

1. Software Version

This manual is for iPhone and Android software version 1.0.0.

2. iPhone User Manual

2.1 Sign in/Sign out

You must sign-in before you can use the application and make or receive calls. When the application launches for the first time, you will be presented with a sign-in view for you to sign in. After you sign in successfully, your username and password will be remembered automatically until you sign out. When you launch the application the next time, the application will sign you in automatically.

If you do not want the application to remember your username and password, you can sign out from the Settings tab. After you sign out, you will not be able to make or receive calls, and you will be required to sign in the next time you launch the application.

2.1.1 Register

#	Action/Event	Screenshot	Application Response
1	Launch application before signing in successfully.	No SM T 12:15 PM 100% THE PROPERTY NetTalk by True 100 Password NetTalk Smart Choice Calling If you have a NetTalk Account, please sign in If you are a new user, register here to get a NetTalk Account. O Sign in Register P A S D F G H J K L T X C V B N M 23 123 ## Space return	Splash screen appears for a while, followed by Sign In view and a Register pop-up. You can sign-in with an existing account or register for a new account.
2	Tap "Sign In" to sign in if you already have a NetTalk account.	No SM \$ 8:49 AM 98% CO PROPERTY NETTAIK by True Sopn in Username Password NetTalk Q W E R T Y U I O P A S D F G H J K L C Z X C V B N M CX 123 \$ space return	Sign In view appears. To sign-in, follow procedures in 2.1.2. To register, tap "Register" on top-left corner.
3	Tap "Register" to register for a NetTalk account if you do not have a NetTalk account.		Browser loads and goes to registration website.

2.1.2 **Sign In**

#	Action/Event	Screenshot	Application Response
1	Launch application after signing in successfully.	NetTalk by True Username Password NetTalk Q W E R T Y U I O P A S D F G H J K L 2 X C V B N M 3 123 space return	Splash screen appears for a while, followed by Sign In view.
2	Enter your username and password.	NetTalk by True Sgn in 9810005367 NetTalk by True Sgn in 9810005367 1 2 3 4 5 6 7 8 9 0 - / : ; () \$ & @ " ABC Space return	
3	Ensure that you are connected to a Wifi or cellular data network. Tap "Sign In" button at top right.	Talk stM1-3GSM © 11:34 AM Connected 1 2 3 ABC DEF 4 5 6 MNO 7 8 9 TUV WXYZ ** 0 # ** ** ** ** ** ** ** ** **	Network busy indicator appears for a while and then disappears. Application attempts to register with SIP registry server and succeeds. Sign In view disappears. Keypad view appears. SIP status label on top left shows "Connected".

2.1.3 Sign Out

#	Action/Event	Screenshot	Application Response
1	Action/Event Tap "Settings" tab item at bottom right.	Screenshot	Application Response Settings view appears. View title shows username.
		Keypad History Contacts Settings	



2.2 Call

After you sign in successfully, you can make and receive calls. During the call, you can mute/un-mute the microphone, turn the speaker on/off, show/hide the keypad. When you are done with the call, you can end the call.

2.2.1 Make a call

If you are signed-in and the network is available (status shows "Connected"), you can make calls.



Input a number to dial Call view appears. zxcvb with keypad and tap Call identification label shows dialed "Call" button at bottom number. middle. Call status label shows "calling...". **◄**))) П The other party answers Call status label changes to show the the call. duration of the call. *** **◄**))) П

2.2.2 Receive a call

If the application is running in the foreground and is active, the network is available, and you are signed-in (status shows "Connected"), you can receive calls. However, if the application is running in the background, you cannot receive calls. When there is an incoming call, the phone rings, and the incoming call view appears. You can decline or answer the call.

2.2.2.1 Decline a call

#	Action/Event	Screenshot	Application Response
1	Incoming call	जा। M1-3GSM ♀ 11:26 AM ● 93% 🖾	Incoming call view appears.
		zxcvb	Phone rings.
		Decline Answer	

Tap "Decline" to decline the call.

Call is declined. Phone stops ringing. Incoming call view disappears. Keypad view appears.

2.2.2.2 Answer a call

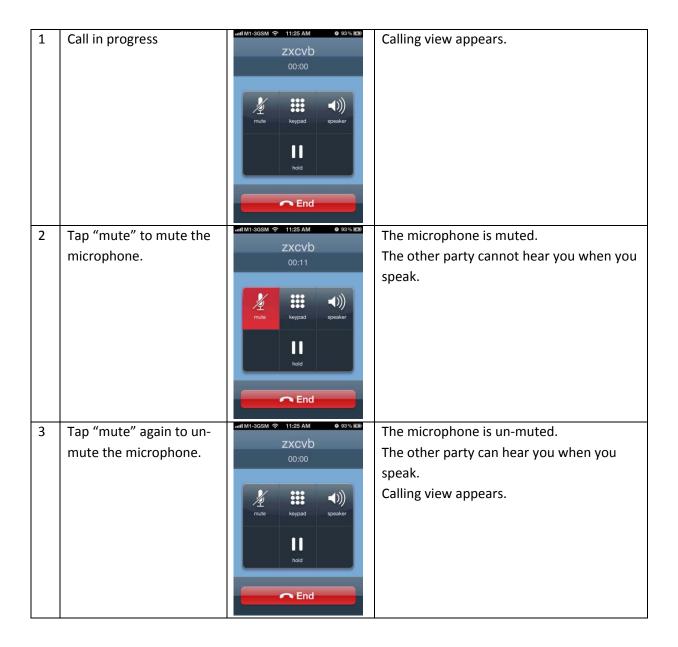
#	Action/Event	Screenshot	Application Response
1	Incoming call	anti M1-3GSM ♀ 11:26 AM	Incoming call view appears. Phone rings.
2	Tap "Answer" to answer the call.	Decline Lanswer Land M1-3GSM 11:25 AM 2XCVb 00:00 Land Land Land Land Land Land Land Land	Call is answered. Phone stops ringing. Incoming call view disappears. Call view appears.

2.2.3 During the call

When a call is in progress, you can mute and un-mute the microphone, turn the speaker on and off, and show and hide the keypad.

2.2.3.1 Mute/Un-mute microphone

#	Action/Event	Screenshot	Application Response
	<u>-</u>		• •



2.2.3.2 On/off speaker

#	Action/Event	Screenshot	Application Response	
1	Call in progress	ZXCVb 00:00 will M1-3GSM ZXCVb 00:00 Signature Signature	Calling view appears.	
		← End		



2.2.3.3 Show/Hide keypad

#	Action/Event	Screenshot	Application Response
1	Call in progress	anti M1-3GSM	Calling view appears.
2	Tap "keypad" in the middle to show the phone keypad.	TXCVb O0:16 1 2 3 ABC DEF 4 5 6 GHI JKL MNO 7 8 9 PORS TUV WXYZ ★ 0 ## Hide Keypad	Phone keypad appears.

Tap "Hide Keypad" at the bottom right to hide the phone keypad.

Tap "Hide Keypad" at the bottom right to hide the phone keypad.

Tap "Hide Keypad" at the bottom right to hide the phone keypad.

Phone keypad disappears.

Calling view appears.

2.2.4 End a call

#	Action/Event	Screenshot	Application Response
1	Call in progress	will M1-3GSM	Calling view appears.
2	Tap "End" to end call.	att. M1-3GSM © 11:34 AM © 100% © Connected 1 2 3 DEF 4 5 6 MNO 7 8 9 WXYZ * 0 # + Talk * Talk Kapad Hatery Contacts Settings	Call is ended. Call view disappears. Keypad view appears.

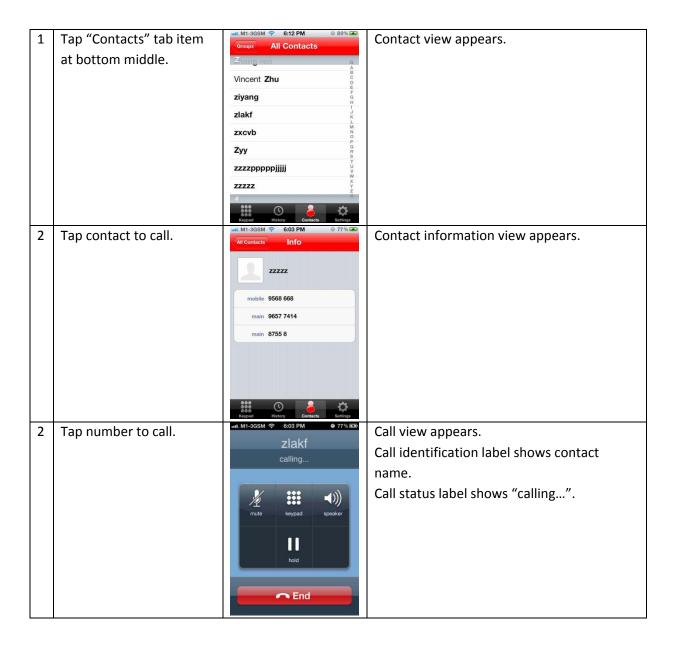
2.3 Contacts

The application includes a contact list that is integrated with the iPhone's contact list. In addition to the standard iPhone contact functions such as viewing, editing, deleting contacts, you can make a VoIP call to a contact, and create a new contact from a VoIP number, or add a VoIP number to an existing contact.

2.3.1 Dial a contact

You can make a call to a contact from the contact list.

# Action Screenshot Expected Result	
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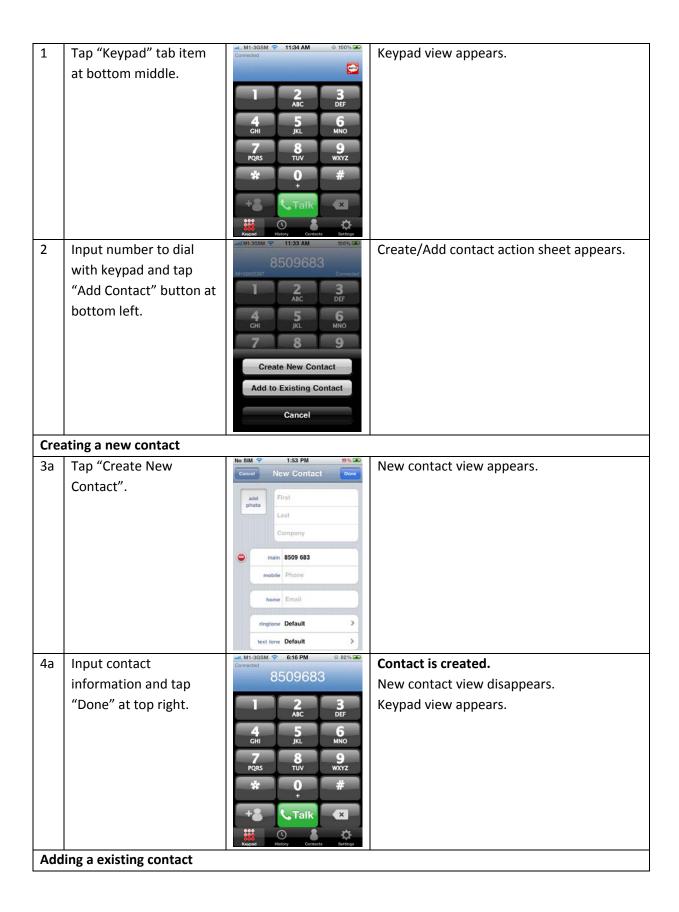
2.3.2 Create a new contact/Add to existing contact

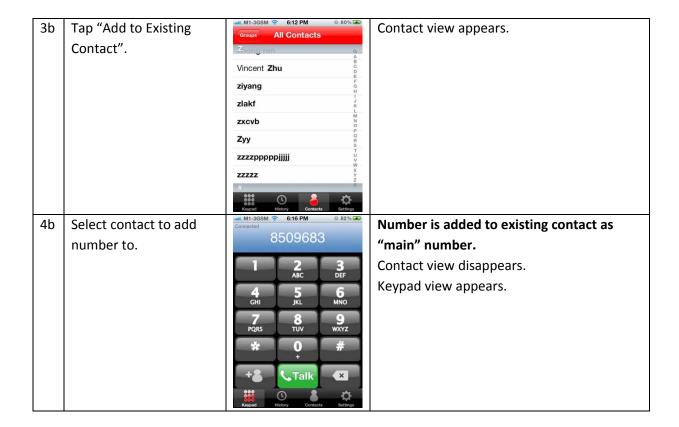
You can add a new contact to the contact list or add a number to an existing contact in contact list. There are several ways to do this.

2.3.2.1 From keypad

You can create a new contact with the number entered into the keypad or add the number to an existing contact.

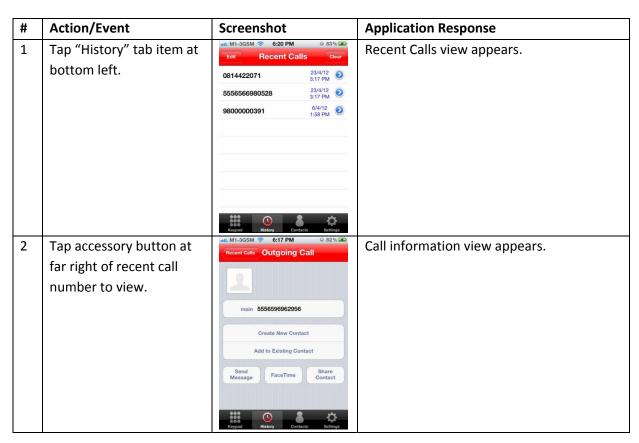
#	Action/Event	Screenshot	Application Response
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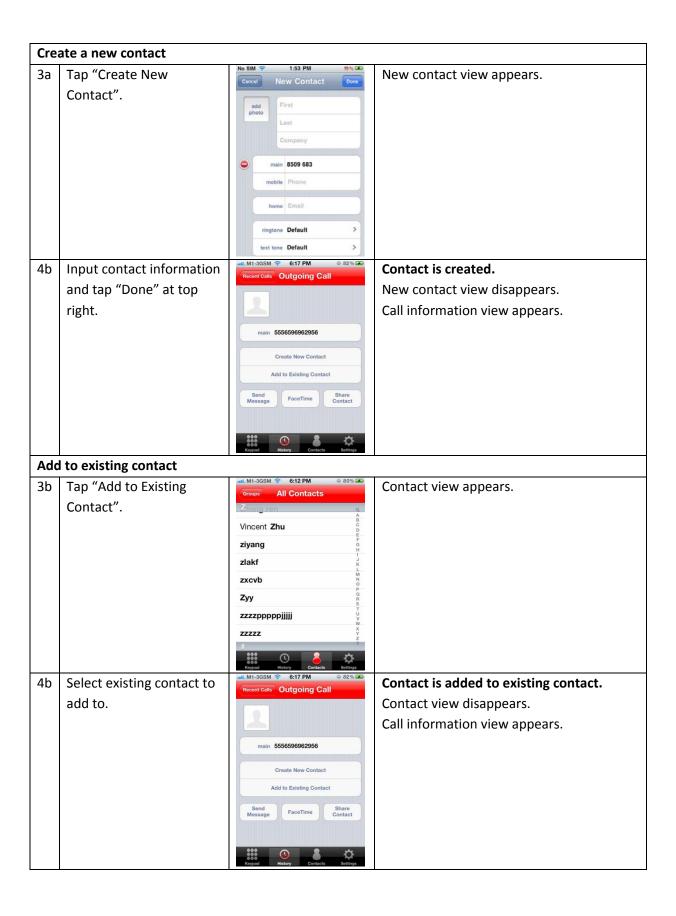




2.3.2.2 From recent calls list

You can create a new contact with a number in the recent calls list or add the number to an existing contact.

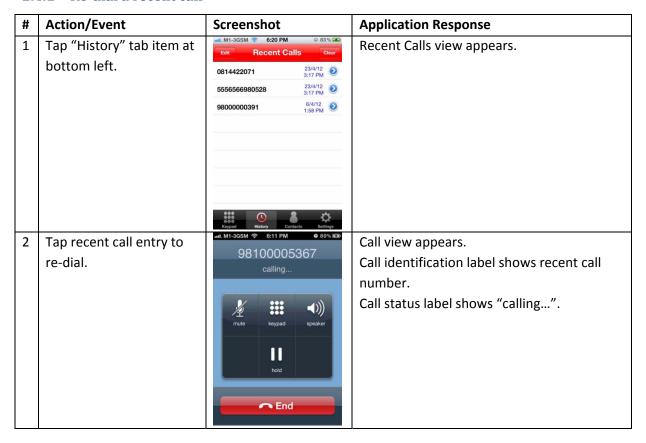




2.4 Recent Calls

The application includes a recent calls list. Whenever a call is completed, it is added to the recent calls list. You can re-dial a recently received/dialed number, view a recent call, edit a recent call, and clear all recent calls.

2.4.1 Re-dial a recent call



2.4.2 View a recent call

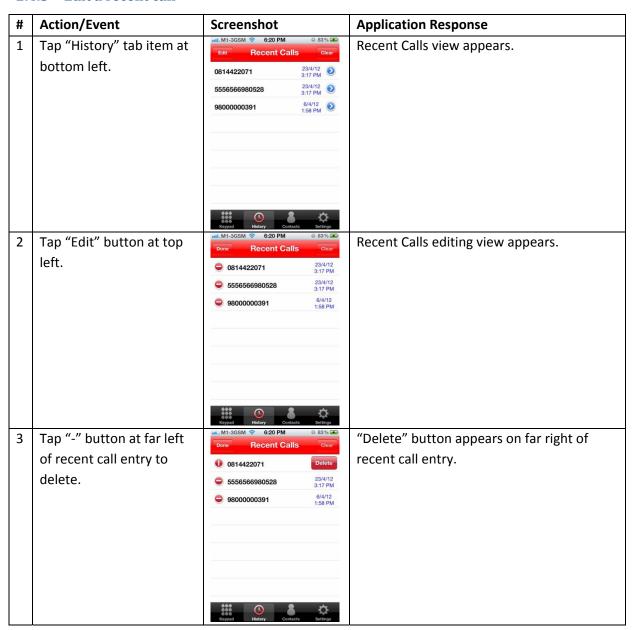
#	Action/Event	Screenshot		Application Response
1	Tap "History" tab item at	etil. M1-3GSM © 6:20 PM © 83 % SE Edit Recent Calls Clear		Recent Calls view appears.
	bottom left.	0814422071	23/4/12 3:17 PM	
		5556566980528	23/4/12 3:17 PM	
		98000000391	6/4/12 1:58 PM	
		Keypad History Con	tacts Settings	

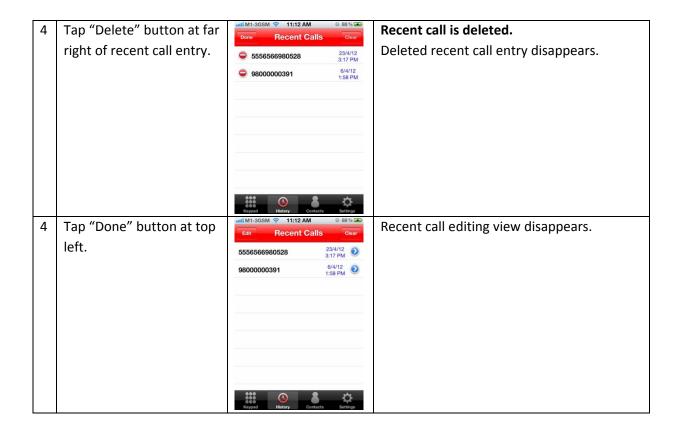
2 Tap accessory button at far right of recent call entry to view.

Call information view appears.

Call information view appears.

2.4.3 Edit a recent call





2.4.4 Clear recent calls

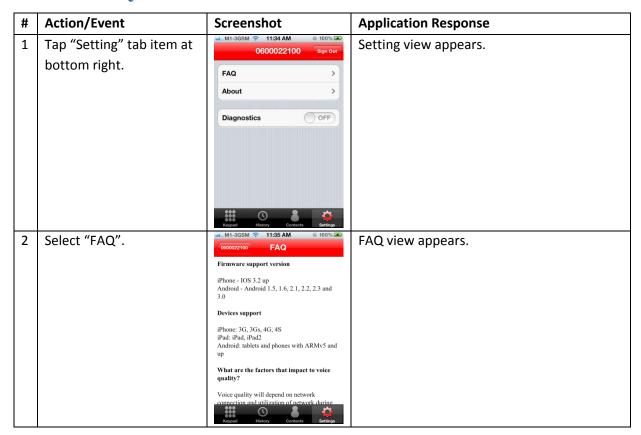
#	Action/Event	Screenshot	Application Response
1	Tap "History" tab item at bottom left.	Mail. M1-3GSM	Recent Calls view appears.
2	Tap "Clear" button at top right.	### MIN-305M © 11:13 AM	Clear All Recents action sheet appears.



2.5 FAQ

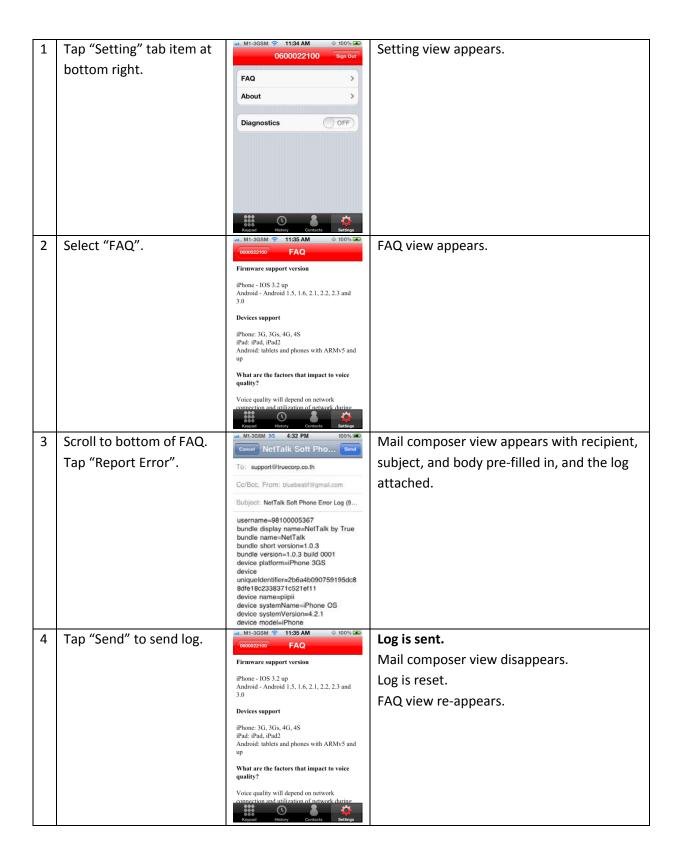
An FAQ is included with the application. The FAQ contains answers to frequently asked questions. If you encountered a problem with the FAQ, you should consult the FAQ first to see if it answers your question. If the FAQ does not address your problem, and an error was generated by the application, you can send an error report. You need to set up your mail accounts on the phone first before sending an error report.

2.5.1 View FAQ



2.5.2 Report Error

#	Action/Event	Screenshot	Application Response
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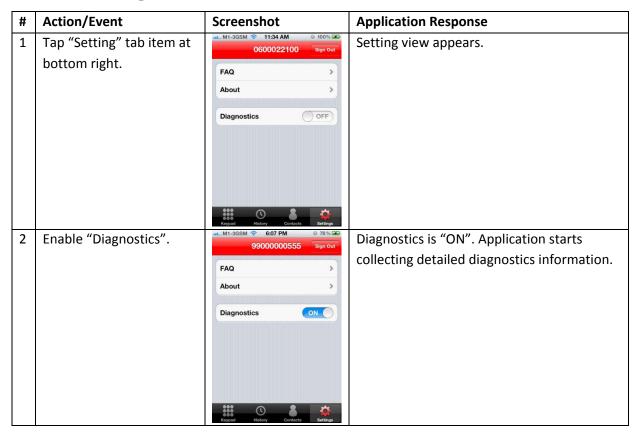


2.6 Diagnostics

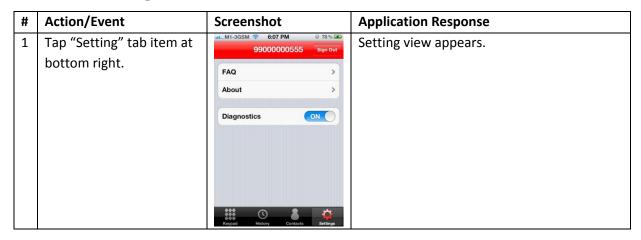
By default, the application only collects error information in the log. If there are no errors, the log will be empty. However, if you encounter problems with the application, you can enable diagnostics to gather more information in the log, including detailed SIP protocol messages (including IP

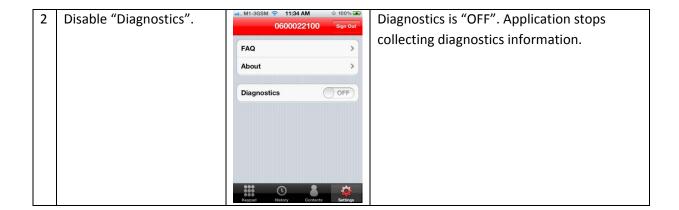
address), call media statistics, network information, and device information. When diagnostics is enabled, detailed diagnostics information is collected and written to the log until diagnostics is disabled. You should only enable diagnostics only if you encounter problems with the application, as gathering diagnostics information consumes resources such as memory, storage space, and battery. You should remember to disable diagnostics after you have gathered the necessary information for the problem you encounter.

2.6.1 Enable diagnostics



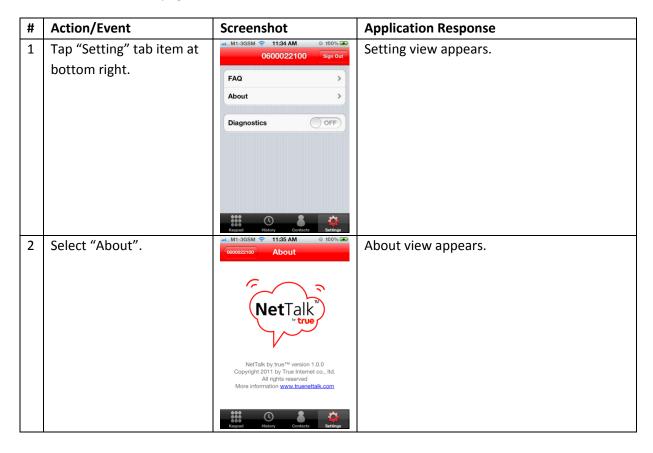
2.6.2 Disable diagnostics





2.7 About

You can view information about the application, including the software version, and a link to the website, on the about page.



2.8 Volume

These control the volume amplification and the microphone sensitivity amplification. Changes in these values will be stored in the phone's memory.

2.9 Troubleshooting

#	Signs and Symptoms	Screenshot	Possible problems	Solution
Sig	gn-In			
1	"You must be connected to a Wi-Fi or cellular data network." alert view pops up.	NetTalk by True Sign In 98100005367 You must be connected to a Wi-Fill or ceilular data network. OK	No network available	Connect to a Wifi or cellular data network.
2	"Incorrect username or password" alert view pops up.		Incorrect username or password	Input correct username and password.
M	ake a call			
1	Call cannot be made. "No Network Available" alert view pops-up.	Min-acsM Genvice Unavailable 1 2 3 ABC No Network Available You must be connected to a Wi-Fi or cellular data network to make a call. OK Talk Agond Astron. Costacts Costacts Costacts Costacts	No network available.	Connect to a Wifi or cellular data network.
Re	port error		<u> </u>	<u>-</u>
1	"No error log found" alert view pops up.	Section 2016 And Sectio	Error log is empty because there is no error or error report has already been sent.	Enable diagnostics to collect more information.

2.10 FAQ

#	Question	Answer	
1	Firmware support	iPhone - IOS 3.2 up	
	version	Android - Android 1.5, 1.6, 2.1, 2.2, 2.3 and 3.0	
2	Devices support	iPhone: 3G, 3Gs, 4G, 4S	

		iPad: iPad, iPad2
		Android: tablets and phones with ARMv5 and up
3	What are the factors that impact to voice quality?	Voice quality will depend on network connection and utilization of network during your call. Normally WiFi is the best network connection for VoIP call then 3G, Edge and GPRS in order. Sometime you may experience voice quality on Edge connection better than WiFi connection. This is because during your conversation, WiFi connect utilization may greater than Edge connection which effect to voice quality.
4	I have problems when registering.	Step 1 - Checking network connection make sure you already connected to internet via GPRS/Edge/3G/WiFi Step 2 - Checking your username/password is correct. To verify your username/password you can connect to http://www.truenettalk.com/truetalk/index consumer.jsp and fill-in username/password. If your username/password is correct you will get account profile screen Step 3 - If you still can't sign on, please contact call center 1686
5	There is a "Service Unavailable" status message.	The application cannot contact the server. This may be because your phone's network connection is unavailable or because the server is unavailable. Make sure your phone's network connection is available. The application will contact the server automatically when the network connection is available.
6	I cannot hear the other party clearly.	Your phone's network connection may be be weak at your current location. The application depends on the quality of the network connection (3G, EDGE, GPRS or WiFi). Try to get a better network connection for your phone.
7	Call disconnect when you moving on your location	The application does not support hand-over between network, if your network connection change from 3G to WiFi, the call will be disconnected.
8	The application does not respond when I touch a button.	The application is slow to respond. Your phone might be slow and/or you have too many open applications and your phone is running out of memory. Wait for the application to respond. If application does not respond after a long time, close and restart the application. To free up memory for your phone, close some applications.
9	The application does not work.	Close and restart the application. If the application still does not work, please send an error report.

2.11 Diagnostics information

If the application is in diagnostic mode (**Enable Diagnostic** is enabled in the settings menu), the application will add detailed information to the log file.

2.11.1 Software Version and Device Information

Software version and device information is added the error report email whenever an error report is sent.

Sample Software Version and Device Information in the Error Report

```
Version: 1.0.6 build 0003

Information about your device:

Device unique identifier: fe0825521fd4aabf99d8364ef6900e5ef2a96bc6

Device system software: iPhone OS 4.2.1

Device model: iPhone

Device localized model: iPhone

Device platform: iPhone 3G
```

2.11.2 Connectivity information

When diagnostics is enabled, the application logs the current connectivity status, and whenever the connectivity changes. The possible connectivity types are listed below:

Connectivity Entries in the Log File

```
Connectivity type: Not available Connectivity type: Wifi Connectivity type: GPRS/EDGE/3G
```

2.11.3 Call statistics

When diagnostics is enabled, call statistics are logged every time the call session ends.

Sample Call Statistic Entry in the Log File

```
Line 1: Call time: 00h:00m:14s, 1st res in 21180 ms, conn in 21276ms
Line 2: SRTP status: Not active Crypto-suite: (null)
Line 3: #0 PCMA @8KHz, sendrecv, peer=85.17.186.6:54790
        RX pt=8, stat last update: 00h:00m:03.158s ago
Line 4:
               total 359pkt 57.4KB (71.8KB +IP hdr) @avg=36.5Kbps/45.7Kbps
Line 5:
Line 6:
               pkt loss=0 (0.0%), discrd=0 (0.0%), dup=0 (0.0%), reord=0 (0.0%)
Line 7:
                                                              dev
                     (msec) min
                                      avg
                                              max
                                                      last
Line 8:
               loss period: 0.000 0.000 0.000 0.000 0.000
Line 9:
               jitter : 1.250 24.051 92.125 16.750 12.611
Line 10:
            TX pt=8, ptime=20ms, stat last update: 00h:00m:05.286s ago
Line 11:
Line 12:
            total 633pkt 101.0KB (126.3KB +IP hdr) @avg 64.3Kbps/80.4Kbps
                 pkt loss=0 (0.0%), dup=0 (0.0%), reorder=0 (0.0%)
Line 13:
                     (msec) min avg max last dev
             loss period: 0.000 0.000 0.000 0.000 0.000 jitter : 55.250 76.625 98.000 98.000 21.374
Line 14:
Line 14: loss per
Line 15: jitter
Line 16: RTT msec
                          : 584.000 584.000 584.000 584.000 0.000
```

Line	Description
1	The duration of the call media session: 00h:00m:14s
	The delay before the first non-100 SIP response was received: 21180 ms
	The delay before call was confirmed (established): 21276ms
2	Shows the Secure RTP status. This is only applicable for secured calling.
3	Codec being used for the media session: PCMA @8KHz (G.711)
	'peer' is the address where RTP packets were transmitted: 85.17.186.6:54790
Line	4-9 show the statistics of the received (RX) RTP packet stream.
4	The last time the RX statistics was updated: 00h:00m:03.158s ago
5	Total number of RTP packets received: 359
	Total RTP payload size received: 57.4KB
	Total RTP payload + IP/UDP/RTP header size received: 71.8KB
	Average bitrate for the received audio: 36.5kbps
	Average bitrate including received packet headers: 45.7Kbps
6	Lost packets: 0 (0.0%)
	Discarded packets: 0 (0.0%)
	Duplicate packets: 0 (0.0%)
	Out-of-order packets: 0 (0.0%)
7	-
8	Shows the minimum, average, maximum, and last duration (in milliseconds) of packet loss burst.
9	Shows the minimum, average, maximum, and last jitter value (in milliseconds) of incoming RTP packets.
calcu	10-15 shows the statistics of the transmitted (TX) packet stream. TX statistics are actually late by the remote party, so if the remote party does not support RTCP, or if RTCP SR/RR ets were not received, TX statistics will be all zero.
10	The last time TX statistics was updated: 00h:00m:05.286s ago
11	Total number of RTP packets transmitted: 633
	Total RTP payload size transmitted: 101.0KB
	Total RTP payload + IP/UDP/RTP header size transmitted: 126.3KB
	Average bitrate for the transmitted audio: 64.3Kbps
	Average bitrate including transmitted packet headers: 80.4Kbps
12	Lost packets: 0 (0.0%)
	Duplicate packets: 0 (0.0%)
	Out-of-order packets: 0 (0.0%)
13	-
14	Shows the minimum, average, maximum, and last duration (in milliseconds) of packet
	I

	loss burst.
15	Shows the minimum, average, maximum, and last jitter value (in milliseconds) of incoming RTP packets
16	RTT is the round trip time, which is the time taken (in milliseconds) for a packet to travel from one end point to the other and return. This is a measure of latency. RTT measurement requires RTCP to be enabled. If no RTCP packets are sent/received, the RTT will show up as 0. The call session duration also has to be long enough for RTCP packets to be sent/received in order for the RTT to be computed.

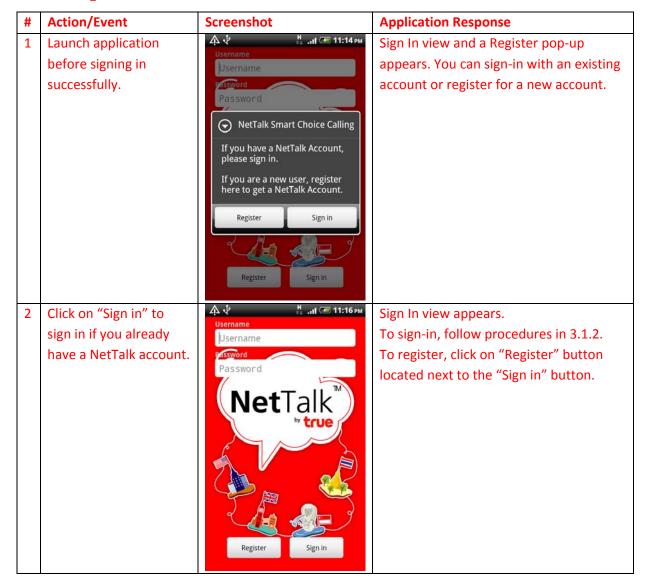
3. Android User Manual

3.1 Sign in/Sign out

You must sign-in before you can use the application and make or receive calls. When the application launches for the first time, you will be presented with a sign-in view for you to sign in. After you sign in successfully, your username and password will be remembered automatically until you sign out. The application supports background call receiving, so you will be registered with the SIP server even if the application is not in the foreground. When the phone is rebooted, the application will register with the SIP server automatically.

If you do not want the application to remember your username and password, you can sign out via the options menu. After you sign out, you will not be able to make or receive calls, and you will be required to sign in the next time you launch the application.

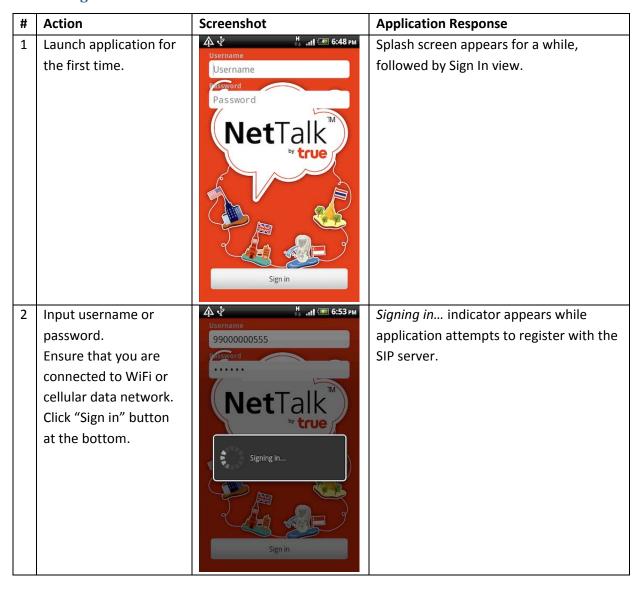
3.1.1 Register



3 Click on "Register" to register for a NetTalk account if you do not have a NetTalk account.

Browser loads and goes to registration website.

3.1.2 **Sign In**





3.1.3 Sign Out



Click "OK" on alert view.

Signing out... indicator appears for a while and then disappears.

Application attempts to de-register with SIP registry server and succeeds.

Sign In view appears.

3.2 **Call**

After you sign in successfully, you can make and receive calls. During the call, you can mute/un-mute the microphone, turn the speaker on/off, show/hide the keypad. When you are done with the call, you can end the call.

3.2.1 Make a call from Number Keypad

If you are signed-in and the network is available (status shows "Registered"), you can make calls.



2 Input number to dial with keypad and click "Call" button at bottom.



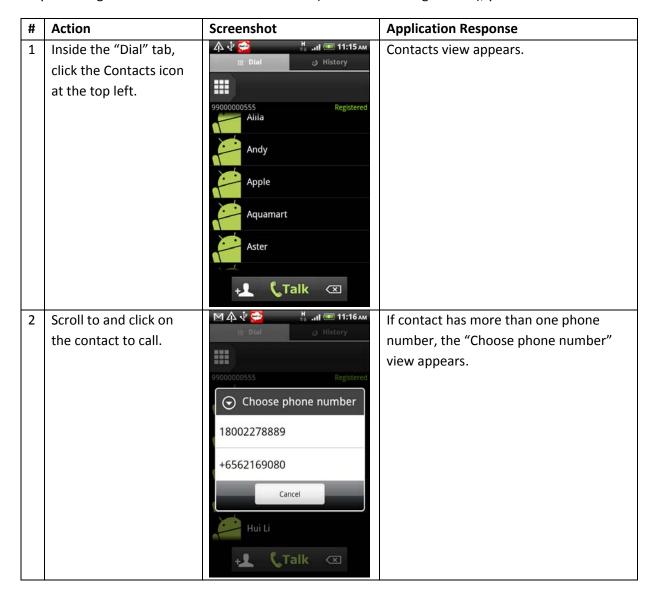
Call view appears.

Call identification label shows dialed number.

Call status indicator on Android status bar shows "Call in Progress".

3.2.2 Make a call from Contact List

If you are signed-in and the network is available (status shows "Registered"), you can make calls.



3 Click on a phone number on the "Choose phone number" view then click the "Call" button.



Call view appears.

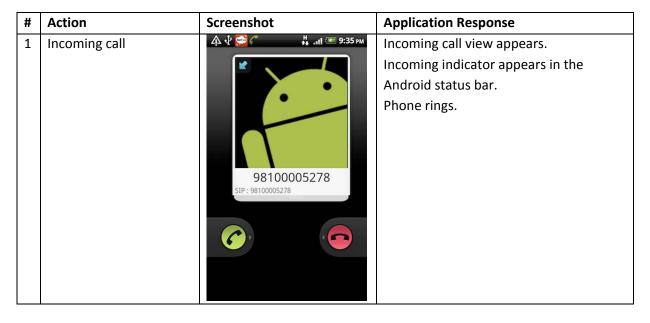
Call identification label shows dialed account.

Call status indicator on Android status bar shows "Call in Progress".

3.2.3 Receive a call

If the network is available, and you are signed-in (status shows "Registered"), you can receive calls. The application allows background call receiving, and does not need to be in the foreground to receive incoming VOIP calls. When there is an incoming call, the phone rings, and the incoming call view appears. You can decline or answer the call.

3.2.3.1 Decline a call



2 Slide the red "Ignore call" icon from right to left to decline the call.



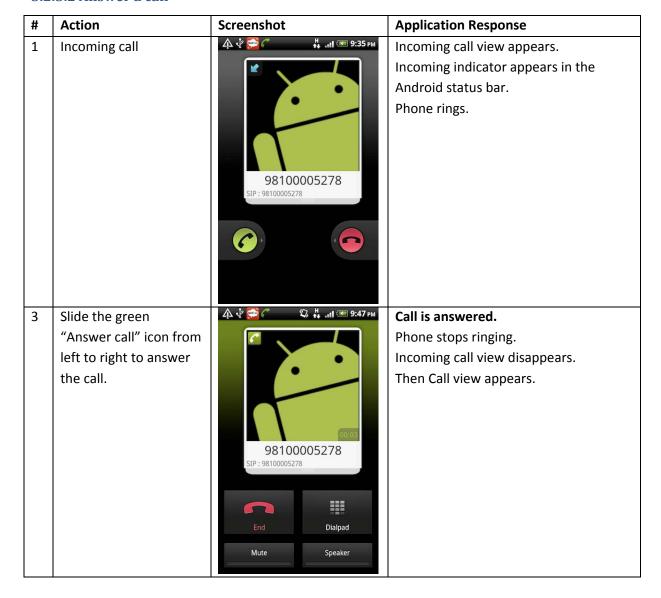
Call is declined.

Phone stops ringing.

Other SIP client receives notice that test account has declined the call.

Then Incoming call view disappears.

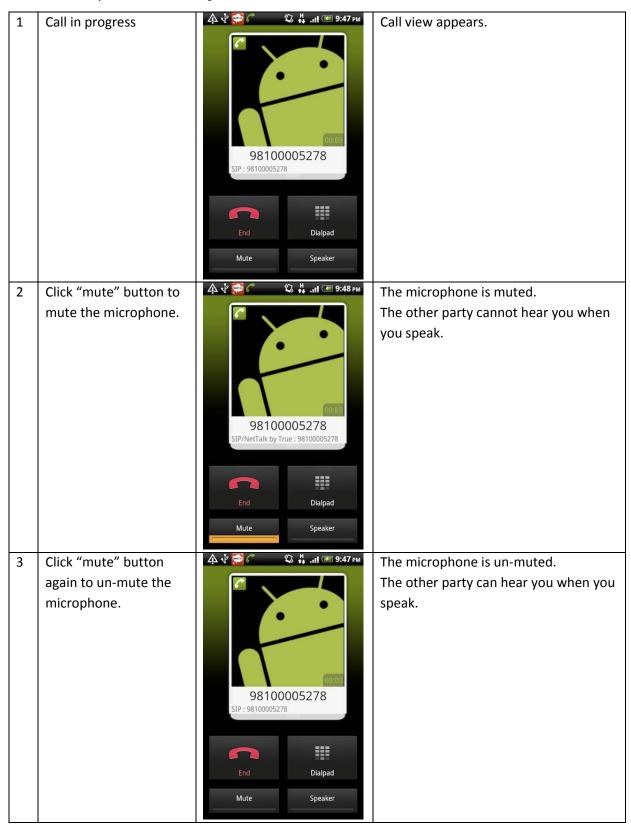
3.2.3.2 Answer a call



3.2.4 During the call

When a call is in progress, you can mute and un-mute the microphone, turn the speaker on and off, and show and hide the keypad.

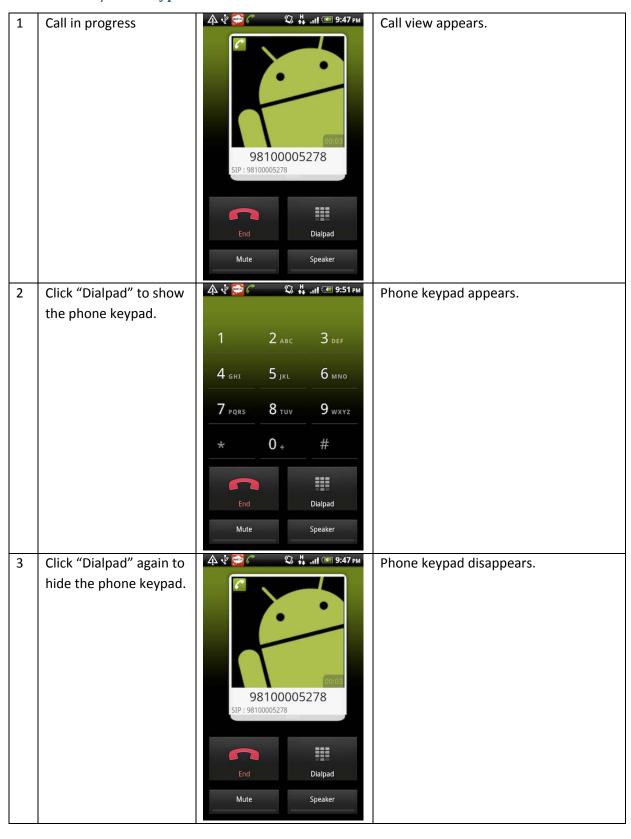
3.2.4.1 Mute/Un-mute microphone



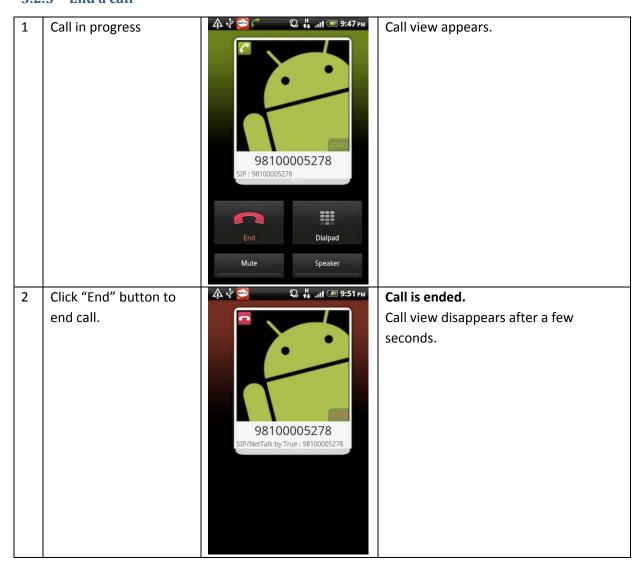
3.2.4.2 On/off speaker



3.2.4.3 Show/Hide keypad



3.2.5 End a call

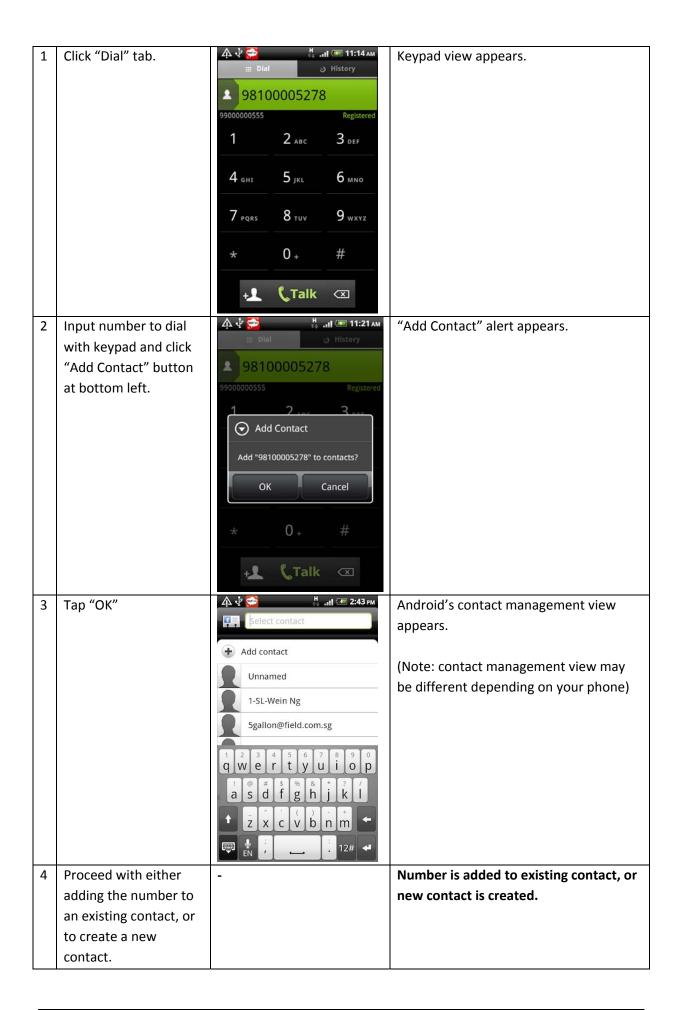


3.3 Contacts

The application includes a contact list that is integrated with the Android's contact list. In addition to dialing a contact from the contact list, you can create a new contact from a VoIP number, or add a VoIP number to an existing contact.

3.3.1 Create a new contact/Add to existing contact

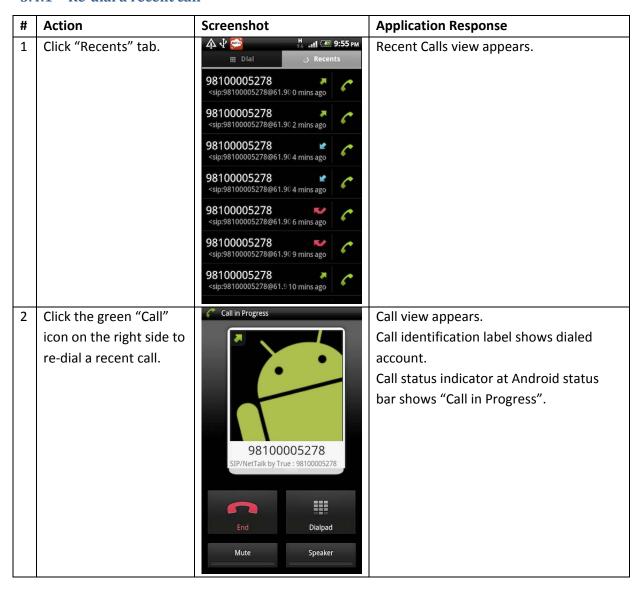
#	Action	Screenshot	Application Response
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3.4 Recent Calls

The application includes a recent calls list. Whenever a call is completed, it is added to the recent calls list. You can re-dial a recently received/missed/dialed number, delete a recent call, or delete all recent calls.

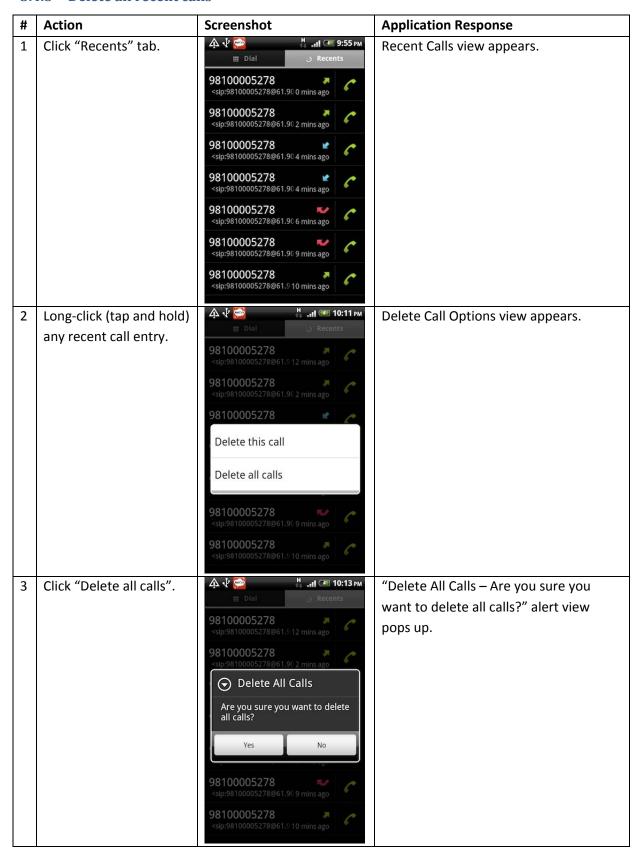
3.4.1 Re-dial a recent call

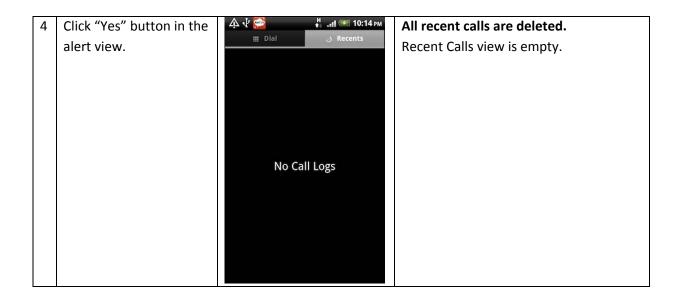


3.4.2 Deleting a recent call

#	Action	Screenshot	Application Response
1	Click "Recents" tab.	## Dlal	Recent Calls view appears.
2	Long-click (tap and hold) a recent call entry.	## Dial	Delete Call Options view appears.
3	Click "Delete this call".	-	The recent call entry is deleted from the Recent Calls view.

3.4.3 Delete all recent calls

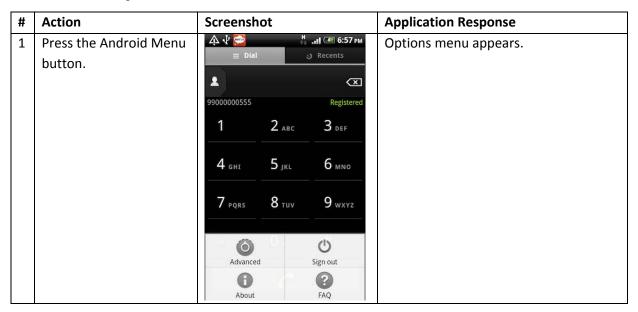


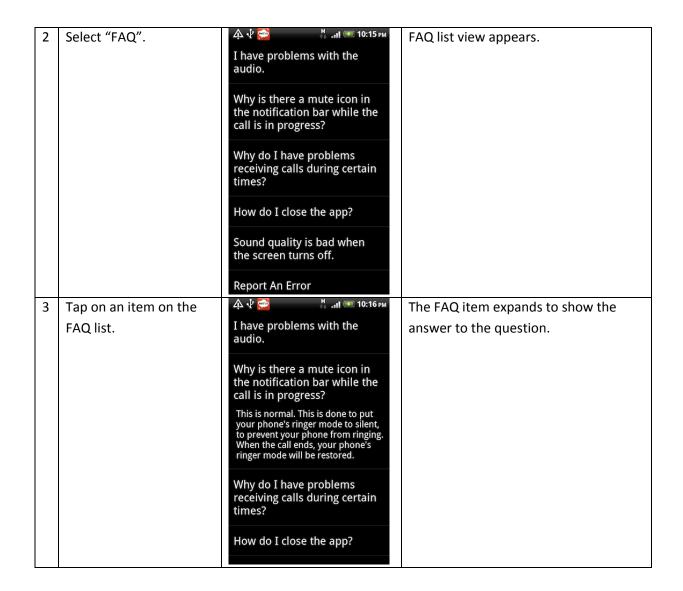


3.5 FAQ

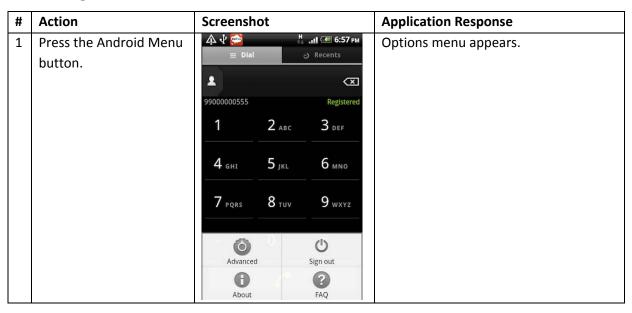
An FAQ is included with the application. The FAQ contains answers to frequently asked questions. If you encountered a problem with the application, you should consult the FAQ first to see if it answers your question. If the FAQ does not address your problem, and an error was generated by the application, you can send an error report. You need to set up your mail accounts on the phone first before sending an error report.

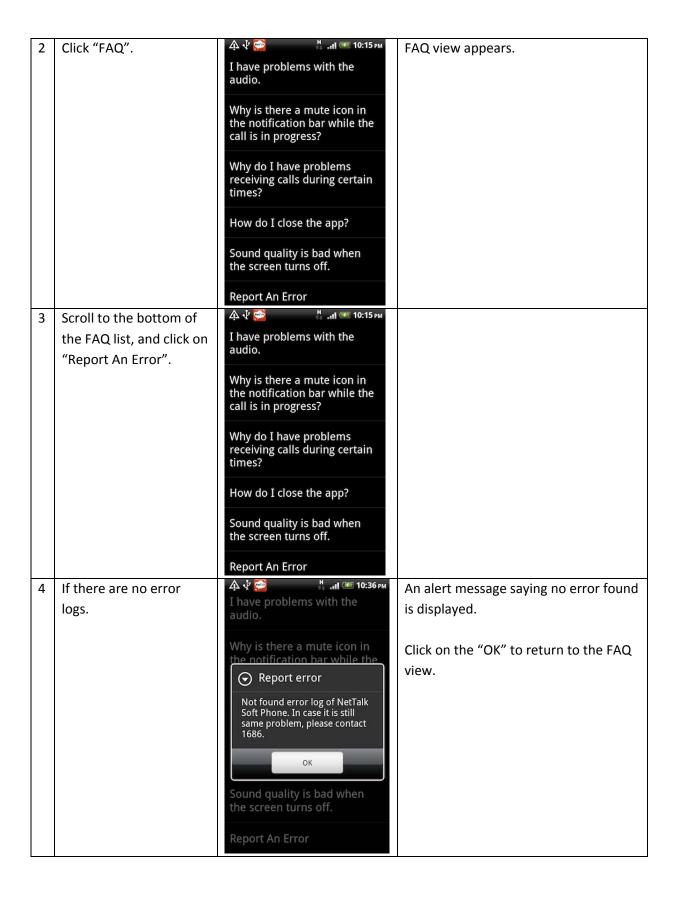
3.5.1 View FAQ





3.5.2 Report Error





Н 📶 🍱 🕲 1:54 ам However, if there are "Complete action using" alert view error logs. appears. the screen turns off I have problems when (This step happens only when there are Complete action using error logs). M Gmail Yahoo! Mail Use by default for this action. Device Model Support **♦** Н ... 🗷 10:19 рм Click "Gmail" or any Mail composer view appears with other email app to send recipient, subject, and body pre-filled From htatlau@gmail.com error report by email. in, and the log attached. Email contains <support@truecorp.co.th>, information about device. rror Log (Username - 99000000555) logs_11-14-11_221733.txt User can write additional error <<<Ple><<<Ple>ease add error description
here>>> descriptions in the email message. Version: 1.01 r1087 Information about your device : android.os.Build.BOARD : bravo User then clicks on "Send" to send the android.os.Build.BRAND: email to the support team. htc_asia_wwe android.os.Build.DEVICE : bravo android.os.Build.ID : FRF91 android.os.Build.MODEL: HTC (This step happens only when there are Save as draft Discard error logs).

3.6 Advanced settings

The Advanced settings menu has 3 sections:

3.6.1 Performance

These settings control the system options that will affect the performance of the VoIP application. These settings are pre-set to the default values that are suitable for most Android phones. However, in certain situations the user may want to experiment with these settings to improve the VoIP performance.

3.6.1.1 Keep WiFi active

Enabling this option will prevent the phone's WiFi from going idle when a VoIP calling is active. If the WiFi goes into idle mode when a call is active, the call quality may be seriously impacted. The default setting for most phones is enabled.

3.6.1.2 Enable WiFi speed boost

Certain Android phones have the option of setting the WiFi into high performance mode. Turn on this option may improve WiFi connectivity and call quality. However, be this option may cause the phone battery to drain faster. In phones that don't support WiFi speed boost, this setting is hidden from the user.

3.6.1.3 Keep CPU active

Enabling this option will prevent the phone from going into idle mode. This will enable the VoIP application to receive calls even when the phone becomes idle. However, even with this enabled, Android phones can still go into deep sleep mode when not being used for a long time. This option is disabled by default because it will cause faster battery drain.

3.6.1.4 Keep screen active

Enabling this option will prevent the phone's screen from turning off when the phone is near the caller's ear during a VOIP call. The default value this setting is disabled for most phones. However, in some Android phones, the default setting is enabled. This is because in some phone models, the WiFi quality will degrade when the screen is turned off.

3.6.2 Volume

These control the volume amplification and the microphone sensitivity amplification. Changes in these values will be stored in the phone's memory.

3.6.3 Miscellaneous

3.6.3.1 Invert proximity sensor

The proximity sensor detects when a person puts the phone near his or her ear/face. The proximity sensor is used by the VoIP application to turn off the phone's screen (if Keep screen active is not enabled) when the user puts the phone near his or her ear/face. In some Android phone models, this option has to be enabled for the proximity sensor to work correctly.

3.6.3.2 Diagnostics

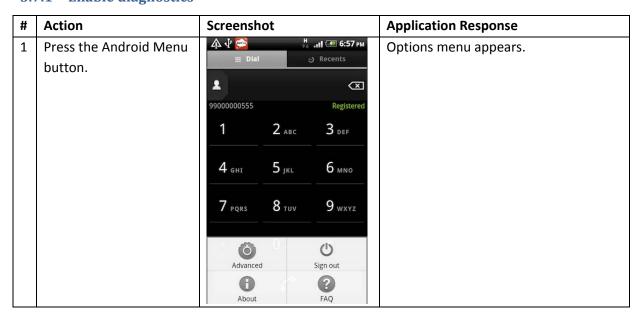
If this setting is enabled, the VoIP application is in Diagnostic mode. In this mode, the VoIP application will store all SIP packets and call statistics into the Android system log. Please see next section for details.

3.7 Diagnostics

By default, the application only collects error information in the log. If there are no errors, the log will be empty. However, if you encounter problems with the application, you can enable diagnostics to gather more information in the log, including detailed SIP protocol messages (including IP address), call media statistics, network information, and device information. When diagnostics is enabled, detailed diagnostics information is collected and written to the log until diagnostics is disabled. You should only enable diagnostics only if you encounter problems with the application, as gathering diagnostics information consumes resources such as memory, storage space, and battery. You should remember to disable diagnostics after you have gathered the necessary information for the problem you encounter.

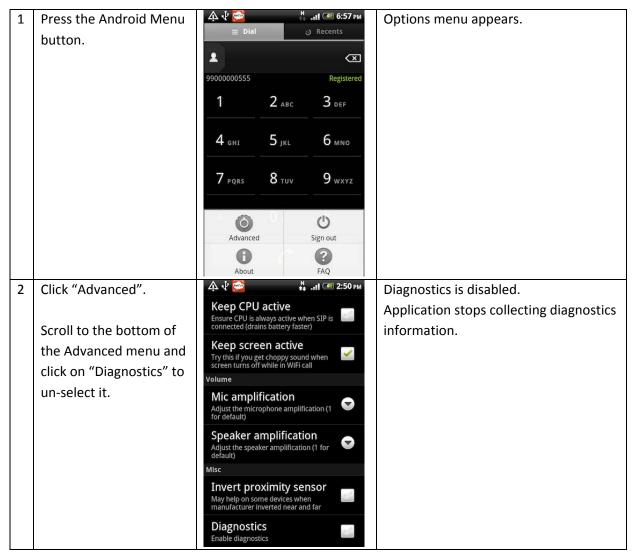
The Android system log buffer space is limited, so the amount of log messages is also limited. If the log buffer is filled, the oldest log messages are discarded.

3.7.1 Enable diagnostics



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 Click "Advanced". Diagnostics is enabled. Keep CPU active Application starts collecting detailed Ensure CPU is always active when SIP is connected (drains battery faster) Scroll to the bottom of diagnostics information. Keep screen active the Advanced menu and Try this if you get choppy sound when screen turns off while in WiFi call click on "Diagnostics" to select it. Mic amplification Speaker amplification 0 Adjust the speaker amplification (1 for default) Invert proximity sensor May help on some devices when manufacturer inverted near and far Diagnostics

3.7.2 Disable diagnostics



3.8 Diagnostics information

If the application is in diagnostic mode (**Diagnostics** is enabled in the **Advanced** settings menu), the application will add detailed information to the log file.

3.8.1 Software Version and Device Information

Software version and device information is added the error report email.

Sample Software Version and Device Information in the Error Report

Version: 1.08 r1087 → NetTalk application version Information about your device: → Phone hardware board name android.os.Build.BOARD : bravo → Phone hardware brand android.os.Build.BRAND: htc asia wwe → Phone hardware device name android.os.Build.DEVICE : bravo android.os.Build.ID: FRF91 → Phone hardware ID android.os.Build.MODEL: HTC Desire → Phone hardware model name android.os.Build.PRODUCT : htc_bravo → Phone hardware product name android.os.Build.TAGS: release-keys android.os.Build.VERSION.INCREMENTAL: 274424 → Android software incremental number android.os.Build.VERSION.RELEASE: 2.2 → Android OS version android.os.Build.VERSION.SDK: 8 → Android SDK version number

3.8.2 Connectivity information

When diagnostics is enabled, the application logs the current connectivity status, and whenever the connectivity changes. The possible connectivity types are listed below:

Sample of Connectivity Entries in the Log File

```
Connectivity: DISCONNECTED.
Connectivity: CONNECTED. Type: WIFI-
Connectivity: CONNECTED. Type: mobile-HSDPA
Connectivity: CONNECTED. Type: mobile-EDGE
```

3.8.3 Call statistics

When diagnostics is enabled, call statistics are logged every time the call session ends.

Sample Call Statistic Entry in the Log File

```
Call time: 00h:00m:14s, 1st res in 21180 ms, conn in 21276ms
Line 1:
Line 2:
          SRTP status: Not active Crypto-suite: (null)
Line 4: RX pt=8, stat last update: 00h:00m:03.158s ago
Line 5: total 350pkt 57 400 05
Line 3:
          #0 PCMA @8KHz, sendrecv, peer=85.17.186.6:54790
             total 359pkt 57.4KB (71.8KB +IP hdr) @avg=36.5Kbps/45.7Kbps
Line 6:
              pkt loss=0 (0.0%), discrd=0 (0.0%), dup=0 (0.0%), reord=0 (0.0%)
Line 7:
                    (msec) min avg
                                                    last
                                             max
              loss period: 0.000 0.000 0.000 jitter : 1.250 24.051 92.125
Line 8:
                                                    0.000
                                                            0.000
                             1.250 24.051 92.125 16.750 12.611
Line 9:
Line 10:
            TX pt=8, ptime=20ms, stat last update: 00h:00m:05.286s ago
Line 11:
               total 633pkt 101.0KB (126.3KB +IP hdr) @avg 64.3Kbps/80.4Kbps
Line 12:
                pkt loss=0 (0.0%), dup=0 (0.0%), reorder=0 (0.0%)
Line 13:
                     (msec) min
                                      avg
                                              max
                                                      last
             loss period: 0.000 0.000 0.000 0.000 0.000
Line 14:
                jitter : 55.250 76.625 98.000 98.000 21.374
Line 15:
         RTT msec
Line 16:
                           : 584.000 584.000 584.000 584.000
```

Line Description

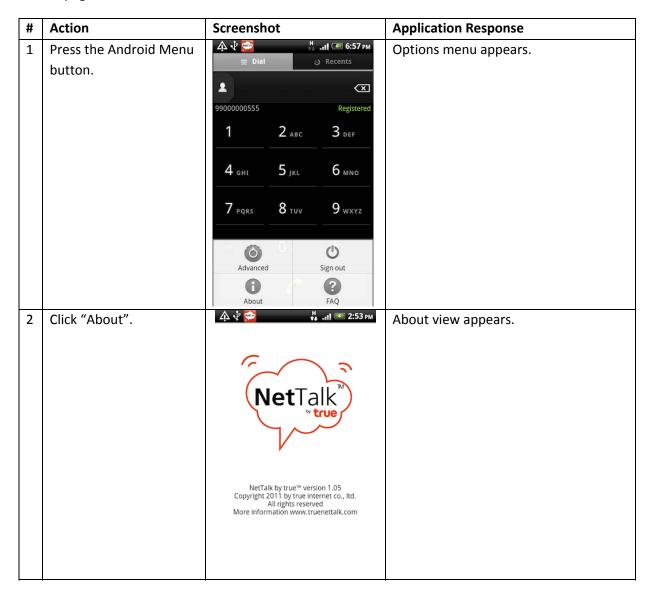
1	The duration of the call media session: 00h:00m:14s				
	The delay before the first non-100 SIP response was received: 21180 ms				
	The delay before call was confirmed (established): 21276ms				
2	Shows the Secure RTP status. This is only applicable for secured calling.				
3	Codec being used for the media session: PCMA @8KHz (G.711)				
	'peer' is the address where RTP packets were transmitted: 85.17.186.6:54790				
Line	Line 4-9 show the statistics of the received (RX) RTP packet stream.				
4	The last time the RX statistics was updated: 00h:00m:03.158s ago				
5	Total number of RTP packets received: 359				
	Total RTP payload size received: 57.4KB				
	Total RTP payload + IP/UDP/RTP header size received: 71.8KB				
	Average bitrate for the received audio: 36.5kbps				
	Average bitrate including received packet headers: 45.7Kbps				
6	Lost packets: 0 (0.0%)				
	Discarded packets: 0 (0.0%)				
	Duplicate packets: 0 (0.0%)				
	Out-of-order packets: 0 (0.0%)				
7					
8	Shows the minimum, average, maximum, and last duration (in milliseconds) of packet				
	loss burst.				
9	Shows the minimum, average, maximum, and last jitter value (in milliseconds) of				
	incoming RTP packets.				
Line	10-15 shows the statistics of the transmitted (TX) packet stream. TX statistics are actually				
	late by the remote party, so if the remote party does not support RTCP, or if RTCP SR/RR				
pack	ets were not received, TX statistics will be all zero.				
10	The last time TX statistics was updated: 00h:00m:05.286s ago				
11	Total number of RTP packets transmitted: 633				
	Total RTP payload size transmitted: 101.0KB				
	Total RTP payload + IP/UDP/RTP header size transmitted: 126.3KB				
	Average bitrate for the transmitted audio: 64.3Kbps				
	Average bitrate including transmitted packet headers: 80.4Kbps				
12	Lost packets: 0 (0.0%)				
	Duplicate packets: 0 (0.0%)				
	Out-of-order packets: 0 (0.0%)				
13	-				
14	Shows the minimum, average, maximum, and last duration (in milliseconds) of packet				
	loss burst.				
15	Shows the minimum, average, maximum, and last jitter value (in milliseconds) of				
	incoming RTP packets				

RTT is the round trip time, which is the time taken (in milliseconds) for a packet to travel from one end point to the other and return. This is a measure of latency.

RTT measurement requires RTCP to be enabled. If no RTCP packets are sent/received, the RTT will show up as 0. The call session duration also has to be long enough for RTCP packets to be sent/received in order for the RTT to be computed.

3.9 About

You can view information about the application, including the software version and website on the about page.



3.10 Troubleshooting

#	Signs and Symptoms	Screenshot	Possible	Solution			
			problems				
Sign-In							
1	"Can't sign in - You must be connected to a WiFi or data network." alert view pops up.		No network available	Connect to a WiFi or cellular data network.			
2	"Can't sign in - Incorrect username or password" alert view pops up.		Incorrect username or password	Input correct username and password.			
M	ake a call		I				
1	Call cannot be made. "No Network Available – You must be connected to a WiFi or data network" alert view pops- up.	981-000-05278 9900000555 Inactive 1 2 3 No Network Available You must be connected to a WiFi or data network 7 OK #	No network available.	Connect to a Wifi or cellular data network.			

"Inactive" Status Message

1 The status message near to top right corner says "Inactive". This means that the app is currently not connected to the VoIP service, and calls cannot be made and received.



No network available.

Connect to a Wifi or cellular data network.

Report error

"Report error – Not found error log of NetTalk Softphone. In case it is still same problem, please contact 1686" alert view pops up.



Error log is empty because there is no error or error report has already been sent. Enable diagnostics to collect more information.