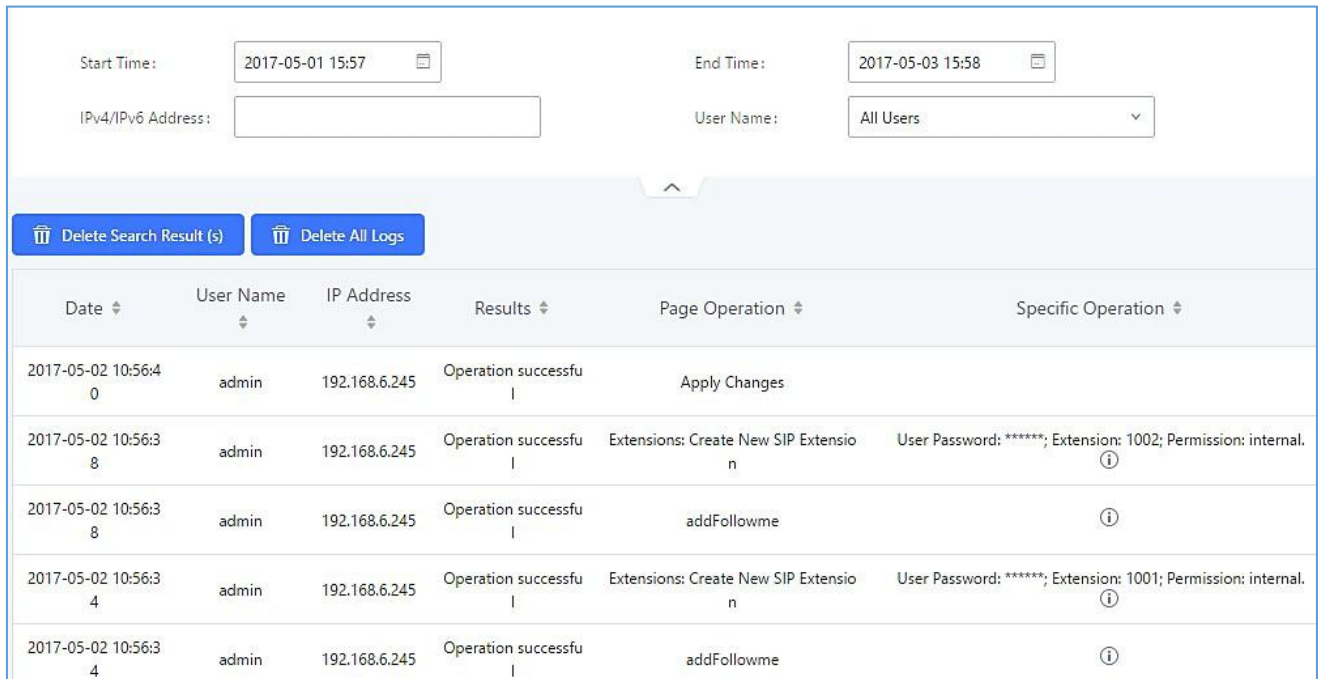
**Specific Operation**

 Click on  to view the options and values configured by this operation.

**Remark**

Allows users to add notes and remarks to each operation

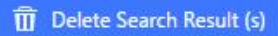

User could also filter the operation logs by time condition, IP address and/or username. Configure these conditions and then click on .



Date	User Name	IP Address	Results	Page Operation	Specific Operation
2017-05-02 10:56:40	admin	192.168.6.245	Operation successful	Apply Changes	
2017-05-02 10:56:38	admin	192.168.6.245	Operation successful	Extensions: Create New SIP Extension	User Password: *****; Extension: 1002; Permission: internal.
2017-05-02 10:56:38	admin	192.168.6.245	Operation successful	addFollowme	
2017-05-02 10:56:34	admin	192.168.6.245	Operation successful	Extensions: Create New SIP Extension	User Password: *****; Extension: 1001; Permission: internal.
2017-05-02 10:56:34	admin	192.168.6.245	Operation successful	addFollowme	

**Figure 330: Operation Logs Filter**

The above figure shows an example that operations made by user “support” on device with IP 192.168.40.173 from 2014-11-01 00:00 to 2014-11-06 15:38 are filtered out and displayed.

To delete operation logs, users can perform filtering first and then click on  to delete the filtered result of operation logs. Or users can click on  to delete all operation logs at once.

## Upgrading

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



## Upgrading Via Network

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

### Examples of valid URLs:

firmware.grandstream.com/BETA

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

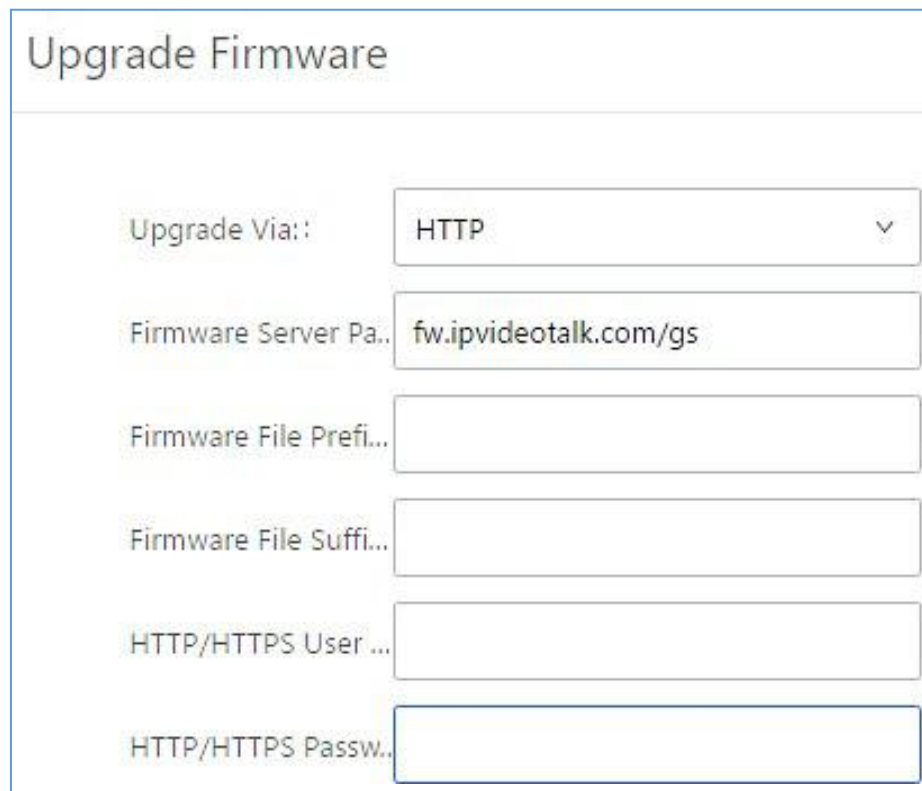


Figure 331: Network Upgrade

Table 146: Network Upgrade Configuration

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



**HTTP/HTTPS Password** The password for the HTTP/HTTPS server.

Please follow the steps below to upgrade the firmware remotely.

1. Enter the firmware server path under Web GUI→**Maintenance**→**Upgrade**.
2. Click on "Save". Then reboot the device to start the upgrading process.
3. Please be patient during upgrading process. Once done, a reboot message will be displayed in the LCD.
4. Manually reboot the UCM6200 when it is appropriate to avoid immediate service interruption. After it boots up, log in the Web GUI to check the firmware version.

### Upgrading Via Local Upload

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

1. Download the latest UCM6200 firmware file from the following link and save it in your PC.  
<http://www.grandstream.com/support/firmware>
2. Log in the Web GUI as administrator in the PC.
3. Go to Web GUI→**Maintenance**→**Upgrade**, upload the firmware file by clicking on "choose file to upload" and select the firmware file from your PC. The default firmware file name is ucm6200fw.bin

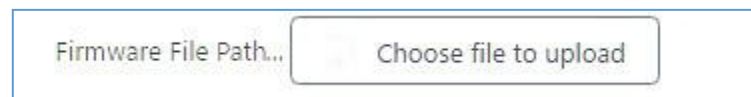
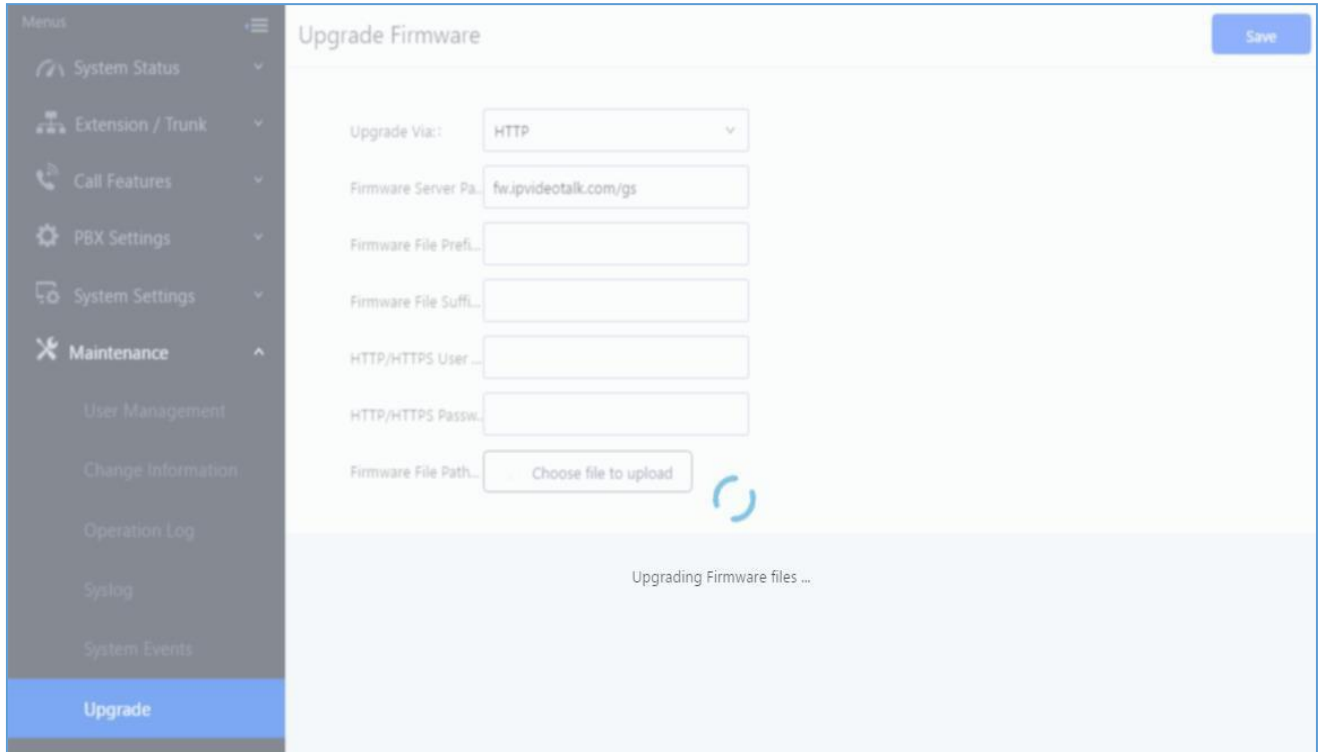


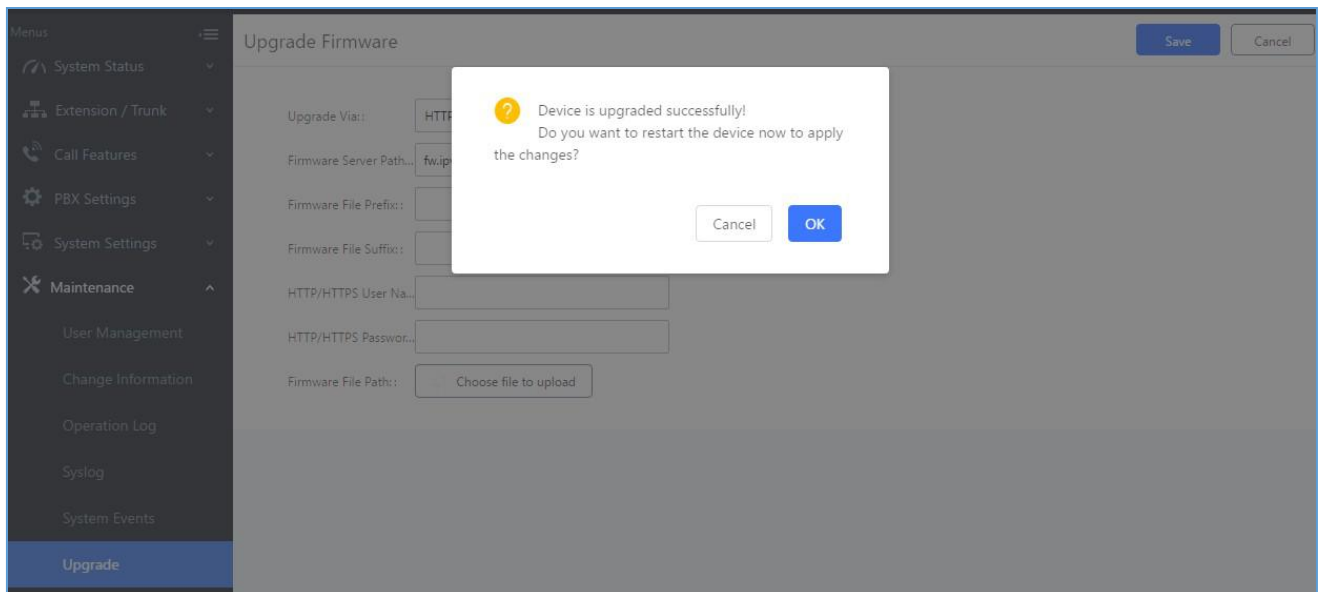
Figure 332: Local Upgrade





**Figure 333: Upgrading Firmware Files**

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



**Figure 334: Reboot UCM6200**

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



**Notes:**

- Please do not interrupt or power cycle the UCM6200 during upgrading process.
  - The firmware file name allows the use of the special characters besides the following restricted characters: # \$ ^ & \* + ( ) [ ] / ; ' | , < > ?
- 

## No Local Firmware Servers

Service providers should maintain their own firmware upgrade servers. For users who do not have TFTP/HTTP/HTTPS server, some free windows version TFTP servers are available for download from [http://www.solarwinds.com/products/freetools/free\\_tftp\\_server.aspx](http://www.solarwinds.com/products/freetools/free_tftp_server.aspx)  
<http://ftpd32.jounin.net>

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

Instructions for local firmware upgrade via TFTP:

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

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

## Backup

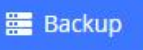
The UCM6200 configuration can be backed up locally or via network. The backup file will be used to restore the configuration on UCM6200 when necessary.

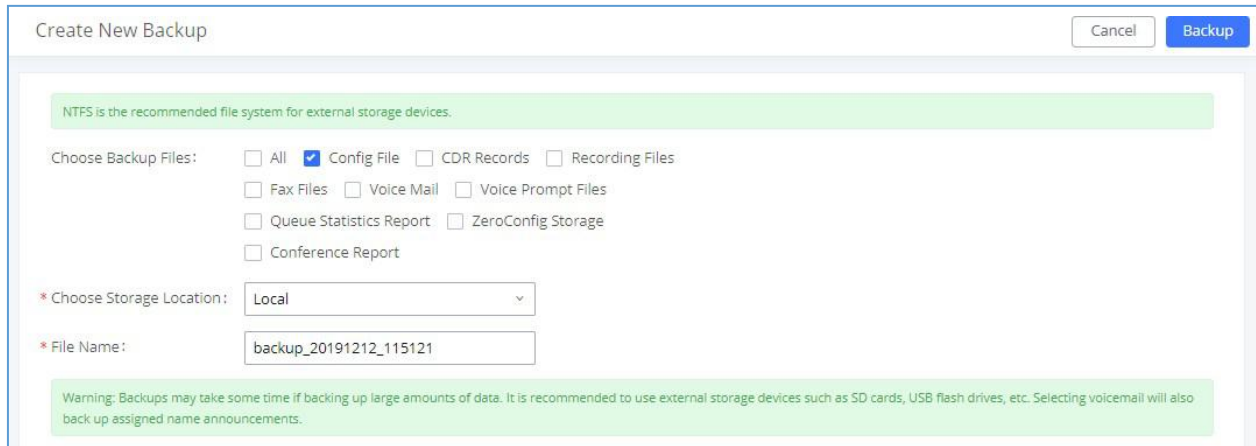




## Backup/Restore

Users could backup the UCM62xx configurations for restore purpose under **Web GUI → Maintenance →**

**Backup → Backup/Restore.** Click on  **Backup** to create a new backup. Then the following dialog will show.







**Figure 335: Create New Backup**

1. Choose the type(s) of files to be included in the backup.
2. Choose where to store the backup file: USB Disk, SD Card, Local or NAS.

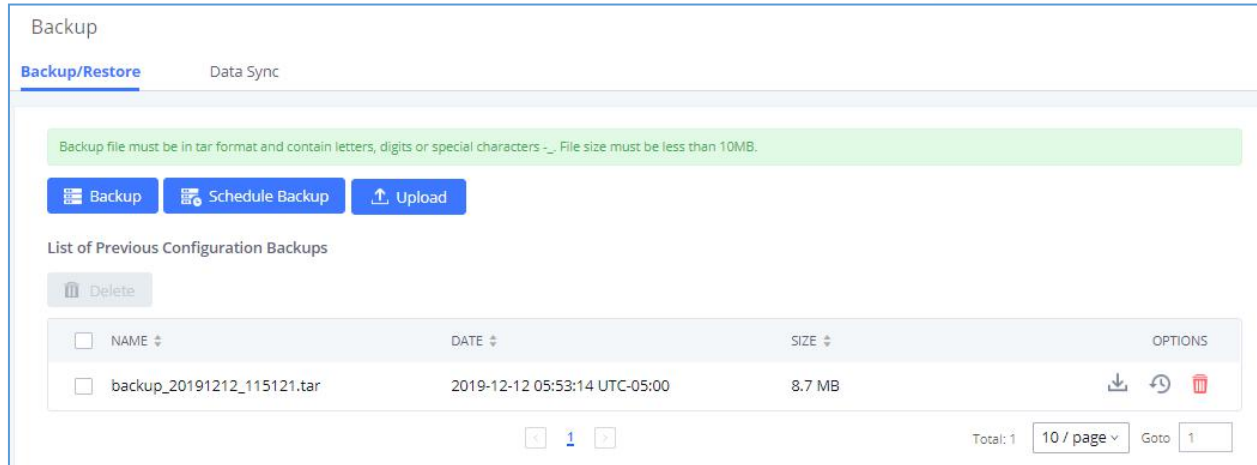
**Note:** USB Disk or SD card options will show only if plugged; NAS server will show only if configured and status is available. Refer to [PBX Settings/NAS].

3. Name the backup file.
4. Click on "Backup" to start backup.

Once the backup is done, the list of the backups will be displayed with date and time in the web page. Users can download , restore , or delete  it from the UCM62xx internal storage or the external device.

Click on  **Upload** to upload backup file from the local device to UCM62xx. The uploaded backup file will also be displayed in the web page and can be used to restore the UCM62xx.



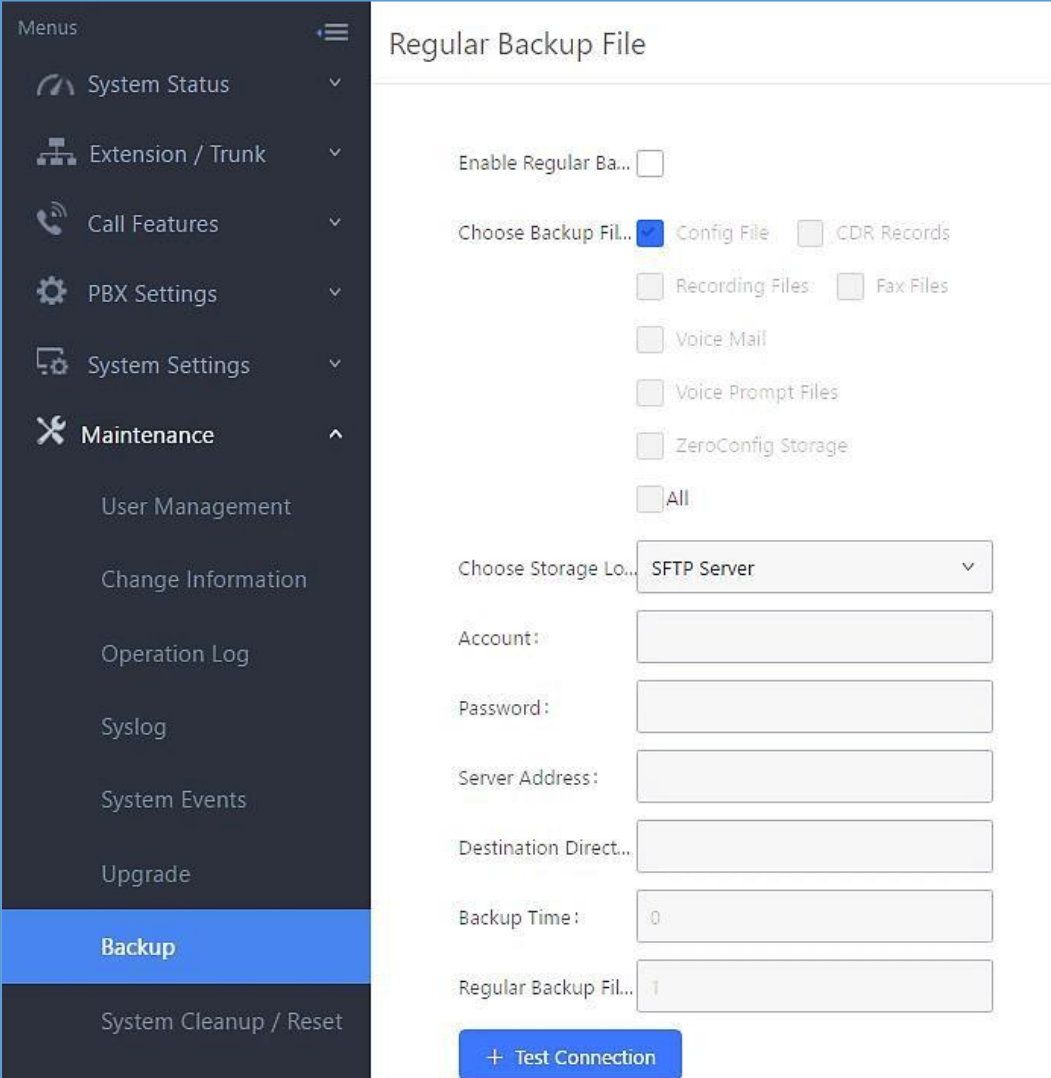


**Figure 336: Backup / Restore**

The **Schedule Backup** option allows UCM62xx to perform automatically backup on the user specified time. The backup file can only be stored in USB / SD card / SFTP server / NAS. User is allowed to set backup time from 0-23 and how frequent the backup will be performed.

The user can test the connection with the SFTP server by clicking on the **+ Test Connection** button.





**Figure 337: Local Backup**

## Data Sync

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

The client account supports special characters such as @ or ".". Allowing the use email address as SFTP accounts. It allows users as well to specify the destination directory on SFTP server for backup file. If the directory does not exist on the destination, UCM6200 will create the directory automatically



Backup

Backup/Restore Data Sync

Use SFTP to automatically sync CDR, recordings, voicemail, CDR, and fax every day.

**Data Sync Configuration**

Enable Data Sync:

Choose Data Sync Files:  CDR Records  Recording Files  
 Voice Mail  Fax

\* Account:

Password:

\* Server Address:

Destination Directory:

\* Sync Time:

+ Test Connection
+ Synchronize All Data

**Data Sync Log**

Clean

No record to view

**Figure 338: Data Sync**

**Table 147: Data Sync Configuration**

<b>Enable Data Sync</b>	Enable the auto data sync function. The default setting is "No".
<b>Account</b>	Enter the Account name on the SFTP backup server.
<b>Password</b>	Enter the Password associate with the Account on the SFTP backup server.
<b>Server Address</b>	Enter the SFTP server address.
<b>Destination Directory</b>	Specify the directory in SFTP server to keep the backup file. Format: 'xxx/xxx/xxx', If this directory does not exist, UCM will create this directory automatically.
<b>Sync Time</b>	Enter 0-23 to specify the backup hour of the day.



Before saving the configuration, users could click on + Test Connection. The UCM6200 will then try connecting the server to make sure the server is up and accessible for the UCM6200. Save the changes and all the backup logs will be listed on the web page. After data sync is configured, users could also manually synchronize all data

by clicking on + Synchronize All Data instead of waiting for the backup time interval to come.



## Restore Configuration from Backup File

To restore the configuration on the UCM6200 from a backup file, users could go to Web GUI→Maintenance→Backup→Backup/Restore.

- A list of previous configuration backups is displayed on the web page. Users could click on  of the desired backup file and it will be restored to the UCM6200.
- If users have other backup files on PC to restore on the UCM6200, click on "Upload Backup File" first and select it from local PC to upload on the UCM6200. Once the uploading is done, this backup file will be displayed in the list of previous configuration backups for restore purpose. Click on  to restore from the backup file.
- User could also restore using the backup file saved in SD card or USB device plugged into the UCM6200 or the backup available in his NAS server connected to UCM.

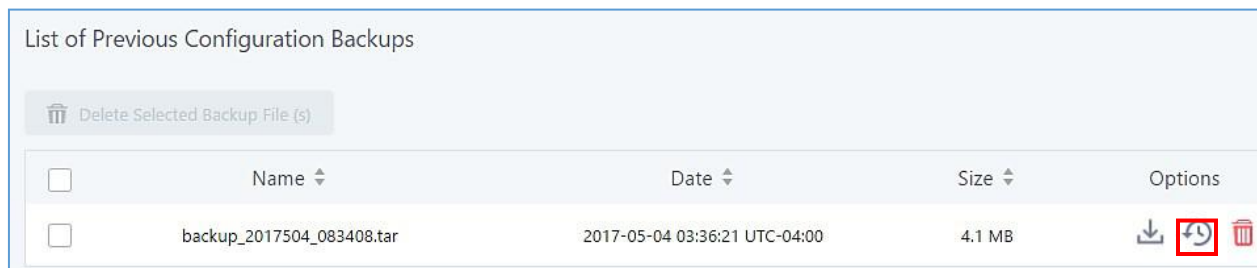


Figure 339: Restore UCM6200 from Backup File

### Note:

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

## System Cleanup/Reset

### Reset and Reboot

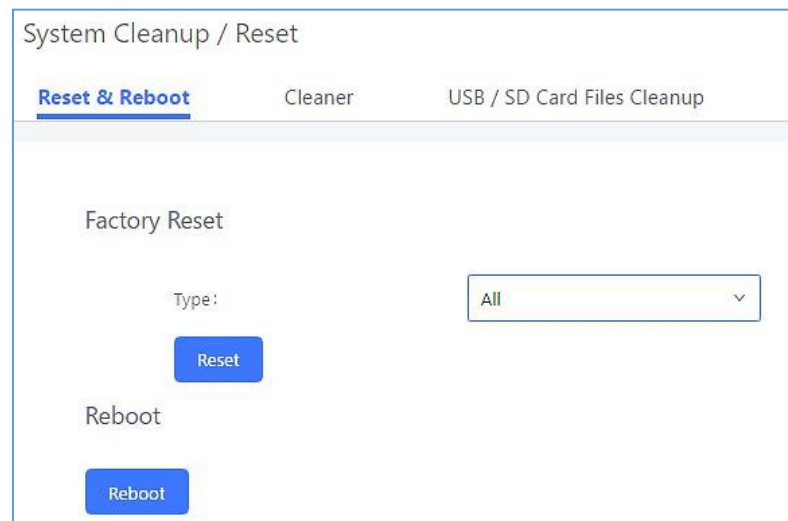
Users could perform reset and reboot under Web GUI→Maintenance→System Cleanup/Reset→Reset and Reboot.

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

- **User Data**  
All the data including voicemail, recordings, IVR Prompt, Music on Hold, CDR and backup files will be cleared.



- **All**  
All the configurations and data will be reset to factory default.



**Figure 340: Reset and Reboot**

## Cleaner

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

The following screenshot show the settings and parameters to configure the cleaner feature on UCM6200.



System Cleanup / Reset
Reset & Reboot   **Cleaner**   USB Disk/SD Card File Management
Cancel   Save

Clean CDR, recordings, voicemail, and fax automatically.

**CDR Cleaner**

Enable Cleaner:

Clean Time:

Cleaning Conditions:

Clean Interval:

**Queue Statistics Report Cleaner**

Enable Cleaner:

Clean Time:

Cleaning Conditions:

Clean Interval:

**Conference Call Statistics Report Cleaner**

Enable Cleaner:

Clean Time:

Cleaning Conditions:

Clean Interval:

**File Cleaner**

Enable Cleaner:

Clean Files in External Storage:

Choose Cleaner Files:

<input type="checkbox"/> Basic Call Recording Files	<input type="checkbox"/> Conference Recording Files
<input type="checkbox"/> Call Queue Recording Files	<input type="checkbox"/> Voicemail Files
<input type="checkbox"/> Fax	<input type="checkbox"/> Emergency Calls Recording Files
<input type="checkbox"/> SCA Recording Files	<input type="checkbox"/> Backup Files

Clean Time:

Cleaning Conditions:

File Clean Threshold:

Keep Last X Days:

**Cleaner Log**

No record to view

**Figure 341: Cleaner**



**Table 148: Cleaner Configuration**

<b>CDR Cleaner</b>	
<b>Enable Cleaner</b>	Enable the CDR Cleaner function.
<b>Clean Time</b>	Enter 0-23 to specify the hour of the day to clean up CDR.
<b>Cleaning Conditions</b>	<ul style="list-style-type: none"> <li>• <b>By Schedule:</b> If the clean interval is 3, cleaning will be performed every 3 days to remove all records that were generated 3 days ago.</li> <li>• <b>Keep Last X Records:</b> If the max number of CDR has been reached, CDR will be deleted starting with the oldest entry at the configured cleaning time. (Note: The amount of records displayed on the page of call queue statistics is not one-to-one with the actual amount of records in the database.)</li> <li>• <b>Keep Last X Days:</b> Delete all entries older than X days.</li> </ul>
<b>Clean Interval</b>	Enter 1-30 to specify the day of the month to clean up CDR when <b>By Schedule</b> is selected as <b>Cleaning Conditions</b> .
<b>Max Entries</b>	Set the maximum number of CDR entries to keep when <b>Keep Last X Records</b> is selected as <b>Cleaning Conditions</b> . Default is 50000. Valid range: 10000 – 100000.
<b>Keep Last X Days</b>	Enter the number of days of call log entries to keep when <b>Keep Last X days</b> is selected as <b>Cleaning Conditions</b> . Default is 30. Valid range: 1 – 100.
<b>Queue Statistics Report Cleaner</b>	
<b>Enable Cleaner</b>	Enable scheduled queue log cleaning. By default, is disabled.
<b>Clean Time</b>	Enter the hour of the day to start the cleaning. The valid range is 0-23.
<b>Cleaning Conditions</b>	<ul style="list-style-type: none"> <li>• <b>By Schedule:</b> If the clean interval is 3, cleaning will be performed every 3 days to remove all records that were generated 3 days ago.</li> <li>• <b>Keep Last X Records:</b> If the max number of Queue Statistics Report entries has been reached, Queue Statistics Report entries will be deleted starting with the oldest entry at the configured cleaning time. (Note: The amount of records displayed on the page of call queue statistics is not one-to-one with the actual amount of records in the database.)</li> <li>• <b>Keep Last X Days:</b> Delete all entries older than X days.</li> </ul>





<b>Clean Interval</b>	Enter how often (in days) to clean queue logs when <b>By Schedule</b> is selected as <b>Cleaning Conditions</b> . The valid range is 1-30.
<b>Max Entries</b>	Set the maximum number of Queue Statistics Report entries to keep when <b>Keep Last X Records</b> is selected as <b>Cleaning Conditions</b> . Default is 50000. Valid range: 10000 – 100000.
<b>Keep Last X Days</b>	Enter the number of days of Queue Statistics Report entries to keep when <b>Keep Last X days</b> is selected as <b>Cleaning Conditions</b> . Default is 30. Valid range: 1 – 100.
<b>Conference Call Statistics Report Cleaner</b>	
<b>Enable Cleaner</b>	Enable scheduled Conference log cleaning. By default, it is disabled.
<b>Clean time</b>	Enter the hour of the day to start the cleaning. The valid range is 0-23.
<b>Cleaning Conditions</b>	<ul style="list-style-type: none"> <li>• <b>By Schedule:</b> If the clean interval is 3, cleaning will be performed every 3 days to remove all records that were generated 3 days ago.</li> <li>• <b>Keep Last X Records:</b> If the max number of Conference Call Statistics Report has been reached, Conference Call Statistics Report will be deleted starting with the oldest entry at the configured cleaning time. (Note: The amount of records displayed on the page of call queue statistics is not one-to-one with the actual amount of records in the database.)</li> <li>• <b>Keep Last X Days:</b> Delete all entries older than X days.</li> </ul>
<b>Clean Interval</b>	Enter how often (in days) to clean queue logs when <b>By Schedule</b> is selected as <b>Cleaning Conditions</b> . The valid range is 1-30.
<b>Max Entries</b>	Set the maximum number of CDR Conference Call Statistics Report entries to keep when <b>Keep Last X Records</b> is selected as <b>Cleaning Conditions</b> . Default is 50000. Valid range: 10000 – 100000.
<b>Keep Last X Days</b>	Enter the number of days of Conference Call Statistics Report entries to keep when <b>Keep Last X days</b> is selected as <b>Cleaning Conditions</b> . Default is 30. Valid range: 1 – 100.
<b>File Cleaner</b>	
<b>Enable Cleaner</b>	Enter the files Cleaner function.
<b>Clean Files in External Device</b>	If enabled the files in external device (USB/SD card) will be automatically cleaned up as configured.



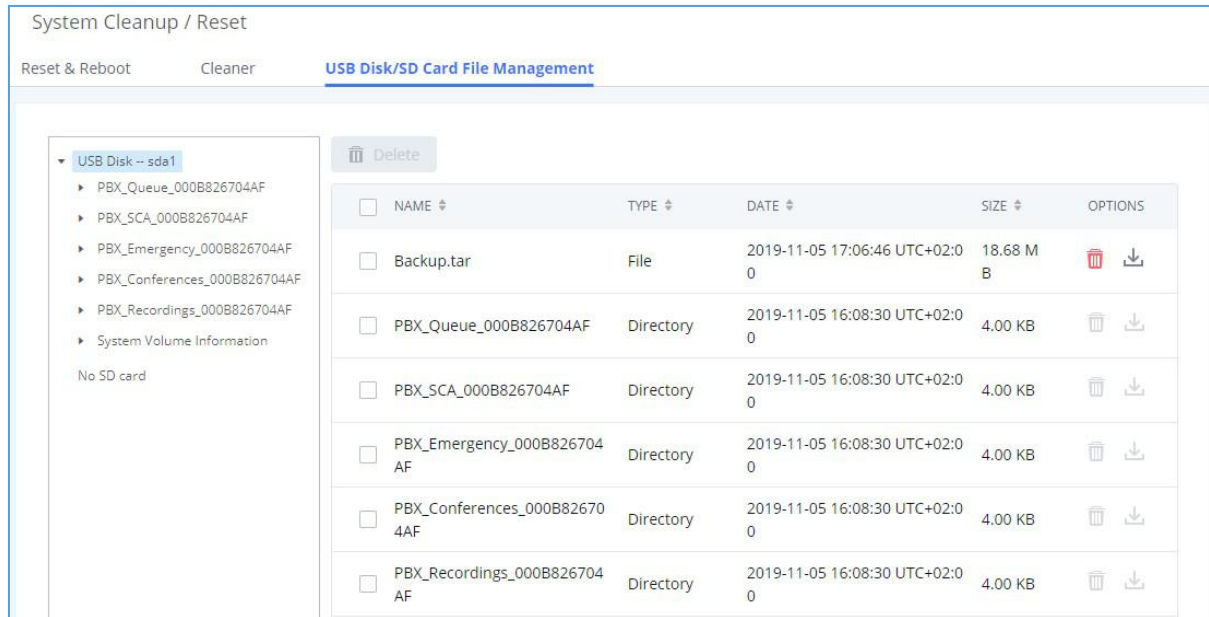
<b>Choose Cleaner File</b>	Select the files for system automatic clean. <ul style="list-style-type: none"> <li>• Basic Call Recording Files.</li> <li>• Conference Recording Files.</li> <li>• Call Queue Recording Files.</li> <li>• Voicemail Files.</li> <li>• Emergency Calls Recording Files.</li> <li>• Fax.</li> <li>• Backup Files.</li> <li>• SCA Recording Files.</li> </ul>
<b>Clean Time</b>	Enter the hour of the day to start the cleaning. The valid range is 0-23.
<b>Cleaning Conditions</b>	<ul style="list-style-type: none"> <li>• <b>By Schedule:</b> If the clean interval is 3, cleaning will be performed every 3 days to delete all files.</li> <li>• <b>By Threshold:</b> Check at the configured cleaning time every day to see if the storage threshold has been exceeded and perform cleaning of all files if it has.</li> <li>• <b>Keep Last X Days:</b> Delete all files older than X days.</li> </ul>
<b>File Clean Interval</b>	Enter 1-30 to specify the day of the month to clean up the files.
<b>File Clean Threshold</b>	Enter the internal storage disk usage threshold (in percent). Once this threshold is exceeded, the file cleanup will proceed as scheduled. Valid range is 0-99.
<b>Keep Last X Days</b>	Automatically delete all recordings older than this x days when the threshold is reached. If not set, all data is cleared. Valid range: 1 – 100.
<b>Cleaner Log</b>	
<b>Cleaner Log</b>	Press Clean “button” to clean cleaner log.

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

### USB/SD Card Files Cleanup

Users could configure to clean or download the Call Detail Report/Voice Records/Voice Mails/FAX automatically under Web GUI→**Maintenance**→**Cleaner**→**USB / SD Card Files Cleanup**.





**Figure 342: USB/SD Card Files Cleanup**

**Table 149: USB/SD Card Files Cleanup**

<b>Current Path</b>	Displays the current path.
<b>Directory</b>	Select the directory user want to clean.
<b>Delete Selected File</b>	Select multiple entries to delete from USB or SD card.

## System Recovery

In some cases (for example after wrong upgrading procedure where the user doesn't follow the correct steps to perform an upgrade) the system may go into some hardware/software issues where the web UI access is lost as well as SSH, in this case the only solution would be to perform a full system recovery in order to reset or update the software version of the device in order to use it again.

1. To access recovery mode on UCM, please follow below steps:
2. Remove the power from the unit and keep the network cable connected.
3. Press using a PIN the reset button and keep holding.
4. Plug back the power supply while maintaining the reset button pressed.
5. Wait for couple of seconds until you hear a click sound.
6. Release the reset button, and the system should display on the LCD a message "Recovery Mode" along with an IP address.

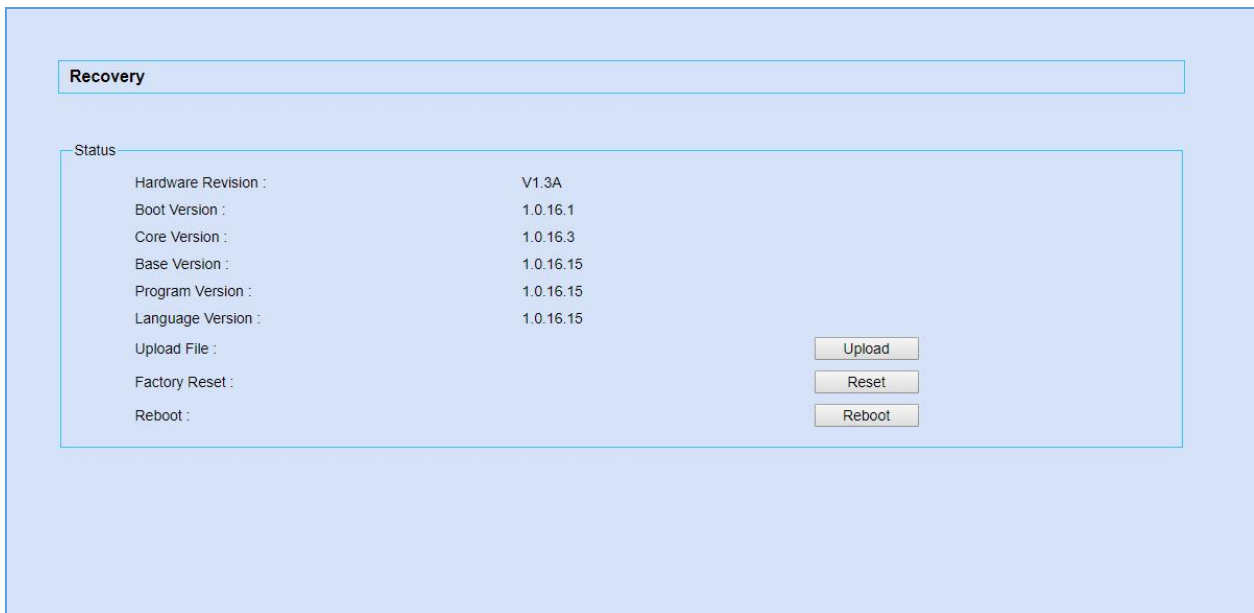
Once at this stage, the administrator can access the recovery mode web portal by typing in either the IP0 address (typically WAN) or IP1 address (typically LAN) into a browser address bar. The following page should appear:





**Figure 343: UCM6202 Recovery Web Page**

Make sure to enter the correct admin password, and press login to access the recovery mode page :



**Figure 344: Recovery Mode**

From here, the user can either upload a firmware file, factory reset or just reboot the device.



## Syslog

On the UCM6200, users could dump the syslog information to a remote server under Web GUI→**Maintenance**→**Syslog**. Enter the syslog server hostname or IP address and select the module/level for the syslog information.

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

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

- **pbx**: This module is related to general PBX functions.
- **chan\_sip**: This module is related to SIP calls.
- **chan\_dahdi**: This module is related to analog calls (FXO/FXS).
- **app\_meetme**: This module is related to conference room.



### Note:

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

The reserved size for Syslog entries on the cache memory of the UCM is 50M, once this sized is reached the UCM will clean up 2M of the oldest Syslog entries to allow to save new logs.

---

## Network Troubleshooting

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

The following sections shows the steps to capture different types of traffic traces for analysis purposes.

### Ethernet Capture

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



### Network Troubleshooting

Ethernet Capture
IP Ping
Traceroute

Interface Type: WAN v

Storage to External D.

USB Disk

Enable SFTP Data Sy...

Capture Filter:

Start
Stop
Download

Output Result

**Figure 345: Ethernet Capture**

**Table 150: Ethernet Capture**

<b>Interface Type</b>	Select the network interface to monitor.
<b>Enable SFTP Data Sync</b>	Check this box to save the capture files in the SFTP server. Please make sure the configuration of data synchronization works before.
<b>Storage to External Device</b>	Check this box to activate storage of the capture either on the USB or SD Card.
<b>Capture Filter</b>	Enter the filter to obtain the specific types of traffic, such as (host, src, dst, net, proto...).
<b>Start</b>	Click to start the trace.
<b>Stop</b>	Click to stop the trace.
<b>Download</b>	Click to download the trace if trace is stored locally.

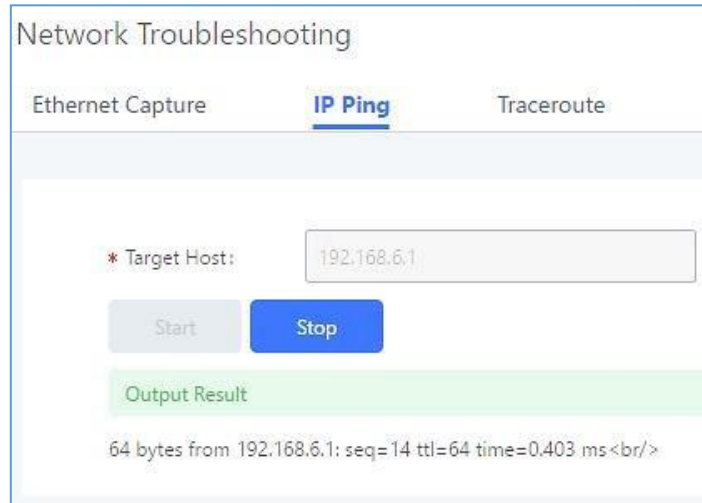
The output result is in .pcap format. Therefore, users could specify the capture filter as used in general network traffic capture tool (host, src, dst, net, protocol, port, port range) before starting to capture the trace.

**Note:** Capture files saved on external devices will now have “capture” prepended to file names.

## IP Ping

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

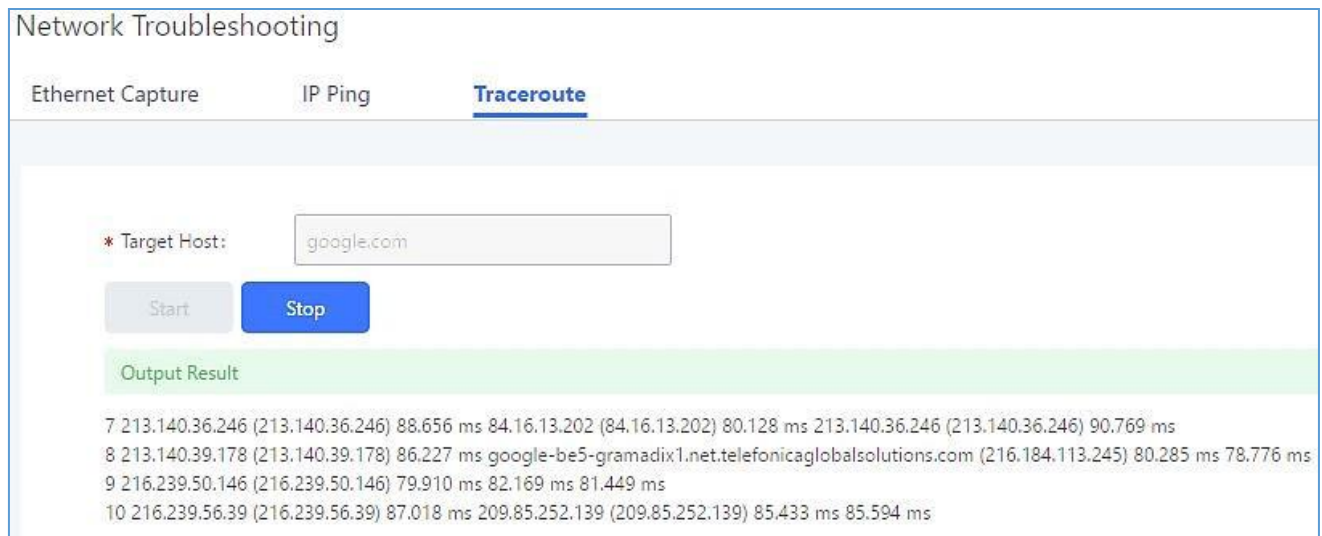




**Figure 346: Ping**

## Traceroute

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



**Figure 347: Traceroute**



## Signaling Troubleshooting

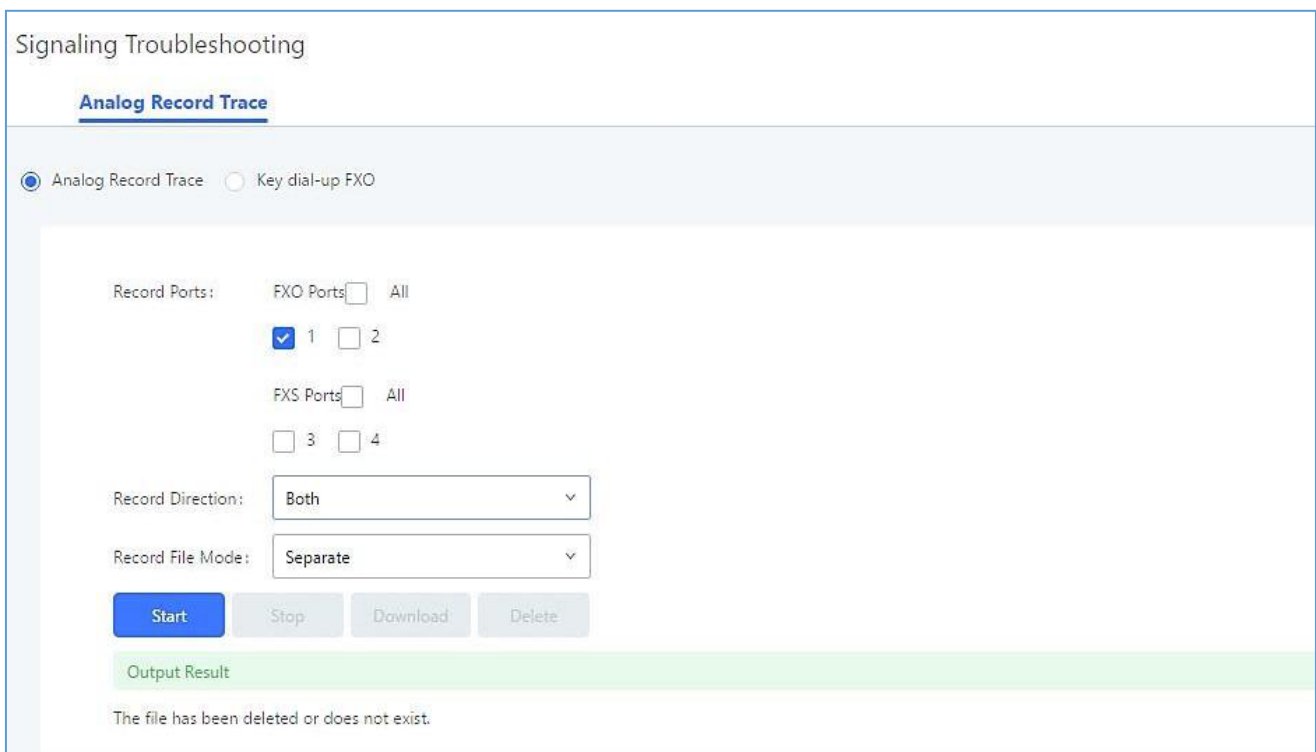
### Analog Record Trace

- **Analog Record Trace**

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

Here is the step to capture trace:

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



The screenshot shows the 'Analog Record Trace' configuration page in the Grandstream Web GUI. The page title is 'Signaling Troubleshooting' and the sub-page is 'Analog Record Trace'. There are two radio buttons: 'Analog Record Trace' (selected) and 'Key dial-up FXO'. Below this, there are sections for 'Record Ports' and 'Record Direction'. Under 'Record Ports', there are checkboxes for 'FXO Ports' and 'All'. Under 'FXO Ports', there are checkboxes for '1' (checked) and '2'. Under 'FXS Ports', there are checkboxes for '3' and '4'. Below this, there are dropdown menus for 'Record Direction' (set to 'Both') and 'Record File Mode' (set to 'Separate'). At the bottom, there are four buttons: 'Start' (highlighted in blue), 'Stop', 'Download', and 'Delete'. Below the buttons is a green bar labeled 'Output Result' and a message: 'The file has been deleted or does not exist.'

Figure 348: Troubleshooting Analog Trunks



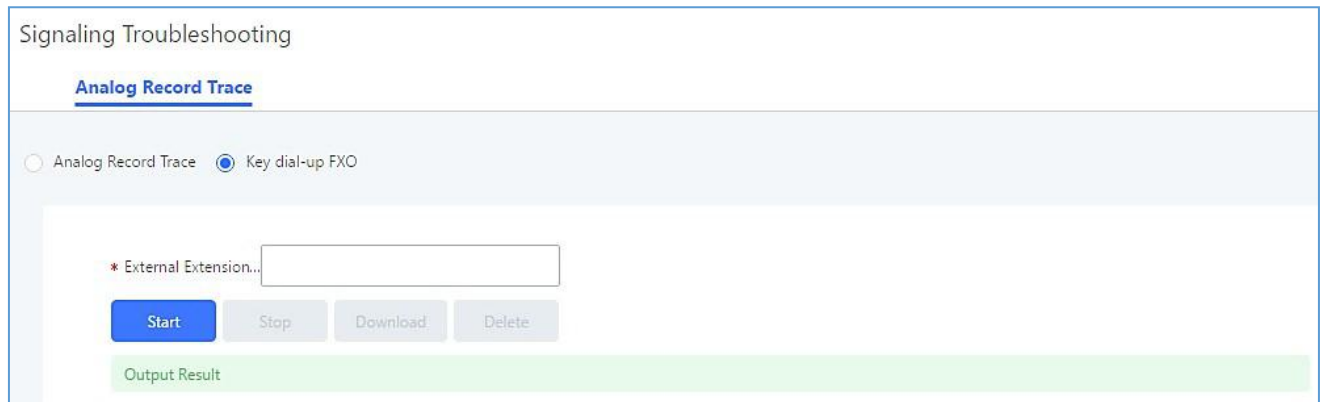


- **A key Dial-up FXO**

Users can directly set a PSTN number on the “**External Extension**” text box to troubleshoot issues related to the analog trunk easily, the following steps shows how to use this feature:

1. Configure analog trunk on UCM, including outbound route.
2. Enter a reachable external number in “**External Extension**”.
3. Press “**Start**” button. The call will be initiated to the external number.
4. Answer and finish the call before pressing “**Stop**” button.

The trace will be available for analysis to download after output result shows “Done! Click on Download to download the captured packets”.



**Figure 349: A Key Dial-up FXO**

**Note:** When using a Key Dial-up FXO feature the outbound trunk for the analog trunk need to have internal permission. As well as it should be the trunk with the highest outbound route priority.

After capturing the trace, users can download it for basic analysis. Or you can contact Grandstream Technical support in the following link for further assistance if the issue is not resolved.

<http://www.grandstream.com/index.php/support>

## Service Check

Enable Service Check to periodically check UCM6200. Check Cycle is configurable in seconds and the default setting is 60 sec. Check Times is the maximum number of failed checks before restart the UCM6200. The default setting is 3. If there is no response from UCM6200 after 3 attempts (default) to check, current status will be stored and the internal service in UCM6200 will be restarted.



### Service Check

Enable Server Check:

\* Check Cycle:

\* Check times:

**Figure 350: Service Check**

## Network Status

In UCM6200 Web GUI→**System Status**→**Network Status**, the users can view active Internet connections. This information can be used to troubleshoot connection issue between UCM6200 and other services.

#### Network Status

**Active Internet Connections (Servers And Established)**

Proto	Recv-Q	Send-Q	Local-Address	Foreign-Address	State
tcp	0	0	0.0.0.0:7681	0.0.0.0:*	LISTEN
tcp	0	0	0.0.0.0:7777	0.0.0.0:*	LISTEN
tcp	0	0	0.0.0.0:389	0.0.0.0:*	LISTEN
tcp	0	0	0.0.0.0:2000	0.0.0.0:*	LISTEN
tcp	0	0	0.0.0.0:8888	0.0.0.0:*	LISTEN

**Active Unix Domain Sockets (Servers And Established)**

Proto	RefCnt	Flags	Type	State	I-Node
unix	2	[ACC]	STREAM	LISTENING	8487
unix	2	[ACC]	STREAM	LISTENING	8491
unix	2	[ACC]	STREAM	LISTENING	8494
unix	2	[ACC]	STREAM	LISTENING	8498
unix	2	[ACC]	STREAM	LISTENING	8501

**Figure 351: Network Status**



## EXPERIENCING THE UCM6200 SERIES IP PBX

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

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

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

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

**\* Asterisk is a Registered Trademark of Digium, Inc**

