

# VoIP User Guide

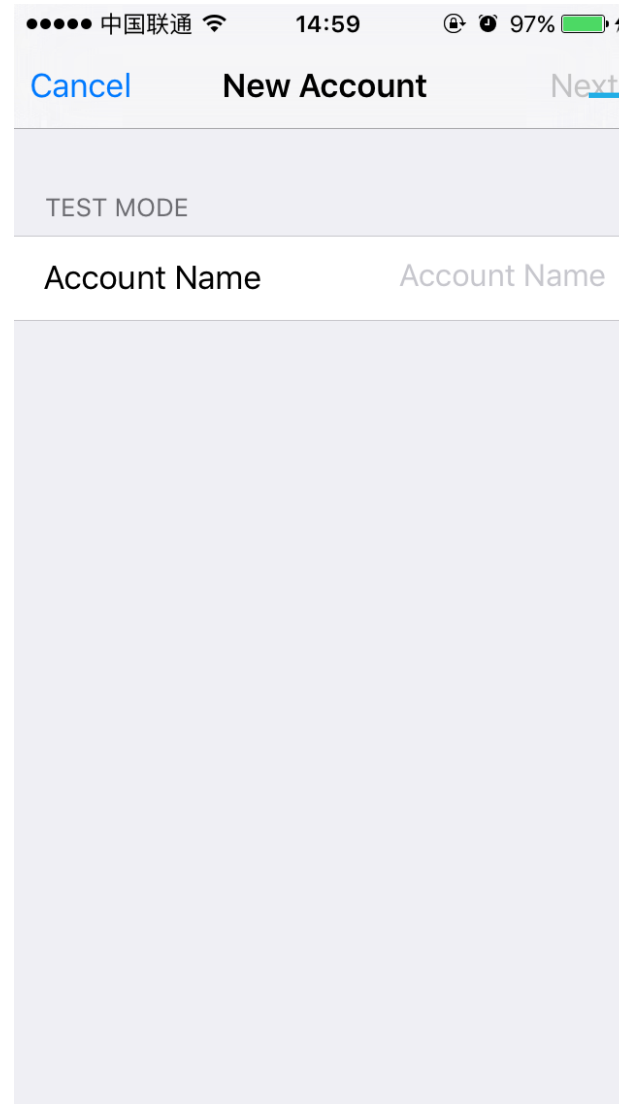
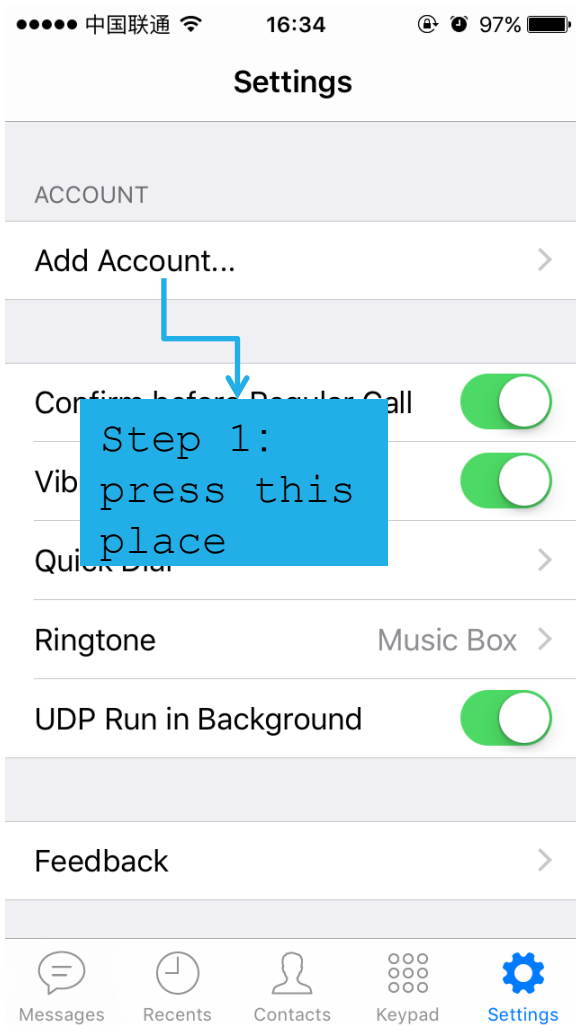
*Version No.: v3.0-20160126*

*Date: Jan 3, 2016*

# Intro

- Common setting
- Audio And Video Parameters selection
- Voice Parameters Intro
- Video Parameters Intro
- Statistics detection

# Common setting



# Common setting

17:48 97%

Cancel +8616000000489 Save

Account Name +8616000000489

Username +8616000000489

Password ●●●●●●

Authname 460070000000489@ims.mn...

Nickname +8616000000489

Server Address 120.197.90.65

Realm gd.ims.mnc007.mcc460.3gppn...

Register Type Hotfixes >

Register No Digest

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< Back Advanced

SIP

Transport Type TCP >

Server Port 5260

AUDIO

Codec >

AGC(send)

AGC(recv)

ANS(send)

ANS(recv)

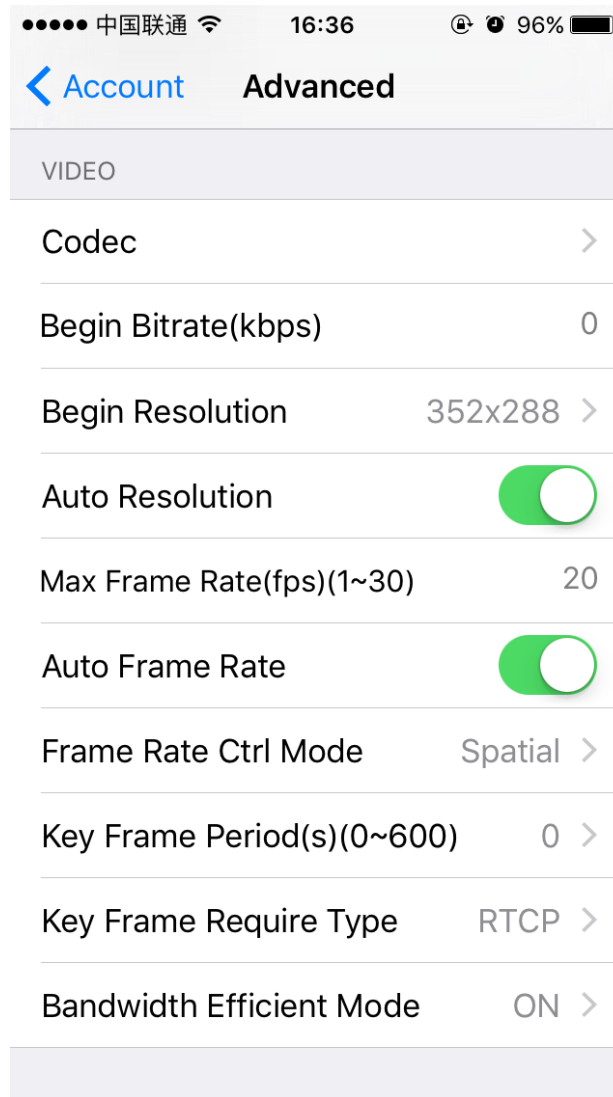
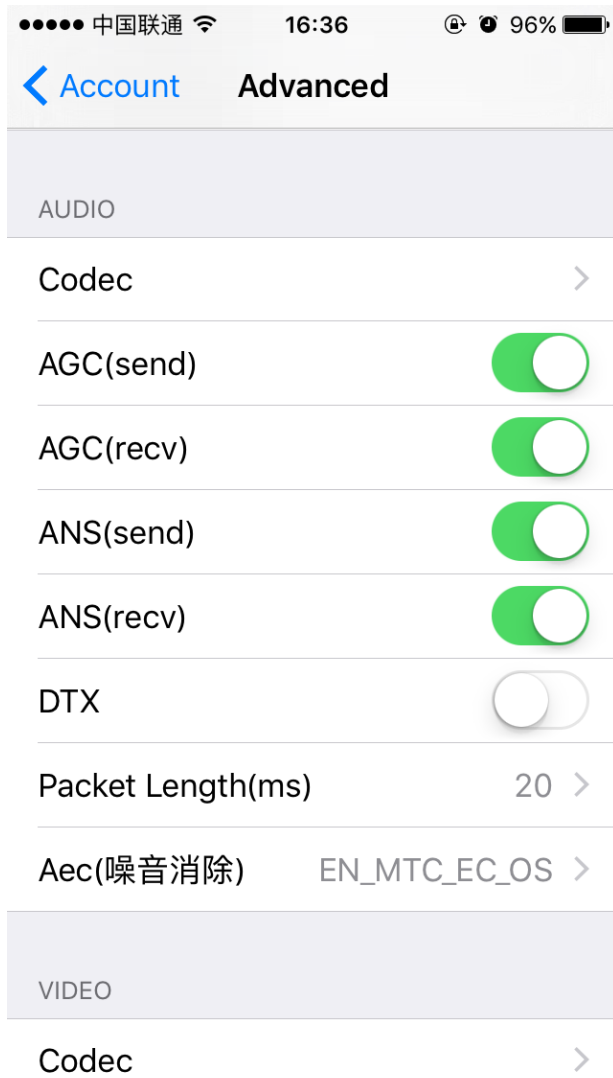
DTX

Packet Length(ms) 20 >

set the  
Username, Password,  
Authname, Server  
Address, Realm.

Set Register  
Type to Hotfixes

# Audio And Video Parameters selection



select the codecs and set the parameters

## Voice Parameters Intro

- Local Port, Remote Port: set local and remote transmission ports
- RTCP Mux: RTCP Port multiplexing
- Codec: select voice codec
- Payload: RTP payload value (not recommended to change)
- Bit rate: set bit rate (not recommended to change)
- Packet Length: set packet length (not recommended to change)
- AEC: acoustic echo canceller, including normal, OS, adaptive-FDE, adaptive-SDE, fixed delay mode  
FDE: formant delay estimate    SDE: spectrum delay estimate
- ANS: auto noise suppression
- AGC: auto gain control
- VAD/DTX: voice activity detection and discrete transmission
- AGC Target(dB): AGC target value with -10dB by default

## Video Parameters Intro

- Local Port, Remote Port: set local and remote transmission ports
- RTCP Mux: RTCP port multiplexing
- Codec: select video codec
- Payload: RTP payload value (not recommended to change)
- Bit rate: set bit rate
- Frame Length: set video frame rate
- Resolution: select video resolution
- Camera: select camera(front/back/off)
- ARS: auto rate sensing to make bit rate adaptive to network
- Resolution control: auto control resolution according to network
- Framerate control: auto control frame rate according to network
- Save mode: save network band width mode

# Video Parameters Intro

## **Transmission enhancement:**

- Color Enhancement: close by default
- FEC/RED: forward error correction
- NACK: Negative Acknowledgement
- TMMBR: temporary maximum media stream bit rate request, need be used with ARS
- FIR: full intra request
- VP8 RPSI: reference picture selection for VP8 codec
- H.264 small nalu: small network abstract layer unit for H.264, which means a small nalu packet is less than a UDP packet
- Key Period: interval between two key frame, use values by default
- Render Mode: video render mode(surface/openGL ES 2.0)



# Settings

## ACCOUNT

+8616000000489



Add Account... >

Confirm be

Vibrate whi

Quick Dial >

Ringtone

Music Box >

UDP Run in Background



Feedback >



Messages



Recents



Contacts



Keypad



Settings

Press this account to login

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+8616000000489

✓ +86 160000000486 

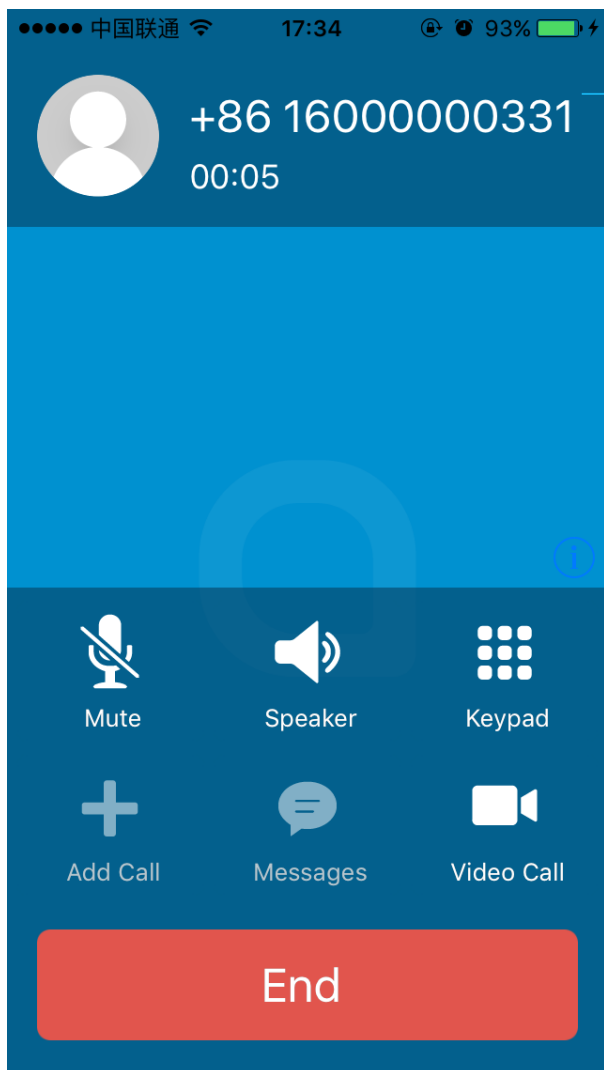
Add to Contacts



Messages Recents Contacts **Keypad** Settings

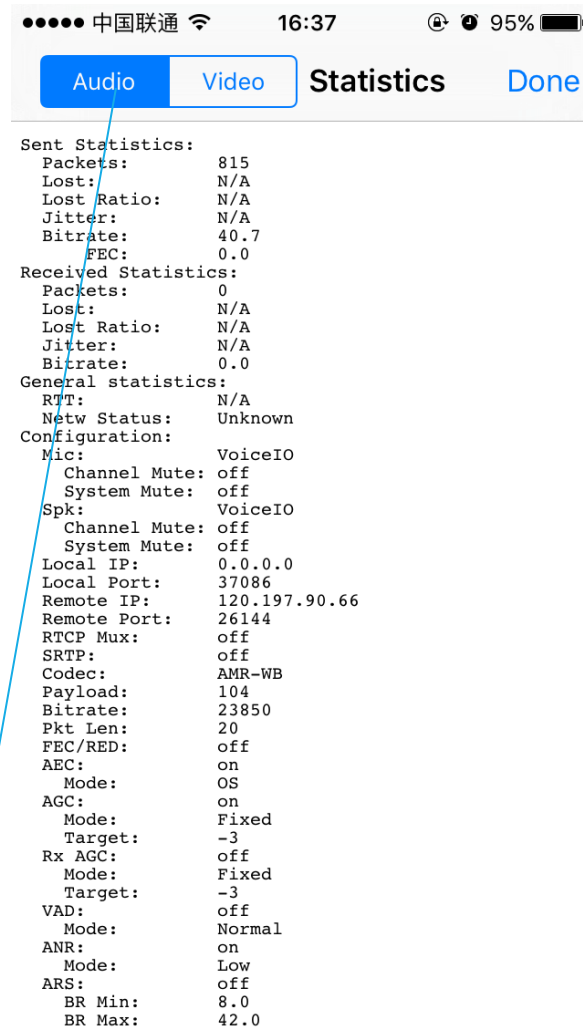
Input the called party sip account. Press button to video call.

# Statistics detection



Press this button to check out the data like bitrate, packet lost, and resolution, etc., when the demo is running

Press this button to switch to voice data.



**Thank you**