Qtopia®

SIP based VoIP
for
Qtopia Phone

User Manual

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October 2005

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1. Scope

The scope of this document is to provide guidelines to the user for using the VoIP settings application and the VoIP call functionality of Qtopia Phone.

2. Overview

The VoIP functionality of Qtopia allows the user to make or receive SIP based VoIP calls. It requires the user to have an account with a SIP service provider. Using account details from the SIP service provider, the user is required to configure the phone for VoIP functionality using the VoIP settings application provided with Qtopia phone. The user is then required to register or login into the ISP SIP server using the VoIP settings application. Once that is done, the user can make VoIP calls using either Quick Dialler, Contacts application or Call History. Once registered with the server, the user can also receive VoIP calls from other VoIP users like a normal call.

This manual provides step by step instructions about configuring and using VoIP functionality of Qtopia phone.

3. User Guide

The User Guide has been divided in the following 4 sections:

1. Configuring VoIP Functionality
2. Contacts (Adding, Editing VoIP Contacts)
3. Phone Call (Make or Receive VoIP Calls)
4. Troubleshooting

4. Configuring VoIP Functionality

VoIP functionality can be configured using the VoIP settings application. Once the settings are complete the user is required to register to a SIP server before any VoIP calls can be initiated or received.

4.1 Selecting and Starting VoIP Application

1. On the Applications page, Select and click on VoIP to start the VoIP Settings Application.

Figure 1 – Launching VoIP Application

2. The VoIP Main Screen is displayed as shown in Figure 2 as follows:

- **Identity** : Name of the user and the identity created to connect to the SIP server
- **Server Name** : Name of the SIP server
- **Registration Status** : Shows “Unregistered” if the user has not registered with the SIP server and “Registered” if the registration is complete
- **VoIP Status** : Status of the user as either:
    - Available - Ready to receive a VoIP call
    - Unavailable - the user has chosen to appear offline to other VoIP users.

By default “Identity” and “Server Name” are blank and “Registration Status” is Unregistered when the application is launched for the first time.

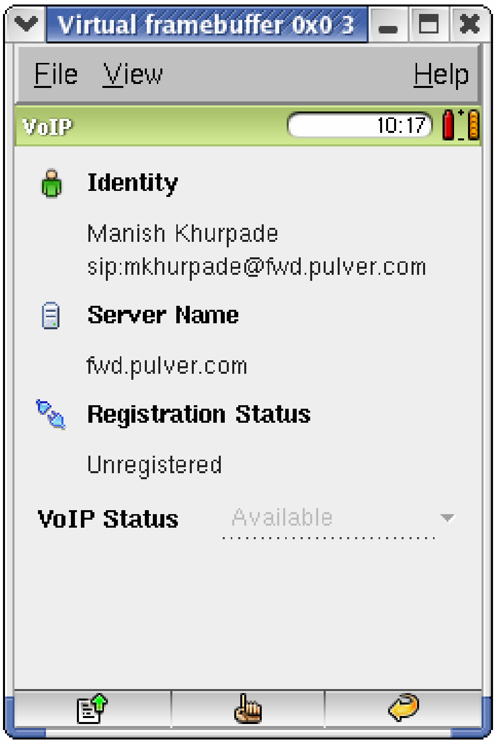


Figure 2 - VoIP Main Screen

3. Click on the Context Menu on VoIP Main Screen. The Context menu options are:

- **Register:** Select this option to register with the SIP Server. This option is disabled ( ) for first time registration
- **Identity:** Select this option to create a new VoIP identity or edit an existing VoIP identity
- **Settings:** Select this option to enter or edit SIP settings
- **Help:** Select this option to view Help.

4.2 Configuring Identity

1. Select Identity from the context menu.
2. The "VoIP Identity page" is displayed.
3. The user can create an identity for using the VoIP Application by entering following details:
   - **Full Name**: Enter your full name
   - **SIP URI** : Enter SIP URI. The URI consists of two parts:
     - o the user login Id
     - o the host - address of the SIP server, with which you want to connect
   - **Password:** Enter the password for SIP server
   - **Auto Register** : If this option is selected and the identity is properly set, then registration with the SIP Server is automatic when "qpe" starts.

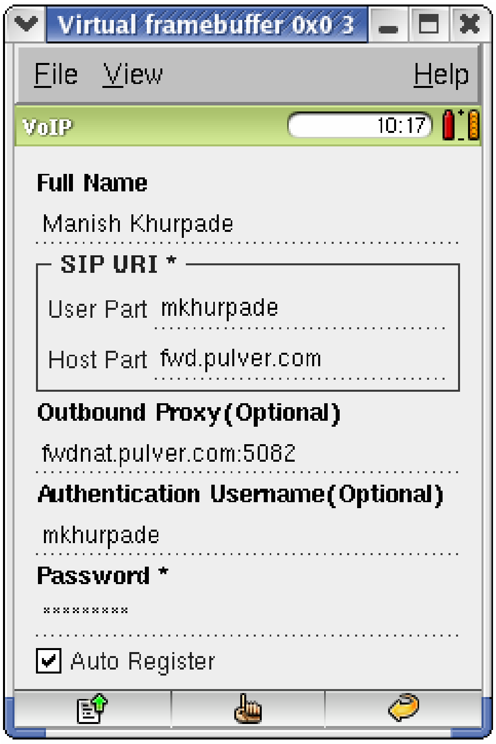


Figure 3 - VoIP Identity Page

4. After the required information is filled, use the back button  to save the changes

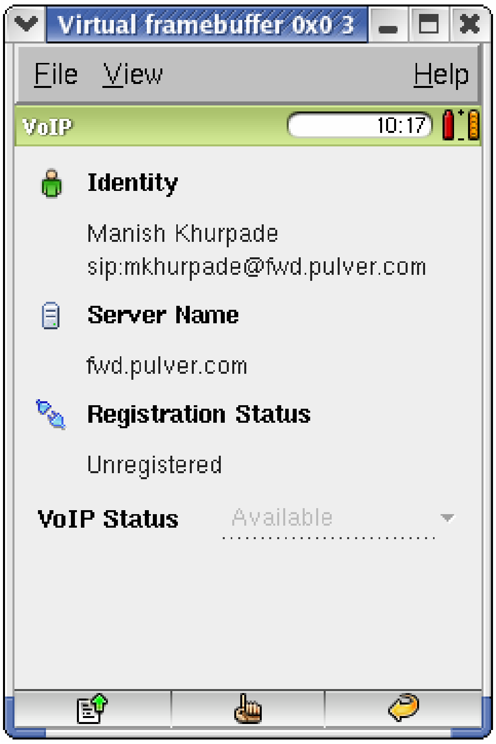
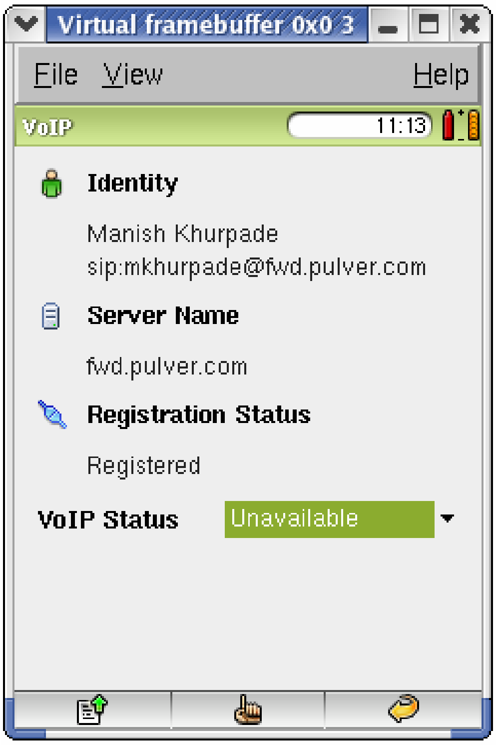


Figure 4 - VoIP Main Screen updated with new identity

4.3 Registering

1. Select the context menu on the VoIP Identity Page
2. Select Register
3. The Registration process starts and Registration Status shows “Contacting Server”
4. If the registration process is successful, the Registration status changes to “Registered”
5. If the registration is unsuccessful , then an error message is displayed
6. On successful registration, VoIP Status defaults to Unavailable.



4.4 Editing Identity Settings

1. Select the Identity  option of context menu from the VoIP Main Screen
2. Make the required changes and use the back button  to save.
3. You need to restart *qpe* for changes to take effect

4.5 Changing VoIP Status

1. Go to the VoIP Main screen
2. Go to “VoIP Status” field on this page
3. Using a drop down menu, change the status to “Available” or “Unavailable”
4. Available Status means that you are shown as online to other VoIP users
5. Unavailable Status means that you are shown as offline to other VoIP users. Other VoIP users are still able to call you.

4.6 Settings

1. Select Settings from the context menu on the VoIP Main page
2. The VoIP Settings page displays
3. Settings include SIP, Audio, Socket and Call Settings.
4. Use tabs to move from one setting to another.

4.6.1 SIP

The SIP settings include:

- **Hide Via** – Select one of the options : Don’t Hide, Request next Hop and Request Full Route
- **Max Forwards**: User can set the maximum number of forwards permitted
- **Expire Time of Registration(sec)**: User can set the timeout duration for registration request.
- **Expire Time of Presence Subscription**: User can set the timeout for Presence Subscription

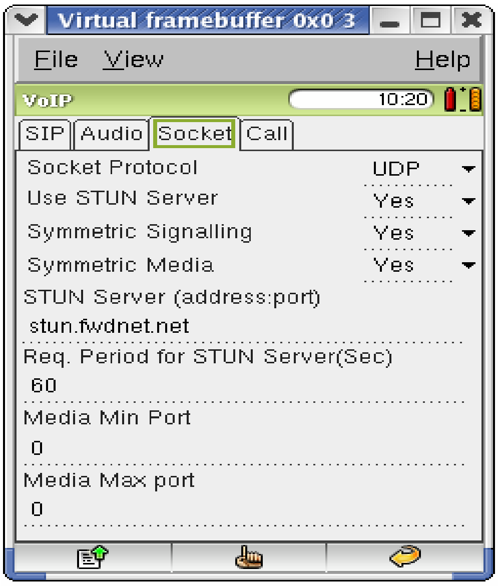
4.6.2 Audio

The audio settings include:

- **OSS Device Mode**: User chooses either ReadWrite or ReadOnly/WriteOnly
- **Device for WrOnly or ReadWrite:** User specifies the device to be used e.g. /dev/dsp
- **Device for RdOnly:** User specifies the device to be used e.g. /dev/dsp
- **Preferred Codec**: User selects a preference for codec from G711u, G711a, GSM or iLBC
- **Size of Payload**: User chooses either 80 or 160 or 240 as payload size depending upon the codec chosen.

4.6.3 Socket

- **Socket Protocol**: User chooses UDP or TCP
- **Use STUN Server:** User specifies whether to use STUN Server or Not (Yes/No)
- **Symmetric Signalling:** User specifies whether to use Symmetric Signaling or Not (Yes/No)
- **Symmetric Media:** User specifies whether to use Symmetric Media or Not (Yes/No)
- **STUN Server (address:port):** Specify STUN server address if "Use STUN Server" is set to Yes
- **Req. period for STUN Server (sec):** Timeout for STUN server
- **Media Min Port and Media Max Port:** User specifies Media Min and Media Max port to be used.



4.6.4 Call

Busy Message: Not implemented

4.7 Exiting the SIP Server or Disabling VoIP Calls

Assuming you are already registered with SIP Server:

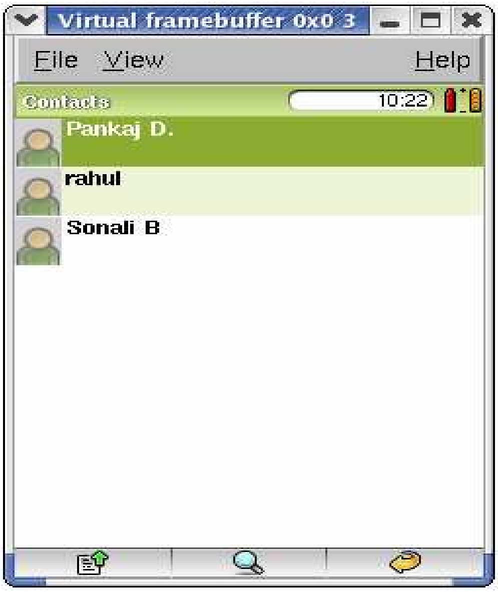
1. Go to the main VoIP application page by launching VoIP application
2. Select “Logout “ from the context menu
3. This will unregister the user from the VoIP server
4. Press back button to exit from the VoIP application

5. Contacts

The user can use the Qtopia Contact application to add new VoIP contacts, update existing contacts, include a VoIP Id for the contact or delete the contact from the list.

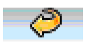
1. Select Contacts from Applications

2. Contacts page is displayed showing a list of all contacts



5.1 Adding New Contacts

1. From the context menu, Select New Contact
2. New Contact page is displayed:

3. Enter the following details for the new contact:
    - **Name** : Name of the new contact
    - **E-mail address** : e-mail address of the new contact
    - **VoIP ID** : VoIP Id of the new contact
4. Press  to save entered information
5. The new contact now appears in the Contact List

5.2 Editing Existing Contacts

1. Select a contact that you want to edit from the Contacts List
2. Click on the contact
3. The Contact Details page is displayed

4. Select "Edit" from the context sub-menu
5. Make the necessary changes to the contact details
6. Press  to save entered information.

5.3 Deleting Contacts

1. Select a contact that you want to delete from the Contacts List
2. Click on the contact
3. The Contact Details page is displayed
4. Select "Delete" from the context sub-menu
5. The contact does not now appear in the Contact list

5.4 Viewing Contact Status

1. View Contacts list
2. Select a contact from the list
3. The Contacts details page is displayed
4. Look at the icon adjacent to the contact name
5. A green icon indicates that the VoIP contact is “Available” for a call
6. A violet icon indicates that the VoIP contact is “Not Available” for a call.

6. Phone Call

6.1 Outgoing Call

A VoIP phone call can be made using either the Contacts Application, Quickdialer or Call History.

6.1.1 Call Using Contacts Application

1. Select a contact from the Contacts list
2. The Contacts Details page is displayed
3. VoIP Status of the contact should be "Available" (green icon)

6.1.1.1 Call Using a Landline Number

1. On the contacts details page, Select the landline number that you wish to dial
2. Using context menu , you can either make a **GSM call or a VoIP call** if you are registered with the VoIP server

3. Select the type of call and Press “Call”

4. The outgoing call is initiated to the remote user

5. Ringing tone can be heard

6. The call will be established if the remote user accepts the call

7. The call will get terminated if the remote user disconnects the call.

6.1.1.2 Call Using VOIP ID

1. On the contacts details page, Select the VoIP Id of the contact

2. Click to initiate the outgoing call to the remote user

3. Ringing tone can be heard

4. The call will be established if the remote user accepts the call

5. The call will get terminated if the remote user disconnects the call.

6.1.2 Call Using Quick Dialler

6.1.2.1 Call Using Landline Number

1. Go to the Quick Dialler pager
2. Enter a landline number that you wish to dial
3. Using context menu , you can either make a **GSM call or a VoIP call** if you are registered with the VoIP server
4. Select the type of call and Press "Call"
5. The outgoing call is initiated to the remote user
6. Ringing tone can be heard
7. The call will be established if the remote user accepts the call
8. The call will get terminated if the remote user disconnects the call.

6.1.2.2 Call using VOIP ID

1. Go the Quick Dialler page
2. Enter the VoIP Id that you wish to dial
3. Press "Call" to initiate the outgoing call to the remote user
4. Ringing tone can be heard
5. The call will be established if the remote user accepts the call
6. The call will get terminated if the remote user disconnects the call.

6.1.3 Call using Call History

1. Go to Applications Page
2. Select Call History
3. Select a number from Received, Dialled or Missed call list
4. Press Dial.
5. For landline numbers you can choose a VoIP call or a GSM call.

6.2 Incoming call

1. When you receive an incoming call, the Calls page is displayed

2. You can accept the call or reject the call using “Accept” or “Reject” buttons respectively

3. If you do not accept or reject the call within some time, the call will be treated as a “Missed Call”.

4. You can see the list of accepted, rejected or missed calls from the “Call History”

7. Troubleshooting

7.1 Registration related issues

7.1.1 Unable to register or Registration Timeout

- Verify that you have IP network connection
- Verify that your account details are typed in correctly
- Verify that your password is typed in correctly.

7.2 Call related issues

7.2.1 Unable to make a VOIP Call

- Verify that you have registered with the SIP server using correct VoIP account details. You can check this by launching VoIP application. The registration status should show Registered with icon as
- Verify that the VoIP ID you are trying to call is correct and the user is online.

7.2.2 Unable to make a landline call

- If you are unable to make a landline call using VoIP, ensure you are following the scheme prescribed by your service provider. For example, some SIP service providers require you to prepend '*' to a landline number.
- Verify that your subscription allows you to make landline calls. Generally this is a paid service and requires subscription.
- Verify that the number dialled is correct.

7.2.3 Poor Voice quality or the call aborts

- This problem could occur due to network congestion or heavy load on server. Report the problem to your SIP service provider.