Grandstream Wave Lite

Android & iOS

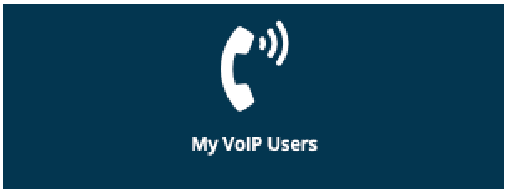
User Guide

Contents

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 your 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



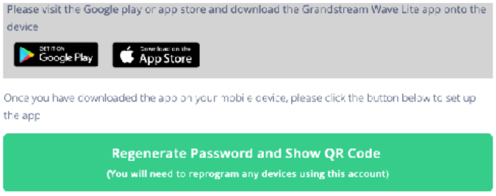
Go to My VOIP Users

Then go to Edit on your VOIP User account

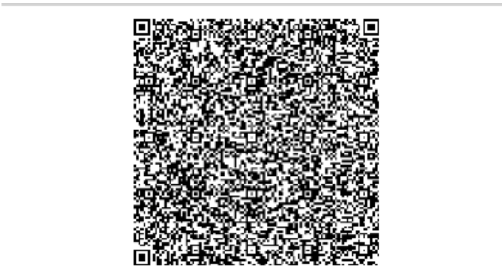
Go to Setup My Device

Then go to Mobile/Softphone

Click on Grandstream Wave Lite

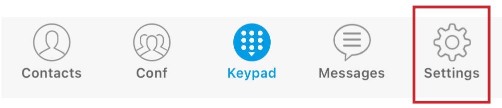


Then go to Regenerate New Password

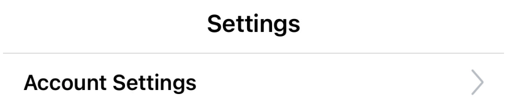


You should now see your QR code you need to scan into the app

How to scan the QR code into Wave Lite

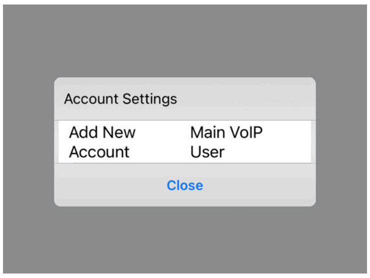


Open up the Grandstream Wave Lite app and allow all permissions and then go to Settings on the app.

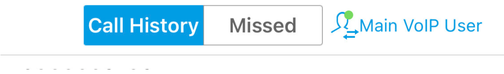


Then click on Account Settings

Click on the + icon on the top right hand side of the screen



Once scanned your need to click on add new account



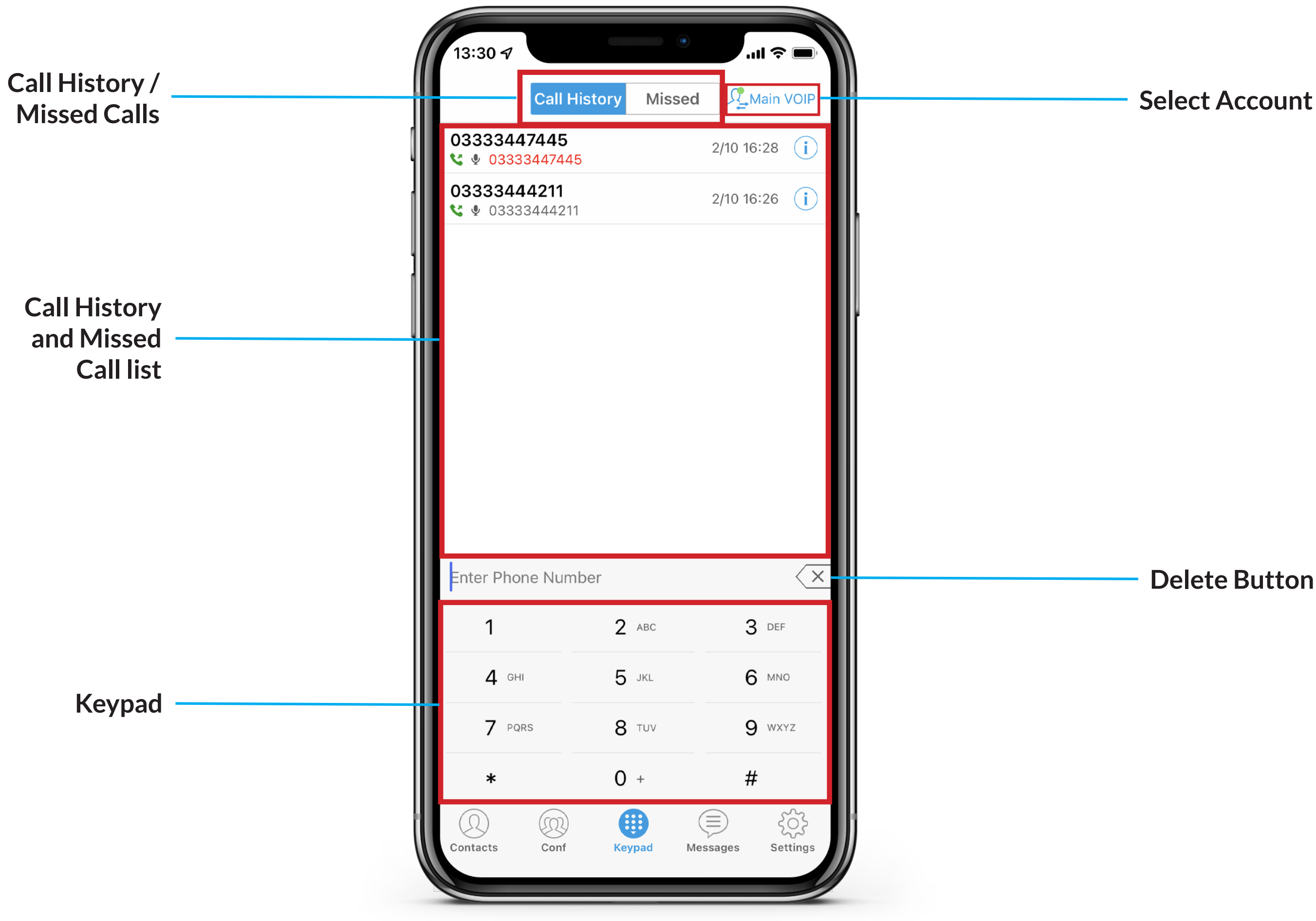
Once you have done that please go to the Keypad and you will see on the top right that the VOIP user is now active.

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

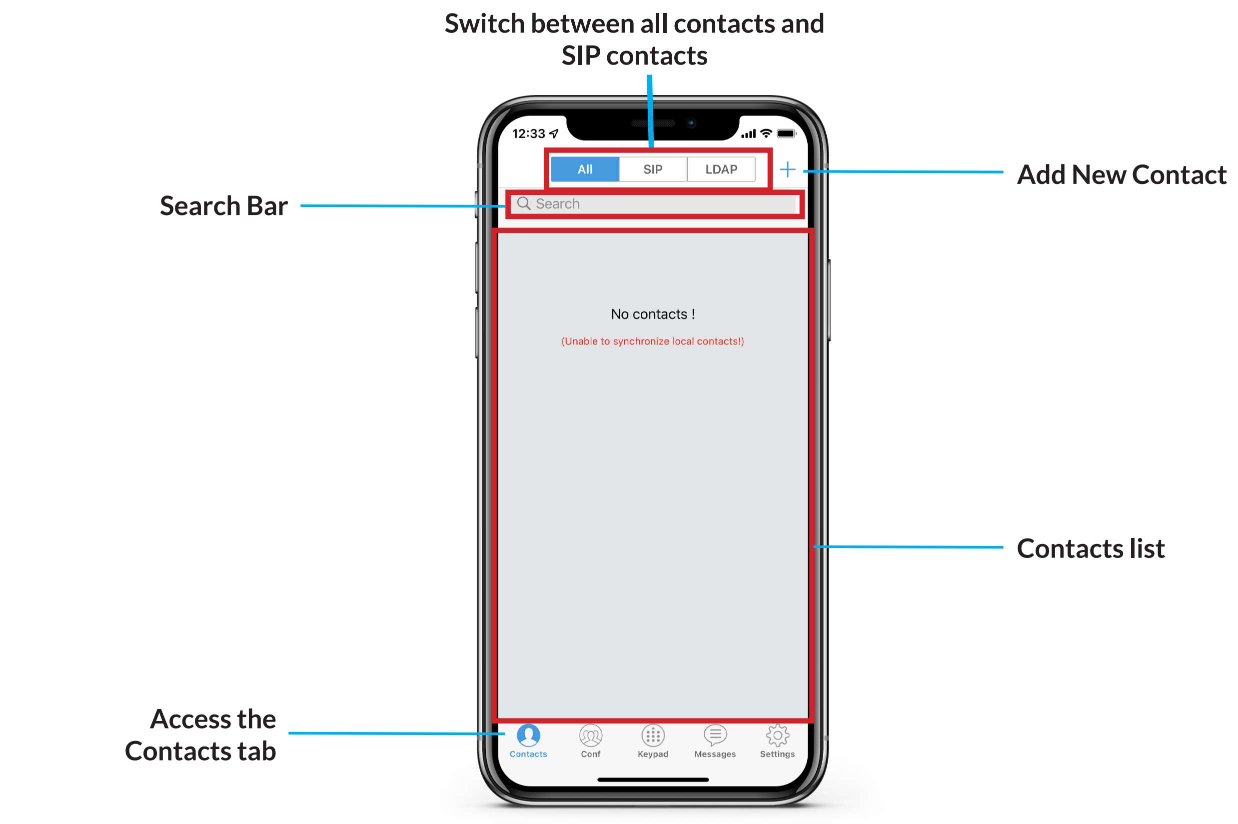
The Grandstream Wave Lite has the same the layout design for both iOS and Android, please note that the app will function differently between OS. We will notify of anything that is slightly different throughout the guide.

Keypad function layout

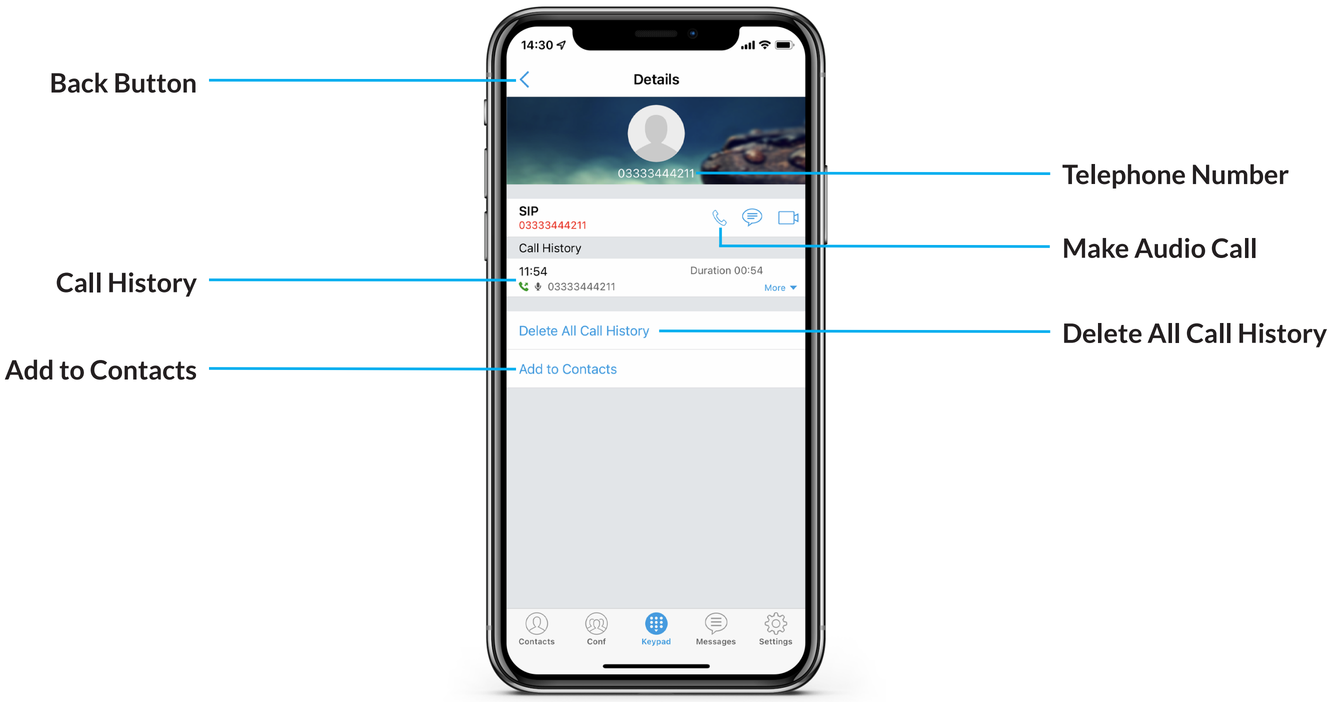


Contact layout

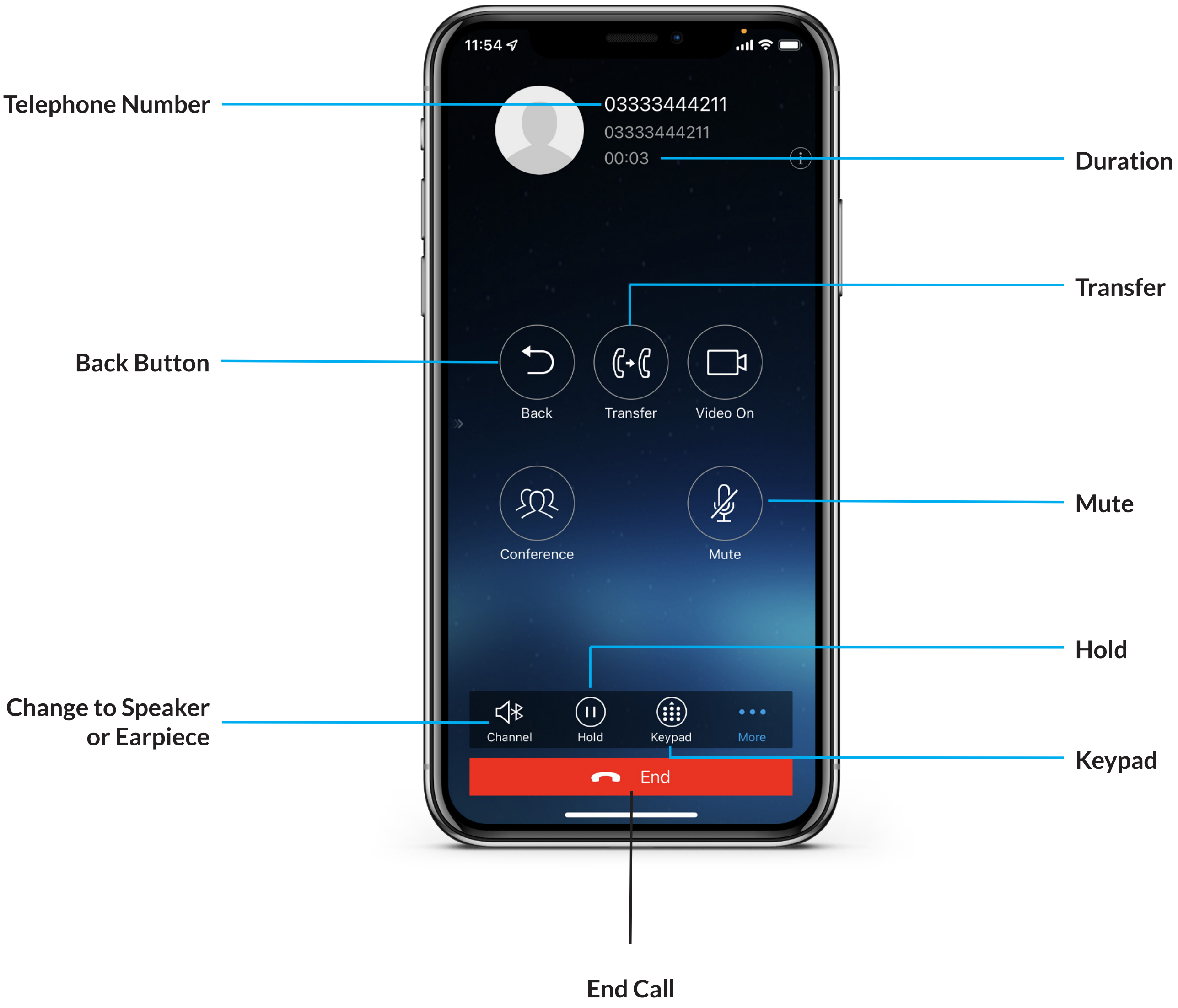
Contact screen



Contact Details Screen

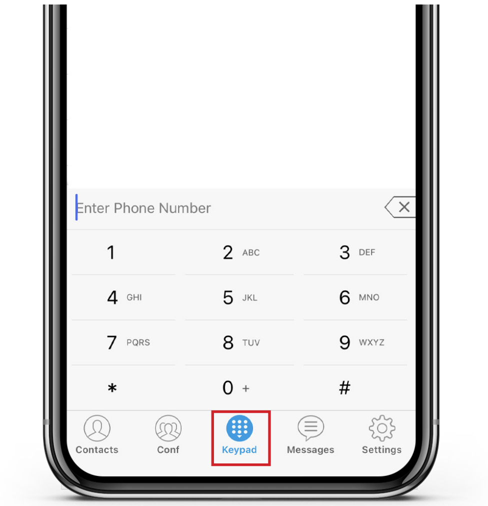


Answering/Making a call layout

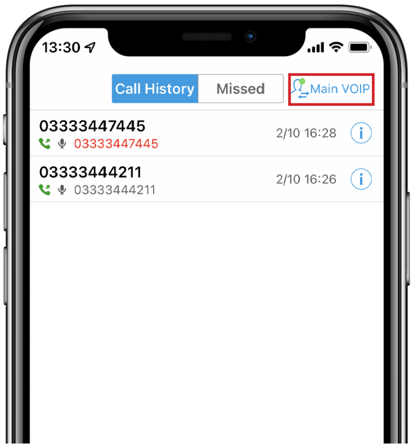


How to make calls with GS Wave

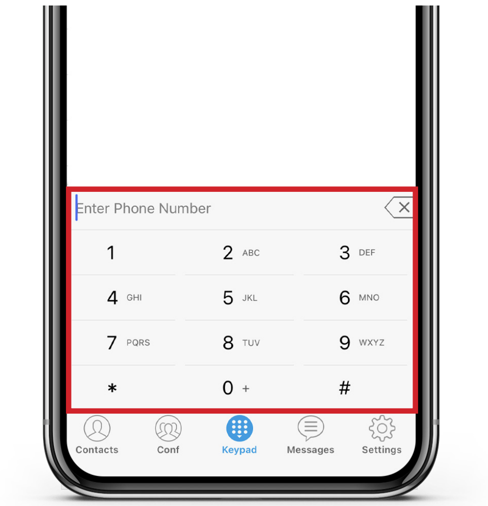
To call from your VOIP Extension



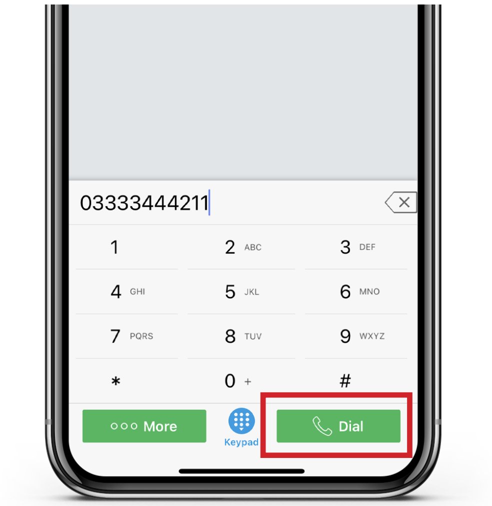
First go to the Keypad



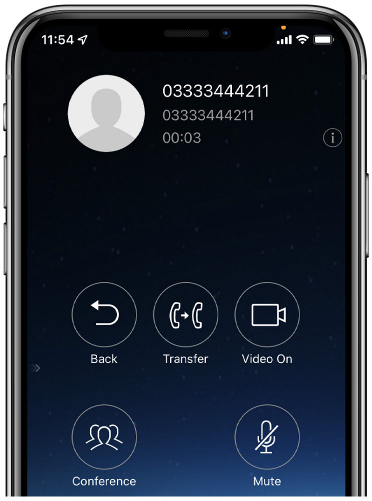
Make sure your VOIP account is active or has a green dot next to the name.



Dial the number on the keypad

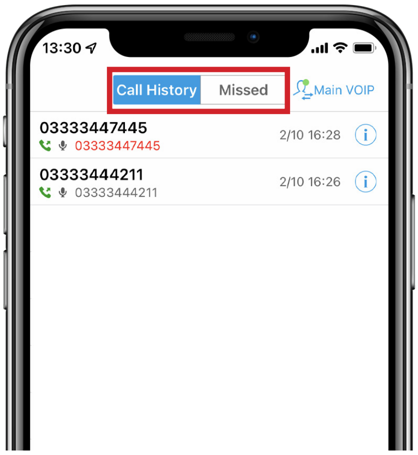


Once you have put your number in press dial button to begin the call

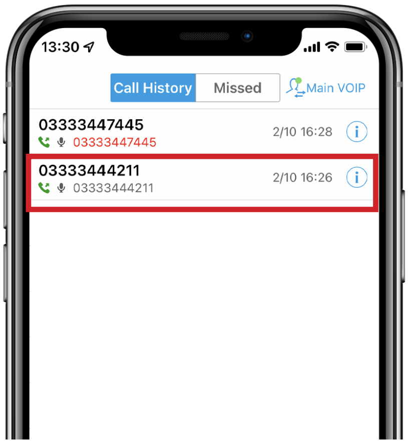


When you press dial it will take you to the call screen where you can speak to the person you are wanting to speak to.

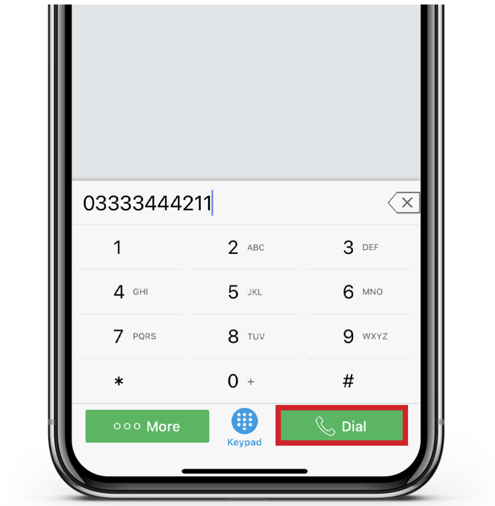
To redial a number from call history or missed calls



First choose to find your number in your Call History or Missed Call List

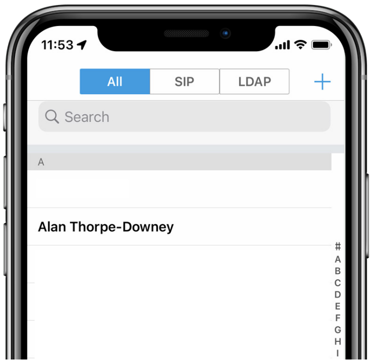


Then pick the number you want and select it by tapping it.

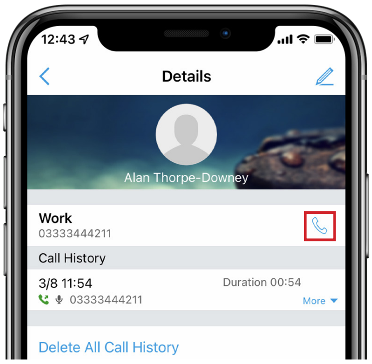


Then with the number selected press the Dial button to call the number.

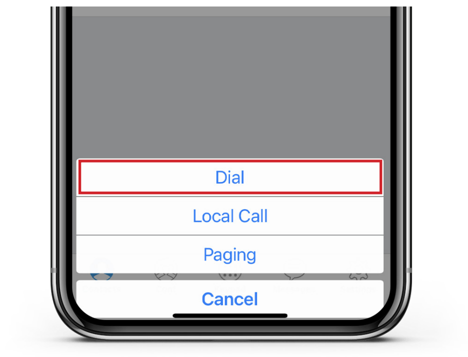
Calling from the phone book



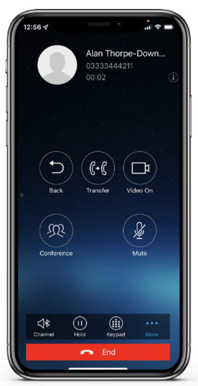
First choose the contact from your contact list



Click on the phone icon



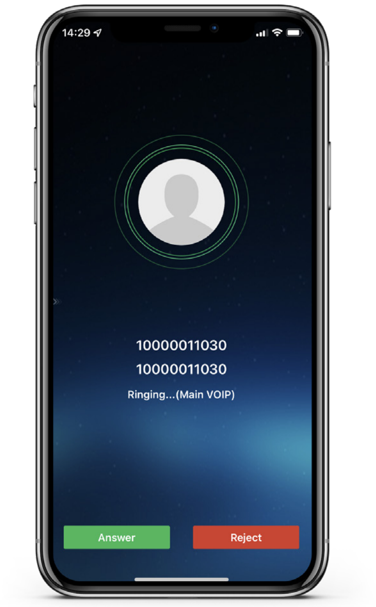
Then press the Dial button to begin calling the number.



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

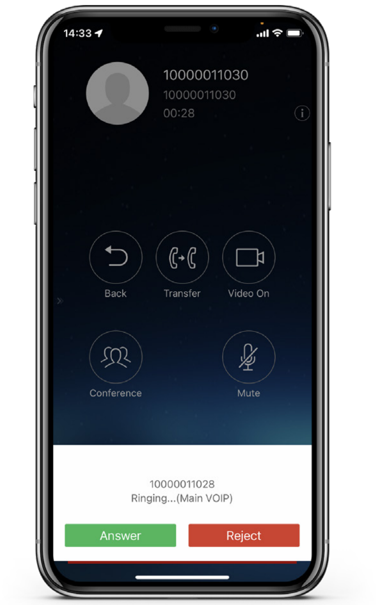
How to answer calls with GS Wave Lite

When a call comes in to 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.

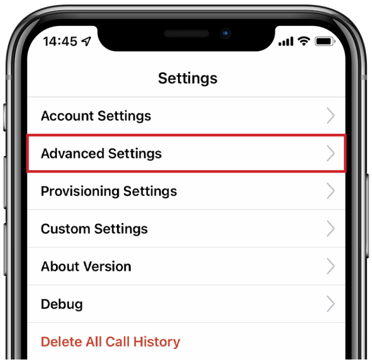
Multiple Incoming calls



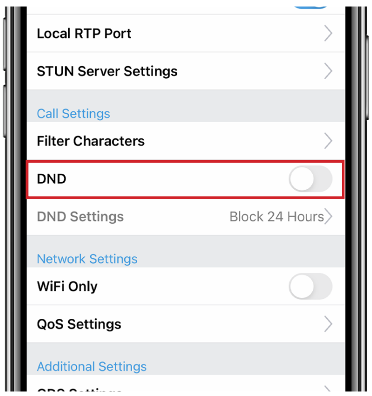
If another call comes in you will hear a beep in your audio earpiece and a notification will pop up at the bottom of the phone to say who is calling you

Putting the phone on DND mode

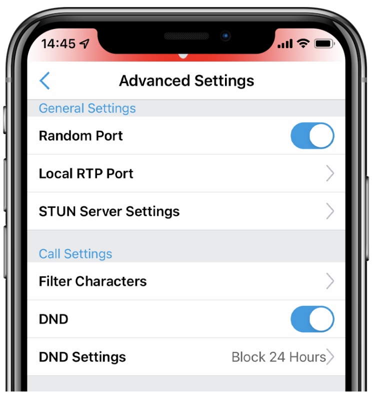
First go to the Settings tab



Then go to the Advanced Settings



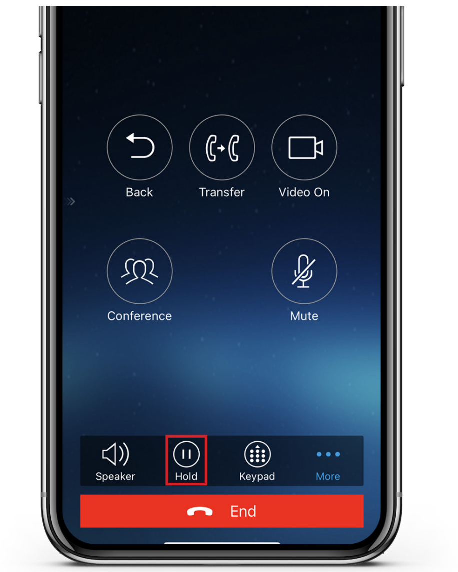
Scroll down to you see DND



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

Phone function during a call

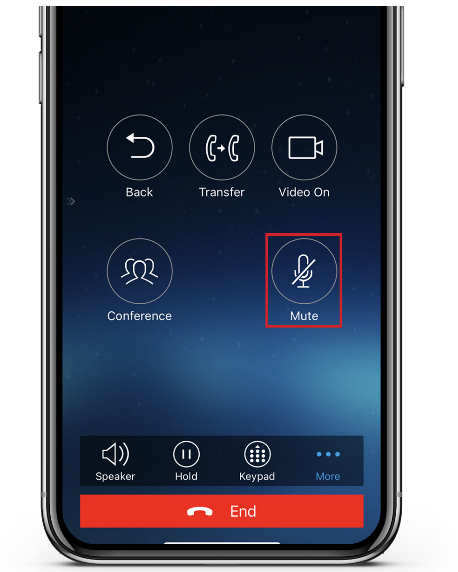
Holding a call



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

To unhold 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

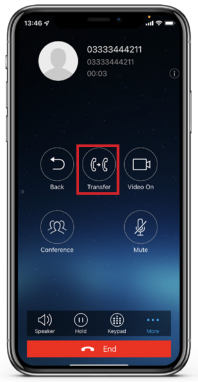


During a call you can put somebody on mute by pressing the mute button to put the active caller on mute.

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

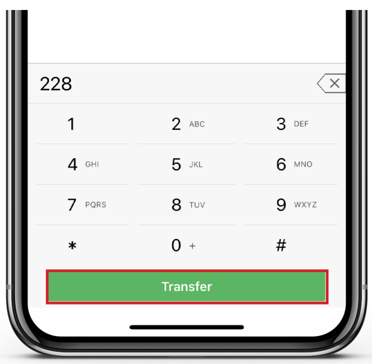


During a call if you want to transfer the call to another member of your team you would first need to press the transfer key.

Blind transfer

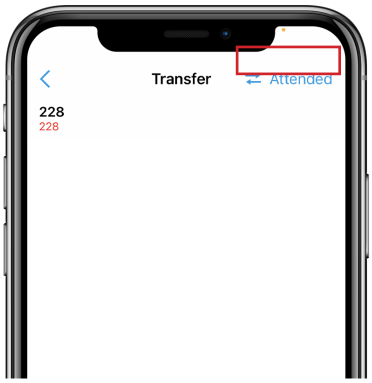
Make sure that Blind is selected

Dial in the extension or number you want to call

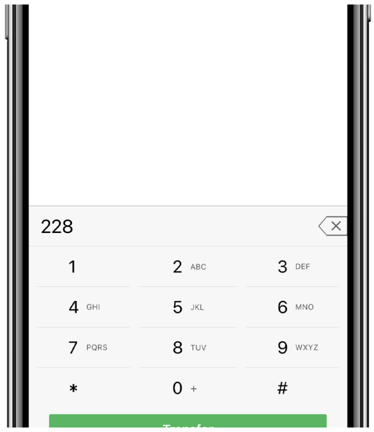


Now press the transfer button and the call will be forwarded to the extension or number you have dialled.

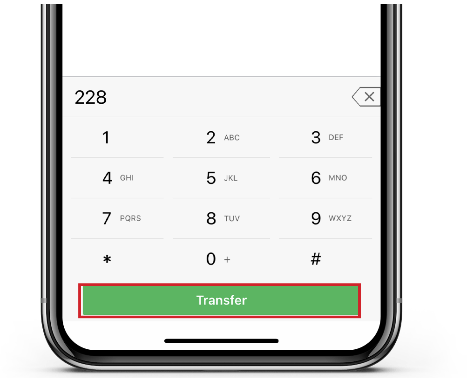
Attended transfer



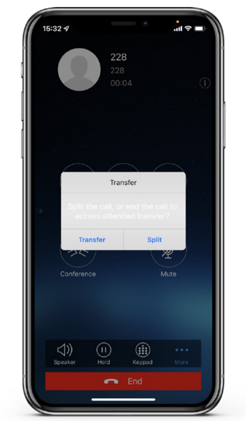
Make sure that Attended is selected in the top right



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 dialled.



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 dialled.

If there is no answer then press the end 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 of 6 the next account you add will ask you to overwrite any accounts you have on the app.

General Settings - Account Settings

Item	Description
Activate Account	Activate or deactiave the SIP account
Edit Account	Edit the VOIP user details - Please see next page these options
Delete Account	Delete the VOIP User account on GS Wave Lite

Account Settings - Edit Account

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

Account Settings - Call Settings Options

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

Account Settings - SIP Settings

Item	Description
SIP Port	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
Transmission Protocol	Used to config the transmission protocol to transmit SIP info. You would be able to choose between UDP or TCP
Registration Expiration (m)	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
Unregister Before New Registrations	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.
Only Accept SIP Requests from Know Servers	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.
Check SIP User ID for Incoming Invite	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
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. The Secondary Outbound Proxy will be used if the primary outbound proxy fails
DNS Mode	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 “NATPTR/SRV” - Please speak to support first before trying this

Item	Description
NAT Traversal	Enable or disables the NAT Traversal mechanism, the default setting is "Keep Alive" 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 The app will send SDP packet to the SIP server periodically to keep the NAT Ports open and if its configured to "Keep Alive" Configure this to be "NAT NO" if outbound proxy is used Configure this to be "UPnP" if router supports UPnP If set to Auto the phone will use all traversal methods above to find which one is best.
Proxy required	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.

Account Settings - Codec Settings

Item	Description
DTMF	Users can specifies the mechanism to transmit DTMF digits, there are three supported modes: **In Audio** - DTMF combined in the audio signal (not reliable with low bit codecs) **RTP (RFC2833)** - 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 **SIP Info** - Uses SIP Info to carry DTMF. The defect of this mode is that its easily to cause desynchronized of DTMF and media packet for the reason the SIP and RTP transmitted respectability
Preferred Vocoder	Select which codecs will be used on WI-FI, 2G, 3G and 4G. Multiple vocoder types are supported on the app e (PCMU, PCMA, OPUS, G722, G726_32, iLBC, G729 and GSM).
H.264 Image size	Configures default images sizes in different network environment, Wifi 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)
Video Bit Rate	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
H.264 Payload Type	Configure the H.264 codec payload type. The valid range is from 96 to 127 - default value is 99.
SRTP Mode	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
Random Port	Forces random generation of both the local SIP and RTP ports when this set to yes.
STUN Server Settings	Configures the IP address or domain name of STUN server. Only non symmetric NAT routers work with STUN

Advanced Settings - Network Settings

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

Advanced Settings - Additional Settings

Item	Description
GDS Settings	The settings are for the GDS Door System where you would find the Name, Number and Password
LDAP Settings	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.
BLF	Configures whether to enable SIP contacts to have status detection in the BLF list.
BLF list	You will be able to add the contacts that you want the BLF status to monitor.
Vibrate When Ringing	Configure if you want the app to vibrate when ringing, this only applies to incoming calls only.
Default Account Registration Notifications	The is to enable default account registration notifications about any account changes. This will appear as a push notifications 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
Receiver is not picking up any sound	First, check if the app VoIP user is active on the keypad, 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. Check if the speaker phone is working. If there is no sound please check if the Microphone is enabled on the phone app settings
GS Wave is not ringing	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'
VOIP User is showing red	If the VOIP user is showing red then please try the following settings below: Go to Settings > Account Settings > YOUR VOIP USER Account and go to Transmission Protocol and change that from UDP to TCP to see if that connects If that does not then please disable the WIFI only on the app by going to Settings > 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 If this does then you have an issue with the router, if this does not connects then please re-scan the QR code for your VOIP User account.

vivi

the IP phone network