



**IPVoice – VoIP Test Software for iPhone, iPod Touch, and iPad
Quick Start Guide**

**Adaptive Digital Technologies, Inc.
November 29, 2012
Issue 3**

Revision History

<u>Issue</u>	<u>Revision History</u>	<u>Date</u>	<u>Initials</u>
1	Initial Version	2/16/2011	SDK
2	Updated at release of V1.20	4/1/2011	SDK
3	Updated to V4 and added SIP	11/29/2012	PPS

Table of Contents

1. Introduction.....	1
2. Features	2
3. Initial Setup.....	3
3.2 Application Settings.....	3
4. Running a Test	4
4.1 SIP Call	4
4.2 VoIP Conference.....	5
4.3 IP Phone Test	5
5. Internal Operation (for those who are interested)	7

1. Introduction

IPVoice test software, which runs on iPhone, iPod Touch, and iPad, exercises Voice over IP (VoIP) functionality over a WiFi or mobile data network. For the purpose of this document, when iPhone is used, it refers to iPhone, iPod Touch, or iPad interchangeably. IPVoice enables the user to set up a VoIP voice connection between multiple iPhones using direct IP to IP connections over WiFi. It also enables the user to make and receive SIP voice calls over WiFi or mobile data network. It is intended to be used as test equipment rather than as a full featured SIP phone application.

The IPVoice test software can be used as a piece of test equipment to facilitate the testing of VoIP phones and gateways.

Note: This application is not a typical mass-market consumer application. It is intended to be used by engineers who develop VoIP equipment or by engineering students.



Figure 1-1 below shows IPVoice in a configuration where it is used to test a wired IP Phone.

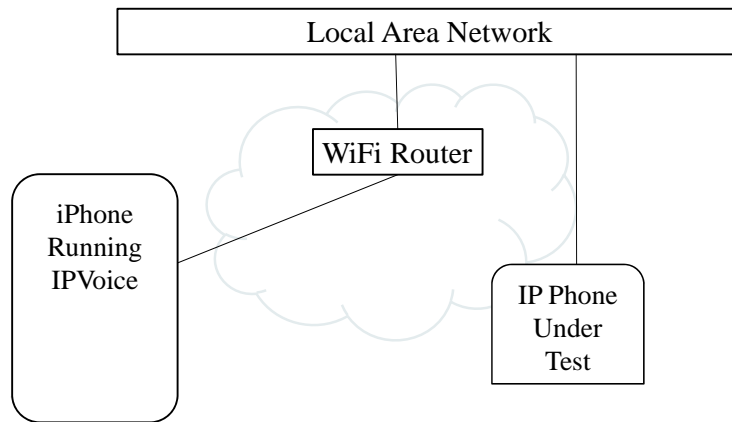


Figure 1-1 IP Phone Test Configuration

Figure 1-2 below shows IPVoice in a configuration where it demonstrates iPhone to iPhone VoIP communication via WiFi.

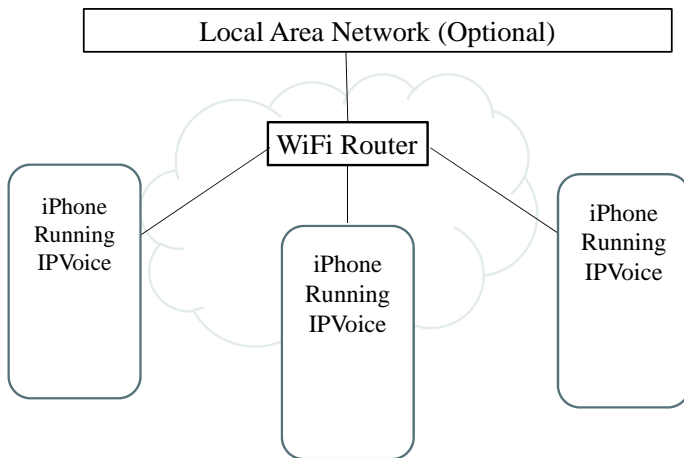


Figure 1-2 IPVoice Conference Configuration

2. Features

The following VoIP features are included in IPVoice:

- SIP
- RTP/Jitter Buffer
- SRTP
- Conferencing (up to 4 users)
- G.711 mu-law and a-law with packet loss concealment
- G.729A 8 kbps speech compression
- G.722 16 kbps speech compression
- Noise Reduction
- Acoustic Echo Cancellation
- Automatic Gain Control
- Tone Generation
- 10, 20, and 40 msec packet times

The following test features are included in IPVoice:

- Tone transmit
- Tone receive
- CSS transmit
- CSS receive
- Acoustic Delay Measurement

3. Initial Setup

3.1 WiFi Network Preparation

WiFi network preparation can be skipped if the IPVoice application is used only for SIP calls over the mobile data network.

1. Set up the network as shown in figure 1-2
2. Join the WiFi network
 - a. Go to Settings->WiFi.
 - i. Follow steps as indicated in below illustration.



Note: Images for sample set-up only, your IP information will vary.

If the phone is already connected to the desired network, touch that network and record the phone's IP address and skip to Application Settings.

- b. Choose the appropriate network. If the network is encrypted choose Other.
- c. Enter the name and security password for that network. Then click join
- d. Once you have joined, touch the network and record the IP address of the phone.

3.2 Application Settings

1. Make sure the IPVoice application is installed on the iPhone.
2. Run the IPVoice application.
3. Select the Settings tab.
4. In the General category, enter a value for the User Name. This value is used to identify the iPhone to other iPhones running the IPVoice application on the WiFi network.
5. In the General category, modify the following values only if needed. The default values are usually sufficient.
 - Peer Port: The UDP port number used for IPVoice to IPVoice communication. This value must be the same on all iPhones.
 - RTP Port: The UDP port number used by this iPhone for sending and receiving RTP packets. This is the port other devices should send to and expect to receive from when communicating with this iPhone.
 - SIP Port: The UDP port number used by this iPhone for sending and receiving SIP messages.
6. Enter SIP account information in the SIP category. This step can be skipped if not making or receiving SIP calls.
 - a. Enter the SIP server's name or IP address in the Server field.
 - b. Enter the SIP server's port number in the Port field. Use a value of 5060 unless known to be otherwise.



- c. Enter the SIP domain name in the Domain field.
 - d. Enter the SIP User Id in the User field.
 - e. Enter the SIP user's password in the Password field.
7. In the VoIP Engine category, modify the following values only if needed. The default values are usually sufficient.
- JitterBuf Mode: The jitter buffer mode (0 = static, 1 to 12 = adaptive).
 - JitterBuf Size: The jitter buffer size (units of msec).
 - JitterBuf Min: The minimum adaptive jitter buffer size (units of msec).
 - JitterBuf Max: The maximum adaptive jitter buffer size (units of msec). The largest value is 500.
 - Send Tone Level: The power level for tones generated toward other devices in units of dBm (-20 to 0).
 - User Tone Level: The power level for tones generated toward the earpiece/speaker in units of dBm (-20 to 0).
 - AGC Power Low: The automatic gain control's target power level low limit in units of dBm (-30 to 0).
 - AGC Power High: The automatic gain control's target power level high limit in units of dBm (-30 to 0).
 - AGC Max Gain: The automatic gain control's maximum gain in units of dB (0 to 23).
 - AGC Max Loss: The automatic gain control's maximum loss in units of dB (0 to 23).
 - AGC Low Signal: The automatic gain control's low signal level threshold in units of dBm (-65 to -20).
 - SRTP Mast Key: A string of 32 hexadecimal characters representing the 128 bit SRTP Master key. Each character represents 4 bits. Invalid or missing characters are interpreted as zero. Excess characters are ignored. This value is used only when SRTP is enabled.
 - SRTP Salt Key: A string of 28 hexadecimal characters representing the 112 bit SRTP Salting key. Each character represents 4 bits. Invalid or missing characters are interpreted as zero. Excess characters are ignored. This value is used only when SRTP is enabled.
 - Log Filter: A collection of bit flags to enable logging of information for test and debug by Adaptive Digital engineers.

4. Running a Test

4.1 SIP Call

1. Start or resume the IPVoice application.
2. Select the SIP tab.
3. Enable or disable use of SIP with the On/Off switch. While enabled, IPVoice will register with the SIP server identified in the Settings tab.
4. Observe the Ready/Not Ready status display to determine if IPVoice successfully registered with the SIP server. It can take several seconds for registration to complete.
5. When SIP is ready, a SIP call can be placed by entering the target SIP User Id (or phone number) and touching the Call button.
6. While SIP is ready, a SIP call can be received. An alert is displayed when an incoming SIP call occurs. Touch Accept to answer the call or Reject to not answer the call.
7. While a SIP call is active;
 - a. Muting of the microphone can be toggled on/off by touching the Mute button.
 - b. Use of the speaker can be toggled on/off by touching the Spkr button.

- c. The keypad can be used for IVR.
- d. The call can be ended (hung up) by touching the End button.

4.2 VoIP Conference

1. Verify WiFi connectivity and start or resume the IPVoice application on all iPhones.
2. Select the VoIP tab.
3. Turn off both Activate switches.
4. Select the transmit packet encoding by selecting Codec in the left column of the picker and the value in the right column. Note: changing this value while a conference is active has no effect.
5. Select the transmit packet time by selecting TxPktTime in the left column of the picker and the value in the right column. Note: changing this value while a conference is active has no effect.
6. Select the Peers tab.
7. The user name and associated RTP transport address of other iPhones running the IPVoice application will be listed. It can take up to 10 seconds for a phone to be listed.
8. A check mark appears to the right of users to which this iPhone is communicating. A maximum of 4 users can be enabled at the same time. All enabled users are conferenced together.
9. Touch a listed user without a check mark to enable communication with that user. That user must also enable communication with you in order to join the conference. An alert is displayed when another user has enabled communication with you. Touch Accept to enable communication with that user or Reject to leave communication disabled.
10. Touch a listed user with a check mark to disable communication with that user. That user's communication with you will be disabled automatically.
11. While a conference is active (at least one enabled user), select the VoIP tab to make the following changes.
 - a. Enable or disable use of the speaker with the Speaker switch.
 - b. Mute or unmute the microphone by selecting TxMute in the left column of the picker and On or Off in the right column.
 - c. Mute or unmute the earpiece/speaker output by selecting RxMute in the left column of the picker and On or Off in the right column.

4.3 IP Phone Test

1. Verify WiFi connectivity and start or resume the IPVoice application.
2. Select the Peers tab.
3. Disable communication with all peers by touching each listed user with a check mark.
4. Select the VoIP tab.
5. Turn off both Activate switches.
6. Enter the IP address and RTP port number of the device under test in the Transport Address field. The format is #.#.#.# (e.g. 172.16.0.120:5004). A second device can be entered forming a conference.
7. Use the picker to select characteristics that can't be changed while activated.
 - a. Select the transmit packet encoding by selecting Codec in the left column and the value in the right column.
 - b. Select the transmit packet time by selecting TxPktTime in the left column and the value in the right column.
 - c. Enable or disable noise reduction by selecting NR in the left column and On or Off in the right column.
 - d. Enable or disable acoustic echo cancellation by selecting AEC in the left column and Native, VE, or Off in the right column. Native uses the AEC built into the iPhone. VE uses the AEC included in IPVoice.

- e. Enable or disable automatic gain control by selecting AGC in the left column and On or Off in the right column. Automatic gain control characteristics are specified with the AGC fields in the VoIP Engine category on the Settings tab.
- f. Enable or disable use of SRTP encryption by selecting SRTP in the left column and On or Off in the right column. The SRTP keys are specified with the SRTP Mast Key and SRTP Salt Key fields in the VoIP Engine category on the Settings tab. The same key values are used for both transmitted and received RTP packets. The SRTP key values must match those of the device under test.
- g. Select the test mode by selecting TestMode in the left column and the value in the right column.
 - Off: Normal voice processing.
 - Noise Sound: Used to determine the acoustic and buffering delay between the speaker output path and the microphone input path. The results are only available using a developer's console in XCode.
 - Rx Tone: Plays a tone out to the speaker at a level of -13 dBm. This can be used for calibration.
 - Tx Tone: Sends a tone to the device under test at a level of -13 dBm. This can be used for Calibration
 - Rx CSS: Sends a noisy signal to the speaker at a level of -10 dBm. This can be used for calibration and frequency response measurements.
 - Tx CSS: Sends a noisy signal to the device under test at a level of -10 dBm. This can be used for calibration and frequency response measurements.
 - Pass Thru: Turns off a number of voice-path features such as gain stages.
 - Tx DTMF Rly: Sends a repeating sequence of DTMF events via RFC2833 RTP packets to the device under test. This can be used to test a VoIP gateway's proper regeneration of DTMF signals.
8. Turn on the Activate switch next to the transport address of the device(s) under test to enable communication.
9. While a a transport address is activated (or prior to activation);
 - a. Enable or disable use of the speaker with the Speaker switch.
 - b. Mute or unmute the microphone by selecting TxMute in the left column of the picker and On or Off in the right column.
 - c. Mute or unmute the earpiece/speaker output by selecting RxMute in the left column of the picker and On or Off in the right column.
 - d. Set the microphone gain by selecting MicGain in the left column of the picker and the value in the right column.
 - e. Set the earpiece/speaker gain by selecting SpkrGain in the left column of the picker and the value in the right column.
10. While a a transport address is activated;
 - a. Select the play mode by selecting PlayMode in the left column of the picker and the value in the right column. The played data replaces the microphone input and is thus sent to the device under test.
 - Off: Microphone input is used (normal mode).
 - Tone: A user defined single frequency tone is played continuously. The frequency is specified with the Play Tone Freq field in the Test category on the Settings tab.
 - File: A user defined PCM data file is played continuously. The file name is specified with the Play File Name field in the Test category on the Settings tab.
11. The following statistics can be viewed in the Statistics category on the Settings tab.
 - RTP Rx Jitter: The most recent RTP receive jitter estimate (units of samples).
 - RTP Rx Delay: The current RTP receive jitter buffer delay (units of samples).
 - RTP Rx Rcvd: The number of RTP packets received.

- RTP Rx Missing: The number of RTP packets missing (expected but not received).
 - RTP Rx Dropped: The number of RTP packets received but discarded due to being invalid or too late.
 - RTP Rx Played: The number of RTP packets decoded and played.
 - RTP Tx Sent: The number of RTP packets sent to the device under test.
 - Port Received: The number of packets received on the RTP port. This value is normally the same as RTP Rx Rcvd. A difference may indicate packets are being sent to the port from a source other than the device under test.
 - Mic Level: The power level measured at the microphone input (units of dBm).
 - Spkr Level: The power level measured at the earpiece/speaker output (units of dBm).
12. Turn off the Activate switch next to the transport address of the device(s) under test to disable communication.

5. Internal Operation (for those who are interested)

Figure 5-1 is a block diagram of the overall IPVoice application.

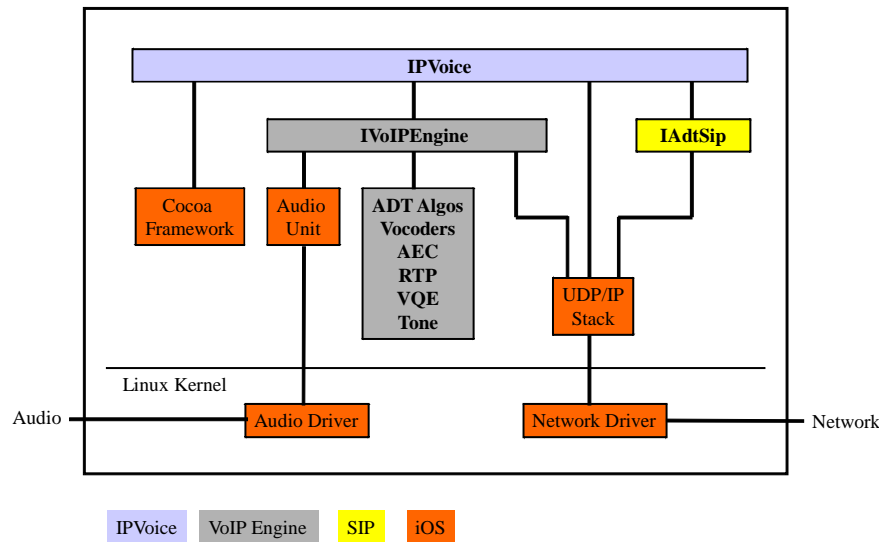


Figure 5-1 – System Architecture

In the receive path (Network to Audio PCM), the Network Driver receives UDP packets. The UDP packets matching the RTP port are passed up through the UDP/IP stack to IVoIP Engine. IVoIP Engine performs an RTP receive function and places the packet in the jitter buffer for later use.

In the transmit path (Audio PCM to Network), audio is fed into the Linux Audio driver as PCM samples. The PCM samples are passed to the iOS Audio Unit. When the Audio Unit buffer is filled, the Audio Unit sends a buffer of PCM data to IVoIP Engine which frames the PCM samples for performing speech compression, echo cancellation, RTP packetization, etc. IVoIP Engine then transmits RTP packets over the network via the UDP/IP stack and Network Driver.

At the same time that a transmit PCM frame is processed, IVolPEngine polls the jitter buffer to determine if there are any packets that contain data that needs to be decoded and played out to the audio interface. If so, IVolPEngine decodes the packet and handles any other necessary DSP functionality. It passes the resulting PCM frame to the audio interface via the Audio Unit and Audio Driver.