



## vivi Android & iOS Grandstream Wave Lite App User Guide

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## vivi Android & iOS Grandstream Wave Lite App

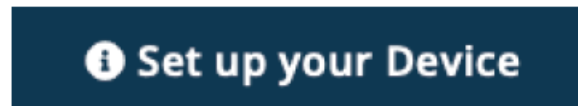
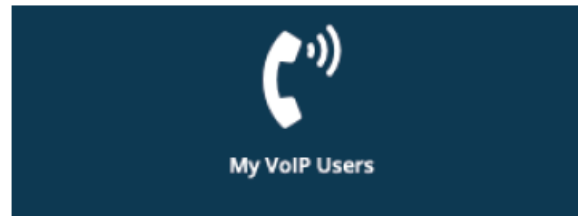


### Connecting GS Wave to your VOIP extension

To connect your Grandstream Wave Lite app to your VOIP account you would need to follow these steps below. You may have been given a QR code by our Vivi support team or you may need to find this through the Vivi Portal under the My VOIP users tab on the dashboard, From there you will be able to get the QR code you need for Grandstream Wave Lite. Please go to step xx if you already have been given a QR code for your extension

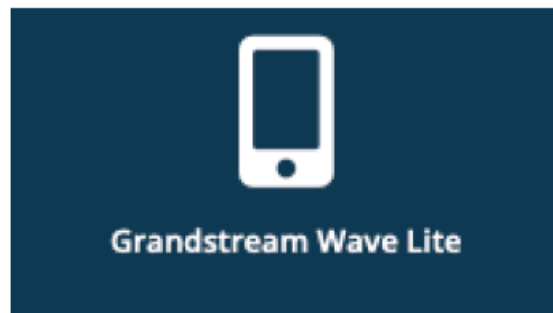
### To get your QR code through the VIVI portal





- Go to My VOIP Users
- Then go to Edit on your VOIP User account
- Go to Setup My Device
- Then go to Mobile/Softphone
-





Please visit the Google play or app store and download the Grandstream Wave Lite app onto the device



Once you have downloaded the app on your mobile device, please click the button below to set up the app

**Regenerate Password and Show QR Code**  
(You will need to reprogram any devices using this account)

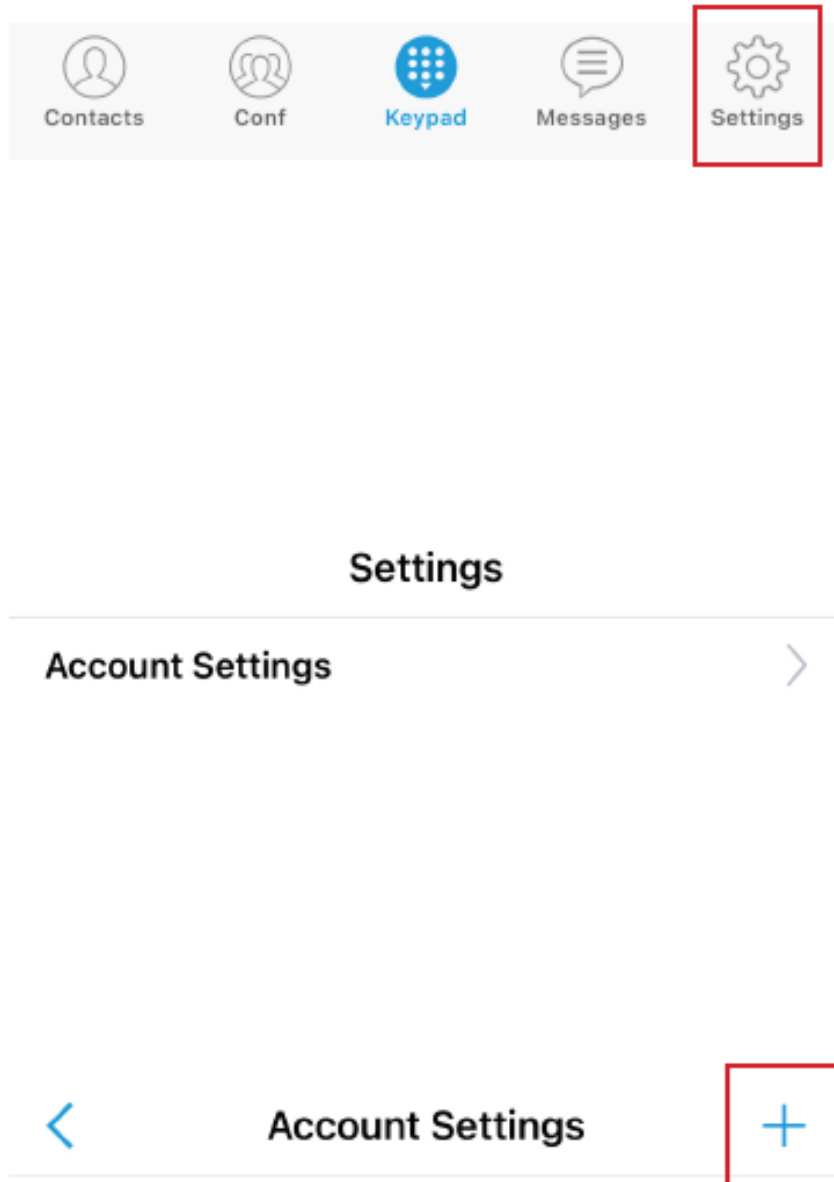


Click on Grandstream Wave Lite

- Then go to Regenerate New Password
- You should now see the QR code you need to scan into the app

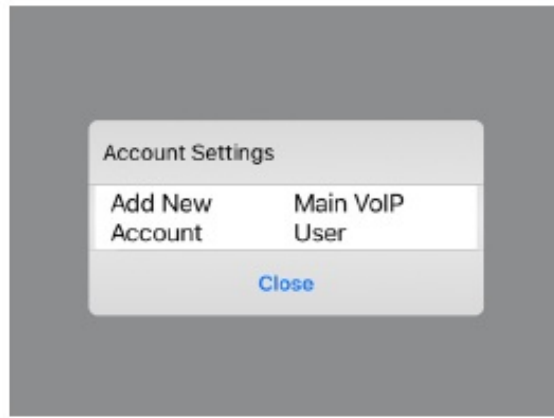


## How to scan the QR code into Wave Lite



- Open up the Grandstream Wave Lite app and allow all permissions and then go to Settings on the app
- Then click on Account Settings
- Click on the + icon on the top right-hand side of the screen
-





Once scanned your need to click on add new account

- Once you have done that please go to the Keypad and you will see on the top right that the VOIP user is now active.

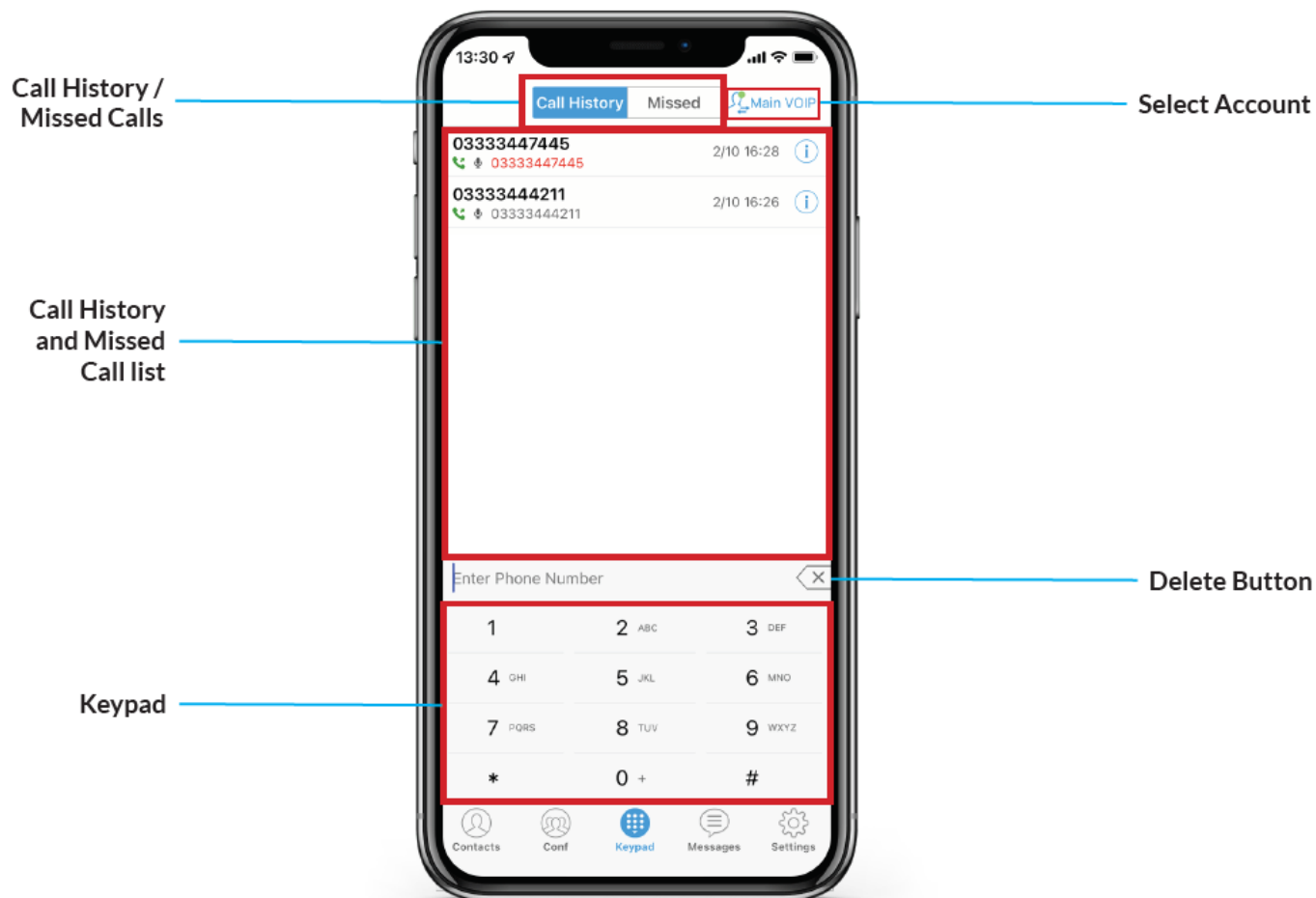
If you have any issues with setting up then please contact our support team who will be happy to provide assistance

## Getting to know GS Wave Lite

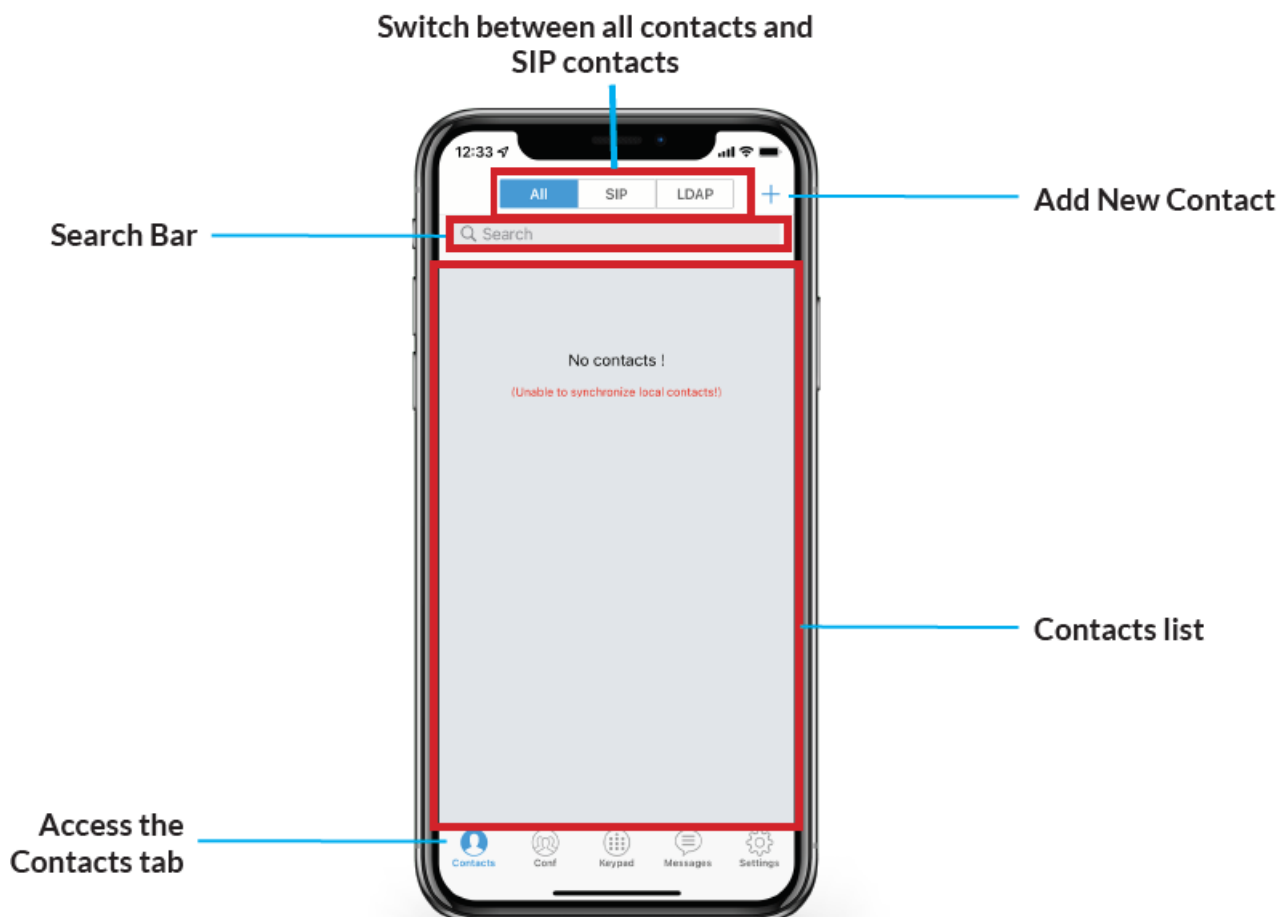
The Grandstream Wave Lite has the same layout design for both iOS and Android, please note that the app will function differently between OS. We will notify you of anything that is slightly different throughout the guide

### Keypad function layout



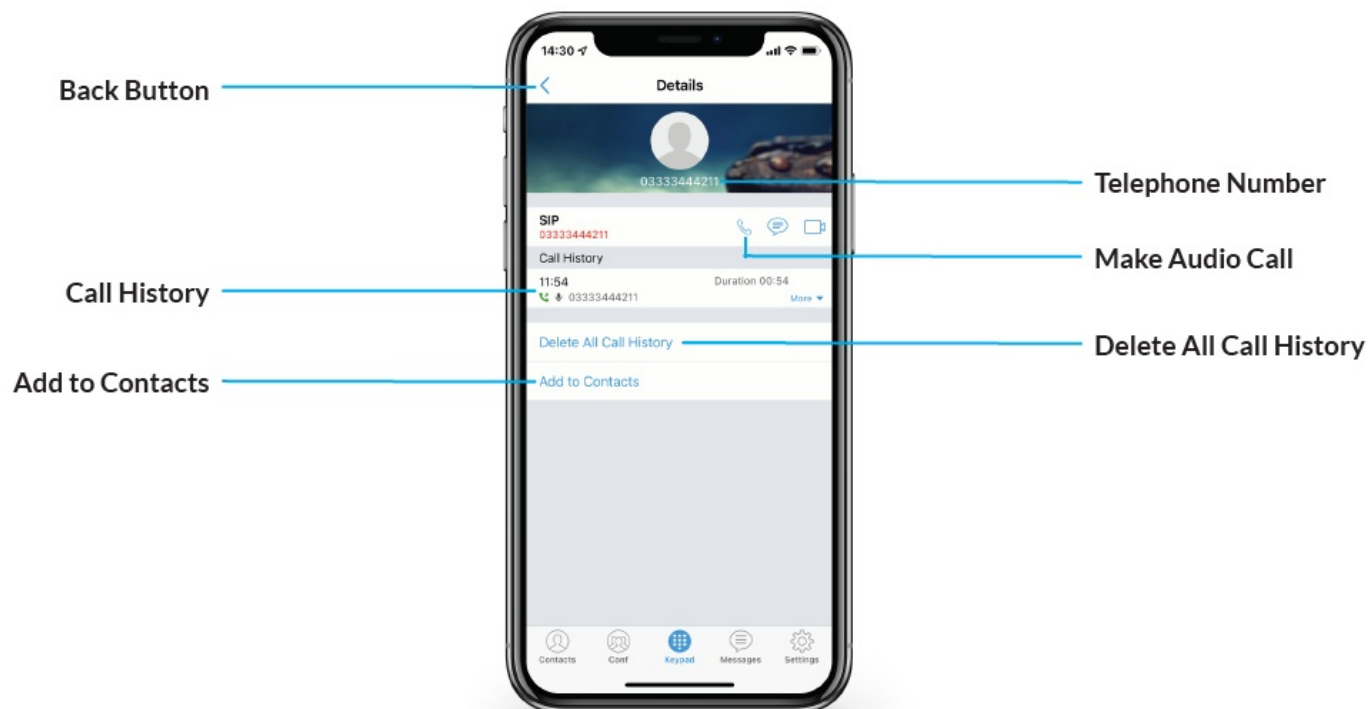


## Contact layout





## Contact Details Screen



## Answering/Making a call layout





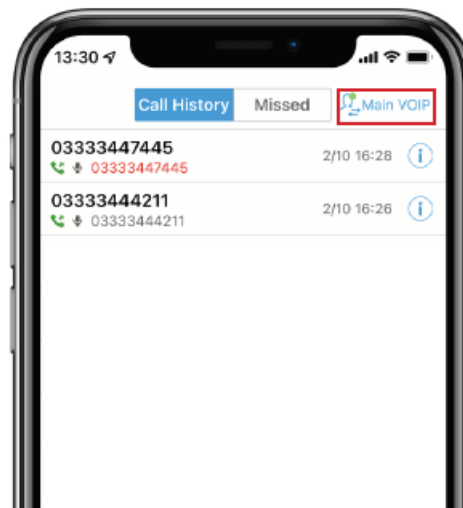
## How to make calls with GS Wave

### To call from your VOIP Extension

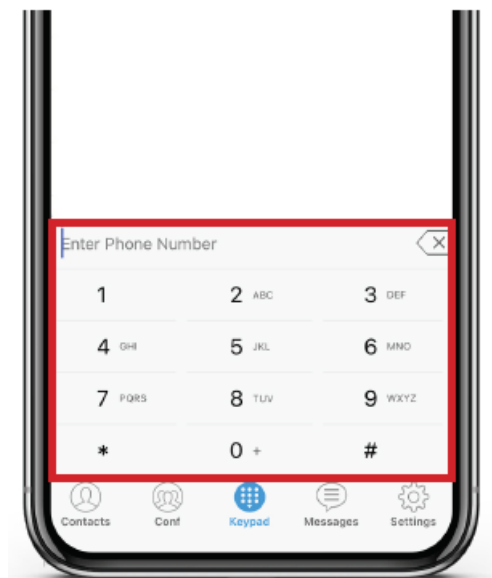
- First, go to the Keypad



- Make sure your VOIP account is active or has a green dot next to the name.

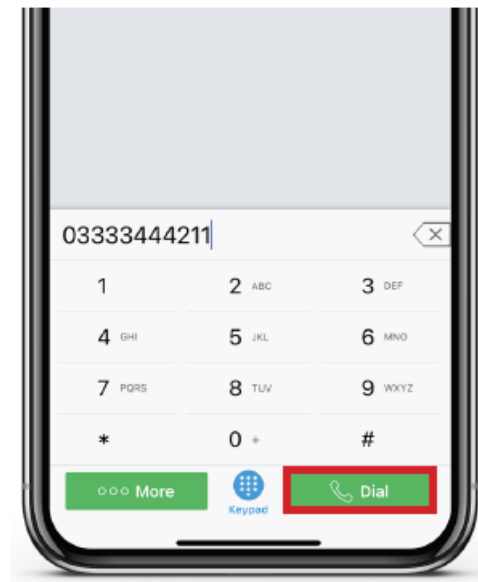


- Dial the number on the keypad

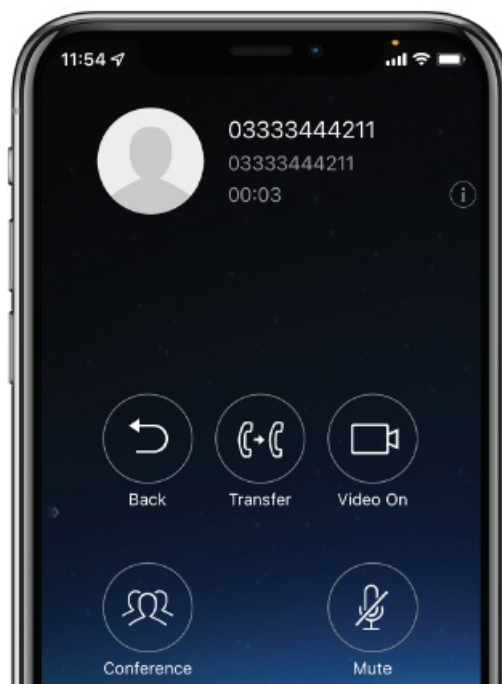


- Once you have put your number in press the dial button to begin the call





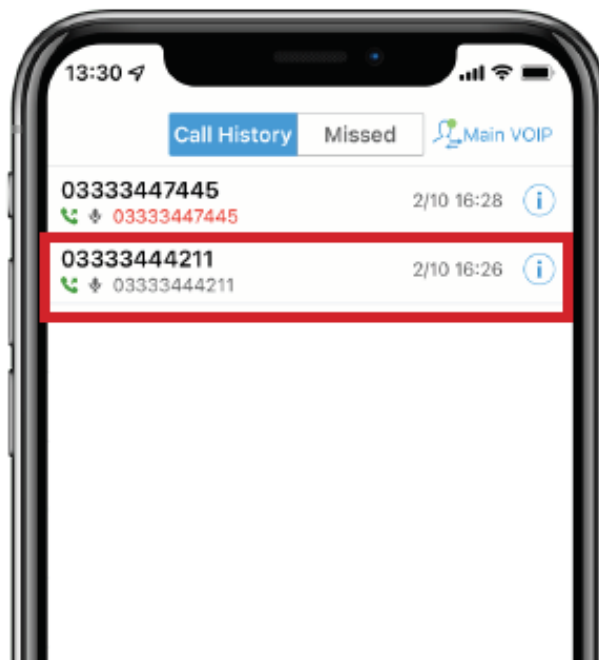
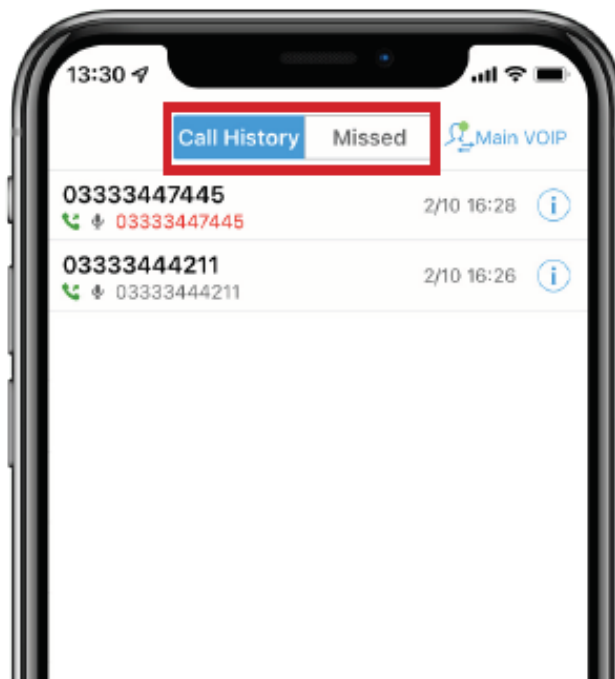
- When you press dial it will take you to the call screen where you can speak to the person you are wanting to speak to



### **To redial a number from call history or missed calls**

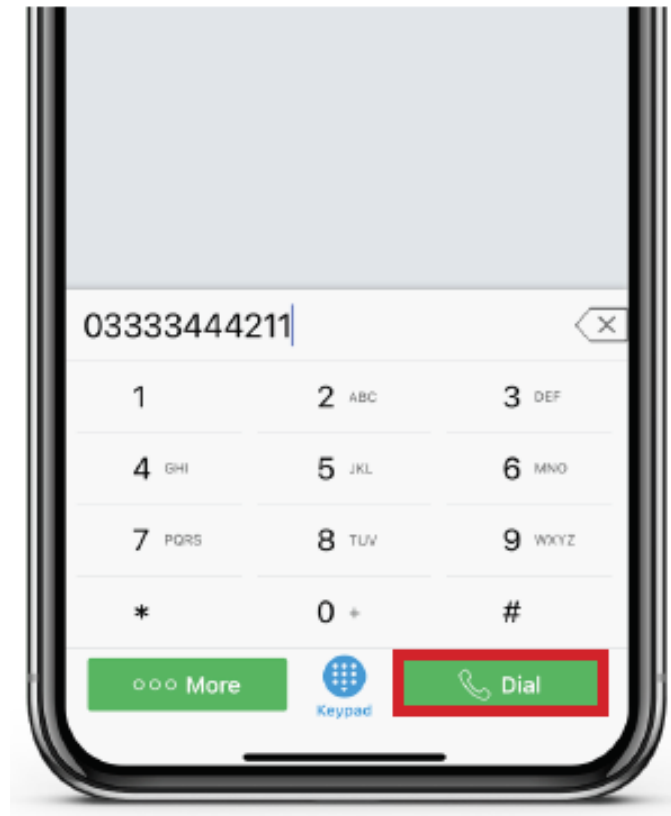
- First, choose to find your number in your Call History or Missed Call List
- Then pick the number you want and select it by tapping it





- Then with the number selected press the Dial button to call the number.

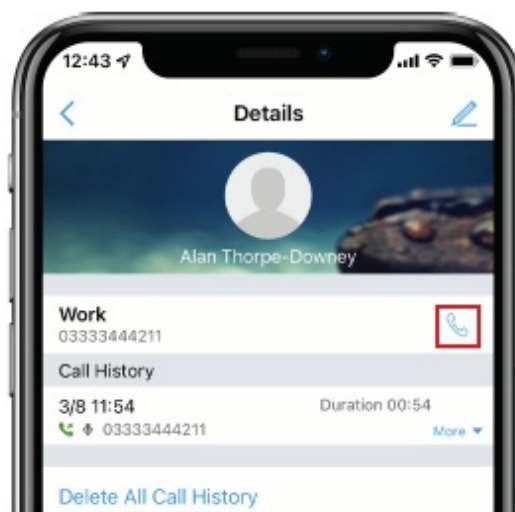
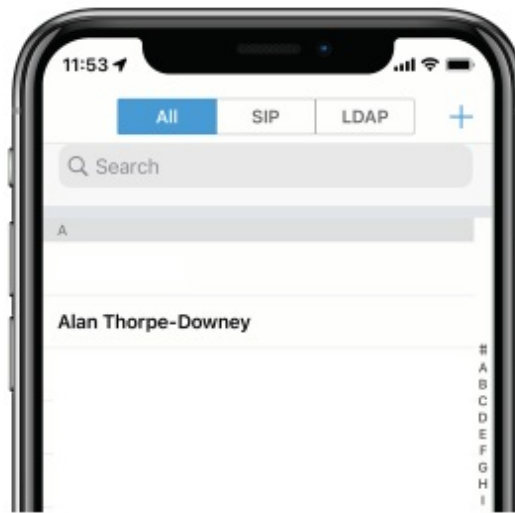




### Calling from the phone book

- First, choose the contact from your contact list
- Click on the phone icon





- Then press the Dial button to begin calling the number.



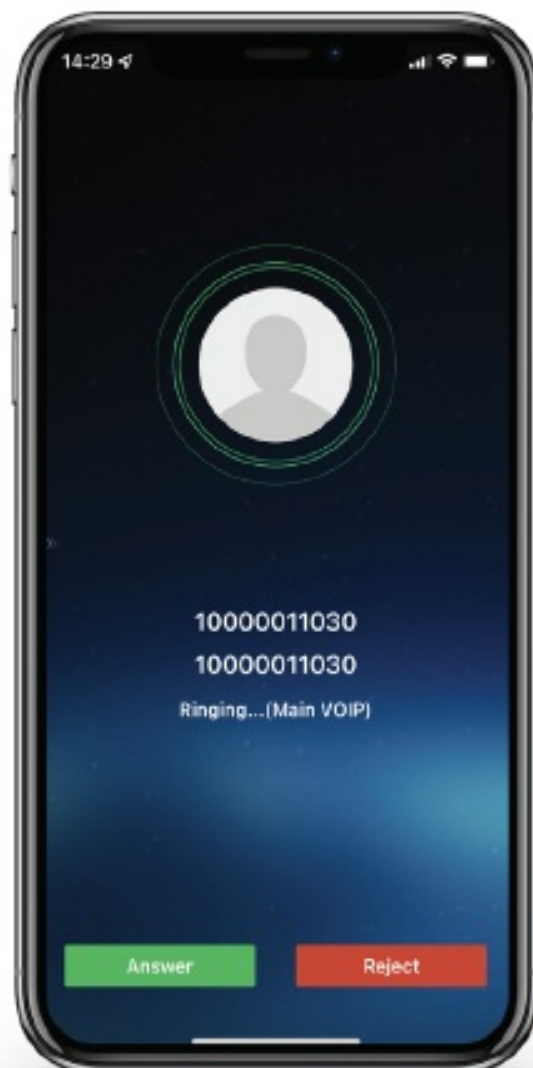
- The app will dial the number and you can speak to the person through the app





## How to answer calls with GS Wave Lite

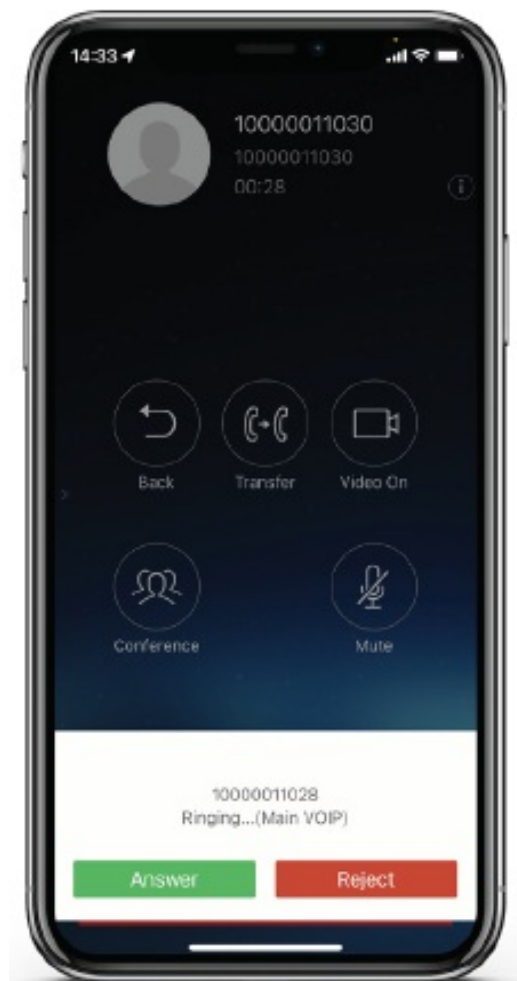
When a call comes on your phone





- When a call comes in you would be given the option to accept or reject the call that comes through from your VOIP extension.

### Multiple Incoming calls



- If another call comes in you will hear a beep in your audio earpiece and a notification will pop up at the bottom of the phone to say who is calling you

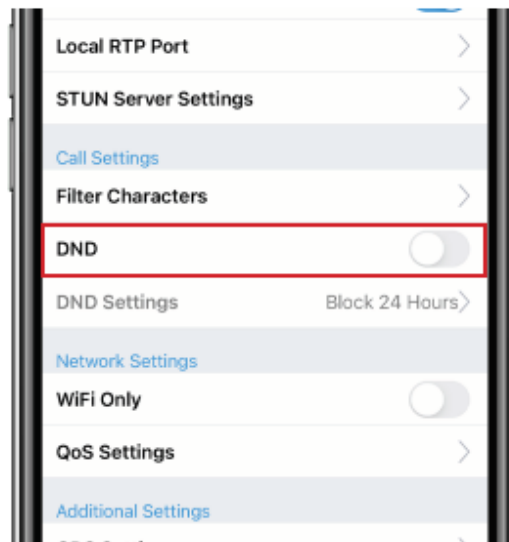
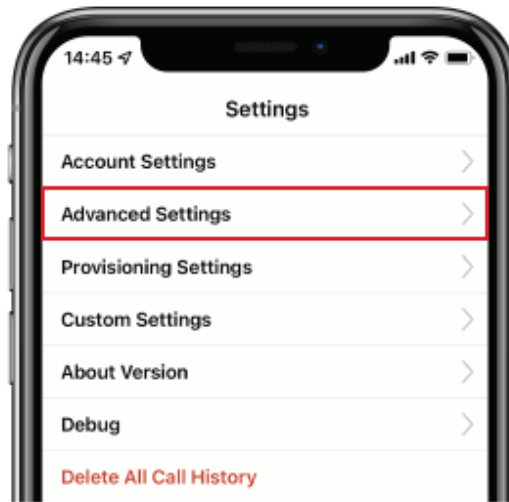
### Putting the phone on DND mode

- First, go to the Settings tab

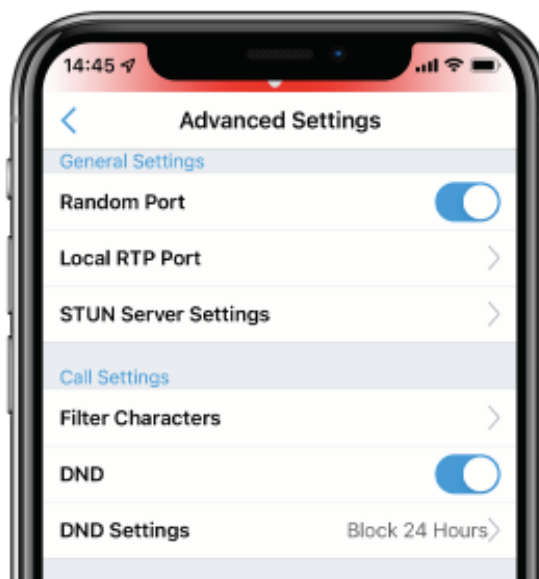


- Then go to the Advanced Settings
- Scroll down to see DND





- Toggle this on and you will see on the top the header has gone to red and no calls will come through while this is enabled



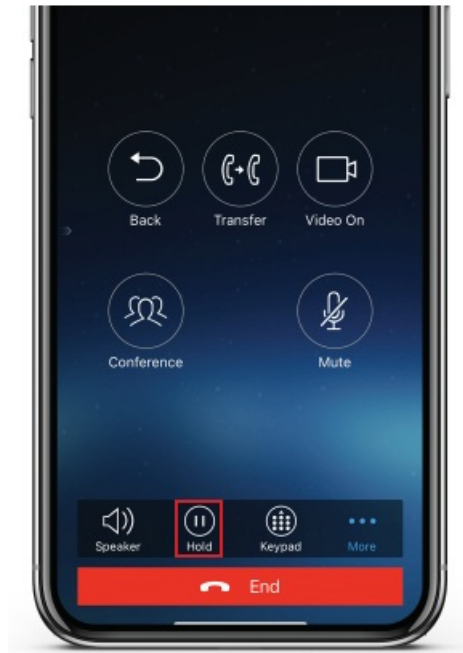
## Phone function during a call

### Holding a call

- During a call, you can put somebody on hold by pressing the hold button to put the active caller on hold

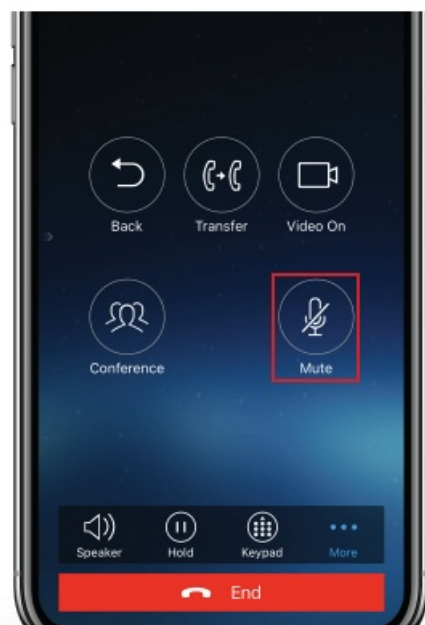


- To uphold the active caller please press the hold button again to take the active caller off hold and speak to the active caller



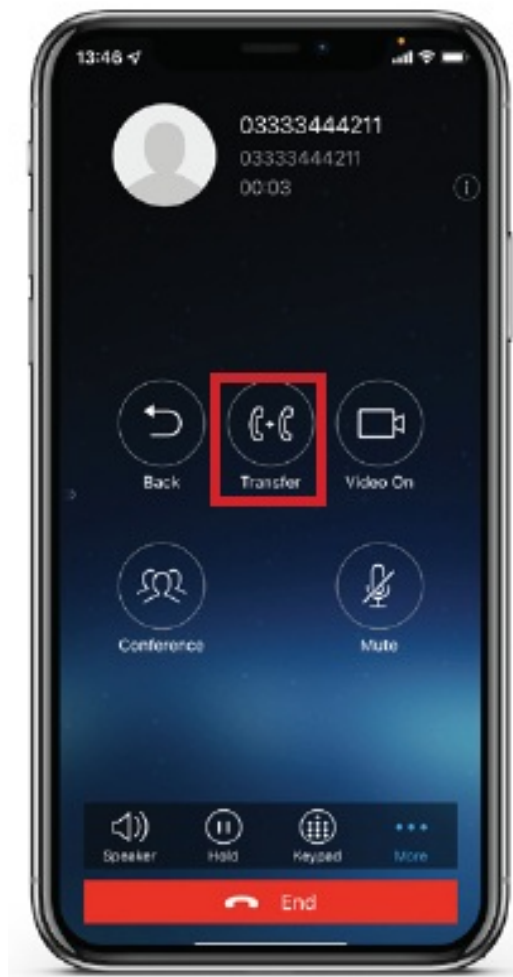
### How to mute the microphone

- During a call, you can put somebody on mute by pressing the mute button to put the active caller on mute.
- Please note the active caller will not hear any hold music when you put the caller on mute.
- To unmute the active caller please press the mute button again to take the active caller off hold and speak to the active caller



### To transfer a caller

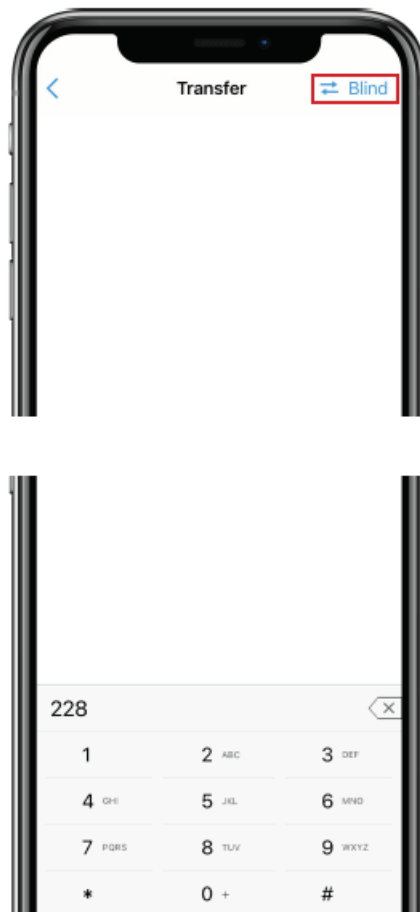




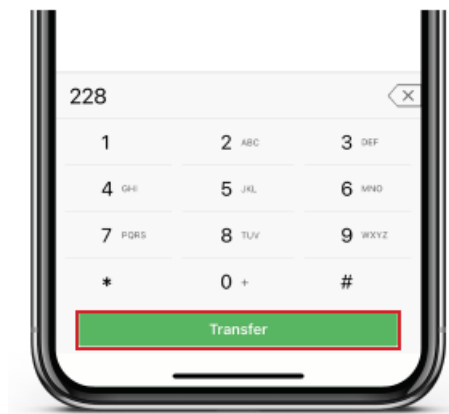
- During a call, if you want to transfer the call to another member of your team you would first need to press the transfer key.

### **Blind transfer**



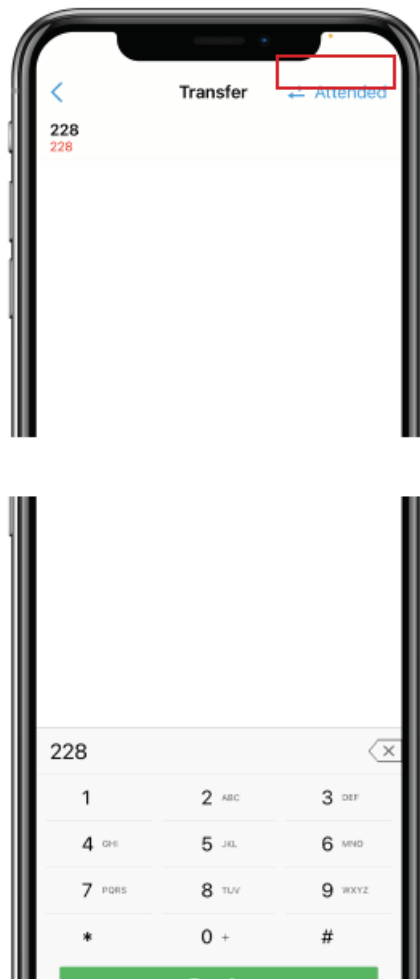


- Make sure that Blind is selected
- Dial-in the extension or number you want to call
- Now press the transfer button and the call will be forwarded to the extension or number you have dialed.

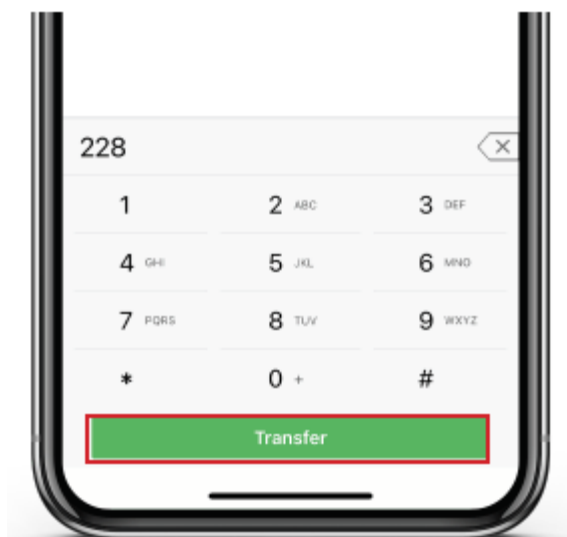


## Attended transfer





- Make sure that Attended is selected in the top right
- Dial-in the extension or number you want to call
- Now press the transfer button and the call will be forwarded to the extension or number you have dialed.



- Now the app will dial the extension or number when the person answers the call you would be given the option to transfer the call to that person or split the call so you and the active caller can speak to the person who you have dialed.





- If there is no answer then press the send button to transfer you back to the active caller

## GS Wave Lite – Account and Advanced Settings

When you first use Wave Lite you will need to go to settings to add your VOIP user account to the GS Wave app as you can see on page 5 on how to do this. On the next few pages, we show each function in Account Settings and Advanced Settings on what each item does, If you are unsure or need additional technical support on which settings need to change then please contact our technical support team.

### Account Settings

The account settings will show you the active and deactivated accounts on this screen, on your VOIP accounts you will see a dot on the left on hand side which will let you know if the account is active or not. If your account is showing a green dot the account is active, if it's showing a red dot then the registration has failed and if it has a black dot then the account is not active.

If you need to delete the account you need to swipe right on the VOIP user account to delete the account from your account settings.

Please note that you can have up to 6 independent SIP accounts on one handset at any time, If you reach the maximum 6 the next account you add will ask you to overwrite any accounts you have on the app.

### General Settings – Account Settings

Item	Description
<b>Activate Account</b>	Activate or deactivate the SIP account
<b>Edit Account</b>	Edit the VOIP user details – Please see next page these options
<b>Delete Account</b>	Delete the VOIP User account on GS Wave Lite

### Account Settings – Edit Account



Item	Description
<b>Account Name</b>	Name of the account
<b>SIP Server</b>	The domain name of the server your are on for your VOIP account
<b>SIP User ID</b>	Your user ID which is your extension number
<b>SIP Authentication ID</b>	Authenticate the SIP user ID which goes back to the server, If this has another user ID the account would not work
<b>Password</b>	Your SIP password, only technical support can only provide you with the SIP password
<b>Voicemail User ID</b>	If you have a voicemail on your SIP account you can put in the number 1571 to dial your voicemail inbox from the app.
<b>Display Name</b>	The name that people will see on the LCD screen when you dial their extension

### Account Settings – Call Settings Options

Item	Description
<b>Ringtone</b>	The ringtone you want GS Wave Lite to play when an inbound call comes in
<b>dialplan Settings</b>	Where you can configure the dial plan settings, This is disabled on default please toggle this to enable the DialPlan Prefix and Settings (only enable this if you been advised to)
<b>DialPlan Prefix</b>	Configure the default prefix, all numbers outbound are 44 for UK but you can change this to the country you want
<b>DialPlan Settings</b>	Configure the Dial Plan setting the default is "{ x+   \+x+   *x+   *xx*x+ }". Please speak to support about changing this setting
<b>Use # as Dial Key</b>	Stead of pressing the dial key you can enable # to dial any numbers
<b>Call Forward</b>	Tap on this to access the call forward options
<b>Call Forward Setting</b>	<p>This is used to specify call forwarding, there are four options to choose from:</p> <p>Unconditional – Send all calls to number or extension</p> <p>Time Based – Send calls either in or out time to extension or number Others – Forward When Busy – Forward calls to extension when you are on a call</p> <p>Others – No Answer – If you do not answer after a time period it would forward to an extension or number.</p> <p>Please note that you can set rules for when calls are not answered on your Vivi portal.</p>
<b>Auto Answer</b>	Enabling this will automatically turn on the speaker phone to answer incoming calls after a short beep.



## Account Settings – SIP Settings

Item	Description
<b>SIP Port</b>	It used define the SIP port that is used to listen and transmit data to either your router or mobile data, The Random Port should be enabled if not then you can choose the SIP port on the Advanced Settings
<b>Transmission Protocol</b>	Used to config the transmission protocol to transmit SIP info. You would be able to choose between UDP or TCP
<b>Registration Expiration (m)</b>	Specifies the amount of minutes that the phone refreshes its registration with the app. Please note that the default is 1 hour (60) you can lower this or increase this up to 64800 mins which is 45 days
<b>Unregister Before New Registrations</b>	If set to "Register All" the sip contact header will use "*" to clear all SIP user's registrations information, If set to "Do Not Register" the app will not clear the current SIP user's info. The default is "Unregistered Single" that means do not cancel SIP user's registration information.
<b>Only Accept SIP Requests from Known Servers</b>	Once enabled your SIP account will only accept requests from known servers like the SIP server that your account is hosted. This is set as off by default.
<b>Check SIP User ID for Incoming Invite</b>	This will check SIP User ID in the request URI for incoming INVITE. If it does not match the SIP User ID the call will be rejected, Direct IP calling will be disabled.

## Account Settings – Network Settings

Item	Description
<b>Outbound Proxy</b>	Configures the IP address or Domain name of the Primary Outbound Proxy, Media Gateway or Session Border Controller
<b>Secondary Outbound Proxy</b>	Configures the IP address or Domain name of the Secondary Outbound Proxy, Media Gateway or Session Border Controller. The Secondary Outbound Proxy will be used if the primary outbound proxy fails
<b>DNS Mode</b>	Controls how the search appliance looks up an IP address for host names. There are three modes – A Record, SRV, NAPTR/SRV. The default settings is set to "A Record" if you need to locate a different server by DNS SRV you may need to use "SRV" or "NAPTR/SRV" – Please speak to support first before trying this



Item	Description
<b>NAT Traversal</b>	<p>Enable or disables the NAT Traversal mechanism, the default setting is “Keep Alive”</p> <p>If set to STUN and the STUN Server is configured, the phone will route to the STUN sever. The NAT type will be Full Cone, Address Restricted Cone or Port Restricted Cone, The app will try and use public IP addresses or port number in all SIP messages</p> <p>The app will send SDP packet to the SIP server periodically to keep the NAT Ports open and if its configured to “Keep Alive”</p> <p>Configure this to be “NAT NO” if outbound proxy is used Configure this to be “UPnP” if router supports UPnP</p> <p>If set to Auto the phone will use all traversal methods above to find which one is best.</p>
<b>Proxy required</b>	<p>A SIP Extension to notify the SIP server that the phone is behind a NAT/ Firewall. Do not configure this setting as we do not allow proxy's on our server.</p>

#### Account Settings – Codec Settings



Item	Description
<b>DTMF</b>	<p>Users can specify the mechanism to transmit DTMF digits, there are three supported modes:</p> <p><b>In Audio</b> – DTMF combined in the audio signal (not reliable with low bit codecs)</p> <p><b>RTP (RFC2833)</b> – allows to specify DTMF with RTP packet, Users could know the packet is DTMF in the RTP header as well as the type of DTMF. Default settings is set to this</p> <p><b>SIP Info</b> – Uses SIP Info to carry DTMF. The defect of this mode is that it's easy to cause desynchronization of DTMF and media packet for the reason the SIP and RTP transmitted independently</p>
<b>Preferred Vocoder</b>	<p>Select which codecs will be used on Wi-Fi, 2G, 3G and 4G.</p> <p>Multiple vocoder types are supported on the app (PCMU, PCMA, OPUS, G722, G726_32, iLBC, G729 and GSM).</p>
<b>H.264 Image size</b>	Configures default image sizes in different network environment, Wi-Fi is set to VGA and mobile data is set to QVGA if you need to change you can use (720P, VGA, CIF, QVGA and QCIF)
<b>Video Bit Rate</b>	Configures the video bit rate for different network environment, you can increase the bit rate if network is sufficient and if bandwidth is poor they you will see a reduced video quality during calls.

Item	Description
<b>H.264 Payload Type</b>	Configure the H.264 codec payload type. The valid range is from 96 to 127 – default value is 99.
<b>SRTP Mode</b>	Configure the SRTP mode when is set to either enabled or force, this will enable or force to use SRTP. If set to Enable but not force it will be enabled but will not use SRTP. The default setting is Disable

### Advanced Settings – General Settings

Item	Description
<b>Random Port</b>	Forces random generation of both the local SIP and RTP ports when this set to yes.
<b>STUN Server Settings</b>	Configures the IP address or domain name of STUN server. Only non symmetric NAT routers work with STUN



## Advanced Settings – Network Settings

Item	Description
<b>WIFI Only</b>	In the WIFI environment it will only register with specified connected WiFi
<b>QoS Settings</b>	Configure layer 3 SIP QoS and Layer 3 Audio QoS

## Advanced Settings – Additional Settings

Item	Description
<b>GDS Settings</b>	The settings are for the GDS Door System where you would find the Name, Number and Password
<b>LDAP Settings</b>	Tap to access the LDAP settings which includes:  “Scan QR Code” to scan the QR code that you scanned in with your VOIP User account details on there  “Selected QR code image” To access screen with QR code image which will have your VOIP User account details on there to register to the app.
<b>BLF</b>	Configures whether to enable SIP contacts to have status detection in the BLF list.
<b>BLF list</b>	You will be able to add the contacts that you want the BLF status to monitor.
<b>Vibrate When Ringing</b>	Configure if you want the app to vibrate when ringing, this only applies to incoming calls only.
<b>Default Account Registration Notifications</b>	The is to enable default account registration notifications about any account changes. This will appear as a push notification in the notifications bar on your phone

## Faulty Handset or replacement parts

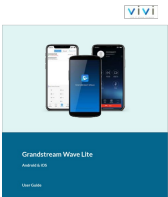
If your handset develops a fault then we will need to do a remote session with you to see what the issue is. You may be asked to move the handset to another ethernet port or swap some parts with another phone that is working to see if the issue persists. You'll also need to reboot the handset, and possibly any networking equipment such as your router or network switches. If the handset is faulty we will email you instructions on where to return the handset

## Faults and Solutions



Issue	Solution
<b>Receiver is not picking up any sound</b>	<p>First, check if the app VoIP user is active on the keypad,</p> <p>Then, dial 121 to do an echo test. While on the call, press the volume key and make sure the volume hasn't been turned down.</p> <p>Check if the speaker phone is working.</p> <p>If there is no sound please check if the Microphone is enabled on the phone app settings</p>
<b>GS Wave is not ringing</b>	You may have DND on the handset. If you do, you'll see an icon on the LCD display. See Page 15 – 'Putting the phone on DND mode'
<b>VOIP User is showing red</b>	<p>If the VOIP user is showing red then please try the following settings below:</p> <p>Go to Settings &gt; Account Settings &gt; YOUR VOIP USER Account and go to Transmission Protocol and change that from UDP to TCP to see if that connects</p> <p>If that does not then please disable the WIFI only on the app by going to Settings &gt; Advanced Settings, Once you have done that turn off WIFI only on the app and turn off WIFI on your phone to see if connects</p> <p>If this does then you have an issue with the router, if this does not connect then please re-scan the QR code for your VOIP User account.</p>

## Documents / Resources

	<p><a href="#">vivi Android &amp; iOS Grandstream Wave Lite App</a> [pdf] User Guide</p> <p>Android iOS Grandstream Wave Lite App, Grandstream Wave Lite App, Grandstream, Wave Lite App</p>
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