



GRANDSTREAM

CONNECTING THE WORLD



UCM6000 & GXP Phones Bundle Solutions





- Founded in 2002
- 30+ employees
- Over 125,000 customers worldwide

Office: Amherst, New York
Contact Us: 1-800-398-VoIP





Since 2002 VoIP Supply has delivered unparalleled service and expertise to over 125,000 customers worldwide.



NEW VoIP Rental Program

our Device as a Service (DaaS) rental program gives your customers the widest variety of VoIP Products for a low monthly payment!

Hardware

featuring over 60 manufacturers that offer over 16,000 products

Experts on Your Side

the most qualified professionals become your extended team!

CloudSpan MarketPlace

a single place to shop various VoIP service providers - finding the perfect match for your client's diverse business needs

Fulfillment

provisioning and professional services from multiple warehouse locations in North America; providing real-time access to manage your projects from order through delivery

Refresh & Reclaim

offering certified reconditioned devices at a fraction of the cost plus offering an outlet for selling off used and excess VoIP equipment



The Simplest way to sell cloud services & make \$

1. Sign VoIP Supply's CloudSpan Reseller Agreement
2. Introductions & training w/CloudSpan vendors
3. Leverage resources for marketing & quoting
4. Cash your commission checks!

Benefits of CloudSpan

- No quotas/commitment levels
- Best in class commissions and spiffs/incentives
- Sell hardware + services
- Programs for single person through SMB to enterprise
- Commissions grow as your business grows, higher than if you go direct



Become a VoIP Supply Partner

Call or Email Joe Shanahan at (716) 531- 4316 jshanahan@voipsupply.com

- Complete partner program agreements & paperwork
- Create an Onboarding Game Plan with your Account Representative (Training & Equipping your Team)
- Marketing & Sales Collateral - <https://www.voipsupply.com/partner-portal-home-page/>
- Opportunity Support – Consulting & Solution Design, Proposal/Pricing



AGENDA

1

INTRODUCTION

2

CALL PARK / CALL RECORDING

3

EVENT-LIST BLF / PRESENCE

4

TRANSFER MODES

5

CONFERENCE BRIDGE CEI

6

CALL CENTER AGENT EASY LOGIN

7

ZERO CONFIG

8

QUESTIONS AND ANSWERS



Grandstream's Long History of Open Source

SIP



ANDROID

Android is a Registered Trademark of Google Inc.



Asterisk is a Registered Trademark of Digium



VoIP Phone Product Category

High-End



GXV3275



GXV3240



GXP2130



GXP2135



GXP2140



GXP2160



GXP2170



DP720/750



GXP1760



GXP1780/82



GXP1610



GXP1620/1625



GXP1628



GXP1630

Low-End



UCM6200 series

IP PBX Appliance



- 2/4/8 FXO trunk port models, 2 FXS ports with lifeline capability.
- Up to 100 concurrent calls and up to 32 conference attendees
- Up to 800 SIP endpoints and up to 50 SIP trunk accounts
- Dual Gigabit network ports, integrated PoE, USB and SD ports
- Zero Configuration endpoint provisioning and no licensing fees



UCM6200 series

Enhanced Features



- Up to 100 concurrent calls and up to 32 conference attendees
 - 50 concurrent calls – UCM6202
 - 75 concurrent calls – UCM6204
 - 100 concurrent calls – UCM6208
- Up to 800 SIP endpoints
- Dual-core 1GHz processor, 1GB RAM for enhanced capacity and speed
- Three sizes – UCM6202, 6204 and 6208



UCM6510

IP PBX Appliance



- **E1/T1/J1 Interface**
- **2 PSTN trunk FXO ports, 2 FXS ports** with lifeline capability
- Up to **2000 SIP endpoints** and up to **50 SIP trunk** accounts
- **Dual Gigabit** network ports, integrated **PoE**, USB and SD ports
- Zero Config endpoint provisioning and no licensing fees





Call Management

Call Park

This feature allows you to place a call on hold, so it can be retrieved from another phone in the system.

To use Call Park: if you are on an active call at your phone, you can park the call in 3 ways:

- Press #72 and the call will be parked.
- Initiate a blind transfer then dial 700 to park the call.
- Press the MPK button configured with call park extension lot

You can retrieve the call in another room or someone else on another phone in your system can then dial the call park extension to retrieve the call.

If the call is not retrieved after the timeout, it will ring back the phone that parked it.





Call Management

Call Park

* Parking Lot Extension:	<input type="text" value="700"/>	* Parking Lot Name:	<input type="text" value="DefaultLot"/>
* Parking Slots:	<input type="text" value="701-720"/>	Use parklot as extension:	<input checked="" type="checkbox"/>
* Parking Timeout (s):	<input type="text" value="300"/>	Music On Hold Classes:	<input type="text" value="Default"/>
Destination When Time Out Call Busy:	<input type="text"/>	Timeout Callback Ringing All:	<input type="checkbox"/>

Feature Codes

Feature Maps

DND/Call Forward

Feature Codes

* Blind Transfer:	<input type="text" value="#1"/>	<input type="text" value="Disable"/>
* Seamless Transfer:	<input type="text" value="*44"/>	<input checked="" type="checkbox"/>
* Call Park:	<input type="text" value="#72"/>	<input type="text" value="Allow Both"/>
* Feature Code Digits Timeout:	<input type="text" value="1000"/>	

A screenshot of a mobile application interface for call management. At the top, it shows a status bar with a signal strength indicator, a battery icon, and the time 08:32 AM. Below the status bar, there's a header bar with a back arrow, a user icon, and the text 'ad mp'. The main content area is divided into two columns. The left column contains a list of call records with details like '1004', '1013', and '00:00:06'. The right column contains a list of call park slots, with 'Park 1' and 'Park 2' highlighted in green. At the bottom, there's a navigation bar with five buttons: 'EndCall', 'NewCall', 'Transfer', 'CallPark', and 'HideLabel'. A hand icon is pointing at the 'CallPark' button.





UC Features

Call Recording

- Recording length can be extended via USB flash drive
- Recording files are wav format and can be played and downloaded from UCM6XXX series web GUI
- Automatic recording can enable for specific extensions, ring groups, call queues, conference rooms, trunks.

Recordings Storage

Enable auto change:

☒

USB Disk:

☐

Local:

☐

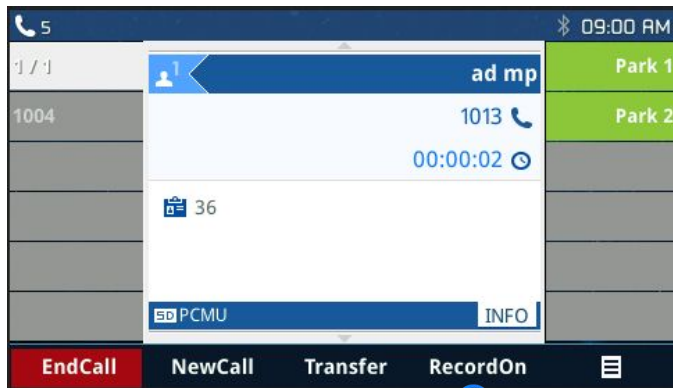
CDR						
<div><div>Delete All</div><div>Download All Records</div><div>Download Search Result (s)</div><div>Automatic Download Settings</div></div>						
Status	Call from	Call to	Action Type	Start Time	Talk Time	Account Code
<div><div></div><div></div></div>	1002	7946541 [Trunk: BranchOffice]	DIAL	2017-05-05 04:59:51	0:00:08	Emily/GSEMEA
<div><div></div><div></div></div>	1002	7654654 [Trunk: BranchOffice]	DIAL	2017-05-05 04:59:12	0:00:06	Jane/GSEMEA
<div><div></div><div></div></div>	1002	7564654 [Trunk: BranchOffice]	DIAL	2017-05-05 04:58:38	0:00:06	John/GSEMEA





UC Features

Call Recording



Session Initiation Protocol (INVITE)

Request-Line: INVITE sip:1005@192.168.22.174:5060 SIP/2.0

Message Header

Via: SIP/2.0/UDP 192.168.22.119:5060;rport;branch=z9hG4bKPj9068ead4-aaac-46c7-9ca5-5b911b96c175

From: "1013" <sip:1013@192.168.22.119>;tag=afb4dd78-4e1e-49b8-a481-68f17947331d

To: <sip:1005@192.168.22.174>

Contact: "1013" <sip:1013@192.168.22.119:5060>

Call-ID: 119f55d6-0b21-4a09-b866-6fa72e78ff03

CSeq: 29345 INVITE

Allow: OPTIONS, INFO, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, REFER, MESSAGE, REGISTER

Supported: 100rel, timer, replaces, norefersub

Session-Expires: 1800

Min-SE: 90

X-UCM-AudioRecord: *3

X-UCM-CallPark: #72

Remote-Party-ID: "1013" <sip:1013@192.168.22.119>;privacy=off;screen=no

Max-Forwards: 70

User-Agent: Grandstream UCM6510V1.2A 1.0.17.10

Content-Type: application/sdp

Content-Length: 388

Feature Codes

Save

Cancel

Feature Maps

DND/Call Forward

Feature Codes

Reset All

Default All

* Blind Transfer:

#1

Disable

* Attended Transfer:

*2

Disable

* Seamless Transfer:

*44

☒

* Disconnect:

*0

Allow Both

* Call Park:

#72

Allow Both

* Audio Mix Record:

*3

Allow Both

* Feature Code Digits Timeout:

1000



UC Features

Eventlist BLF

- UCM allows up to 20 endpoints to subscribe on the single resource list.
- UCM automatically assigns the BLF extensions into the available MPKs/VMPKs slots when Auto Provision Eventlist BLF is enabled on the phone

Auto Provision Eventlist BLFs

☐ Disabled ☒ Enabled



Create New Event List

URI:

BLF_LIST_SAMPLE

Local Extensions:

Available Extensions/Extension Groups

5001 "David Jackson"
5003 "Eric Green"
5005 "Martin Cook"

Selected Extensions/Extension Groups

5000 "John Smith"
5002 "William Thompson"
5004 "Jose Hernandez"

Remote Extensions:

Available Extensions

ting-1 "ting1"
ting-2 "ting2"
ting-3 "ting3"
ting-4 "avaya03"

Selected Extensions

Special Extensions:



UC Features

SIP Presence

- SIP Presence is a feature available to phones with programmable keys such as Multi-Purpose Keys (MPKs) that allows them to monitor the service status of assigned extensions.
- A phone would send a SIP SUBSCRIBE message to the UCM to check for changes in the status of specified extensions, and the UCM would return the status of those extensions in a SIP NOTIFY message back to the phone.



Available	The contact is online and can participate in conversations/phone calls.
Away	The contact is currently away (ex: for lunch break).
Chat	The contact has limited conversation flexibility and can only be reached via chat.
Do Not Disturb	The Contact is on DND (Do Not Disturb) mode.
Custom Presence Status	Please enter the presence status for this mode on the web GUI.
Unavailable	The contact is unreachable for the moment, please try to contact later.





Call Management

Call Transfer

Call Transfer allows a user to transfer calls to another phone. Our UCM supports 3 types of Transfer :

- Blind Transfer : Blind Transfer involves passing a call without notifying the recipient
- Attended Transfer : Attended Transfer involves passing a call by notifying the recipient, first hold initial call then call recipient while first call is waiting, then connect the 2 calls.
- Seamless Transfer : Seamless Transfer allows user to perform blind transfer using UCM feature code without having music on hold presented during transfer process.

Attended Transfer Mode

☐ Static ☒ Dynamic

1 / 1

1004

TRANSFER

Park 1

Park 2

Cancel BlindTrnf AttTrnf HideLabel Target

Feature Codes

Feature Maps DND/Call Forward Feature Codes

Reset All Default All

* Blind Transfer: #1 Disable

* Seamless Transfer: *44 ☒

* Call Park: #72 Allow Both

* Feature Code Digits Timeout: 1000



UC Features

Conference

Conference Bridge feature allows to have several people participating in call. Different Conference Bridges can be defined on UCM6XXX and have following main features:



- ☐ Public or Private
- ☐ Recording Option
- ☐ Bridging of Multiple Parties
- ☐ Caller Menu
- ☐ User Invite
- ☐ Caller Announcements

	UCM6102/6104/6202/6204	UCM6108/6116/6208	UCM6510
Max. Number of Conference Bridges	3	6	8
Max. Number of Participants	25	32	64





UC Features

Conference

CEI allows users to monitor the conference call activity from the LCD screen of the GXP phone



+ Create New Conference Room		⊙ Conference Settings		Enable CEI Notify <input checked="" type="checkbox"/>	
Room	Attendee	Administrator	Start Time	Activity	Options
+ 6300	0	0		--	





UC Features

Call Center

UCM supports lightweight call center features including virtual queue, switchboard 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 once an agent is free.

- ☐ Virtual Queue.
- ☐ Call Queue position announcement
- ☐ Call Queue Statistics
- ☐ Switchboard for all agents





UC Features

Call Center –Web Based Call Center

- A new Web based Call Center solution, it provides every user a bird-view into the phone system
- Extensive call control and visibility features
- Checking the extension status in real-time
- Adjust the agents in queue and monitor/transfer/hang up the call on web

The screenshot displays the Grandstream Web Based Call Center interface. On the left is a dark sidebar menu with options: IVR, Voicemail, Ring Groups, Paging/Intercom, Call Queue (highlighted), Pickup Groups, Dial By Name, Speed Dial, DISA, Callback, Event List, Feature Codes, Fax/T.38, PBX Settings, System Settings, Maintenance, CDR, and Value-added Features. The main area is titled 'Switchboard' and contains a table with columns for status, extension, and call details. Below this are two sections: 'Ringing' and 'Calling', each with a table showing active calls.

Status	Extension	Call Count	Duration	Caller	Time	Options
Ringing	10612	2	9	1970-01-01 00:00:00	1970-01-01 00:05:21	Static
Unavailable	10613	0	0	--	1970-01-01 00:07:07	Static
In Use	10614	0	1	1970-01-01 00:00:00	1970-01-01 00:05:21	Static
Unavailable	10615	0	0	--	1970-01-01 00:07:07	Static
In Use	10616	0	0	1970-01-01 00:00:00	1970-01-01 00:05:21	Static
Unavailable	10617	0	0	--	1970-01-01 00:07:07	Static
In Use	10618	0	1	1970-01-01 00:00:00	1970-01-01 00:05:21	Static
Unavailable	10619	0	0	--	1970-01-01 00:07:07	Static

Status	Caller	Callee	Talk Time	Options
📞	PJSIP/10616-00000527	6500	2017-01-24 13:57:44	📞
📞	PJSIP/10614-00000529	6500	2017-01-24 13:57:50	📞

Status	Caller	Callee	Talk Time	Options
📞	PJSIP/10618-0000052c	PJSIP/10610-0000052e	2017-01-24 13:58:02	⚙️



UC Features

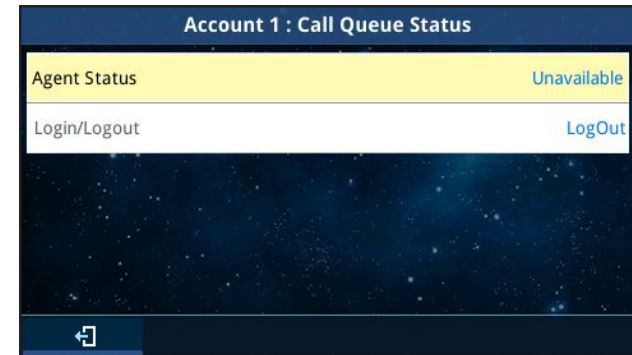
Call Center – Call Queues Agents

- ❑ Call queue Agents are members defined to answer the queue calls. Agents can be either static or dynamic.
- ❑ Agent Easy Login

Enable Agent Login :



UCM Model	Max Agents in Call Queue
UCM6102 and UCM6202	18
UCM6104 and UCM6204	27
UCM6108 and UCM6116	36
UCM6208	60
UCM6510	120



Special Feature

UCM Call Center





Deployment using Zero Config

Setup Made EASY!

- Plug & Play
- Auto Discovery
- Automatic Assignment
- Minimal Manual

Zero Config

Zero ConfigGlobal PolicyGlobal TemplatesModel TemplatesModel UpdateZero Config Settings

Basic Settings

Enable Zero Config:☒

Enable Automatic Configura...☐

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 for the device to download.

Auto Assign Extension: ☐

Zero Config Extension Segm...5000 - 6299Zero Config Extension Segment













Enable Pick Extension: ☐

Pick Extension Segment: 4000 - 4999Pick Extension Segment


Pick Extension P

Network Se

Subnet Whitelis

	MAC Address	IP Address	Extension	Version				
<input type="checkbox"/>	000B822DD46F	192.168.6.196		1.0.8.9				
<input type="checkbox"/>	000B8275CB88	192.168.6.147		1.0.9.4	Grandstream	GXP2130	1	 
<input type="checkbox"/>	000B82836615	192.168.6.185		1.0.8.46	Grandstream	GXP2160	1	 
<input type="checkbox"/>	000B829E25D4	192.168.6.169		1.0.0.48	Grandstream	--	1	 
<input type="checkbox"/>	000B825C6927	192.168.6.157		1.0.9.7	Grandstream	GXP2160	1	 
<input type="checkbox"/>	000B8282C6B7	192.168.6.238		1.0.4.55	Grandstream	GXP1630	1	 
<input type="checkbox"/>	000B822F1066	192.168.6.156		1.0.15.5	Grandstream	GXW4004	1	 

Create New Device



Model: GRANDSTREAM GXP2170

MAC Address: 000B82654F11

IP Address: 192.168.6.145

Version: 1.0.7.91

BasicAdvanced

Accounts

Hot Deskling: No

Account 1: 1001

Account 2: 1001

Auto Discover

The PBX can automatically discover the new devices by ARP or PING. It can scan the entire network segment or a single IP address.

PBX LAN/LAN1 Address: 192.168.2.1

Network Segment: 192.168.2.0 - 192.168.2.255

Broadcast IP: 192.168.2.255

Scan Method: SIP-Message

Scan IP: 192.168.6.137

CancelOK

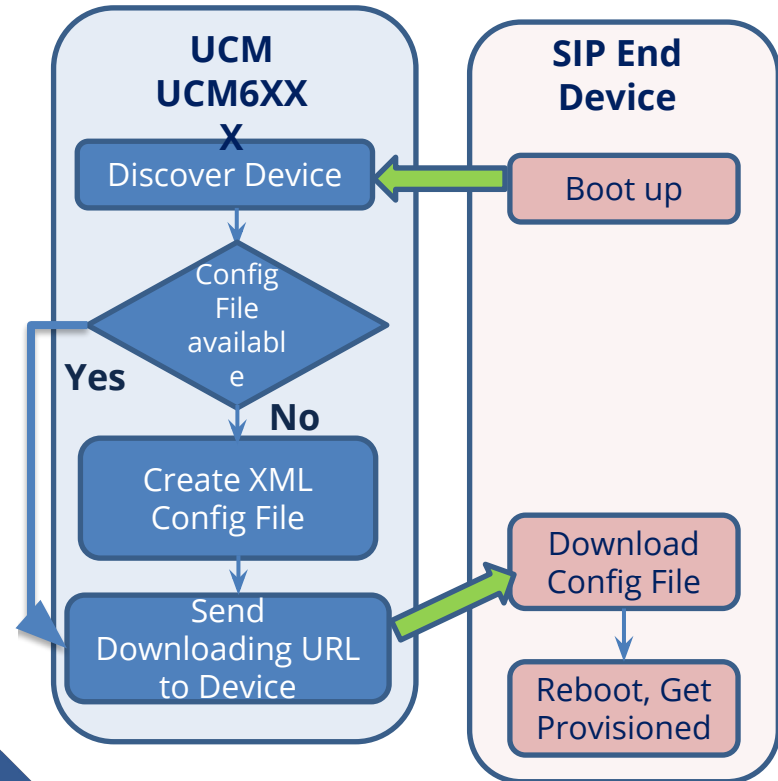


Installation & Deployment

Zero Config - Mechanism

Three methods for the “interaction” between SIP End Device and the UCM:

- **SIP SUBSCRIBE**
- **Option 66** (route mode only)

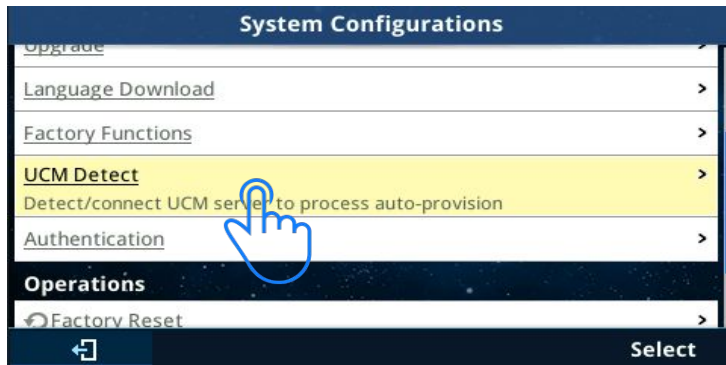




Installation & Deployment

Zero Config – Auto provisioning settings

- Automatic Configuration and extensions' assignment
- Configurable Extensions segment
- Pick Extension feature
- Flexible network segments whitelist



Zero Config

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

Basic Settings

Enable Zero Config: ☒

Enable Automatic Configura... ☐

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 Segm...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](#)

















Installation & Deployment

Zero Config – Discovery

Discovery methods:

- ✓ PING
- ✓ ARP
- ✓ SIP MESSAGE (NOTIFY)

The discovered devices will be displayed in the list.

<input type="checkbox"/>	MAC Address	IP Address	Extension	Version	Vendor	Model	Create Config	Options
<input type="checkbox"/>	000B822DD46F	192.168.6.196		1.0.8.9	Grandstream	GXP1405	1	 
<input type="checkbox"/>	000B8275CBB8	192.168.6.147		1.0.9.4	Grandstream	GXP2130	1	 
<input type="checkbox"/>	000B82836615	192.168.6.185		1.0.8.46	Grandstream	GXP2160	1	 
<input type="checkbox"/>	000B829E25D4	192.168.6.169		1.0.0.48	Grandstream	--	1	 
<input type="checkbox"/>	000B825C6927	192.168.6.157		1.0.9.7	Grandstream	GXP2160	1	 
<input type="checkbox"/>	000B8282C6B7	192.168.6.238		1.0.4.55	Grandstream	GXP1630	1	 
<input type="checkbox"/>	000B822F1066	192.168.6.156		1.0.15.5	Grandstream	GXW4004	1	 

Auto Discover ✕

The PBX can automatically discover the new devices by ARP or PING. It can scan the entire network segment or a single IP address.

PBX LAN/LAN1 Address: 192.168.2.1

Network Segment: 192.168.2.0 - 192.168.2.255

Broadcast IP: 192.168.2.255

Scan Method:

SIP-Message ▾

Scan IP:

192.168.6.137

Cancel

OK

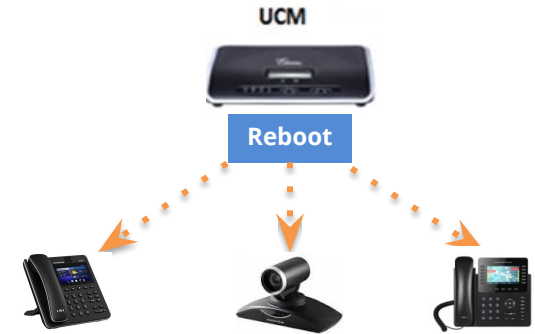


Installation & Deployment

Zero Config – Reboot remote device

The UCM6XXX supports rebooting discovered devices from zero config page.






To use this feature, please navigate to web UI-> Value-added Features -> Zero Config -> Zero Config, and click the reboot button under Options tab.



Zero Config

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

Auto Discover Create New Device Delete Selected Devices Modify Selected Devices Reset All Extensions Filter: Scan Results

	MAC Address	IP Address	Extension	Version	Vendor	Model	Create Config	Options
<input type="checkbox"/>	000B826B24CD	192.168.5.129	1005	1.0.3.171	GRANDSTREAM	GXV3275	Connected	    

Total: 1 < 1 >

30 / page Goto 1



Installation & Deployment

Zero Config – Configuration

- Global Policy
- Global Template
- Model Configuration
- Device Configuration






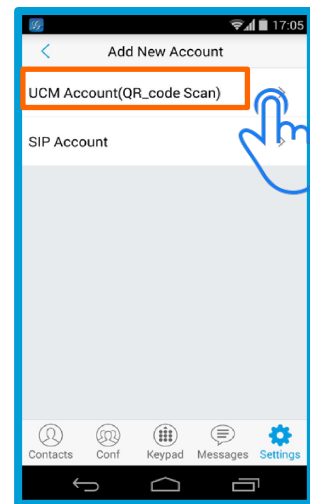
Installation & Deployment

Integration with GS-Wave QR code scan

GSWave is a Grandstream softphone application for iPhone and Android devices on Android 4.0 and higher. It can be automatically provisioned by the UCM using QR code scan.

Steps:

1. Go to the UCM web UI->Extension/Trunk->Extensions. Click on  on
2. The UCM users that have Email address configured in the extensions will receive account registration and LDAP configuration information in Email.
3. On GSWave app, select "UCM Account (QR_code Scan)" to scan and get registered immediately.



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.



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



Questions?

Please submit your questions using the Q/A feature.



jshanahan@voipsupply.com (716) 531-4316

