

Grandstream Networks, Inc.

Grandstream Wave for Android™

User Manual



COPYRIGHT

©2017 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 guide 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.



Table of Contents

CHANGE LOG	9
GS Wave Version 1.0.2.16.....	9
GS Wave Version 1.0.2.2.....	9
WELCOME	10
PRODUCT OVERVIEW	11
Feature Highlights.....	11
Grandstream Wave Technical Specifications	11
Grandstream Wave Android™ Prerequisites	12
Using Grandstream Wave.....	13
Dial Screen	13
<i>Dialing a Number Directly</i>	14
<i>Redial</i>	15
<i>Dialing a Number via Call History</i>	16
<i>Dialing a Number via Contacts</i>	16
Switching Audio Channel during Call.....	17
Answering Calls	18
<i>Single Incoming Call</i>	18
<i>Multiple Incoming Calls</i>	20
Active Call	21
<i>Call Hold/Resume</i>	23
<i>Mute</i>	24
<i>Switching Audio Channel During Call</i>	25
Call Recording	26
Missed Calls.....	27
Call Transfer.....	27
<i>Blind Transfer</i>	28
<i>Attended Transfer After Calling</i>	28
<i>Attended Transfer Before Calling</i>	29
6-Way Conference	30
<i>Add New Conference</i>	32



<i>Initiating Conference</i>	32
<i>Removing Participant from Conference</i>	34
<i>Mute/Unmute Conference</i>	34
<i>Hold/Resume Conference</i>	35
Direct IP Call	36
Voicemail.....	37
Contacts.....	39
<i>Add Contacts</i>	40
<i>Search Contacts</i>	40
<i>View Contact</i>	41
<i>Edit Contact</i>	42
<i>Delete Contact</i>	42
<i>Add Contact to Favorites</i>	44
<i>LDAP Contacts</i>	45
Call History.....	46
Messages.....	48
<i>View Message</i>	49
<i>Create New Message</i>	51
<i>Edit Message</i>	51
SETTINGS	53
Account Settings.....	53
<i>UCM Account (Scan QR Code)</i>	53
<i>UCM Account (Select QR Code Image)</i>	54
<i>SIP Account</i>	55
Advanced Settings	61
<i>General Settings</i>	61
<i>Call Settings</i>	61
<i>Audio Settings</i>	62
<i>Network Settings</i>	63
<i>Additional Settings</i>	63
Custom Settings.....	66
About Version.....	67
Debug	68
Exit	69



CONNECTING GS WAVE WITH GDS3710 DOOR SYSTEM..... 70

EXPERIENCING GS WAVE..... 72



Table of Tables

Table 1: GS Wave Features at a Glance.....	11
Table 2: DP750 Technical Specifications	11
Table 3: Edit Account Parameters	57
Table 4: Account Settings - General Settings Parameters.....	57
Table 5: Account Settings - Call Settings Parameters	57
Table 6: Account Settings - SIP Settings Parameters	58
Table 7: Session Expiration Settings Parameters	59
Table 8: Account Settings - Network Settings Parameters	59
Table 9: Account Settings - Codec Settings Parameters	60
Table 10: Advanced Settings - General Settings Parameters.....	61
Table 11: Advanced Settings - Network Settings Parameters.....	63
Table 12: Advanced Settings - Additional Settings Parameters	63
Table 13: LDAP Settings Parameters	65
Table 14: Custom Settings Parameters	66
Table 15: Debug Settings Parameters	68



Table of Figures

Figure 1: Grandstream Wave Finger Gestures on the Touchscreen	13
Figure 2: Keypad Screen	14
Figure 3: Select Account	15
Figure 4: Dial-up via Call History	16
Figure 5: Dial-up via Contacts.....	17
Figure 6: Switch Call Lines.....	18
Figure 7: Single Incoming Audio Call	19
Figure 8: Single Incoming Video Call.....	19
Figure 9: Multiple Calls - Audio call.....	20
Figure 10: Multiple Calls - Video call.....	21
Figure 11: Audio Call Interface	22
Figure 12 Video Call Interface.....	23
Figure 13: Call UnHold.....	24
Figure 14: Call Mute.....	25
Figure 15: Call via Bluetooth	26
Figure 16: Call Recording	26
Figure 17: Missed Call Screen	27
Figure 18: Call Transfer - Blind Transfer	28
Figure 19: Attended Transfer after Calling - Transferring.....	29
Figure 20: Attended Transfer before Calling - Split.....	30
Figure 21: Grandstream Wave Conference Screen.....	31
Figure 22: Grandstream Wave Conference - Add New Call to the Conference	32
Figure 23: Grandstream Wave Conference - Initiating Conference.....	33
Figure 24: Grandstream Wave Conference - Delete Conference participant.....	34
Figure 25: Grandstream Wave Conference - Mute Conference Participant	35
Figure 26: Grandstream Wave Conference - Hold Conference.....	36
Figure 27: Direct IP Call	37
Figure 28: Configure Voicemail UserID.....	38
Figure 29: Retrieve Voicemail	38
Figure 30: Grandstream Wave Contacts Screen	39
Figure 31: Grandstream Wave Add New Contact.....	40
Figure 32: Grandstream Wave Search Contact.....	41
Figure 33: Grandstream Wave View Contact.....	42
Figure 34: Edit Contact	43
Figure 35: Batch Remove Contact Screen	43
Figure 36: Contact Details Screen	44
Figure 37: Add Contact to Favorites	45
Figure 38: LDAP Contacts Screen - Search Contact.....	46
Figure 39: Grandstream Wave Missed Calls Screen.....	47



Figure 40: Call History Details Screen	47
Figure 41: Call History Details Screen - Edit Contact	48
Figure 42: Grandstream Wave Messages Screen	49
Figure 43: Grandstream Wave Message Details Screen.....	50
Figure 44: Create New Message	51
Figure 45: Grandstream Wave Message Screen - Edit Message	52
Figure 46: Add New Account Screen	53
Figure 47: QR Code Scan Screen	54
Figure 48: Scan QR Code Image Screen	55
Figure 49: Account Settings Screen - Registration Success	56
Figure 50: Slide to Delete Account.....	56
Figure 51: Call Settings Screen	62
Figure 52: DND Settings Screen.....	62
Figure 53: BLF List Screen	64
Figure 54: LDAP Settings Screen	65
Figure 55: About Page	67
Figure 56: Debug	68
Figure 57: Exit the Application	69
Figure 58: Configuring GDS3710 Settings on GS Wave	70
Figure 59: GDS Settings.	71
Figure 60: "OpenDoor" Button	71



CHANGE LOG

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

GS Wave Version 1.0.2.16

- Added video software encode feature.
- Add IP call feature [Direct IP Call].
- Added Speaker / MIC gain settings [Audio Settings].
- Added audio recording feature [Call Recording].
- Add "Check SIP User ID for Incoming INVITE" [Check SIP User ID for Incoming Invite].
- Add GDS Settings and open the door feature [GDS Settings] [CONNECTING GS WAVE WITH GDS3710 DOOR SYSTEM].
- Support audio Codec G.729 [Voice / Video Codecs and Capabilities] [Preferred Vocoder].
- Remove IPVideoTalk account.

GS Wave Version 1.0.2.2

- This is the initial version for GS Wave Android™.



WELCOME

Thank you for using Grandstream Wave. To meet the requirements of our customers, Grandstream Wave emerged on the basis of our existing multimedia VoIP Phones and enable users to move freely and continue to receive calls from any business or residential SIP account. The Grandstream Wave is a free softphone application that allows users to connect to their SIP accounts from anywhere in the world and it supports Android™ 4.0 and higher, and it is compatible with most of Android™ mobile phones and tablets.

By combining powerful phone functions and integration of Grandstream UCM applications, businesses throughout the world can use Grandstream Wave for all communication and productivity requirements with unprecedented high quality experience.

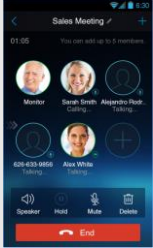


PRODUCT OVERVIEW

Feature Highlights

The following tables contain the major features of the GS Wave IOS™:

Table 1: GS Wave Features at a Glance

 <p>Grandstream Wave IOS™</p>	<ul style="list-style-type: none"> • Support Android™ 4.0 and higher • Standard SIP-based softphone with exceptional voice quality • Strong security features including SIP over TLS and 128 or 256-bit SRTP • Support 6 SIP accounts, up to 6-way audio conferences • Support CID, voicemail and call encryption • Support synchronize with local Contacts and call history on the phone • Enterprise features including UCM integration, BLF, call transfer/pickup, LDAP • Powerful NAT traversal options including automatic NAT discovery, STUN and UPnP • Automatic call forward based on time and location rules • Support G.711, G.726, G.722, iLBC, Opus, and G.729 • Automatic provision including XML provision and QR code scan • Fully customizable skins and themes for optional branding needs
--	--

Grandstream Wave 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 the GS Wave:

Table 2: DP750 Technical Specifications

Lines	6 lines with up to 6 independent SIP accounts
Protocols and Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, DNS(A record, SRV, NAPTR), STUN/ICE, SIMPLE, LDAP, TLS, DTLS, IPv6 (Pending)
Network	Support 2G/3G/4G and WiFi
Graphic Display	800 x 480 resolution or higher
Camera	Support forward or rear facing cameras
Bluetooth	Support making calls with Bluetooth



Voice / Video Codecs and Capabilities	Support G.711μ/a, G.722 (wide-band), G.726-32, iLBC, G.729, GSM, DTMF (In audio, RFC2833, SIP INFO), Opus, HD Audio, H264, video resolution up to 720p HD
Audio Quality	Full-duplex speaker, AEC, AGC, Noise Reduction, PLC, Adaptive JIB
Telephony Features	Call hold, mute, transfer, forward (unconditional/no-answer/busy/time-based) , call park, paging/intercom, DND (Do Not Disturb), busy lamp field (BLF), downloadable phone book (XML, LDAP), call waiting, call history, flexible dial plan, custom ringtones, server redundancy & fail-over, BLF
UCM Integration	Supports many functions like QR code scan
Mobile Device Integration	Supports background mode, proximity sensor for in-call touch screen and keys lock, auto rotation, GPS location based call forward (pending)
Feature Functions	LDAP, MWI (Message Waiting Indicator) , display instant online status, call history and messages
QoS	Layer 3 (ToS, DiffServ, MPLS) QoS
Security	Support AES configuration file, TLS encryption, SRTP encryption (128-bit and 256-bit) , HTTPS
Multi-language	English, Simplified Chinese, Polish, Germany, Russian, Italian, Arabic, Spanish, Portuguese, French, etc.

Grandstream Wave Android™ Prerequisites

The Grandstream Wave is compatible with most of Android™ mobile phones and tablets running Android™ 4.0 or higher version and it supports 2G/3G/4G and WiFi. Users could download Grandstream Wave via scan QR code, or from Google Play store.



Note: When using the Grandstream Wave for the first time, users have to confirm whether allow the application to read local contacts from the phone. If it allows, users could view local contacts on the corresponding Grandstream Wave screen.

To fully manipulate the Grandstream Wave capacitive touch screen, use fingers to operate following the introductions below on the Grandstream Wave icons, buttons, menu items, onscreen keyboard, etc.

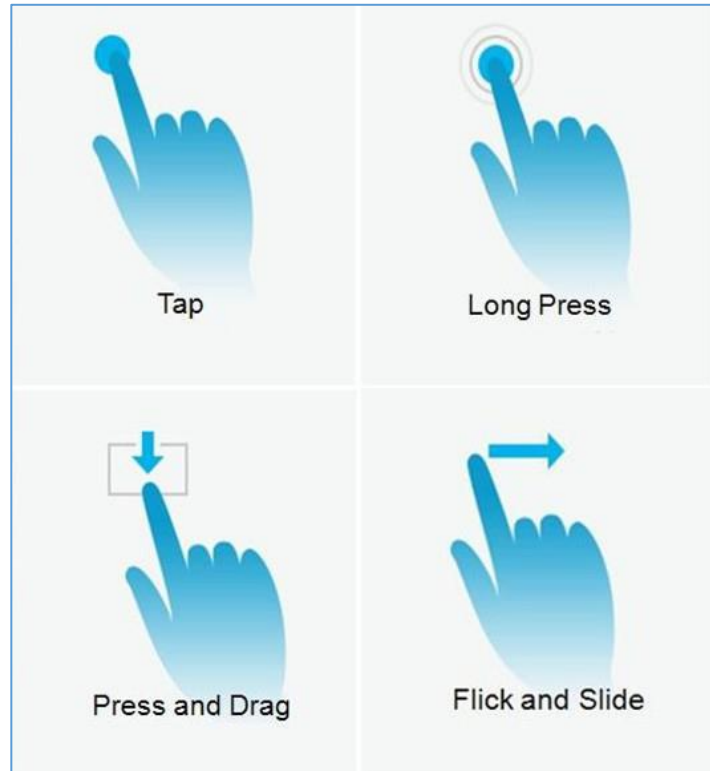



Figure 1: Grandstream Wave Finger Gestures on the Touchscreen

- **Tap:** Slightly touch the screen with fingertip once to initiate menu, options or applications.
- **Long Press:** Touch the screen with fingertip for about 2 seconds without lifting finger from the screen to bring up the context menu for more operations.
- **Press and Drag:** Press the item and move it by dragging the finger up, down, left or right, without lifting finger from the screen.
- **Flick and Slide:** Touch the screen with fingertip and slide over the screen. For example, users could slide up to scroll up the page, slide down to open dropdown menu, slide left to delete an item from the list. If the finger stays on the screen for too long, the item may be selected and sliding will not occur.

Using Grandstream Wave

This chapter provides basic operations on the GS wave, including making / receiving calls, call transfer, conference calls, managing contacts and etc...

Dial Screen

Tap on the keypad button  at the bottom of screen to open dial screen, as shown in figure 2.

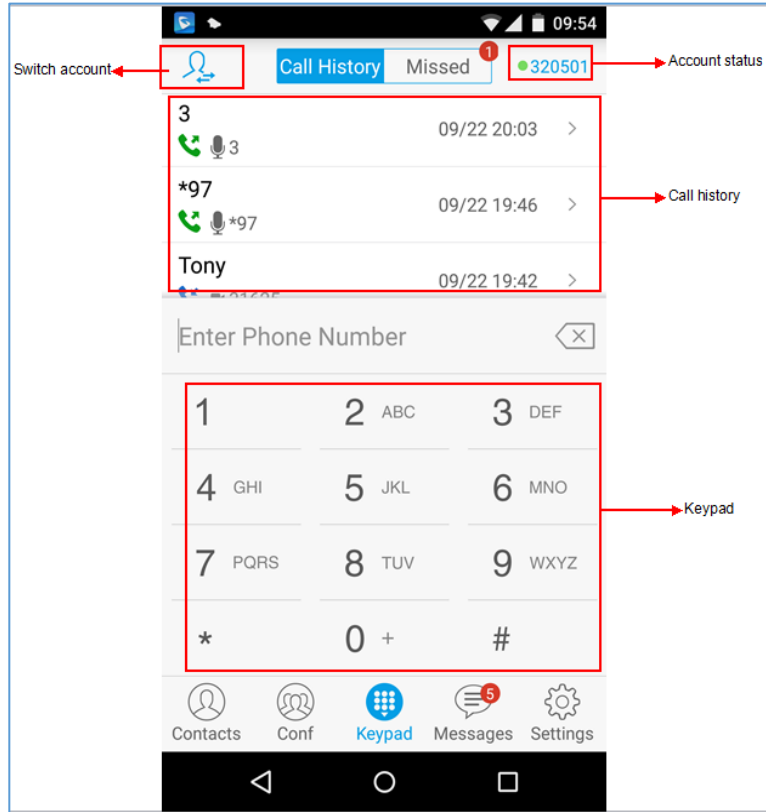



Figure 2: Keypad Screen

Dialing a Number Directly

1. Access the dial screen;
2. Put one finger on left screen edge, and slide to right or tap  on the upper left corner, select the account as shown on the following screenshot;

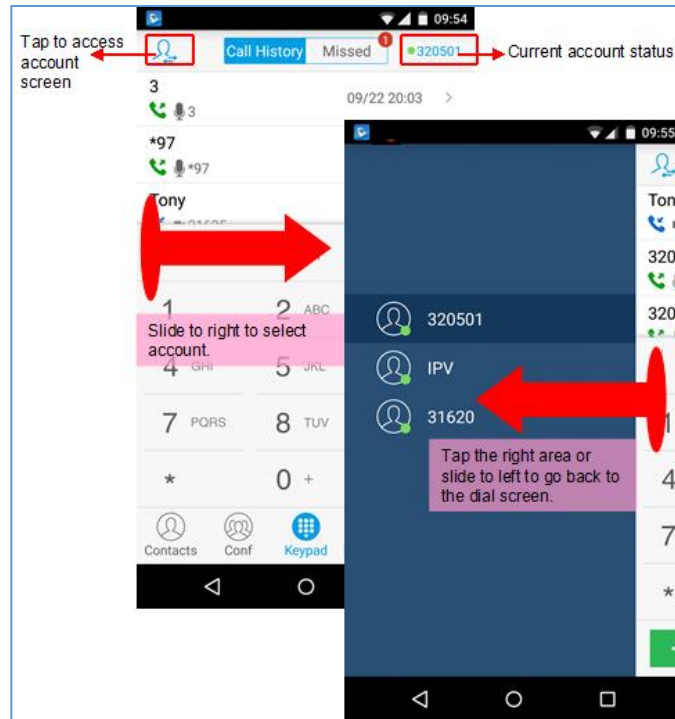




Figure 3: Select Account

3. Tap the right area to go back to the dial screen;
4. Enter the phone number on the keypad;
5. Tap on  to dial out with SIP account; Tap on  and select "Local Call", "Video Call" or "Paging" to dial out via local phone number, or select "New Contact" to add the number as a contact quickly.

Note:


- By default, Grandstream Wave allows users to press # key as SEND key. This behavior can be disabled via set option "Use # as Dial Key" to "No" under Settings->Account Settings.
- If inserting an active SIM card into the phone, users could make calls with the SIM card number but cannot send messages with the local phone number.

Redial

Users can dial out the last dialed number if there is dialed call history.

1. Access the dial screen;
2. Press # key to dial out the last dialed number.

Dialing a Number via Call History

The Grandstream Wave call history is listed on the upper of the dial screen. It displays all call histories (local and SIP account) and missed calls. Navigate on the call history entries by tapping on button  on the bottom of the main screen to slide up/down as displayed on the following figure.

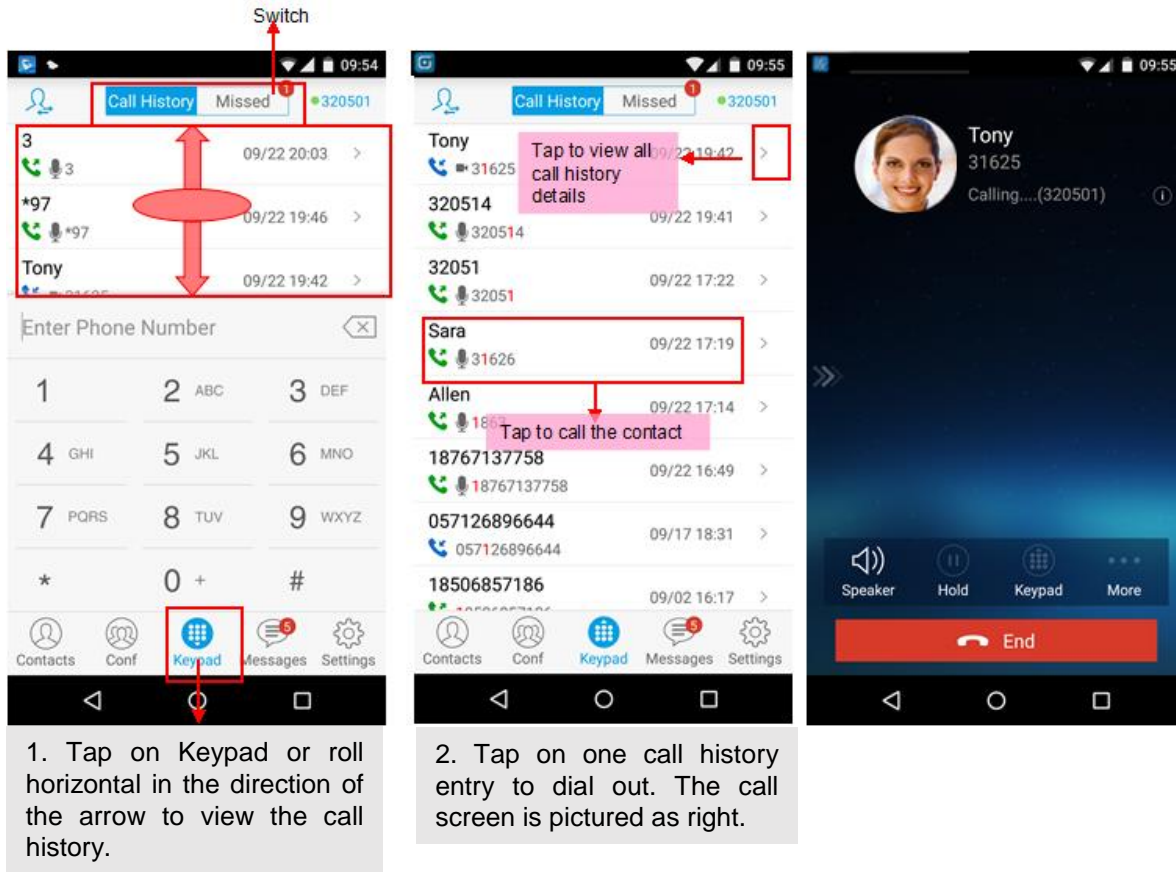



Figure 4: Dial-up via Call History

Note: Dialing out through call history will use the account which made the last call.

Dialing a Number via Contacts

Access Contacts by tapping on  icon on the bottom of the main screen, the SIP contacts and LDAP Contacts (please go to Settings page to configure first) are shown up individually. Follow the steps in figure 6 to dial a number via Contacts.

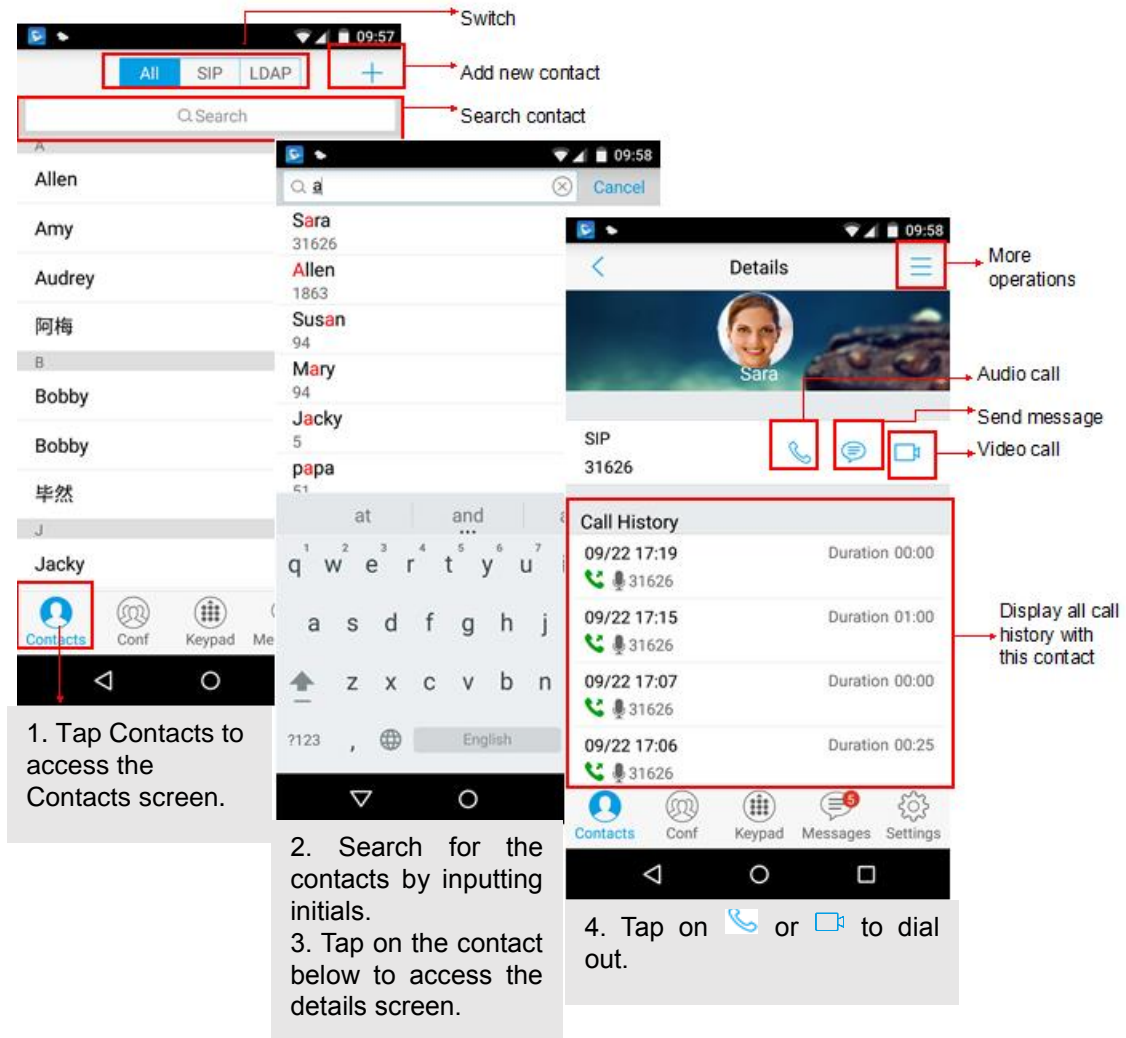


Figure 5: Dial-up via Contacts

Switching Audio Channel during Call

Users could switch lines by sliding the call screen when there are multiple calls, as shown in figure below.

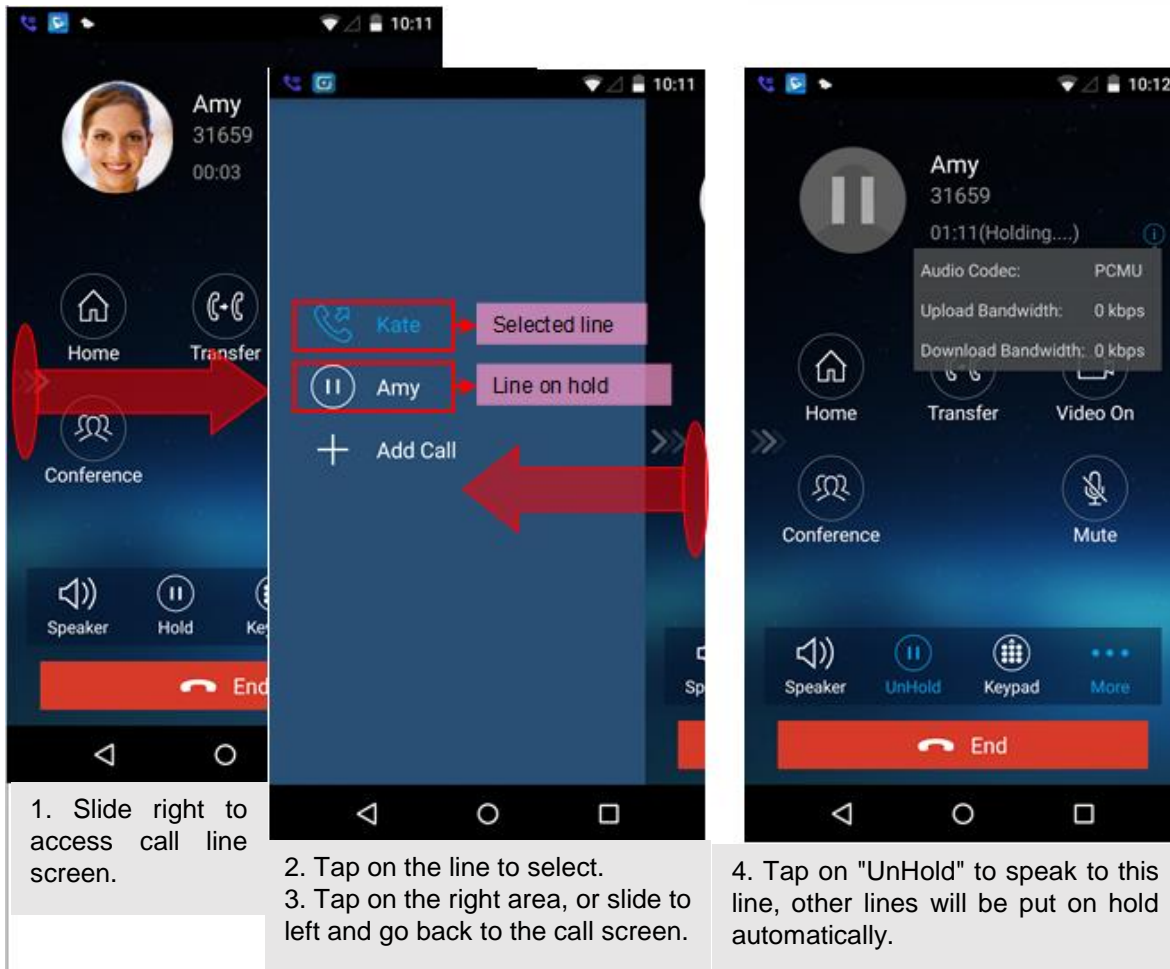

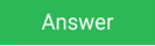



Figure 6: Switch Call Lines

Answering Calls

Single Incoming Call

When the phone is at idle state, and there is an incoming audio call, the status bar will display the icon  and the phone screen is as shown on the figure below.

Tap on button  to answer the call via speaker, or tap on button  to reject the call.

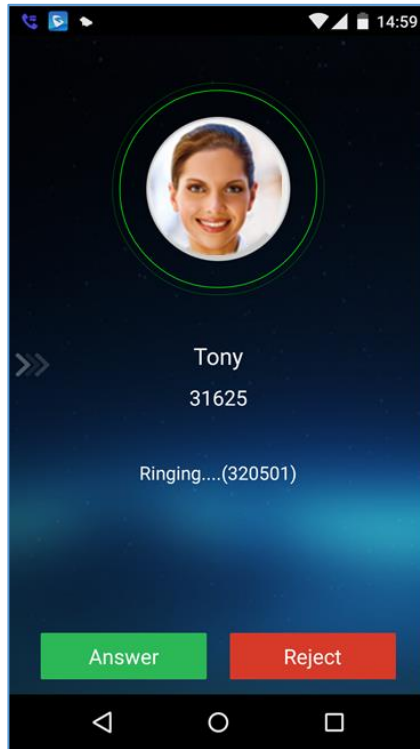


Figure 7: Single Incoming Audio Call

When there is a video call, you can see the screen is as shown below.

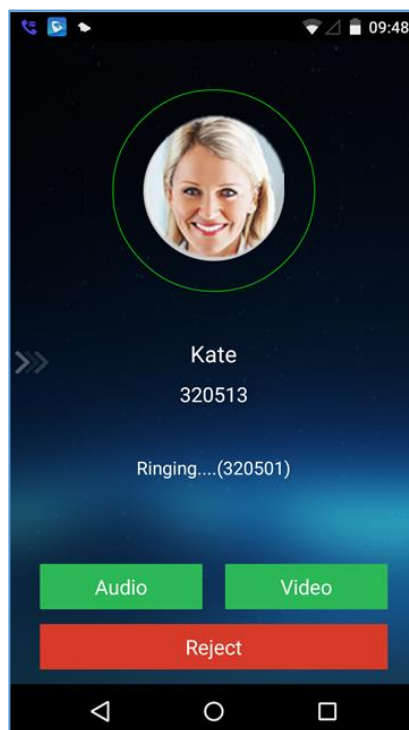

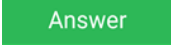



Figure 8: Single Incoming Video Call

Multiple Incoming Calls

When there is another incoming audio call during an active call, the status bar will display the icon , and at the same time, users will hear call waiting tone, with the screen displaying the caller's name or number for the incoming call. A prompt appears for users to confirm as shown on the following figure.

Tap on button  to answer the call, once the new call is answered, the current active call will be placed on hold. If the new call is rejected by tapping on button , the current active call will not be interrupted.

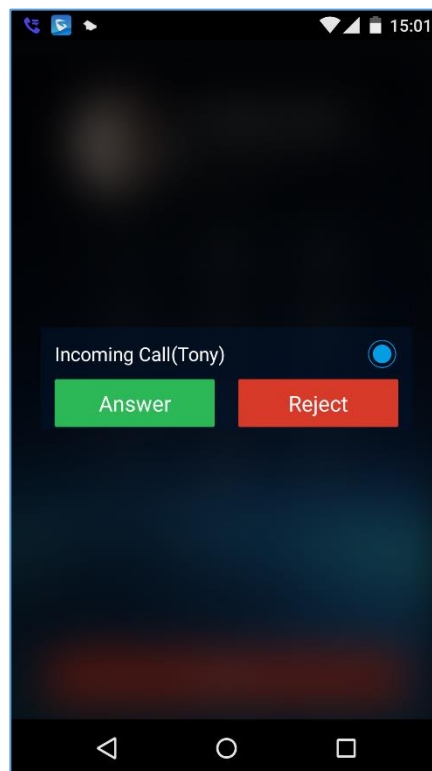





Figure 9: Multiple Calls - Audio call

If the incoming call is video call, the screen is shown as below. Tap on  or  to answer. once the new call is answered, the current active call will be placed on hold. If the new call is rejected by tapping on button , the current active call will not be interrupted.

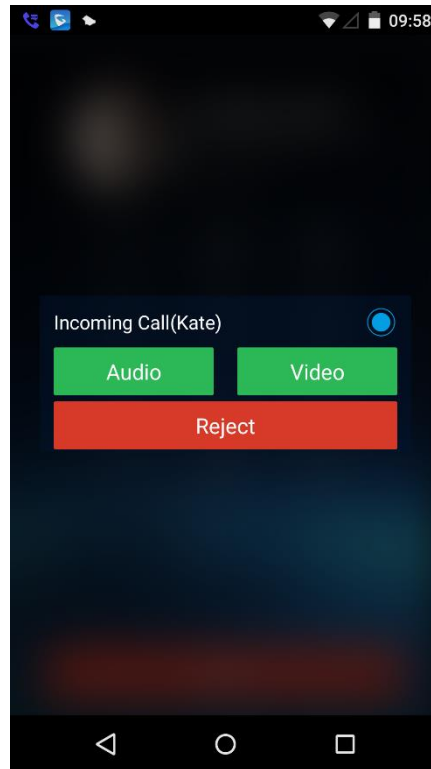



Figure 10: Multiple Calls - Video call

Active Call

During an active call, users could hold/resume call, mute/unmute, input DTMF, add new call, initiate conference, end a call or switch audio channel, turn on/off video, switch front/rear camera. Tap  on left screen, and slide right to bring up the lines list. Users could switch to other lines or add a new call.

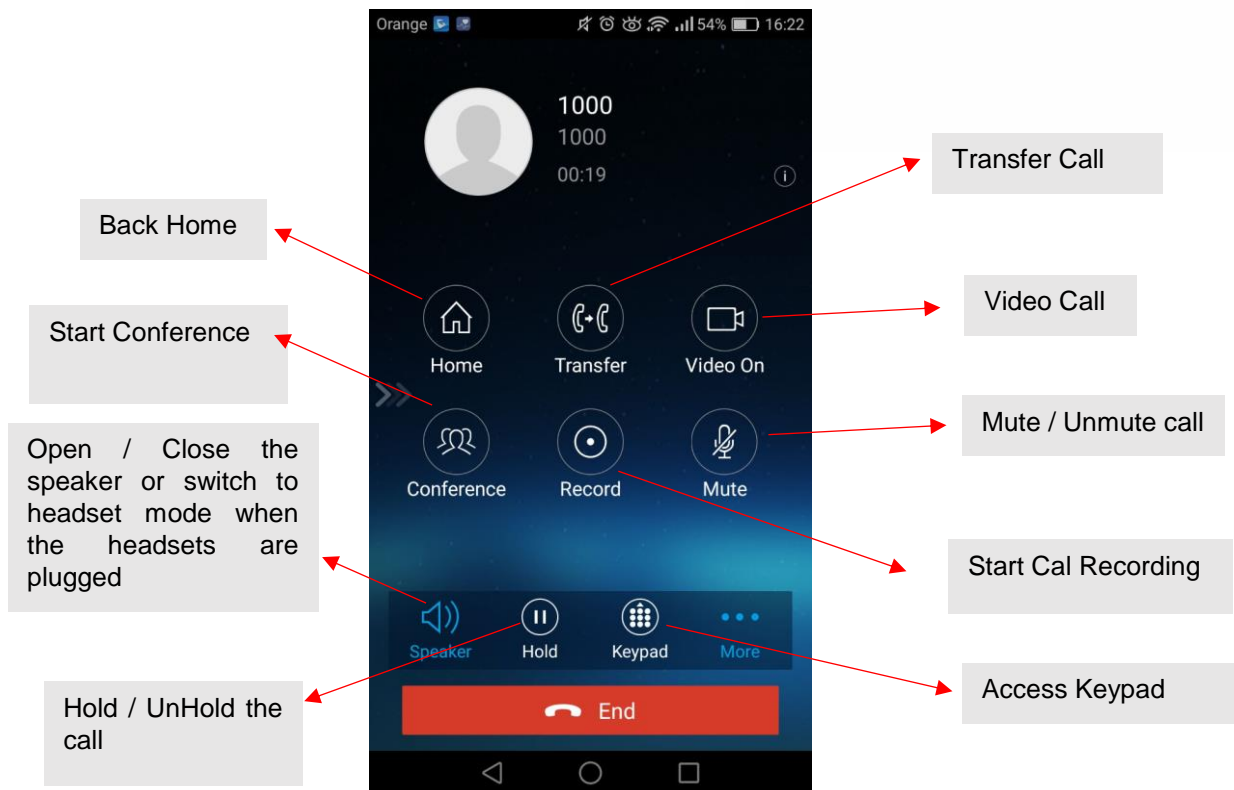




Figure 11: Audio Call Interface

-  : Slide to right to add new call or switch lines.
- **Speaker**: Switch voice channels to speaker or 3.5mm headset if it is plugged in.
- **Hold/UnHold**: During the call, users could press the HOLD button to hold or resume the call at any time.
- **Keypad**: Tap on the icon to bring up digital soft keypad for inputting DTMF.
- **More**: Access more operations including Home, Transfer, Conference and Mute/Unmute.
- **Home**: Back to the home screen (dial screen), the active call interface will be hidden; users could tap on button  at the upper left corner of the screen to go back to the call interface.
- **Transfer**: Switch to the transfer screen. Grandstream Wave supports blind transfer and attended transfer. Please refer to chapter **Call Transfer** for more details.
- **Video On**: Enable video call. Tap to dial up video call to the callee.
- **Conference**: Bring up conference screen.
- **Mute**: Tap on the icon to mute/unmute the call.
- **End**: Tap on the icon to end the call.

The video call screen is shown below; the basic operations are the same as audio call.

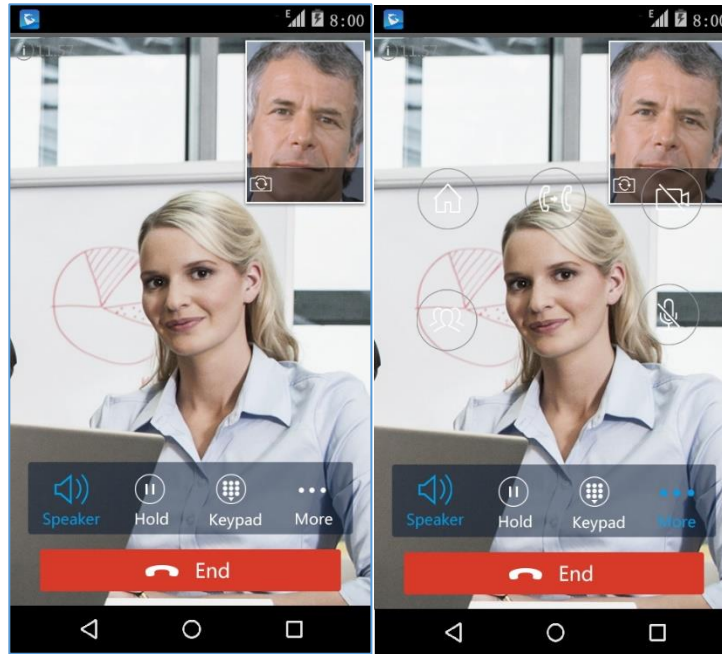


Figure 12 Video Call Interface

Call Hold/Resume

During the active call, press the **HOLD** button to put the call on hold. Users could dial up or answer a new call. The call hold screen is displayed on the following screenshot:

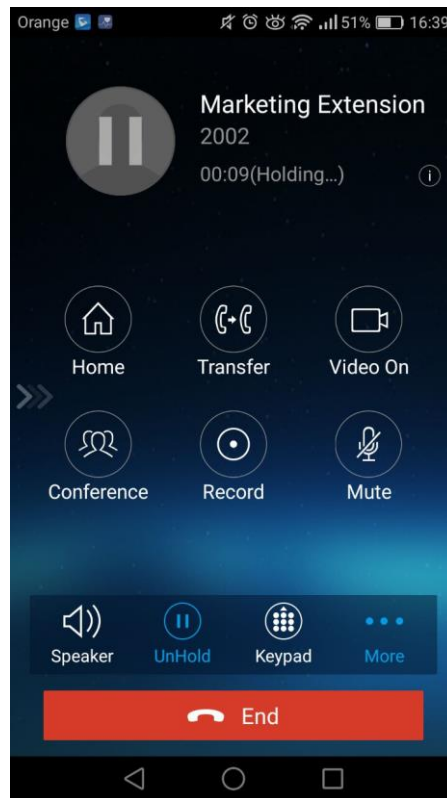



Figure 13: Call UnHold

To resume the call, press the **UNHOLD** button again to resume the call if the current active call is put on hold.

Mute

During an active call, press the mute button  to mute the call. Press the button mute  again to unmute the call. The mute screen is displayed on the following figure:

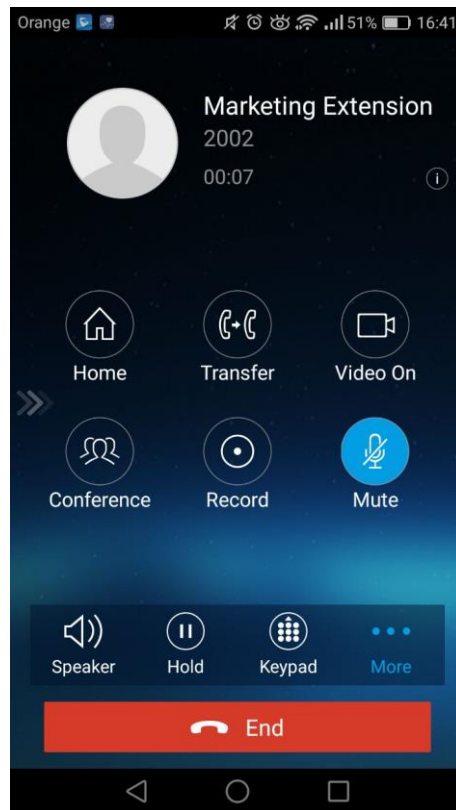


Figure 14: Call Mute

Switching Audio Channel During Call

Grandstream Wave allows users to switch audio channel among handset (if user plugs in headset, the handset status will be turned into headset status), speaker or Bluetooth headset when making calls.


Following screenshots shows the call screen when using the Bluetooth, tap on button  to switch channels.



Figure 15: Call via Bluetooth

Call Recording

During an active call, users can start easily recording the outgoing audio conversation and retrieve the recording stored on the internal memory of the phone under the folder named “Record” as displayed on the following screenshots.

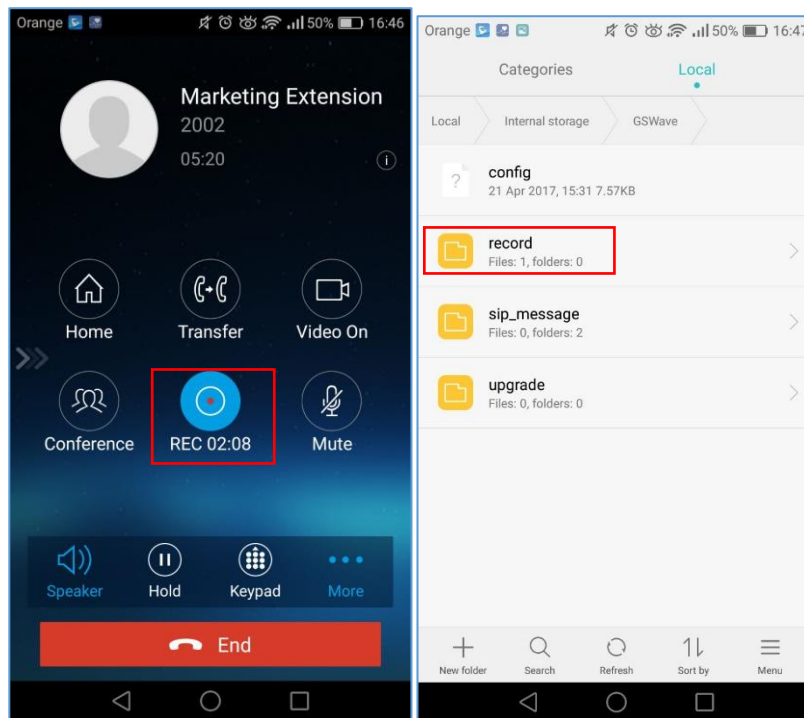


Figure 16: Call Recording

Missed Calls

When there is a missed call, the phone will display  on the status bar and prompt on Grandstream Wave call history list, as shown in figure below.

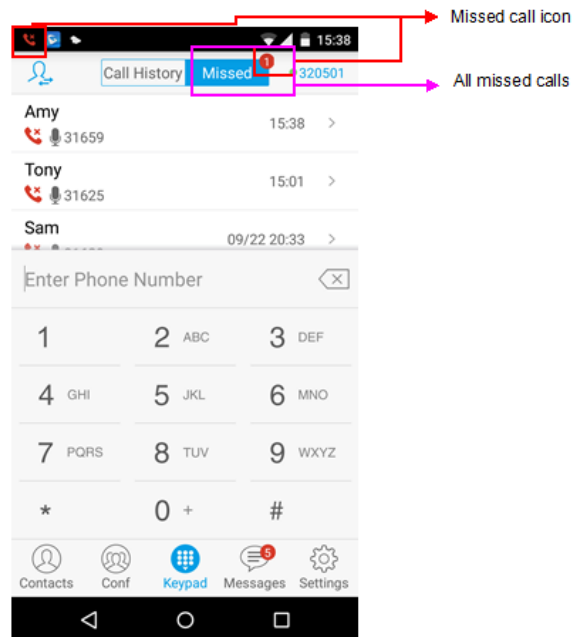


Figure 17: Missed Call Screen

Call Transfer

A call can be transferred to another party during the call. The Grandstream Wave supports blind transfer and attended transfer.

Blind Transfer

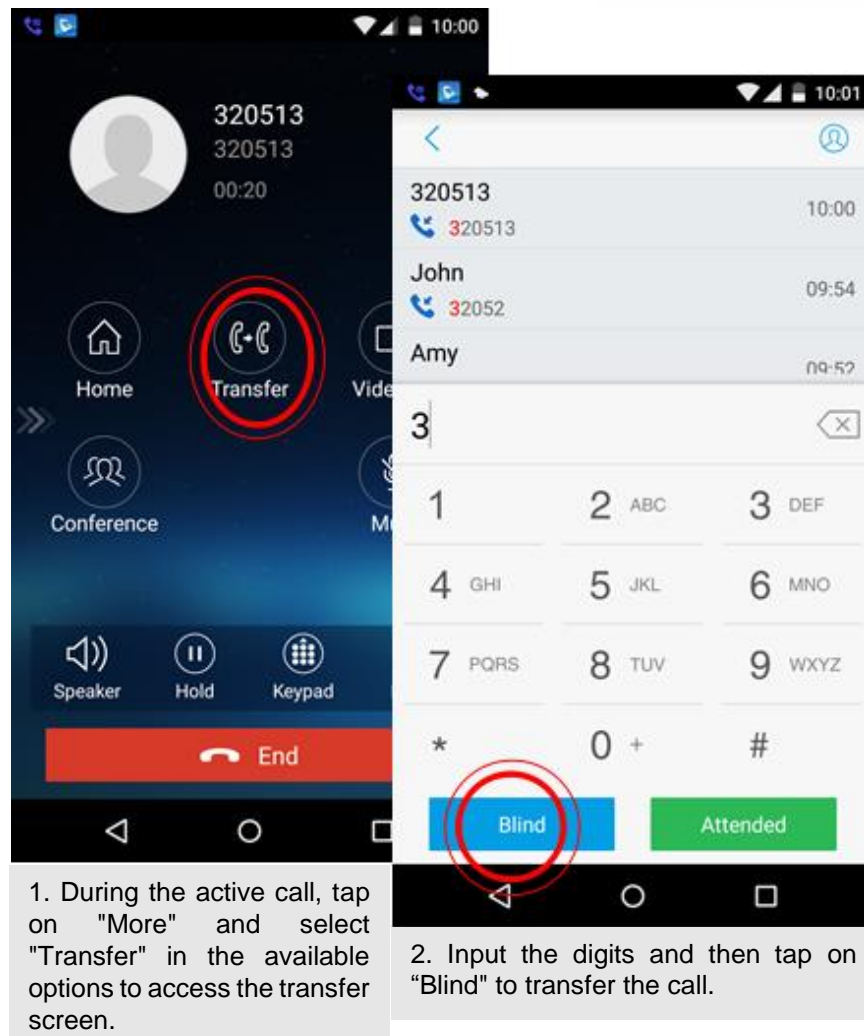


Figure 18: Call Transfer - Blind Transfer

When the ringback tone is played, users will automatically go back to the main screen (dial screen) to complete the transfer after the callee answers the call.

Note: If entered incorrect digits, tap on button  to delete the digits one by one, or long press it to clear all digits.

Attended Transfer After Calling

Grandstream Wave supports attended transfer before or after calling, which provides users a fast and easy way to complete attended transfer. Make an active call first and follow the steps below to transfer the call to the third party.

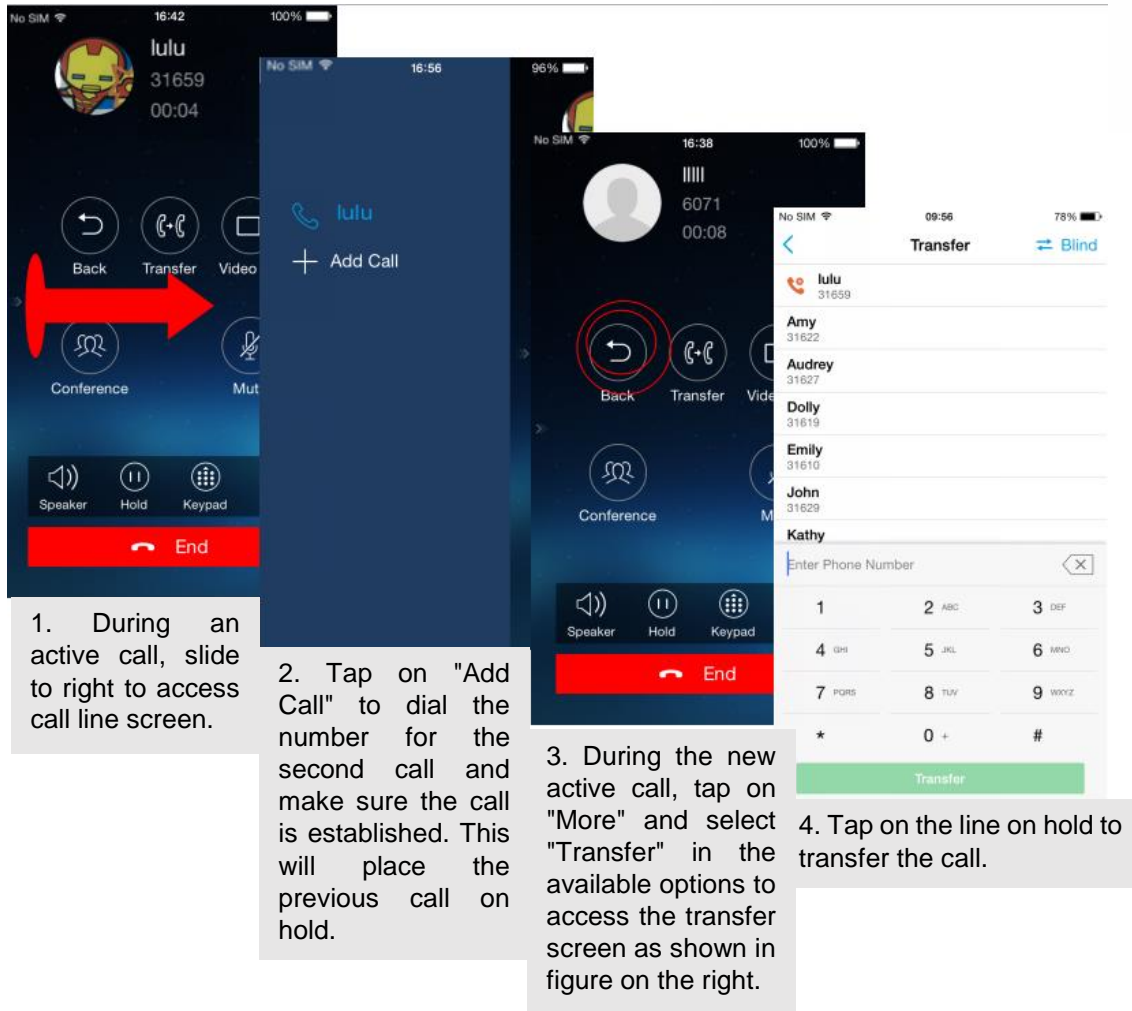


Figure 19: Attended Transfer after Calling - Transferring

Attended Transfer Before Calling

Users can also consult the third party first before transferring the call. The following steps illustrate how to attend before calling:

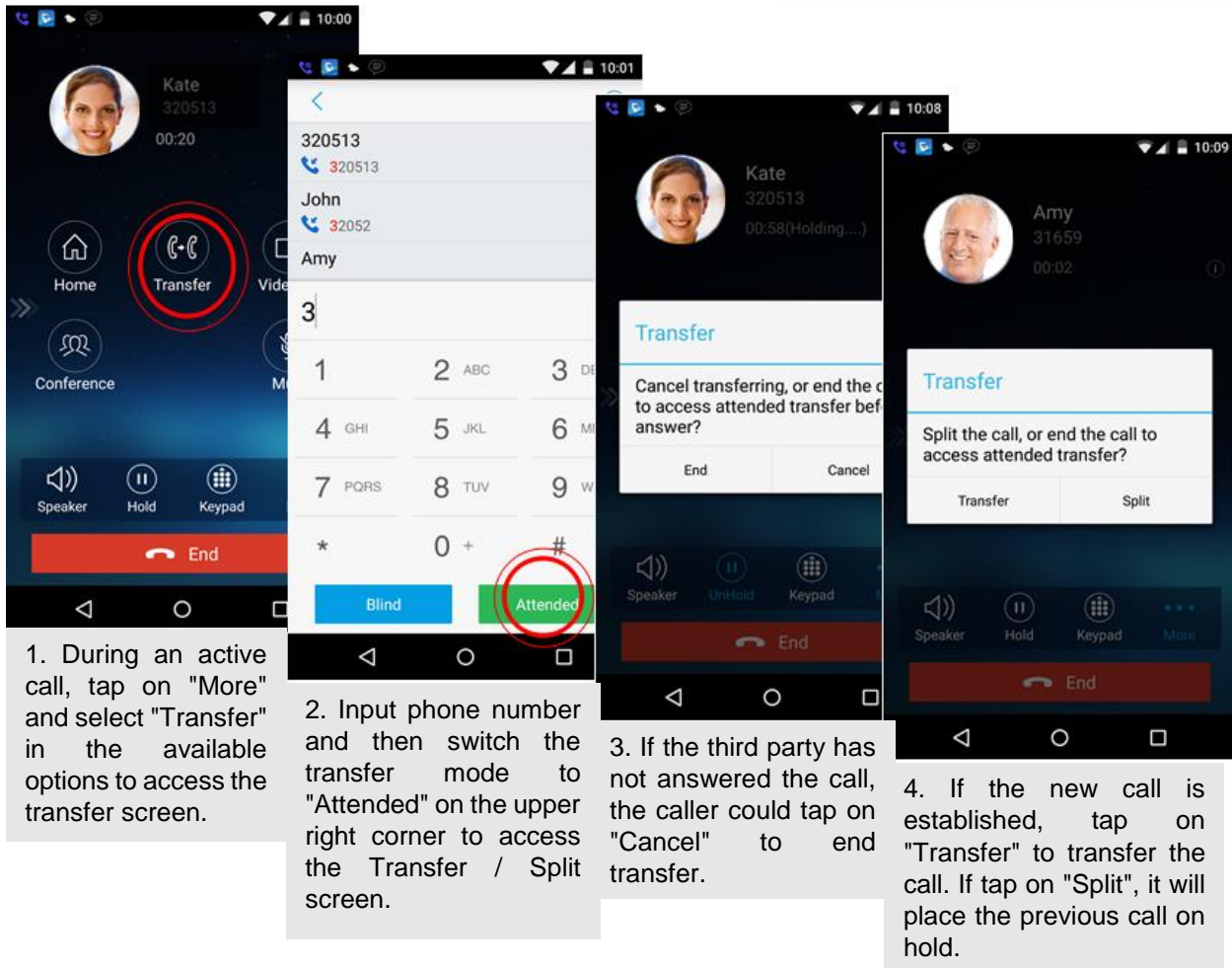


Figure 20: Attended Transfer before Calling - Split

6-Way Conference

Grandstream Wave supports up to 6-way conferencing. The conference screen is displayed on the screenshot below:

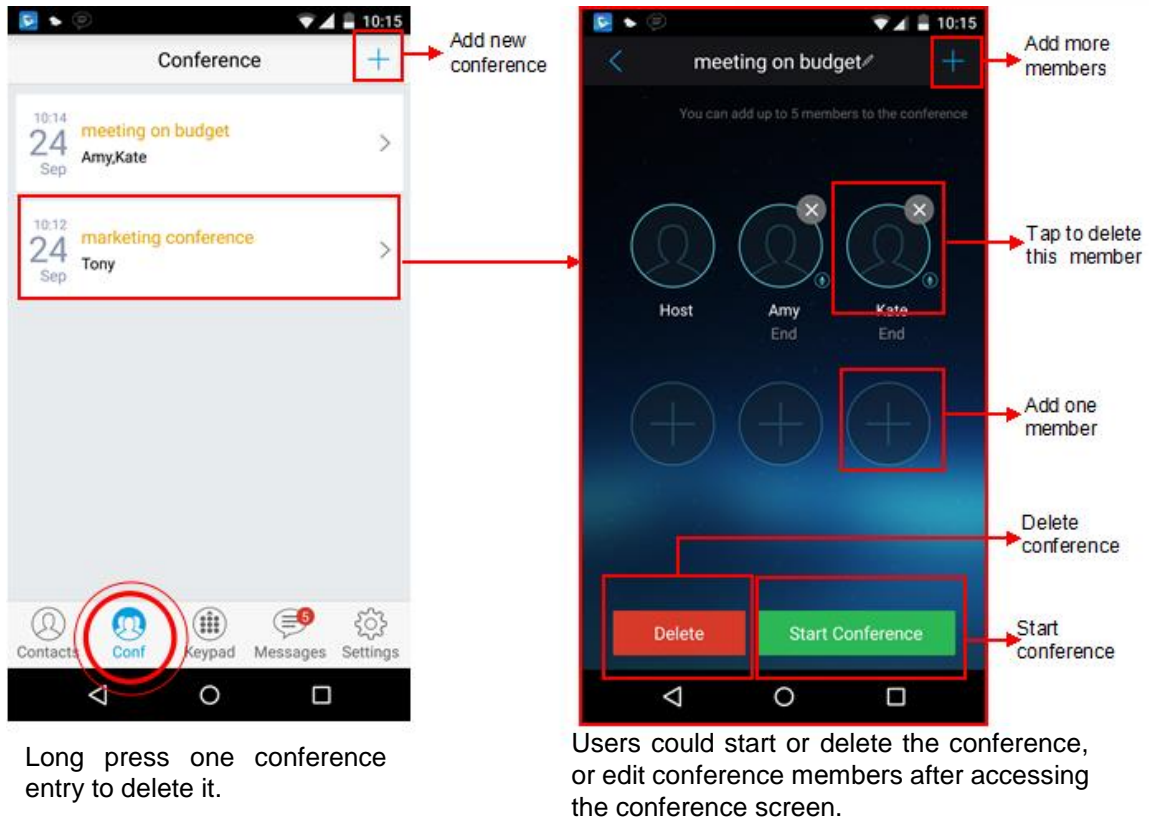


Figure 21: Grandstream Wave Conference Screen

Add New Conference

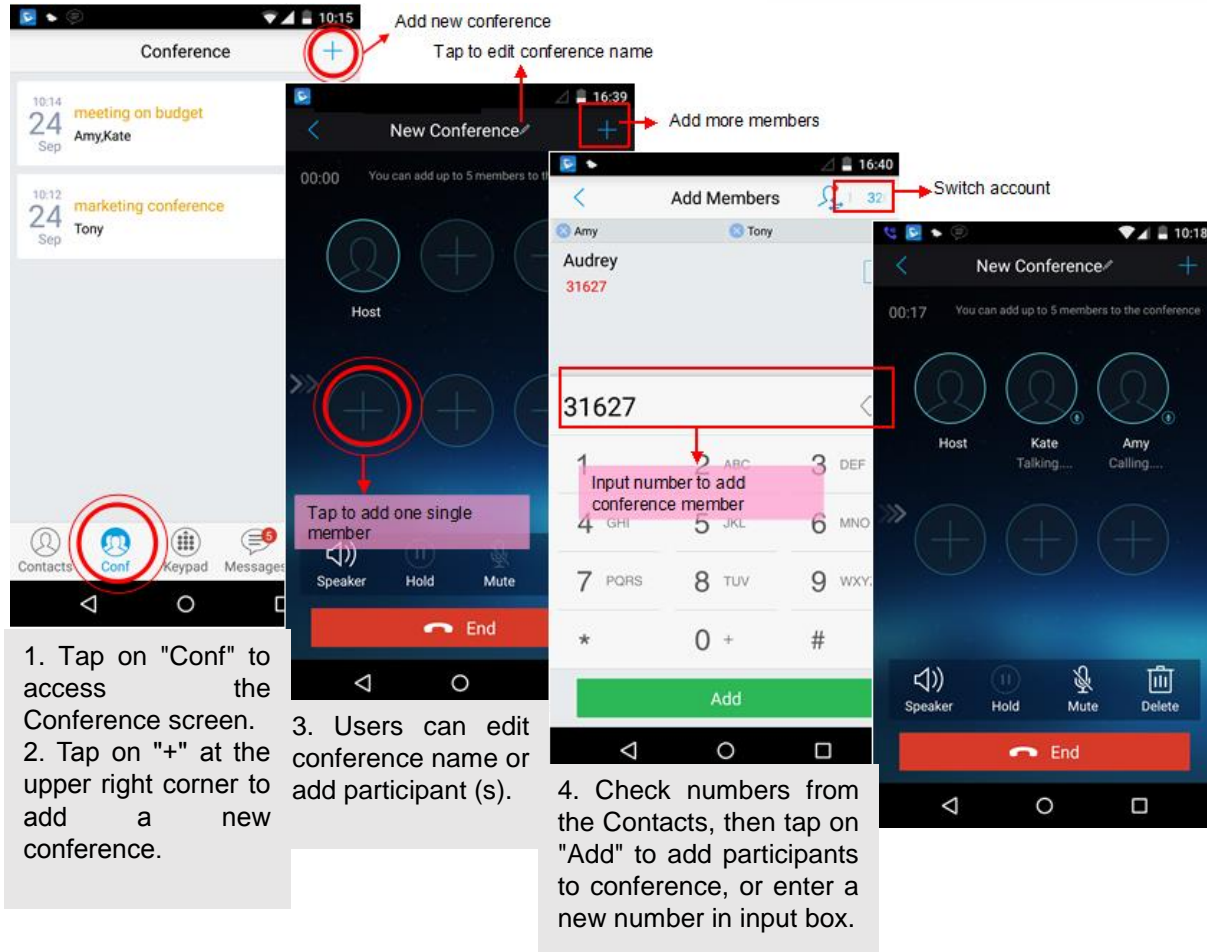


Figure 22: Grandstream Wave Conference - Add New Call to the Conference

Adding a participant to conference via 2 ways:

- Enter phone number in the input box. If this is an existing contact in the Grandstream Wave, it will be shown up. Then, users could add it to the conference.
- If the conference has started and there already exists an existing line, check the line and tap on "Add" to add the line to conference directly.

Initiating Conference

During an active call, tap on "More" and select "Conference" to access conference room. Users could add new participants if there exists an active call.



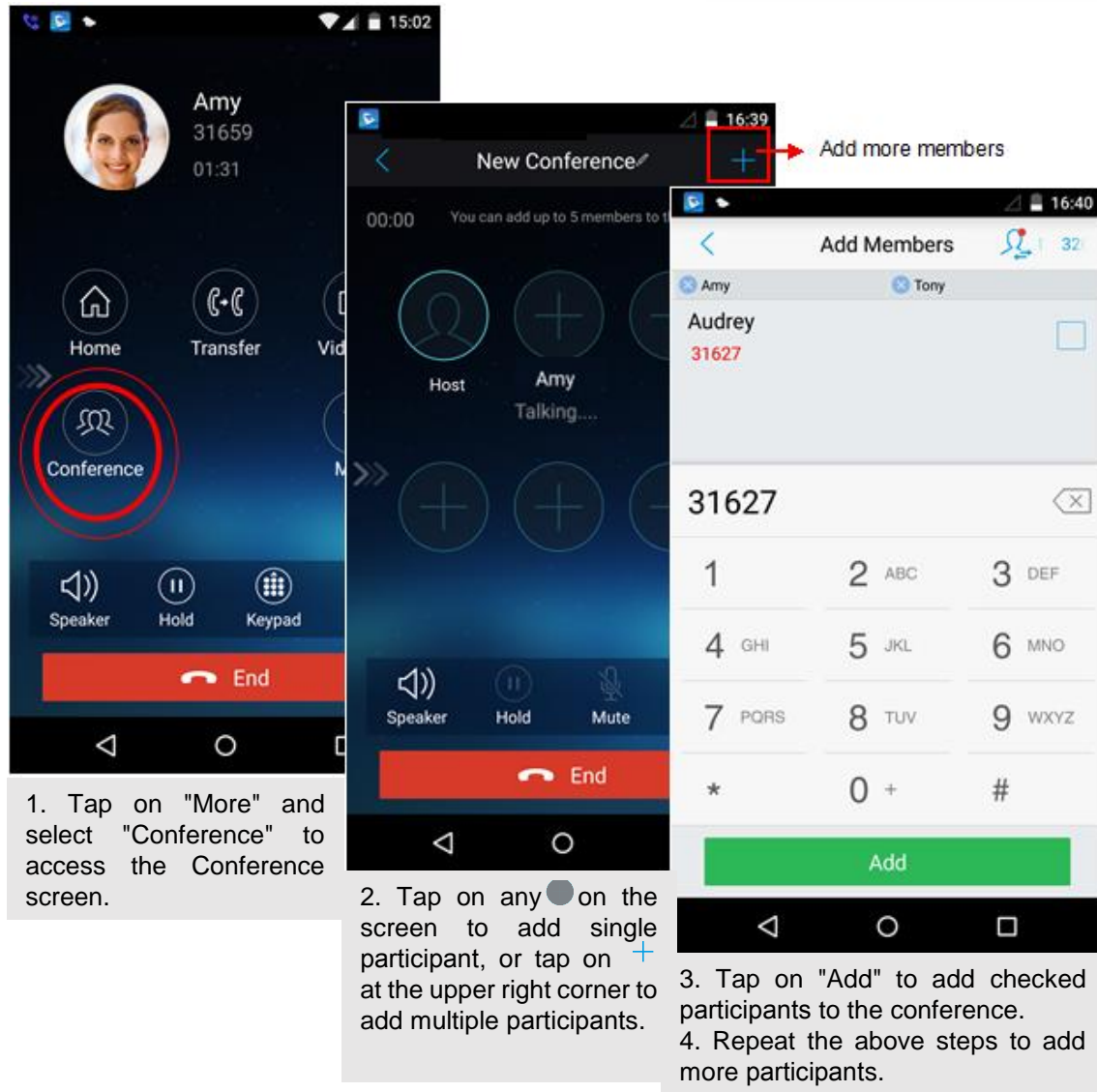




Figure 23: Grandstream Wave Conference - Initiating Conference

While all participants have been in the conference, users could tap on the buttons below to make the corresponding operations.

- **Speaker:** Enable the speaker for the conference.
- **Hold:** Hold the conference.
- **Mute:** Mute the host and each conference participant individually.
- **Delete:** Delete each conference participant.

When the conference participant is disconnected, or the call with the participant is over, tap  on the top right corner of the participant to redial.

Removing Participant from Conference

To remove a participant from the conference, users could press **DELETE** button on phone screen, then tap on  icon at the upper right corner of the participant, and then it will be removed.

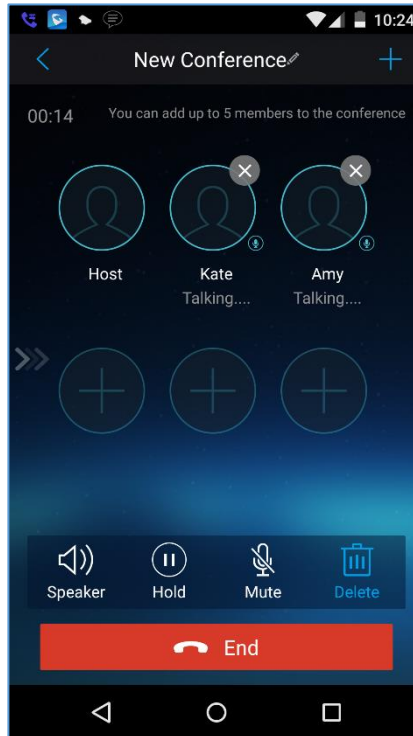




Figure 24: Grandstream Wave Conference - Delete Conference participant

Mute/Unmute Conference

During an active conference, users could press **MUTE** button on phone screen, and then tap on  at the upper right corner of the member to mute the member. The muted member will not be heard by other members, but can hear other members, while it still exists on the conference screen, the muted member in

shown as .

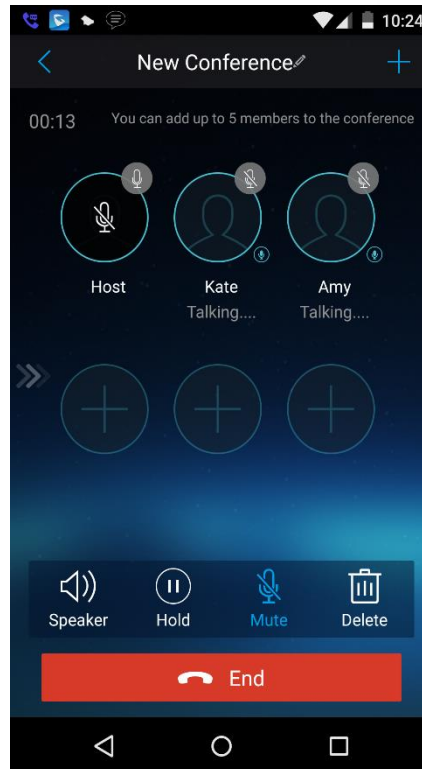


Figure 25: Grandstream Wave Conference - Mute Conference Participant

Hold/Resume Conference

During the conference, users could press the **HOLD** button on phone screen to hold the conference with all participants at any time. If the remote participant presses the **HOLD** button, it will only hold his/her own call from the conference, as shown on the following figure:

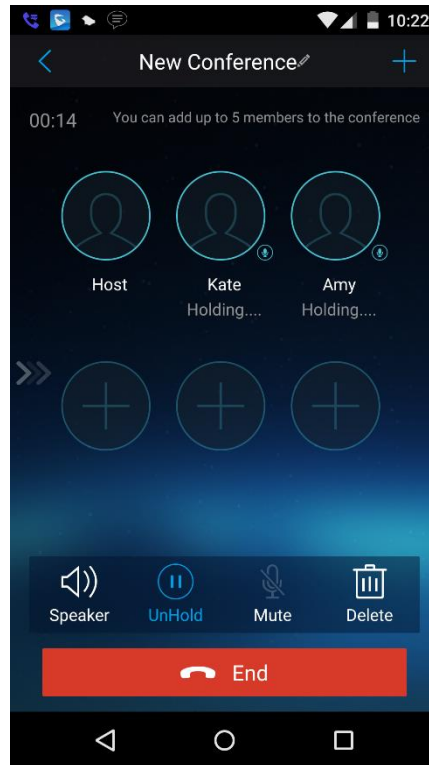



Figure 26: Grandstream Wave Conference - Hold Conference

To end the conference, users could tap  on phone screen to disconnect all the participants from the conference. If the remote participant hangs up the call, it will be disconnected from the conference, but other participants in the conference will stay in the conference.

Direct IP Call

The GS wave support Direct IP Call which allow users to make calls without a SIP proxy, VoIP calls can be made between the phone running the application and the destination phone if:

- The phone running the application and the destination phone have public IP addresses, Or
- Both are on the same LAN/VPN using private or public IP addresses, Or
- Both can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

To make a direct IP call, please follow the steps bellow:

1. Access the dial screen;
2. Input the target IP address. For example, if the target IP address is 192.168.1.60 and the port is 5062 (i.e., 192.168.1.60:5062), input the following: 192*168*1*60#5062
3. Press “Dial” button to initiate an audio call, or press more to access the call menu and choose “Video call” for initiating video calls.



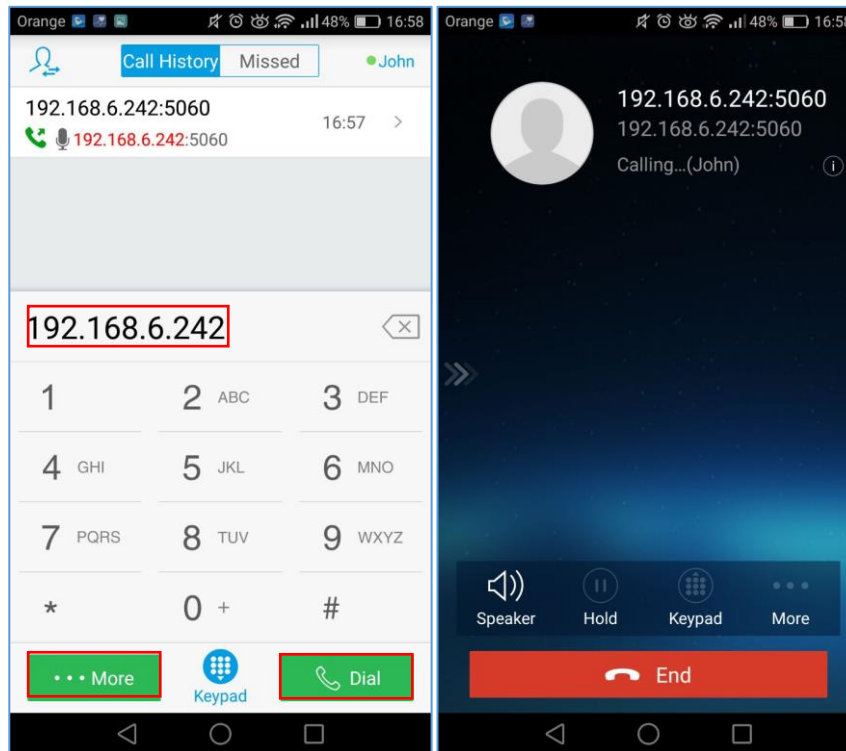


Figure 27: Direct IP Call

Voicemail

When there is a new voicemail, the voicemail icon  will show up on the status bar, and at the same time users could see a new message prompt on the Grandstream Wave messages list.

To configure voicemail UserID, go to **Settings->Account Settings->Edit Account** to fill in the details, as shown on the screenshot below.

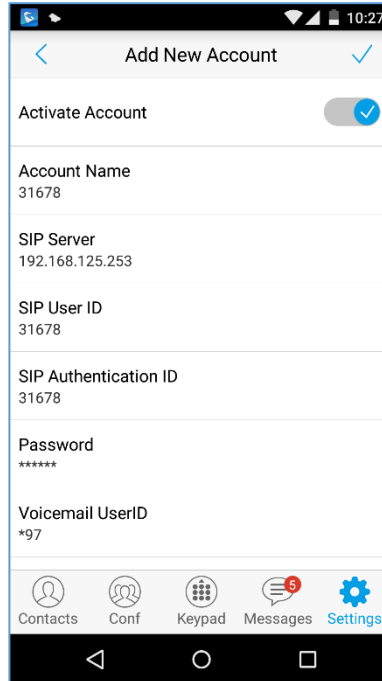


Figure 28: Configure Voicemail UserID

To retrieve the voicemail:

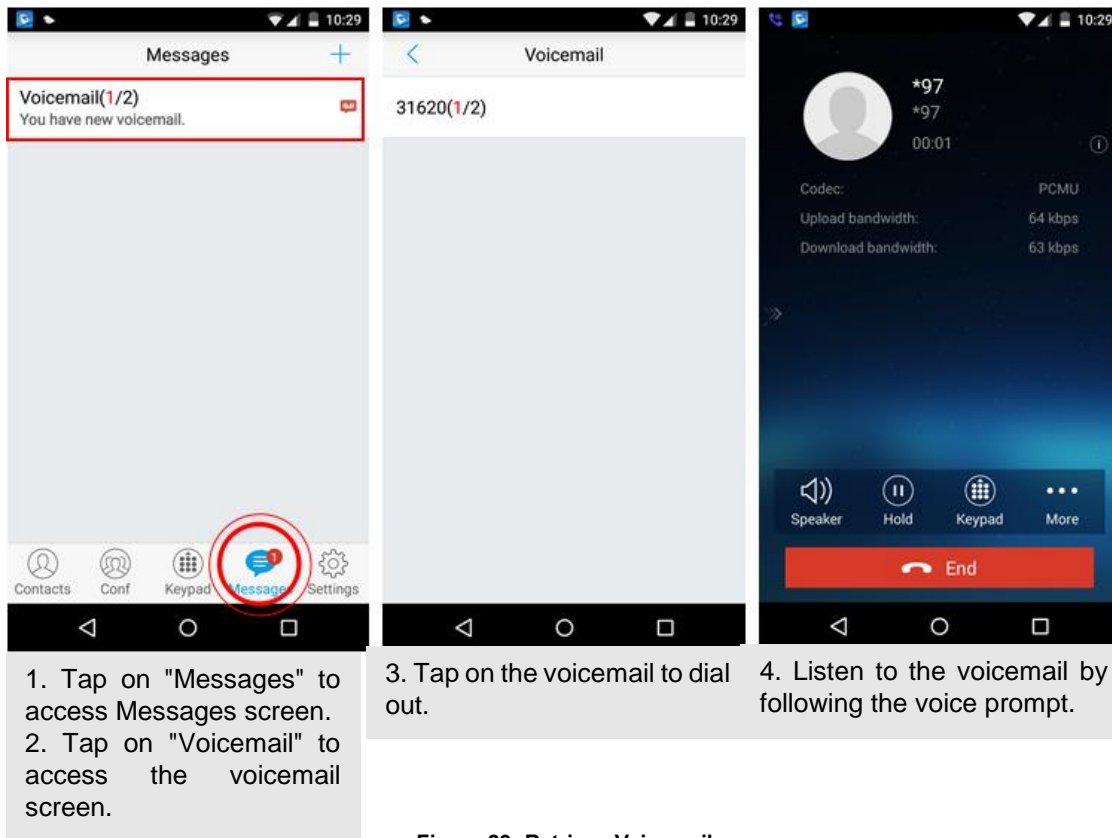



Figure 29: Retrieve Voicemail

Note: To access the voicemail, users will be required to enter the voice mail password, please contact the service provider to obtain the password.

Contacts

Users can manage their phone contacts and SIP contacts in Grandstream Wave Contacts. To access Grandstream Wave Contacts, tap on button  at the bottom of the main screen, as displayed on the following screenshot:

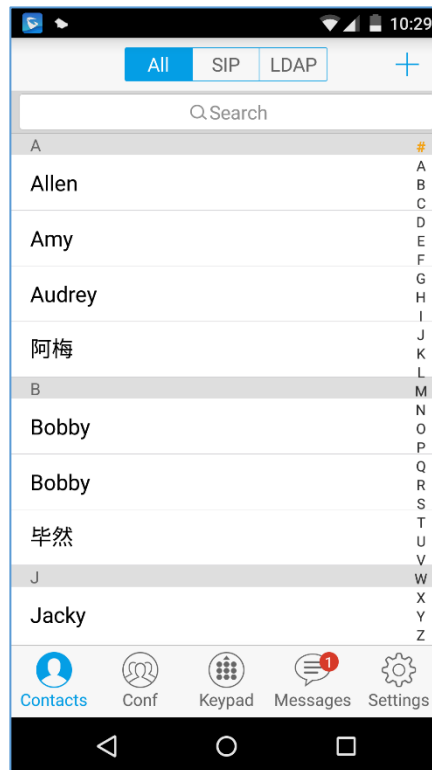


Figure 30: Grandstream Wave Contacts Screen

Add Contacts

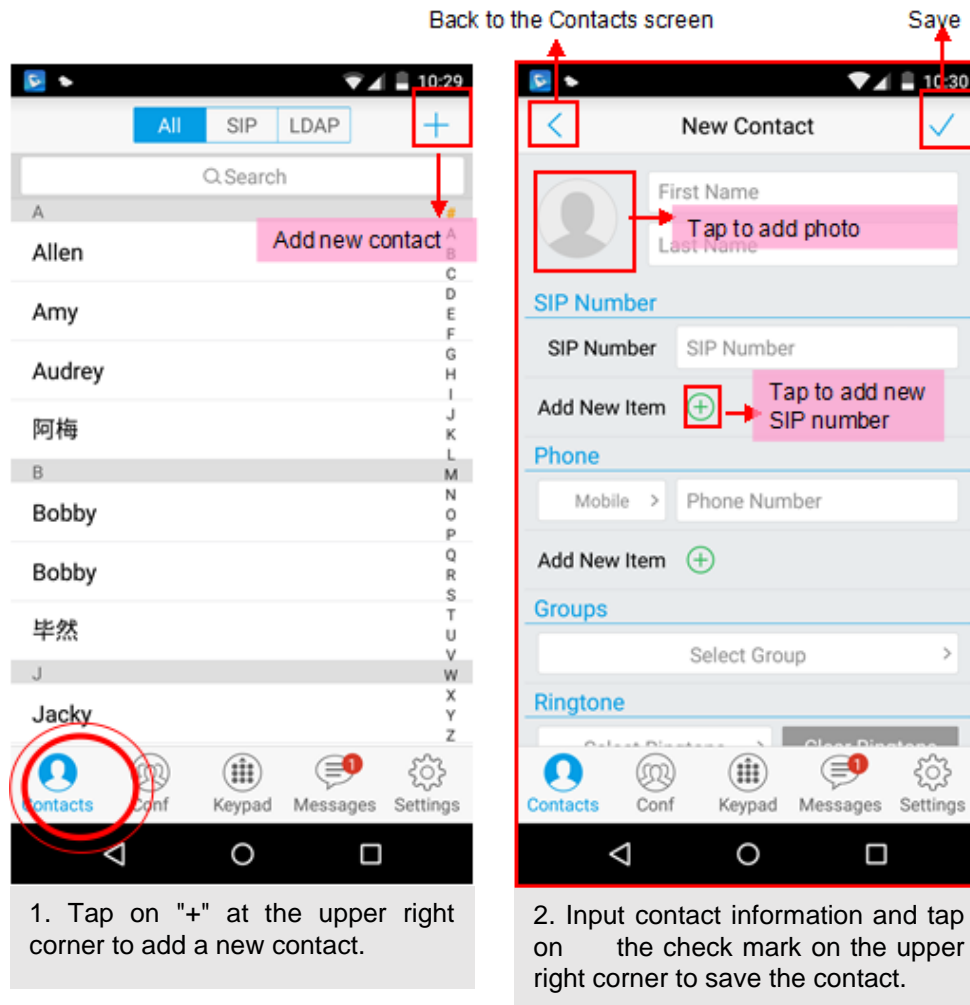


Figure 31: Grandstream Wave Add New Contact

Search Contacts

Tap on the search box on the Contacts screen to access the search screen, as shown on screenshot below:

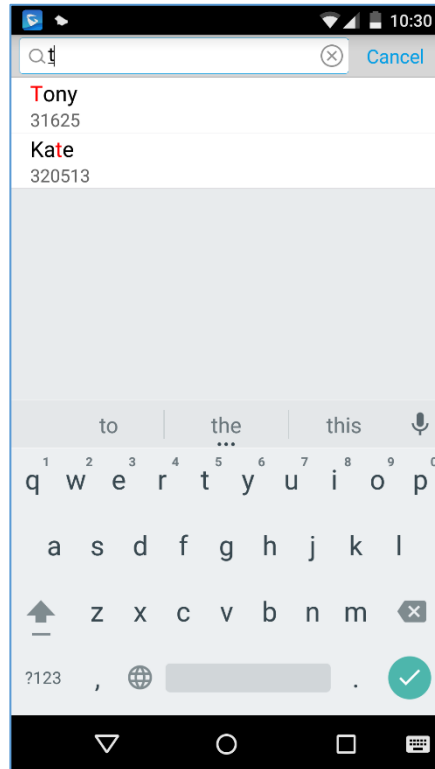





Figure 32: Grandstream Wave Search Contact

Enter the contact name or number to search, the contact will be updated and displayed automatically when entering the initial digits. Tap on the number to view details.

View Contact

Tap on one contact to view details or edit, as displayed on the following screenshot:

-  Dial up the audio call.
-  Access the Messages editing screen.
-  Dial up the video call.

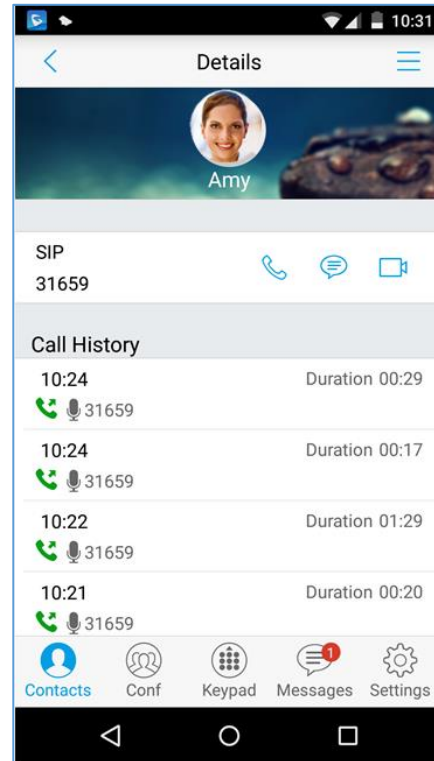



Figure 33: Grandstream Wave View Contact

Edit Contact

Long press the contact on the contact list to bring up the dialog box, tap on **Edit** to access the editing screen;

Or long press the contact to access the details screen, tap on button  at the upper right corner, then select **Edit**.

Delete Contact

Follow one of the following 3 ways to delete contact.

1. Long press one contact on the contact list to bring up the dialog box, select **Delete** to access the editing screen.

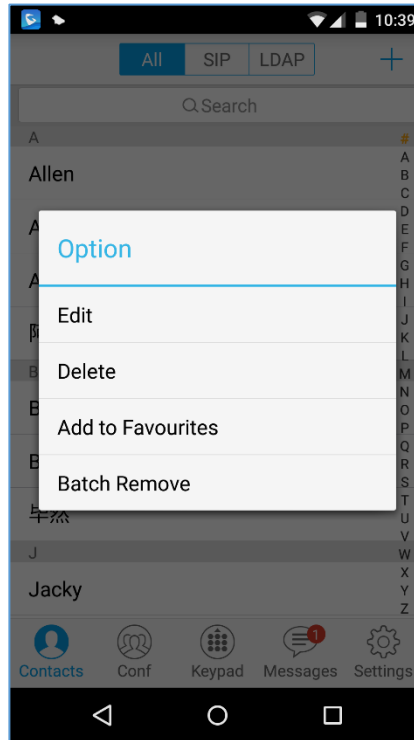



Figure 34: Edit Contact

2. Long press one contact on the contact list to bring up the dialog box, select **Batch Remove** to access batch remove screen, check contacts and tap on  on the upper right corner to delete.

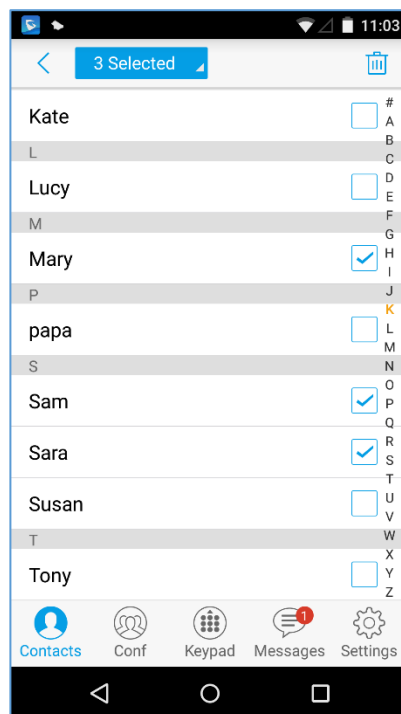



Figure 35: Batch Remove Contact Screen

4. Tap on one contact to access the detail interface, tap on button  at the upper right corner, then select **Delete Contact**, as shown on the following screenshot.

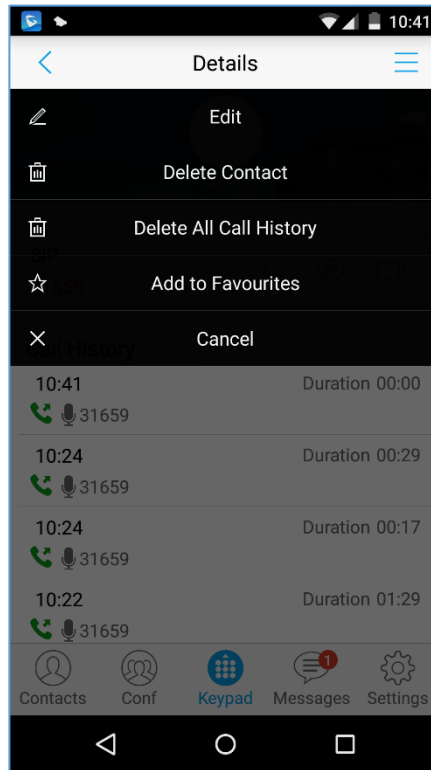


Figure 36: Contact Details Screen

Add Contact to Favorites

To add a contact to favorites, as shown in figure below:

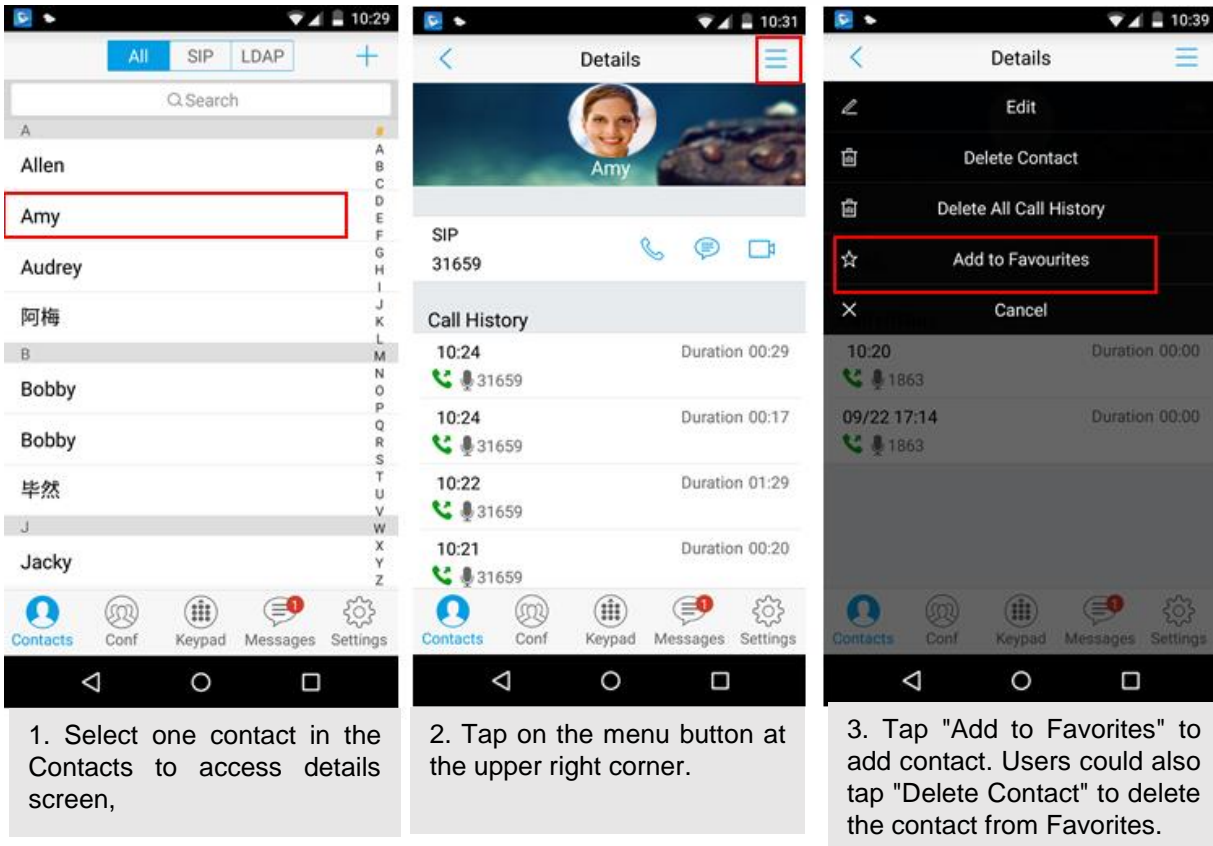


Figure 37: Add Contact to Favorites

Besides the operation mentioned above, users could also select one contact in the **Contacts** and long press it to bring up the dialog box, select **"Add to Favorites"** to add contact to **Favorites**.

Via the similar way, users could delete contacts from Favorites by selecting **"Remove from Favorites"**.

LDAP Contacts

Users could access LDAP screen to search LDAP contacts, view LDAP contact details and add LDAP contact. Please go to Settings->Advanced Settings->LDAP Settings to fill in details.

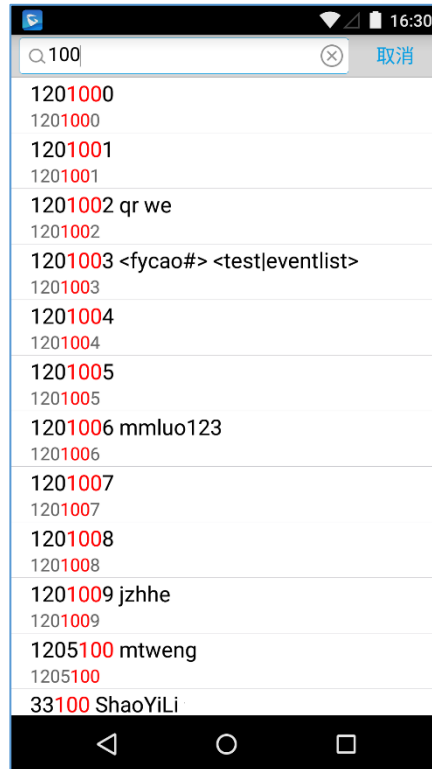








Figure 38: LDAP Contacts Screen - Search Contact

Call History

To view recent call history or view classified call history on Grandstream Wave, tap  on the dial screen or slide down the call history, as shown on screenshot below:

-  Answered calls
-  Dialed calls
-  Missed calls
-  Audio calls
-  Video calls

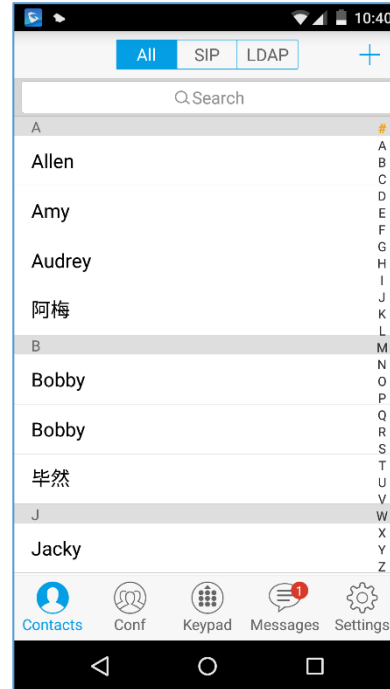



Figure 39: Grandstream Wave Missed Calls Screen

Tap on one call history entry to dial out with the last dial-out account. To access the details for this entry, tap  on the right side of the entry, as shown on following screenshot:

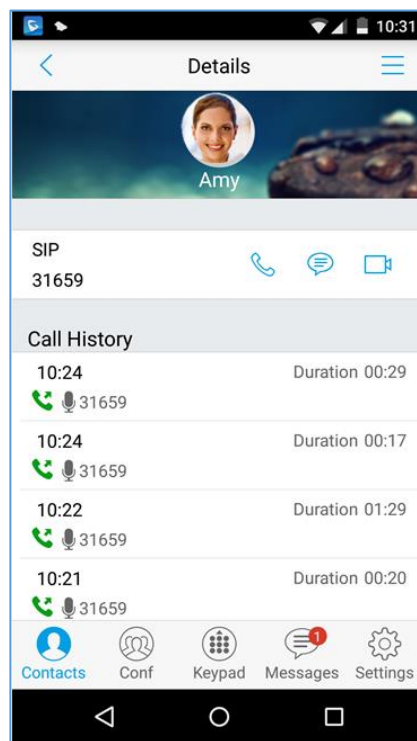



Figure 40: Call History Details Screen

Users could view recent call history of this entry, make calls or send messages to it (not applicable to SIM

card number or anonymous call). Tap on button  at the upper right corner to make operations like **Edit Contact**, **Add to Favorites**, or **Delete All Call History**. If the call is not an existing contact, save it to **Contacts** before making the operations.

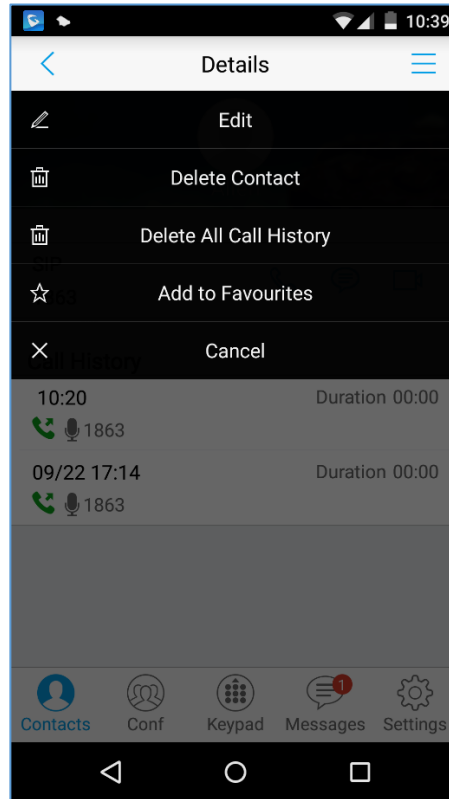



Figure 41: Call History Details Screen - Edit Contact

Messages

Messages function allows users to send/receive messages. Tap on button  to access the Messages screen, as shown on the following screenshot.

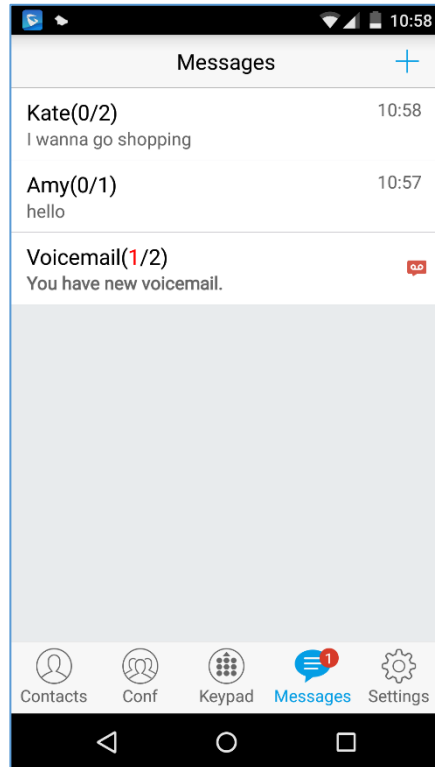


Figure 42: Grandstream Wave Messages Screen

Note: Messages function is not available in all countries and regions. Please contact your service provider for more details.

View Message

The Message screen displays sent & received (draft) messages, the messages are classified by contacts names or numbers while sorted by sent & received time. Tap on one message to check the details, as shown on the following screenshot:

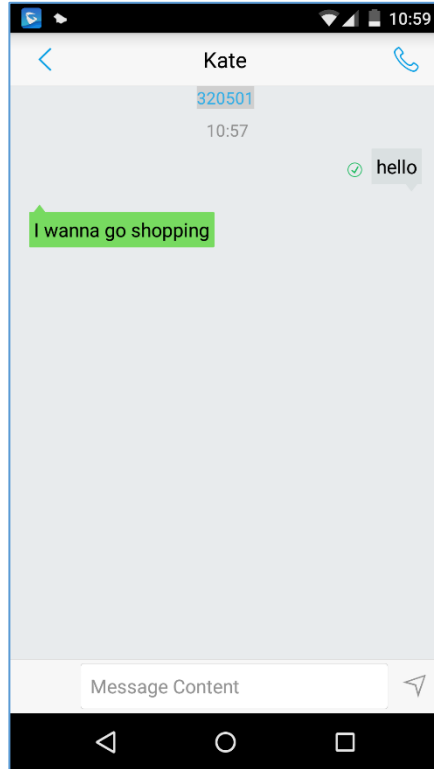


Figure 43: Grandstream Wave Message Details Screen

Create New Message

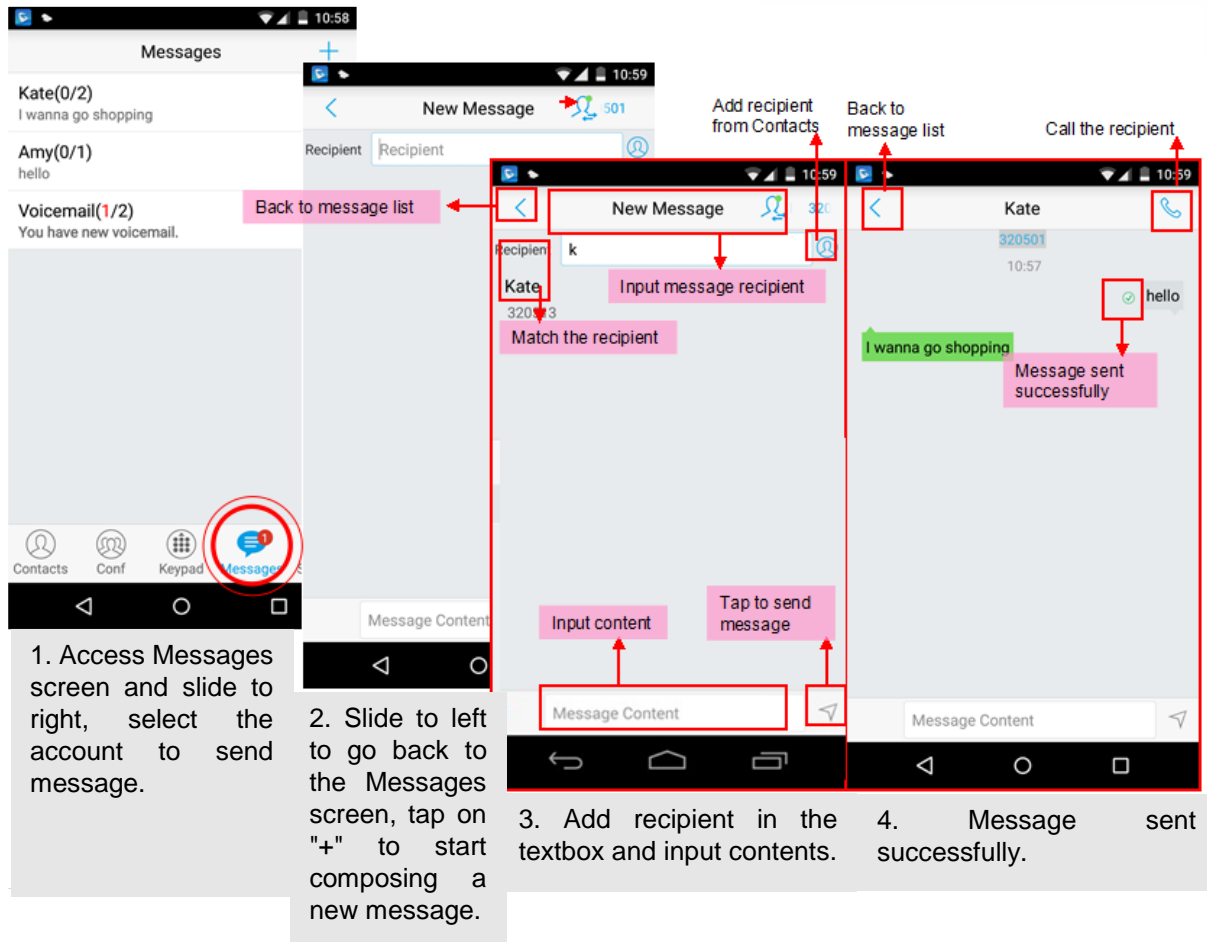




Figure 44: Create New Message

Tap  on the right of the input box to add one contact or more from Grandstream Wave Contacts or input the contact phone number or name in the input box to find the corresponding contact.




If the sent or received message is phone number or Email address, you can tap on the number to dial out directly or tap on the Email address to send an email.

Edit Message

Long press one message on the Messages screen, select "Batch Remove" in the pop up dialog box, and

tap  on the upper right corner to delete all messages with this number. Long press one message content to access the editing screen, users could edit, copy or forward one single message as shown on the figure below.



-  Copy the selected message.
-  Forward the selected message.
-  Delete the selected message.

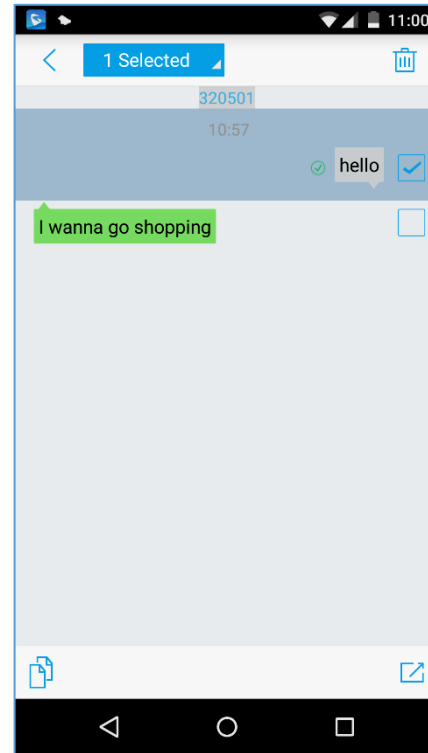


Figure 45: Grandstream Wave Message Screen - Edit Message

SETTINGS

For the first time using Grandstream Wave, go to the **Settings** screen to complete the basic settings, including **Account Settings**, **Advanced Settings**, **Custom Settings**, **About Version**, **Debug**, etc.

Account Settings

Grandstream Wave supports up to 6 independent SIP accounts and 6 lines. Users can make calls after registering the account to the SIP server. Tap on button **+** at the upper right corner of the **Account Settings** screen to add accounts. Users could add account via **Generic Accounts** or **VOIP Providers**. To add generic account, tapping on "**UCM Account (Scan QR Code)**" or "**UCM Account (Select QR Code Image)**", or tap on "**SIP Account**" to add account, as shown on figure below. The way to add VOIP Providers accounts is the same as add generic accounts, just select the providers in the list below and input required information.

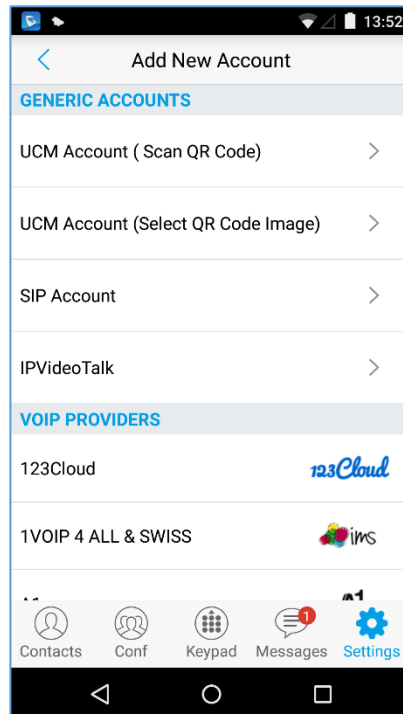


Figure 46: Add New Account Screen

UCM Account (Scan QR Code)

To add account by QR code scan, please follow the steps below as shown in figure below.

1. Tap on "**UCM Account (Scan QR Code)**" to access the scan screen;
2. Scan the QR code containing configuration info sent from the UCM server to the mailbox;
3. Choose whether to overwrite account or add new account, and then the account will be added to the list.

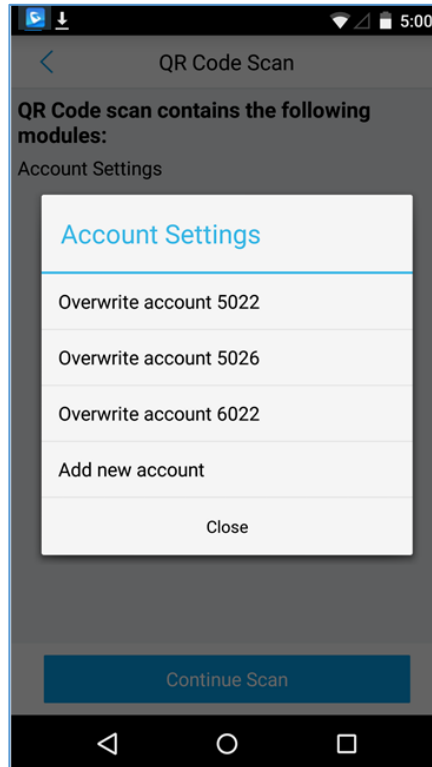


Figure 47: QR Code Scan Screen

Note: Users could add up to 6 accounts, if already reached the limit, you can select overwrite account only.

UCM Account (Select QR Code Image)

1. Tap on "UCM Account (Select QR Code Image)" to access the images screen;
2. Select the QR code image containing configuration info;
3. Choose whether to overwrite account or add account, and then the account will be added to the list.

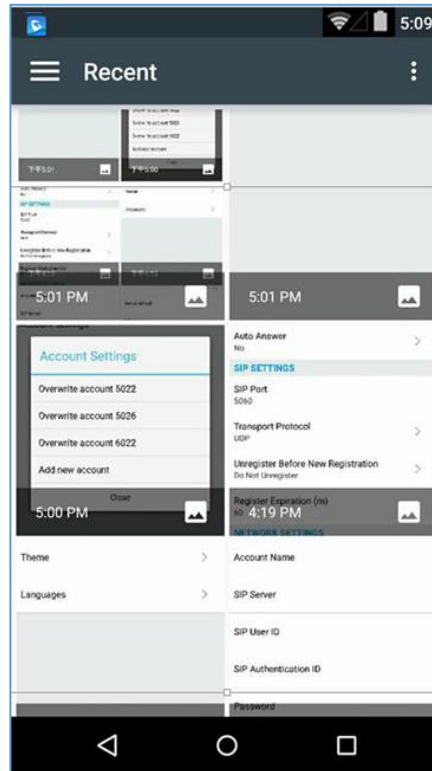





Figure 48: Scan QR Code Image Screen

Note: Users could add up to 6 accounts, if already reached the limit, you can select overwrite account only.

SIP Account

Follow the steps below to add account manually:

1. Tap on "**SIP Account**" to access the **Account Settings** screen, tap on the button on the right of "**Active Account**" to activate the account;
2. Fill in account details and the SIP server address (provided by the service provider);
3. Tap  on the upper right corner to save the configuration and go back to the account settings screen;
4. The following figure 48 shows the accounts are successfully registered, and the account icon is in green . If the account icon is in red , it means the registration failed.

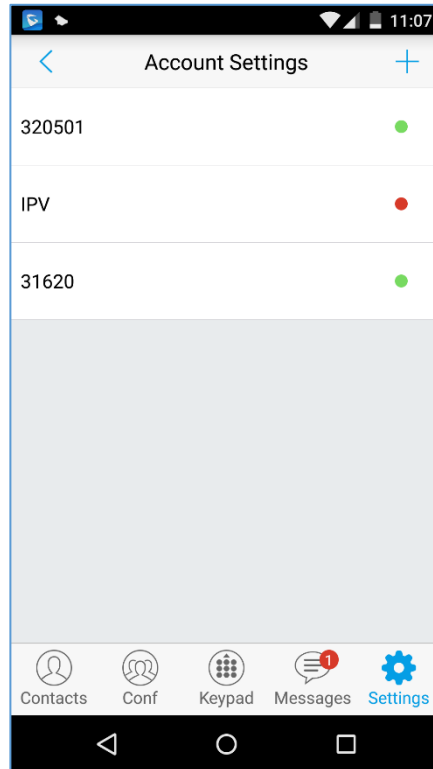


Figure 49: Account Settings Screen - Registration Success

Users could also slide left to delete this account as shown in figure 49.

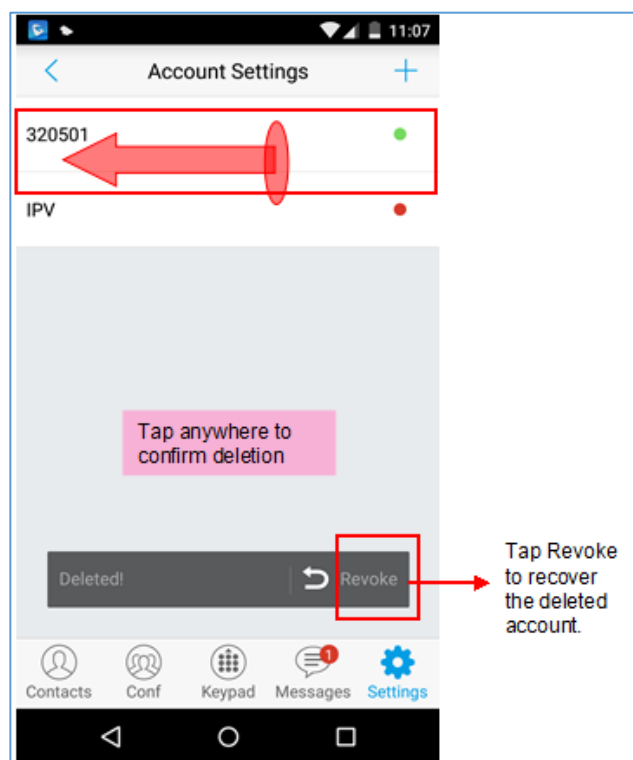


Figure 50: Slide to Delete Account

Table 3: Edit Account Parameters

Activate Account	It is used to define whether to activate account.
Account Name	Defines the name associated to the account to be displayed on the LCD.
SIP Server	Defines the domain name or IP address of your SIP server, provided by your VoIP service provider (ITSP).
SIP User ID	Configures the user account information, provided by your VoIP service provider (ITSP). It's usually in the form of digits similar to phone number or the same as the phone number.
SIP Authentication ID	Configures the SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from the SIP User ID.
Password	Defines the account password required for Grandstream Wave to authenticate with the ITSP (SIP) server before the account can be registered.
VoiceMail UserID	Configure the voicemail user ID to retrieve voicemail by pressing LISTEN button on the message screen. This user ID is usually the VM portal access number. For example, the UCM server voicemail access number is *97.
Display Name	Configures the name to display on LCD when calling, it needs SIP server to support it if this function is enabled.

After configuring the account, users could tap on the existing account for more settings, such as General Settings, Call Settings, SIP Settings, Network Settings and Codec Settings.

Table 4: Account Settings - General Settings Parameters

Activate Account	Activate / deactivate the SIP account.
Set as Default	It is used to set this account as default.
Edit Account	Edits the accounts settings and parameters.
Delete Account	Deletes the current account.

Table 5: Account Settings - Call Settings Parameters

Ringtone	Defines the ringtones played when receiving an incoming call.
DialPlan	Configures to either enable or disable the dial plan.
DialPlan Prefix	Configures the prefix to be added to each dialed number. All numbers use this account will automatically add the prefix. For example, if the prefix is 5, the phone number is 337, thus the dialing number is 5337.
DialPlan Settings	<p>Configures the allowed dial plan for the phone. Default setting is "{ x+ \+x+ *x+ *xx*x+ }". Dial Plan Rules:</p> <ol style="list-style-type: none"> Accepted Digits: 1,2,3,4,5,6,7,8,9,0 , * , # , A,a,B,b,C,c,D,d; Grammar: <ul style="list-style-type: none"> x – any digit from 0-9 X – digits from 0-9, and letters from a-z, A-Z. a) xx+ - at least 2 digit numbers



	<p>b) xx - only 2 digit numbers</p> <p>c) ^ - exclude</p> <p>d) [3-5] - any digit of 3, 4, or 5</p> <p>e) [147] - any digit of 1, 4, or 7</p> <p>f) - replace digit 2 with 011 when dialing</p> <p>g) - the OR operand</p> <p>h) {X123} - match Z123, e123, 5123, ...</p> <p>i) Back slash “\” Character can be used to escape specific letters. e.g. if { \p\al\r\k\+60 } dial plan is configured, park+60 should be able to pass dial plan check. This also can be used to escape Mark and User unreserved characters.</p> <p>Mark = "-" / "_" / "." / "!" / "~" / "*" / "" / "(" / ")"</p> <p>User-unreserved = "&" / "=" / "+" / "\$" / "," / ";" / "?" / "/"</p>
Use # as Dial Key	Configures the "#" key as the "Send" key. If enable, press the "#" key to dial the numbers out immediately; If set to disable, the "#" key will be included in the dialing string.
Call Forward	Tap to access Call Forward Settings screen.
Call Forward Settings	It is used to specify the Call Forward Type from 4 modes: Unconditional, Time Based, Others (Forward When busy and No Answer Forward).
Auto Answer	It is used to configure auto answer mode. If set to "Yes", the phone will automatically turn on the speaker phone to answer incoming calls after a short reminding beep. If set to "Enable Intercom/Paging", it will answer the call based on the SIP info header sent from the server/proxy. Default setting is No.

Table 6: Account Settings - SIP Settings Parameters

Enable Session Expiration	Configures the relevant parameter in "Session Expiration Settings" option below. The default setting is "No".
Session Expiration Settings	Configures the relevant session expiration parameters.
SIP Port	It is used to define the local SIP port used to listen and transmit. If enabled Random Port option on Advanced Settings screen, this option will be unavailable.
Transmission Protocol	It is used to configure the transmission protocol to transmit SIP info. Users could choose TCP/UDP/ TLS. The default is "UDP".
Unregister Before New Registration	If set to "Register All", the SIP contact header will use "*" to clear all SIP user's registration information. If set to "Do Not Register", the phone will not clear the current SIP user's info. The default is "Unregister Single", that means do not cancel the SIP user's registration information.
Register Expiration (m)	Specifies the frequency (in minutes) in which the phone refreshes its registration with the specified registrar. The minimum value is 1 minute while the maximum is 64800 minutes (about 45 days).The default value is 60 minutes (1 hour).



Only Accept SIP Requests from Known Servers	Once enabled, only accept SIP request sent from known servers, the default setting is "Disable".
Check SIP User ID for Incoming Invite	Checks SIP User ID in the Request URI of incoming INVITE; if it doesn't match the base SIP User ID, the call will be rejected. Direct IP calling will also be disabled. Default is No.

Table 7: Session Expiration Settings Parameters

Session Expiration (s)	The SIP Session Timer extension that enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re-INVITE). If there is no refresh of an UPDATE or re-INVITE message, the session will be terminated once the session interval expires. Session Expiration is the time (in seconds) where the session is considered timed out, provided no successful session refresh transaction occurs beforehand.
Min-SE (s)	The minimum session expiration (in seconds). The default value is 90 seconds.
UAC Specify Refresher	As a caller, select UAC to use the phone as the refresher; or select UAS to use the caller or proxy server as the refresher. If set to "Omit", that means do not specify the refresh object. The default setting is "Omit".
UAS Specify Refresher	As a callee, select UAC to use caller or proxy server as the refresher; or select UAS to use the phone as the refresher.
Force INVITE	The Session Timer can be refreshed using the INVITE method or the UPDATE method. Select "Yes" to use the INVITE method to refresh the session timer.
Caller Request Timer	If set to "Yes" and the remote party supports session timers, the phone will use a session timer when it makes outbound calls. The default setting is "No".
Callee Request Timer	If set to "Yes" and the remote party supports session timers, the phone will use a session timer when it receives inbound calls. The default setting is "No".
Force Timer	If Force Timer is set to "Yes", the phone will use the session timer even if the remote party does not support this feature. If Force Timer is set to "No", the phone will enable the session timer only when the remote party supports this feature. To turn off the session timer, set Caller Request Timer, Callee Request Time and Force Timer all to "No".

Table 8: Account Settings - Network Settings Parameters

Proxy-Require	A SIP Extension to notify the SIP server that the phone is behind a NAT/Firewall. Do not configure this parameter unless this feature is supported on the SIP server.
Outbound Proxy	Configures the IP address or Domain name of the Primary Outbound Proxy, Media Gateway, or Session Border Controller.



Secondary Outbound Proxy	Configures the IP address or Domain name of the Secondary Outbound Proxy, Media Gateway, or Session Border Controller. Secondary outbound proxy will be used when the primary outbound proxy fails.
DNS Mode	Controls how the search appliance looks up IP addresses for hostnames. There are three modes: A Record, SRV, NAPTR/SRV. The default setting is "A Record". If the user wishes to locate the server by DNS SRV, the user may select "SRV" or "NAPTR/SRV".
NAT Traversal	<p>Enables or disables the NAT traversal mechanism. The default setting is "Keep-alive".</p> <ul style="list-style-type: none"> • If set to "STUN" and STUN server is configured, the phone will route according to the STUN server; If NAT type is Full Cone, Address-Restricted Cone or Port-Restricted Cone, the phone will try to use public IP addresses and port number in all the SIP&SDP messages. • The phone will send empty SDP packet to the SIP server periodically to keep the NAT port open if it is configured to be "Keep-alive". • Configure this to be "NAT NO" if an outbound proxy is used. • Configure this to be "UPnP" if the router supports UPnP. • If set to "Auto", the phone will try to use all traversal methods mentioned above until find the available one.

Table 9:Account Settings - Codec Settings Parameters

DTMF	<p>Users can choose different ringtones.</p> <p>Specifies the mechanism to transmit DTMF digits. There are 3 supported modes:</p> <ul style="list-style-type: none"> • In audio: which means DTMF is combined in the audio signal (not very reliable with low-bit-rate codecs); • RTP (RFC2833): permits to specify DTMF with RTP packet. Users could know the packet is DTMF in the RTP header as well as the type of DTMF; • SIP INFO: uses SIP INFO to carry DTMF. The defect of this mode is that it's easily to cause desynchronized of DTMF and media packet for the reason the SIP and RTP are transmitted respectively. <p>Default settings is via "RFC2833".</p>
Preferred Vocoder	<p>Selects which codecs will be used on Wi-Fi, 2G, 3G and 4G</p> <p>Multiple vocoder types are supported on the phone (PCMU, PCMA, OPUS, G722, G726_32, iLBC, G729 and GSM). The vocoders in the list is a higher preference. Users can configure vocoders in a preference list that is included with the same preference order in SDP message.</p>
H.264 Image Size	<p>Configures different image size (720P, VGA, CIF, QVGA and QCIF) in different network environment. For Wi-Fi network, the default setting is VGA; For 2G/3G/4G mobile network, the default setting is QVGA.</p>
Video Bit Rate	<p>Configures different video bite rate in different network environment. It is recommended to increase the bit rate if the bandwidth is sufficient, and the video quality will be reduced if the bandwidth is not enough. For Wi-Fi</p>



	network, the default setting is 512kbps; For 2G/3G/4G network, the default setting is 192kbps.
SDP Bandwidth Attribute	<p>Select the SDP bandwidth attribute from "Standard", "Media Level", "Session Level" or "None".</p> <ul style="list-style-type: none"> • Standard: Use AS at the session level and TIAS at the media level. • Media Level: Use AS at the media level. This is the default setting. • Session Level: Use AS at the session level. • None: Don not change the format. Please do not change the format or it may cause decode failure if unclear about what format the server supports.
H.264 Payload Type	Configures the H.264 codec payload type. The valid range is from 96 to 127. The default value is 99.
SRTP Mode	Configures the SRTP Mode, when set to "Enable and Force" it will enable and force to use SRTP and when set to: "Enable but Not Force", it will enable but not force to use SRTP. The default setting is "Disable".
Enable SRTP Key Life Time	It is used to configure whether to enable SRTP key life time. If enabled, the RTP packets received and sent during the call cannot exceed 231. The default setting is "Yes".

Advanced Settings


Advanced Settings include General Settings, Call Settings, Network Settings and Additional Settings.

General Settings

Table 10: Advanced Settings - General Settings Parameters

Edit Before Dial	Configures whether to edit number before dial. If set to "No", tap on the contact or one call history entry on call screen to dial out with the last dial-out account directly. If set to "Yes", when tap on the contact or one call history entry on call screen, the phone will automatically fill in the corresponding number to the input box, users could edit the number before dial out.
Default Account Registration Notification	Defines whether to enable registration notifications for default account. If enabled, users will see the notifications in the status bar once the default account status is changed.
Vibrate When Ringing	Configures whether to vibrate when ringing. It is only applicable to the incoming calls for the GS Wave. The phone settings priority is higher than this option. When set the phone to silent mode, the phone will not vibrate when ringing even set this option to "Yes".
Start on Boot	Configures whether auto start GS Wave when starting up.

Call Settings

Call Settings is mainly used for DND settings and show bandwidth info. When DND is on, the phone will reject calls automatically and the status bar will display the icon . Tap on "DND Settings" to configure



as shown on the figures below:

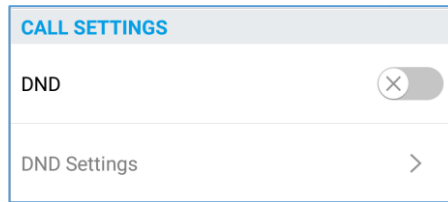


Figure 51: Call Settings Screen

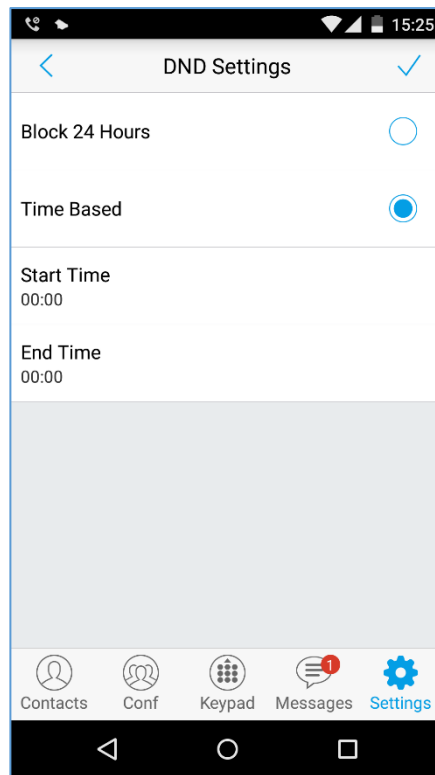


Figure 52: DND Settings Screen

Note:

- When Grandstream Wave is in an active call, turning on/off DND will not affect the current active call. It will take effect on the next incoming call.
- When the DND is on, users could view all the incoming calls in missed call history.

Audio Settings

Noise Reduction Level	Grandstream Wave provides users with multiple noise reduction levels. Choose the level according to the specific environment.
Speaker Gain	Adjusts the speaker gain, available settings are (+6db, 0db and +6db). Default settings is: 0db.
Microphone Gain	Adjusts the microphone gain, available settings are (+6db, 0db and +6db). Default settings is: 0db.



Network Settings

Grandstream Wave supports data communication via **2G/3G/4G** and **WiFi**, you can also configure QoS settings.

Table 11: Advanced Settings - Network Settings Parameters

Network	Users could use Grandstream Wave in 2G/3G/4G/WiFi.
Only Use This WiFi to Register Account	In the WiFi environment, only register account with this specified connected WiFi.
STUN Server Settings	The IP address or URL of the STUN server. Only non-symmetric NAT routers work with STUN.
QoS Settings	Configures layer 3 SIP QoS and layer 3 audio QoS. The valid range is 0-63. The default setting is 48.
Random Port	When set to "Yes", this parameter will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple phones are behind the same full cone NAT. The default setting is "Yes".


Additional Settings


Table 12: Advanced Settings - Additional Settings Parameters


Validate Server Certificate	Validates the servers certificate chain for the server's certificate when enabled. Default settings is "Disabled".	
Config Server Path	Defines IP address or URL for the server. Grandstream Wave could obtain the configurations from the server path automatically.	
Export Configuration	Exports the configuration files to the path on the phone: /sdcard/ GS Wave/ config.	
GDS Settings	GDS Name	Specifies the name to identify the GDS3710. Note: The GS Wave support up 10 GDS items
	GDS Number	Specifies the GDS number which is the SIP user ID configured on GDS3710.
	GDS Password	Determines the GDS password which should match the one configured on "Remote PIN to Open the Door" field on GDS3710 settings.
LDAP Settings	<p>Tap to access the LDAP Settings screen to set up features. Users could set by QR Code Scan, Select QR Code Image or Manual Settings.</p> <ul style="list-style-type: none"> Tap on "Scan QR Code" to access QR code scan screen, scan the QR code which contains LDAP information sent by the UCM server to configure LDAP settings. Tap on "Select QR Code Image" to access screen with QR code image, select the image which contains LDAP information to configure LDAP settings. 	



	<ul style="list-style-type: none"> Tap on "Manual Settings" to access screen as shown in Figure 54. (Please refer to Table 13 for the description of the manual settings)
BLF	Configures whether to enable contacts status detection in BLF list. The status will be shown under SIP option of the Contacts.
BLF List	Add the contacts to monitor its BLF status online in BLF list. The status will be shown in SIP option of the Contacts.

 Offline. Unable to detect the contact status, or the contact's registration is failed.

 Busy. The active line of the contact is busy.

 Online. The call line is in idle.

Note: The BLF function requires that the server supports BLF feature.

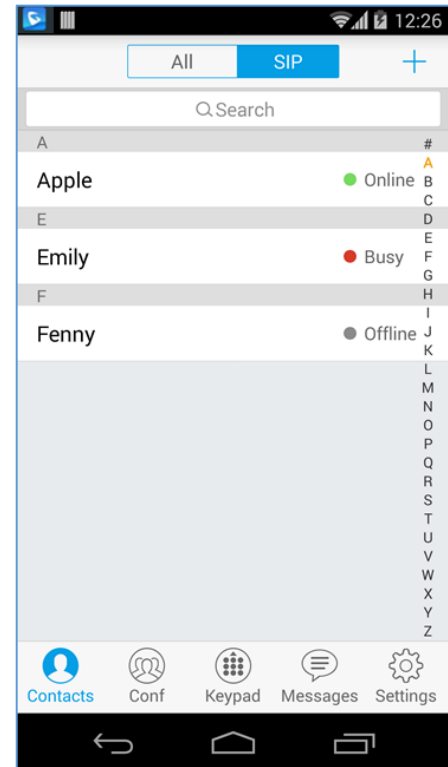


Figure 53: BLF List Screen

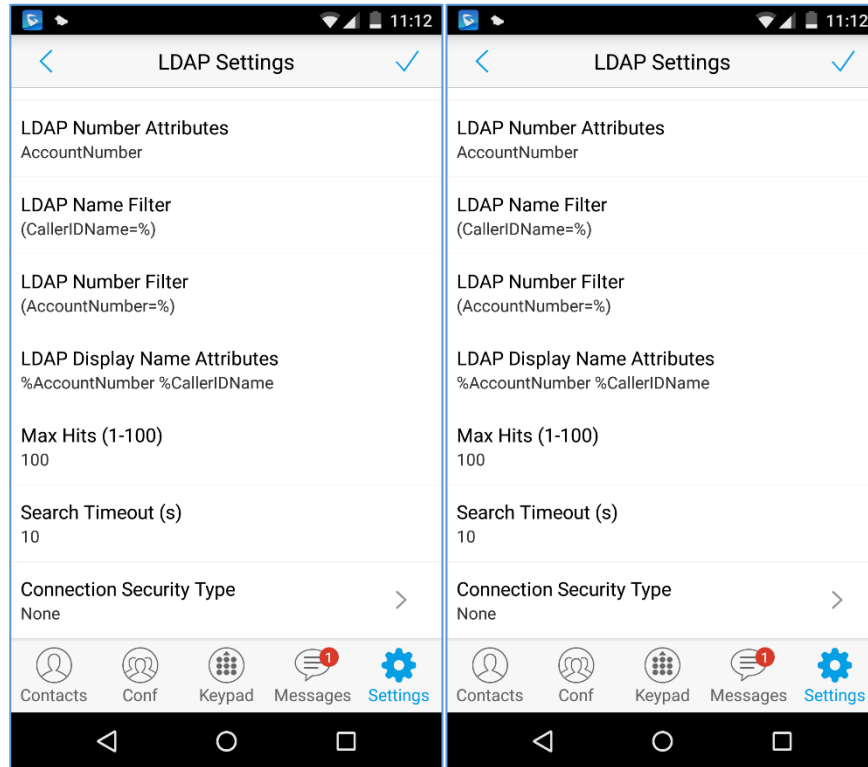


Figure 54: LDAP Settings Screen

Table 13: LDAP Settings Parameters

LDAP Lookup When Dialing	Defines whether to search LDAP when dialing. Default setting is "Yes".
LDAP Lookup When Incoming Call	Defines to search LDAP when there is an incoming call. The default setting is "Yes".
Server Address	Configures the LDAP server URL or IP address.
Port	Configures the LDAP server port. The default value is 389.
Base DN	Configures the base DN which is the root directory of the LDAP server, it means under which directory to search contact.
Username (Binding DN)	Specifies the username to access the LDAP server.
Password	Fill in the password to access the LDAP server.
LADP Name Attributes	Specifies the "name" attributes of each record which are returned in the LDAP search result. <u>Example:</u> gn cn sn description
LADP Number Attributes	Specifies the "number" attributes of each record which are returned in the LDAP search result.

	<p><u>Example:</u></p> <p>telephoneNumber</p> <p>telephoneNumber Mobile</p>
LDAP Name Filter	<p>Configures the filter used for name lookups.</p> <p><u>Examples:</u></p> <p>((cn=%)(sn=%)) returns all records which has the "cn" or "sn" field containing with the entered filter value;</p> <p>(!(sn=%)) returns all the records which do not have the "sn" field containing with the entered filter value;</p> <p>(&(cn=%) (telephoneNumber=*)) returns all the records with the "cn" field containing with the entered filter value and "telephoneNumber" field set.</p>
LDAP Number Filter	<p>Configures the filter used for number lookups.</p> <p><u>Examples:</u></p> <p>((telephoneNumber=%)(Mobile=%) returns all records which has the "telephoneNumber" or "Mobile" field starting with the entered filter value;</p> <p>(&(telephoneNumber=%) (cn=*)) returns all the records with the "telephoneNumber" field starting with the entered filter value and "cn" field set.</p>
LDAP Display Name Attributes	<p>Configures the entry information to be shown on phone's LCD. Up to 3 fields can be displayed. Example: %cn %sn %telephoneNumber</p>
Max Hits (1-100)	<p>The maximum contacts results return to the LDAP server. If set to "1", The server will return all query results. The default setting is 100.</p>
Search Timeout (s)	<p>Specifies the interval (in seconds) for the server to process the request and client waits for server to return. The default setting is 10 seconds.</p>
Connection Security Type	<p>Configures LDAP connection security mode, users could choose None or SSL.</p>

Custom Settings

Users could configure Color and Languages on Custom Settings screen.

Table 14: Custom Settings Parameters

Color	Configures the color of icon, tab bar, navigation bar, send & receive message backgrounds.
Theme	Defines the theme to use for the application. Three themes are supported: white, black and blue. Default theme is: white
Languages	Selects the language on the application to be displayed.



About Version

About version page permits users to verify the version of the application running as shown on the following screenshot, access the Grandstream Privacy Statement web page when clicking on “Privacy agreement” or share this application via the **Bluetooth, Gmail, Google**, etc.

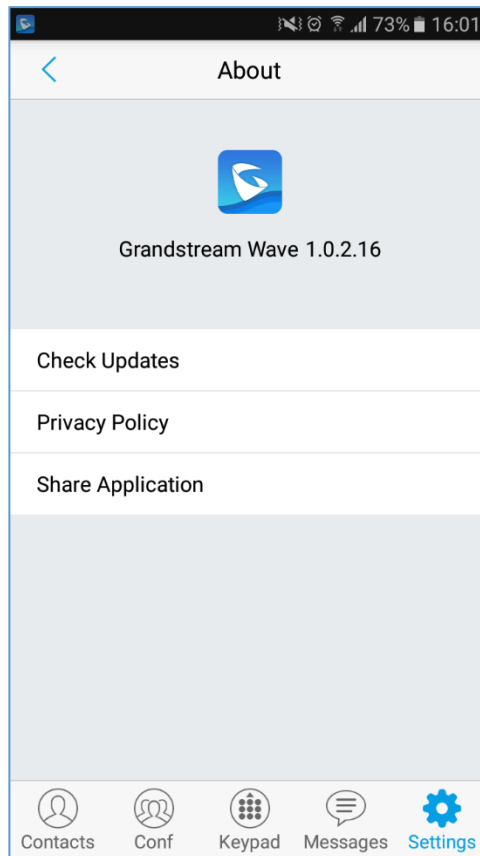


Figure 55: About Page

Debug

Users could trace SIP message with Debug function when coming across software problems.

Table 15: Debug Settings Parameters

Report Bugs	When unexpected crash or accidents occurs, upload the relevant logs to the server, the default setting is "Yes". This function can help users to monitor service condition and locate exception logs.
SIP Message Trace	Save the SIP message on the phone for users to check.
SIP Message Retention Period	It is used to configure the retention period of the SIP message on the phone.

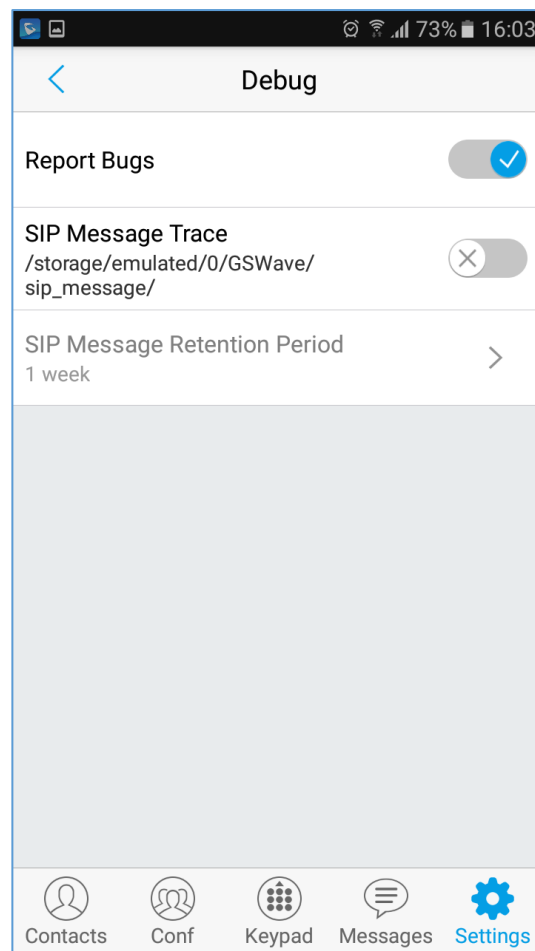


Figure 56: Debug

Exit

Users can easily quit the application by pressing the **Exit** button, and confirming their choice as displayed on the following screenshots. Once confirming, the application will be closed and then they will not receive any calls and messages.

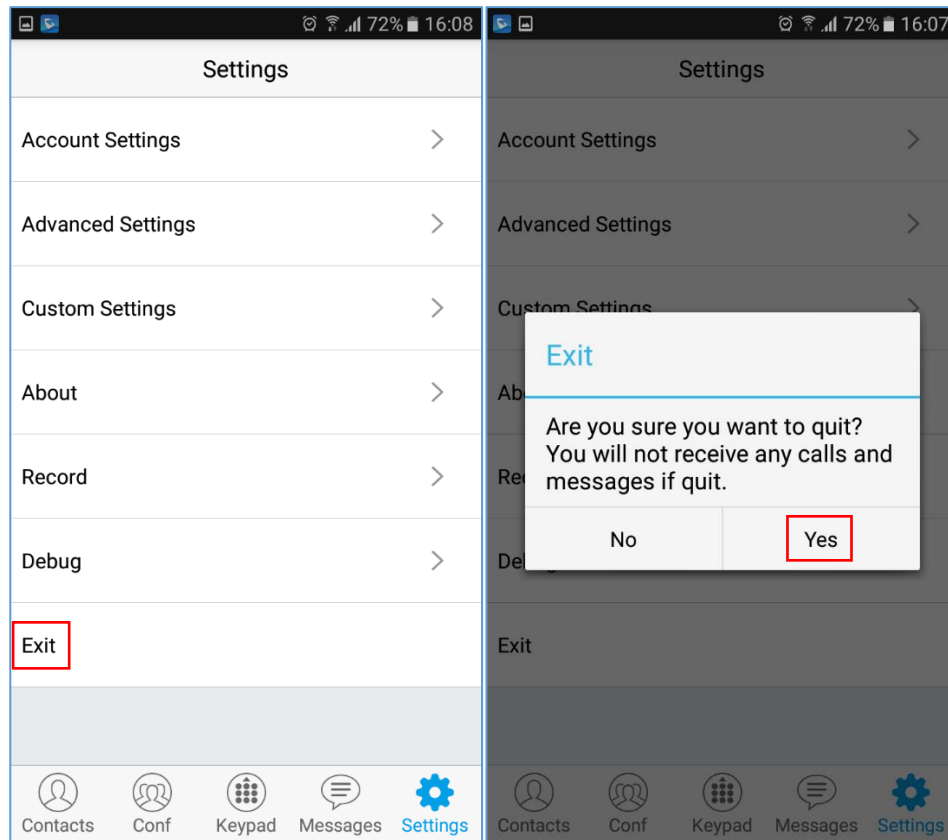



Figure 57: Exit the Application

CONNECTING GS WAVE WITH GDS3710 DOOR SYSTEM

The GS Wave can interact with the GDS3710 Door System to allow users to open door, initiate call to the GDS3710 and get a real-time audio / video stream.

For more details about GDS3710, please refer to [GDS3710 Web page](#).

The following steps illustrate how to configure GDS3710 settings on the GS Wave assuming that the GS Wave and GDS SIP extension are correctly registered.

1. Configure your SIP extension on the GS Wave using the same SIP server on which the GDS extension is registered.
2. Access Settings -> Advanced Settings -> Additional Settings -> GDS Settings.
3. Click on “Add New Item” and configure your GDS settings:
 - **GDS Name:** Specifies the name to identify the GDS3710.
 - **GDS Number:** Specifies the SIP extension number of the GDS3710
 - **GDS Password:** Configure the remote PIN code used on the GDS available under GDS3710 web GUI -> Door System Settings -> Basic Settings -> Remote PIN to open the door.
4. Press on  to save the new GDS settings and add the new item as displayed below.

Note: The GS Wave supports up to 10 GDS3710 door systems.

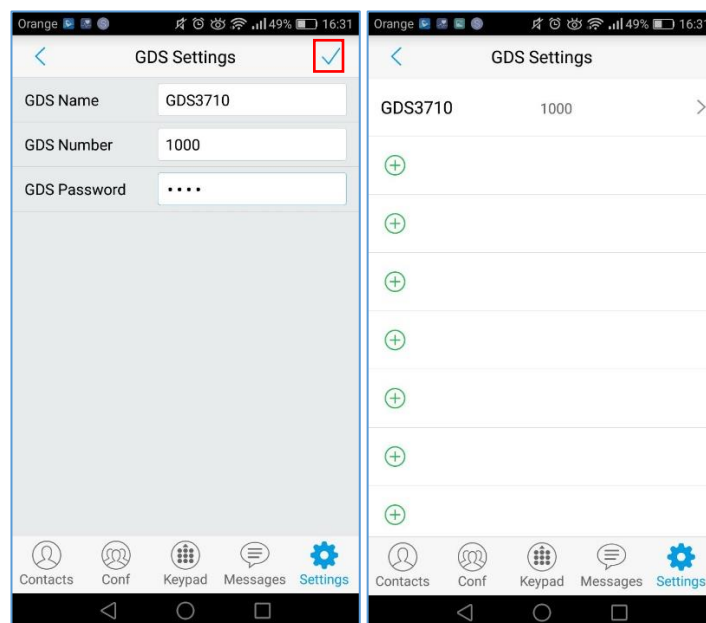


Figure 58: Configuring GDS3710 Settings on GS Wave

5. Access to GDS Web GUI to configure the number called when the doorbell button is pressed under System Settings -> Basic settings -> “**Number Called When Door Bell Pressed**” as displayed on the following screenshot, and configure the remote PIN code if available.

Door System Settings

Unlocking Latency(s)	<input type="text" value="0"/>
Unlock Hold Time(s)	<input type="text" value="5"/>
Swipe Card Intervals(ms)	<input type="text" value="300"/>
Capture Image on Unlock	<input checked="" type="checkbox"/>

Call Mode	SIP Number ▼
Doorbell Mode	Call Doorbell Number ▼
Number Called When Door Bell Pressed	<input type="text" value="1001"/>
Remote PIN to Open the Door	<input type="text" value="....."/>
Local PIN Type	Unified PIN ▼
Local PIN to Open the Door	<input type="text"/>
Enable Guest PIN	<input type="checkbox"/>

Card Issuing State Setting

Enable Card Issuing Mode	<input type="checkbox"/>
--------------------------	--------------------------

Figure 59: GDS Settings.

6. Save and apply the new settings and then when someone presses the doorbell button on the GDS3710, it will initiate a video call to GS Wave extension, once accepting the incoming call, the users can open the door by pressing **“OpenDoor”** button as displayed on the following screenshots.

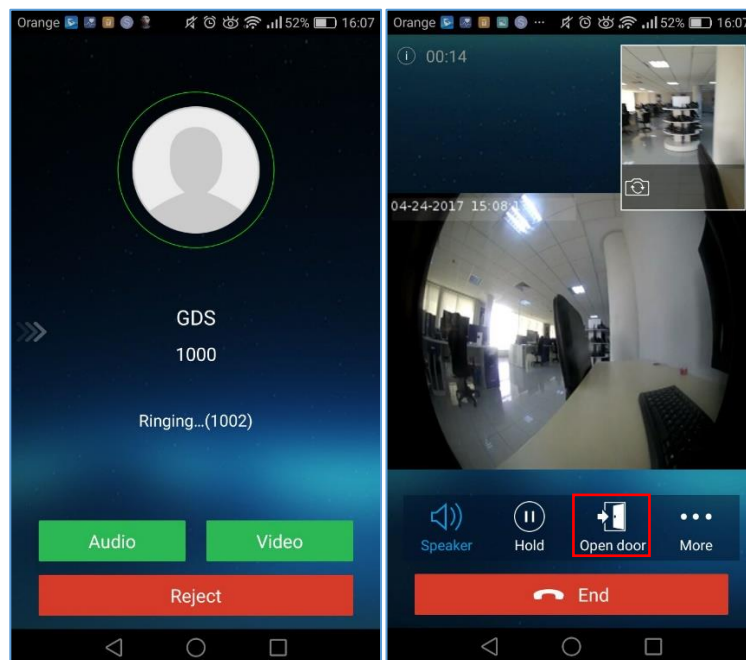


Figure 60: “OpenDoor” Button

EXPERIENCING GS WAVE

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 using Grandstream Wave application, it will be sure to bring convenience to both your business and personal life.

