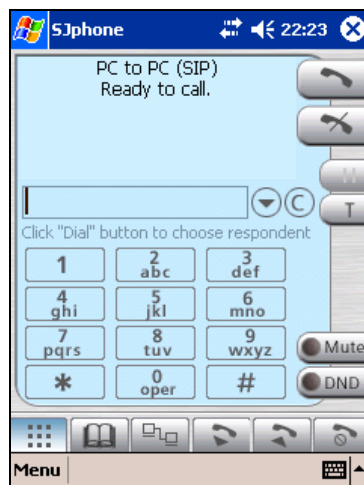


SJphone™

PocketPC2003(SE)/2002 version



User's Guide



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No warranty is made in regard to specifications or features.

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I. Welcome to SJphone™

SJphone™ is a VOIP software client that allows you to speak with any PC, PDA, stand-alone IP-phone and through a VOIP gateway or VOIP service provider with any traditional wired or mobile phone. It supports both SIP and H.323 standards and is fully inter-operable with most major telephony vendors and Internet Telephony Service Providers.

1.1. SJphone™ Features

H.323 Compatibility

This allows you to

- Communicate with other H.323-compatible VoIP programs, such as Microsoft NetMeeting®, Intel IPhone®, Vocatel IPhone®, and many others, or with standalone IP phones.
- Call any host by its name or IP address
- Call a regular telephone using a **Gateway**
- Call an Internet phone with a built-in **Gateway**
- Call a computer or telephone in a private network through a **Gatekeeper**

SIP Compatibility

- This allows you to
- Communicate with other SIP-compatible VoIP programs and systems, such as Vovida VOCAL® SIP User Agent, Cisco Access Server®, Cisco ATA-168 VoIP gateway, Cisco 7940, 7960 (with SIP firmware) IP Phone, Ubiquity SIP User Agent®, and many others.
- Call any host by its name or IP address
- Call a regular telephone using a **SIP Gateway**
- Call an Internet phone with a built-in **SIP Gateway**
- Call a computer or telephone in a private network through a **SIP Proxy**
- DNS support in the SIP stack

Features that Make SJphone™ Easy to Use:

- Service profiles automatically applying provider's or IP-PBX settings
- Contacts and speed dialing
- Call transfer and hold
- Automatic neighborhood discovery
- ILS (Internet user Location Service) directory
- Respondent list allows you to maintain and manage multiple calls
- Dial pad to type in numbers and send DTMF signals.
- Can automatically accept incoming calls
- Can automatically accept incoming calls after specified time
- "Do not disturb" mode
- Skins allow you to change the appearance of the **SJphone™ Main** panel
- Output sound level indication and control
- Automatically adjustable silence detection level
- Voicemail message waiting indication

Features for Advanced Users:

- 3-way conferencing using different VoIP protocols
- Multisession (including conference) calls
- Customized service profiles allow you to create your own profiles for calls through your H.323 Gatekeeper, Gateway, or SIP Proxy
- Various methods for DTMF sending specified in service profiles.
SIP: In-band: DTMF, RFC2833, SIP INFO
H.323: In-band, Q.931 keypad, H.245 alphanumeric, H.245 signal
- Support for extended H.323 address syntax
- Support for SIP URL syntax
- Support for ILS contact syntax
- Lost and out-of-order packet indication

- Manual compression codec selection
- Advanced lost packet recovery, offering better sound quality over a poor connection.
- Remote support console
- Automatic NAT detection (STUN)
- NAT traversal support

Attention! Advanced lost packet recovery is not fully H.323-compatible. Use it only while communicating with other SJphone™-compatible software.

1.2. SJphone™ Features Coming Soon

- Improved sound quality and filtering

To learn more about these features and to obtain the latest version of SJphone™, go to the SJ Labs web page at: <http://www.sjlabs.com>

1.3. System Requirements

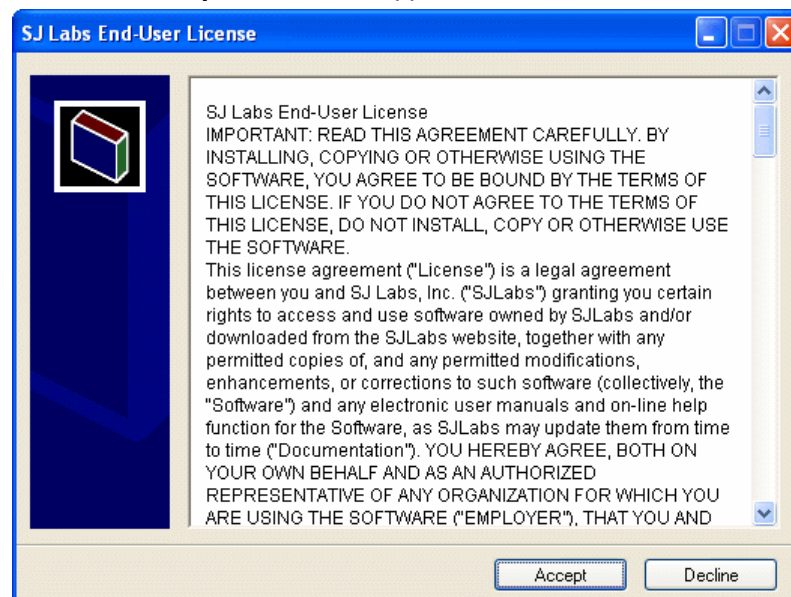
PDA requirements: A Pocket PC 2002/2003/2003SE with the Intel xScale /Strong Arm SA1100 processor, a built-in microphone, speaker or headset, and an Internet connection. Wireless Internet is OK, but quality of sound may vary

1.4. Installing SJphone™

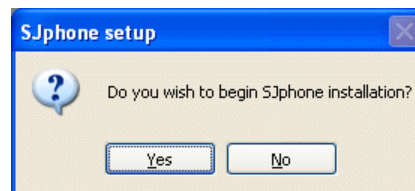
To install SJphone™ using ActiveSync,

- Connect your **Pocket PC device** to your desktop computer and run the installation program on the **desktop** computer.

The **SJphone™ License** and **setup** windows will appear.

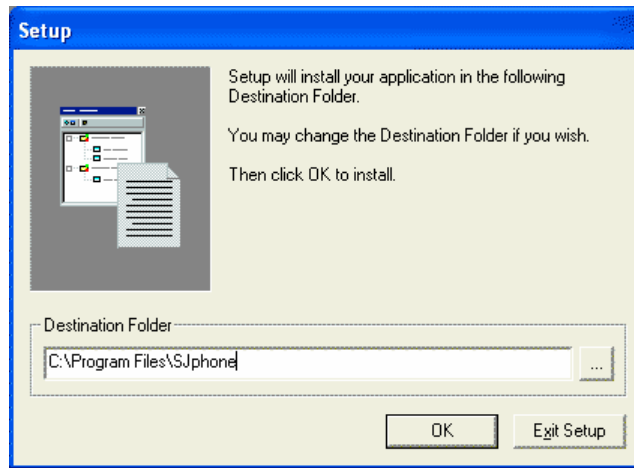


- Click the **Accept** button.



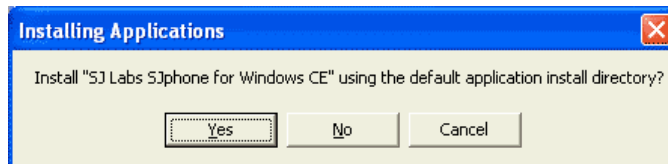
- Click the **Yes** button.

The **Setup** window will ask you about the folder to which it unpacks **SJphone™** installation files.



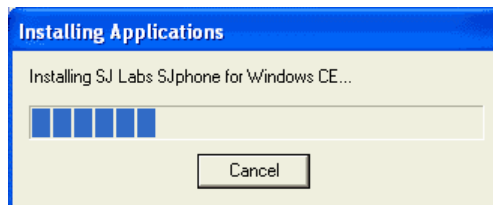
- Specify the folder and click the **OK** button.

The **Installing Applications** window will appear

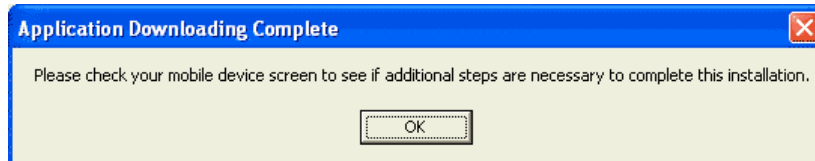


- Click the **Yes** button to install **SJphone™** into the default folder.

Setup will start installing **SJphone™**.



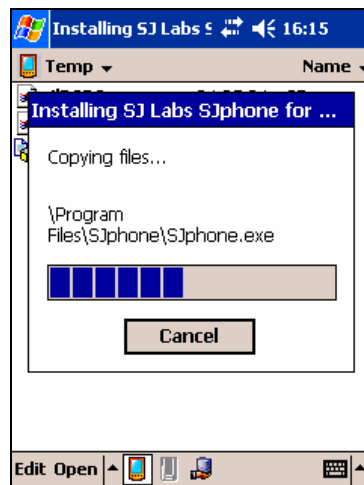
When it finishes, the **Application Downloading Complete** window will appear



Click **OK** and make all necessary actions on your **Pocket PC device**.

To install SJphone™ without ActiveSync

- Download and run the `.cab` file on your **Pocket PC device**.



- Follow the on-screen instructions.
You may need to reset your **Pocket PC** to complete the installation.

To uninstall SJphone™, connect your **Pocket PC device** to your **desktop computer** and use **ActiveSync Tools->Add/Remove Programs...** menu item.

To start SJphone™, go to **Start->Programs->** and tap **SJphone** icon.

1.5. Registering SJphone™

Note: If you have a pre-registered copy of **SJphone™**, you may skip the registration.

You may register your **SJphone™** copy. The registration is free, and you will also be able to receive information about new **SJphone™** releases.

If you do not have the activation key, tap the **Get Key** button and follow the on-screen instructions.

To register your **SJphone™** on-line, do the following:

- Tap **Menu** and select **Options**. The **Options** panel will appear.
- Select the **Support** tab, then tap the **Register** button.

An **SJphone registration** message will appear.

- Enter your activation key and tap the **OK** button. The activation key is case-insensitive.
If you receive the key electronically, you may copy and paste it to the **SJphone registration** message using the **Paste** button.



1.6. Contact Information and Technical Support

Email: phone@sjlabs.com

<http://www.sjlabs.com>

or

- Tap **Menu** and select **Options**.
The **Options** panel will appear.
- Tap the **Web** button.

Bug Report...

This form is used to provide **SJphone™** developers with the information necessary to fix problems that may occur. You may automatically record and submit the information about the problems.

You may also automatically record and submit the information about the problems from the system tray.
To do this,

- Tap **Menu**, select **Bug Reporting**, and then **Start Recording**.
- Reproduce the problem
- Tap **Menu**, and select **Submit...**

The **Submit Bug Report** window will appear

- Describe the problem in the window.
- Submit the bug report either manually or automatically using your default mail program.

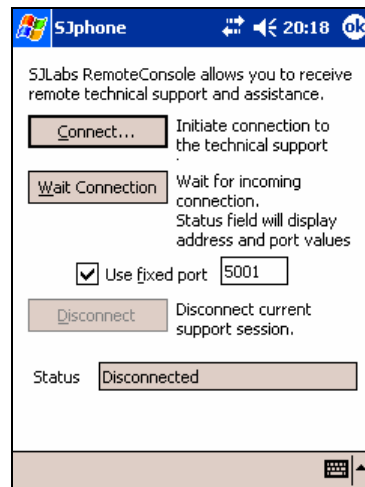
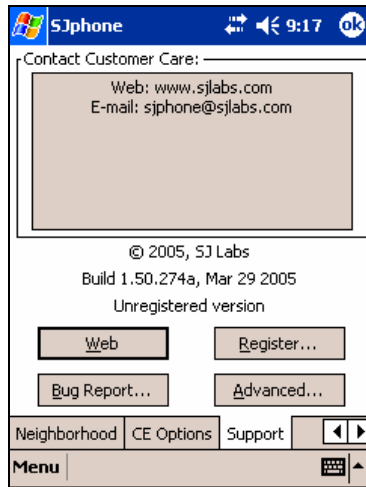
You may cancel bug report by selecting **Cancel Recording**.

Remote Console (available in the Advanced Mode interface only and may be unavailable for some service profiles)

Attention! You need to contact the support team to obtain the information on how to obtain remote on-line support.

Remote on-line support allows the support team to solve your problem remotely. They may review configuration options and immediately correct them if necessary. You need to contact the support team to obtain the information how to receive the remote on-line support.

- Web Tap this button to visit **SJ Labs** home page
- E-mail Tap this button to send an e-mail to **SJ Labs**
- Register Tap this button to register **SJphone™**
- Advanced Tap this button to activate **Remote Console**



II. SJphone™ Panels and User Features

2.1. Controls on the Main Panel



Call to: Type here the IP address, host name, nickname, or telephone number you want to call

Status: Displays the call status

Buttons

Dial Tap this button to dial the address typed in the **Call to:** field
 Hangup Tap this button to end the call
 Hold Tap this button to put the call on hold
 Transfer Tap this button to transfer the call to the address specified in the **Call to:** field

Phonebook Tap this button to activate the **SJphone™ Phonebook, Neighborhood Browser, or Call Logs**

Neighbors

Dialed Calls

Received Calls

Missed Calls

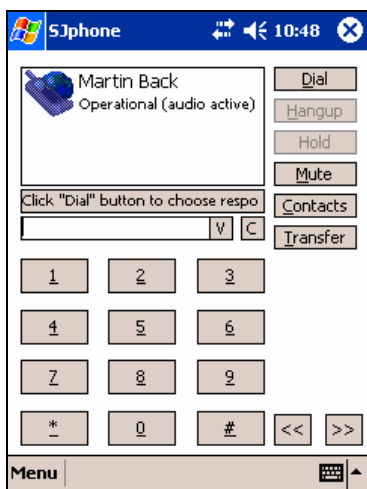
Mute

Speak Tap the **Mute** button to mute your voice. The **Speak** button will appear.

Tap the **Speak** button to resume the conversation. The **Mute** button will appear again.

>> Tap this buttons to go to the next and previous screens of **SJphone™ Main** panel

<< Tap this button to transfer the call to other respondents.

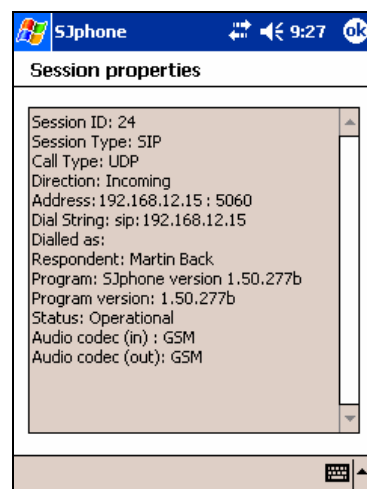


You can select the **Main** panel appearance from available skins. Go to the **Skins** section to learn more about this feature.

(available in the Advanced Mode interface only)

You may look at the respondent properties. Tap and hold the respondent and select the **Properties** item in the pop-up menu.

The **Session Properties** message will appear.



Sound Adjustments:

Generally, you can adjust the **SJphone™** sound characteristics using standard **PocketPC** audio controls.

Some additional audio options may be set on the **Audio** tab.

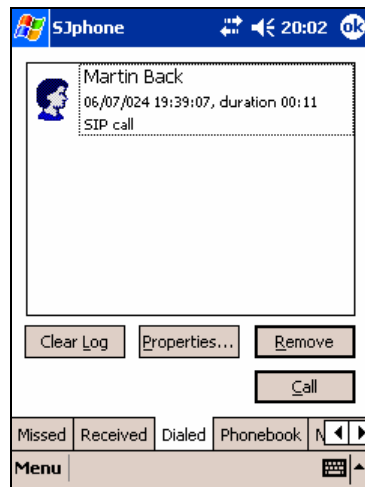
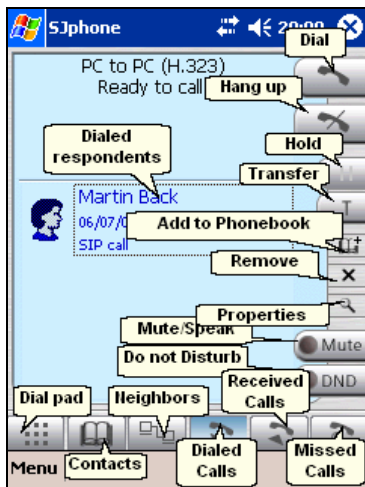
2.2. Contacts

Contacts allow you to store information about respondents you may want to call regularly. It also has the **ILS Directory** and **Neighborhood** browsers, and **logs** for missed, received, and dialed calls.

You may reach **Contacts** either by tapping the bottom buttons on the **Main** panel, or tapping **Menu** and selecting **Contacts**.

Call logs (Dialed, Missed, and Received)

You may see the list of missed, received, and dialed calls. You may repeat calls and see call records.

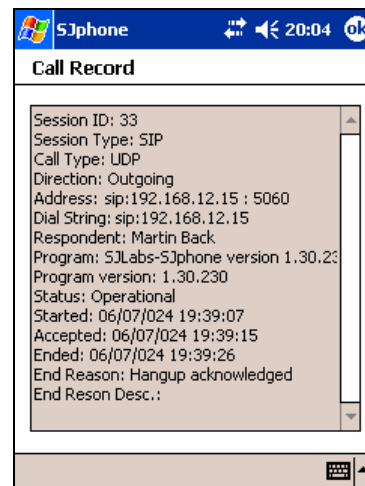


- Call Tap this button to call the selected respondent
- Clear Log Tap this button to remove all records in this log.
- Properties Tap this button to view the information about a selected respondent
- Remove Tap this button to remove the selected record from this log

Tap and hold a record to perform the following operations from a pop-up menu:

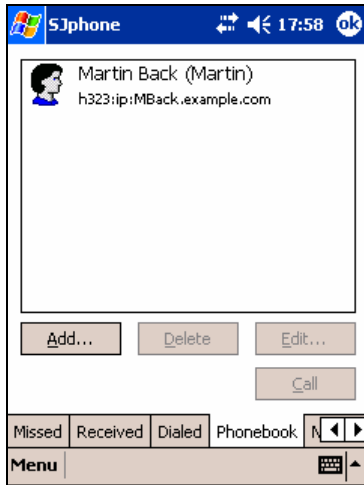
- Call** Select it to call the selected respondent
- Remove** Select it to remove the selected record from this log
- Add to Phonebook** Select it to add a selected respondent to **Phonebook**
- Properties** Select it to view the information about a selected respondent
- Clear Log** Select it to remove all records in this log
- Clear All Logs** Select it to clear all logs
- Export Logs** Select it to export logs to an external file. The default extension is `c:\m`.
- Import Logs** Select it to import logs from an external file. The default extension is `c:\m`.

You may view the information about a call
 To do this, tap and hold the call, or select the call and tap the **Properties** button.
 The **Call Record** message will appear.

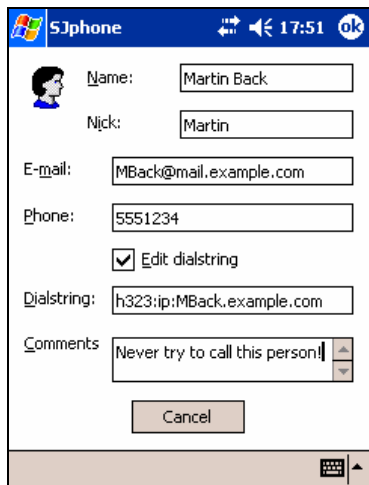


Phonebook

You can use the **Phonebook** panel in **Contacts** to store information about respondents whom you want to call later. You may store either a phone number or an advanced dial string which may be any valid dial string. See the **Place a Call** section for details.

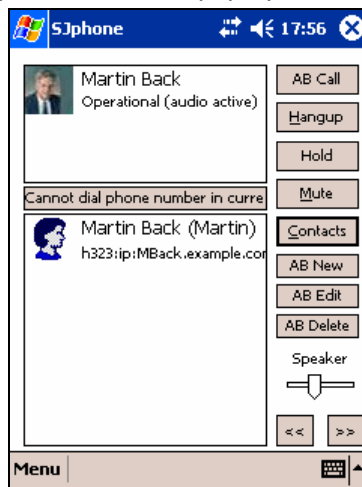
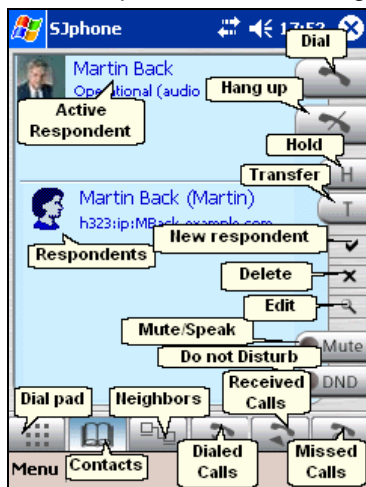


- Call Tap this button to call the selected respondent
- Add Tap this button to add a new record to **Phonebook**
- Delete Tap this button to remove the selected record from **Phonebook**
- Edit Tap this button to edit the selected record



- Name: Field for the respondent's name
- Nick: Field for the respondent's nickname
- E-mail: Field for the respondent's e-mail
- Phone: Field for the respondent's phone number
- Use Advanced Dialing: Select this box if you want to use an advanced dial string. Go to the **Place a Call** section for details
- Dialstring: Field for the advanced dialstring for the respondent.
- Comments: Field for comments

You may also reach **Phonebook** by switching to the **Phonebook** screen on the **Main** panel. Tap and hold a record to perform the following operations from a pop-up menu:

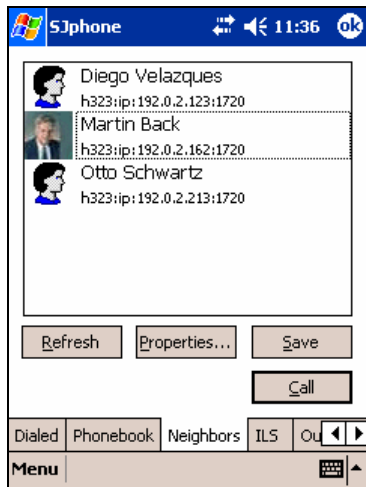


- Call** Select it to call the selected respondent
- Add** Select it to add a new respondent to **Phonebook**.
- Delete** Select it to remove the selected respondent from **Phonebook**
- Edit** Select it to edit the information about a selected respondent

Neighbors

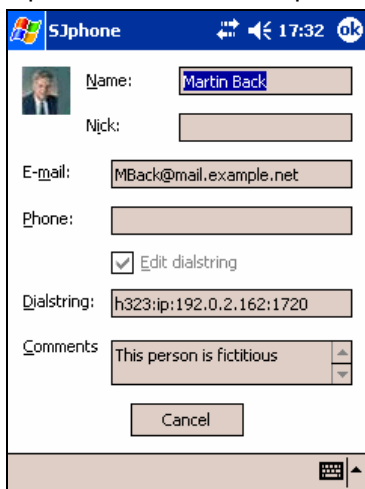
You may see other persons who are using **SJphone™** on your local network. You may call them or add to **Contacts**.

Attention! This feature works only on local area networks!



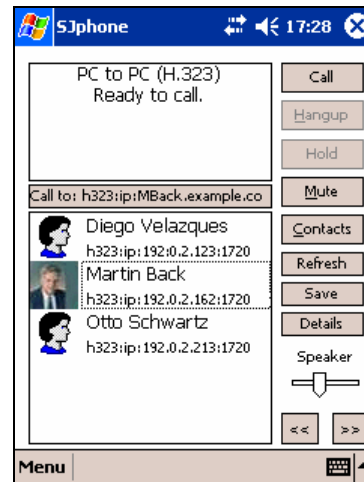
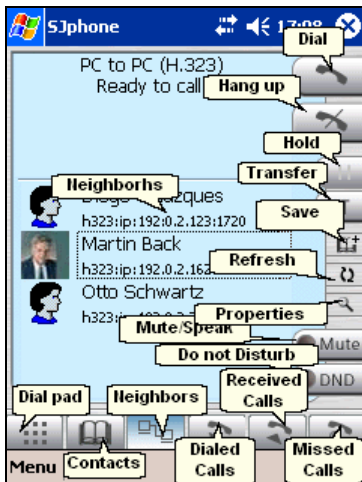
- Refresh Tap this button to refresh the list of neighbors
- Properties Tap this button to view the information about a selected neighbor
- Save Tap this button to add a respondent from the **Neighbors** list to **Phonebook**
- Call Tap this button to call a selected neighbor

Tap and hold a record to perform the following operations from a pop-up menu:



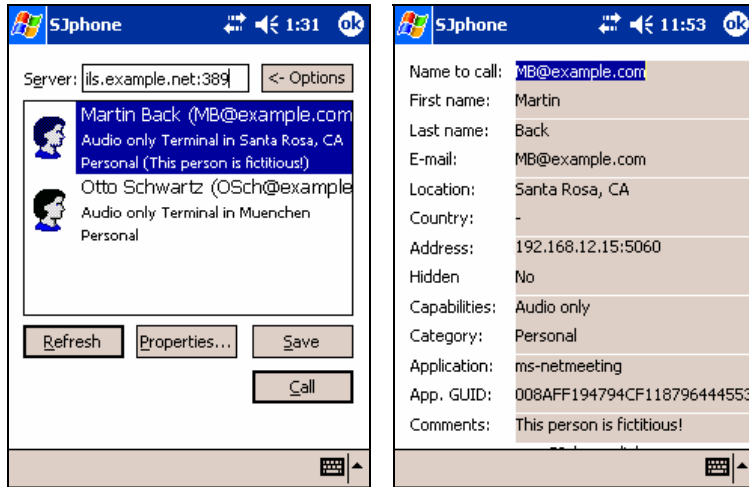
- Call Tap it to call the selected respondent
- Refresh Tap it to refresh the list of neighbors
- Add to Phonebook Tap it to add a selected respondent to **Phonebook**.
- Properties Tap it to view the information about a selected neighbor

You may also reach **Neighbors** by switching to the **Neighbors** screen on the **Main** panel



ILS Directory Browser

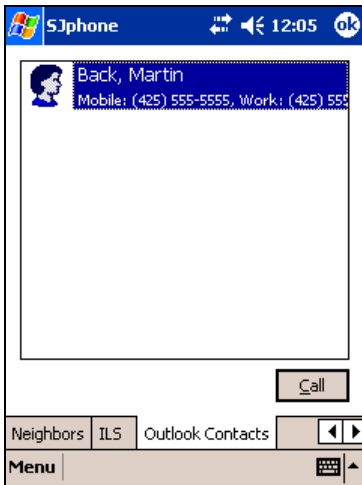
You may use **ILS Directory Browser** to browse through an ILS Directory server to find people. You may log on the ILS Directory specified on the **ILS Directory** tab. Tap the **<- Options** button.



- <- Options** Tap this button to copy ILS options from the **ILS Directory Options** tab.
- Refresh** Tap this button to refresh the **ILS Directory** list.
- Call** Tap this button to call a selected ILS user.
- Save** Tap this button to add a respondent from the **ILS Directory** list to **Phonebook**
- Properties** Tap this button to view the information about a selected ILS user.

Outlook Contacts

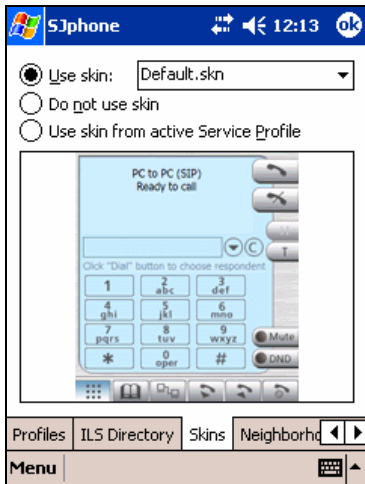
Shows the contacts in the **Contacts** with phone numbers.



- Call** Tap this button to call a selected contact using a currently selected service.

2.3. Skins

You can choose the **SJphone™ Main** panel appearance by using skins.



- To change a skin, tap the arrow and select a skin.
- Do not use skin** Select this option if you do not want to use skins
- Use skin** Select this option if you want to use a skin that you have selected
- Use skin from active Service Profile** Select this option if you want to use a skin specified in the active service profiles

III. Configuring SJphone™

This chapter explains how to configure **SJphone™** basic settings.

Attention! When any of the **Option** tabs is active, **SJphone™** automatically rejects all incoming calls.

Attention! Some tabs are not available on the default **Simple Mode** interface. To reach them, you need to switch to the **Advanced Mode**.

Interface mode switch:

- Tap **Menu** and select either **To Advanced Mode** or **To Simple Mode**

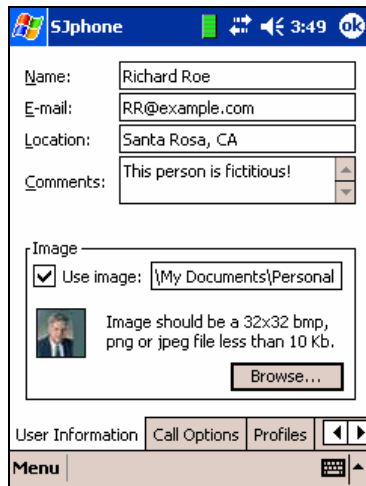
Attention! The **Advanced Mode** is unavailable for some service profiles.

You can reach **SJphone™** configuration controls by taping **Menu** and selecting **Options** on the **Main** panel.

3.1. Entering Your Personal Information

You may personalize your **SJphone™** copy when you start your **SJphone™** for the first time.

Note: This information will be available to your respondents while you are on a call.

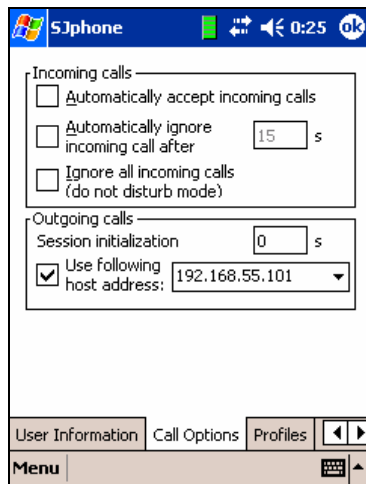


Name	Field for your name
E-mail:	Field for your e-mail
Location:	Field for your location
Comments:	Field for comments
Image	You may add a picture to your personal information. All other SJphone™ users will see it on their main panel.
Use image	Select this box if you want other SJphone™ users to view your picture
Browse	Tap this button to browse for your picture

You may submit this information to **ILS** (Internet user Location Service) if you want other people on the Internet to find you easily. Go to the **ILS Directory Options** section for details.

3.2. Call Options

You may set call options on the **Call Options** tab.



Incoming calls:	
Automatically accept incoming calls:	Select this box if you want SJphone™ to accept all incoming calls automatically
Automatically ignore incoming call after... secs:	Select this box if you want SJphone™ to automatically ignore an incoming call if you do not accept it within the specified time
Ignore all incoming calls:	Select this box if you want SJphone™ to ignore all incoming calls
Outgoing calls:	
Use following host address:	Select this box if your device has several IP addresses and select one from the list of available addresses
Session initialization	Timeout for which SJphone waits for a registration with a SIP proxy. If it fails to register during this timeout, the session will not start. 0 disables this feature

Incoming Calls:

You can configure your **SJphone™** to automatically accept or reject all incoming calls.

Only one of the **Automatically accept incoming calls**, **Automatically ignore incoming call after... secs**, or **Ignore all incoming calls** option can be selected at a time.

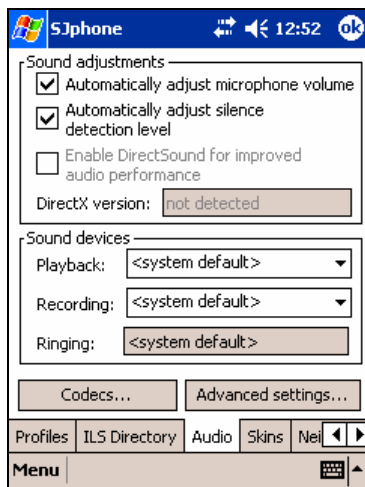
You may also select either **Auto accept calls** or **Do not disturb** from **Menu**.

Outgoing Calls:

If your device has several IP addresses, you need to select the **Use following host address** and an address from the list of available addresses. If this option is not selected, **SJphone™** uses your default address.

3.3. Audio Options

Generally, you can adjust the **SJphone™** sound characteristics using standard **PocketPC** audio controls. Some additional audio options may be set on the **Audio** tab.



Sound adjustments

Automatically adjust microphone volume:

Select this box to let **SJphone™** adjust the microphone volume automatically (currently not implemented)

Automatically adjust silence detection level:

Select this box to enable the automatic adjustment of silence detection level

Enable DirectSound for improved audio performance:

Select this box if your sound system supports **DirectSound**

DirectX version:

Check **DirectX** version installed in your system

Sound devices

Playback:

Shows what audio device **SJphone™** uses to playback audio.

If your system has several audio devices, you may select one from the list of available devices

Recording:

Shows what audio device **SJphone™** uses to record audio.

If your system has several audio devices, you may select one from the list of available devices

Codecs and Advanced Settings:

Tap these buttons to set **SJphone™** advanced audio and compression settings.

Automatically adjust microphone volume

(currently not implemented)

Some sound systems do not have the automatic microphone gain feature. You can select **Automatically adjust microphone volume** option to let **SJphone™** automatically adjust the microphone sound level.

Automatically adjust silence detection level

When the **Automatically adjust silence detection level** is selected, **SJphone™** does not send IP packets when you are silent, or you mute **SJphone™**. That reduces network load. If there is a strong background sound when this feature is selected, **SJphone™** may not detect silence correctly, and fragments of your speech may be lost.

Sound Devices

If your device has several sound devices, you may select which one will be used.

Advanced Audio and Codecs Settings (available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

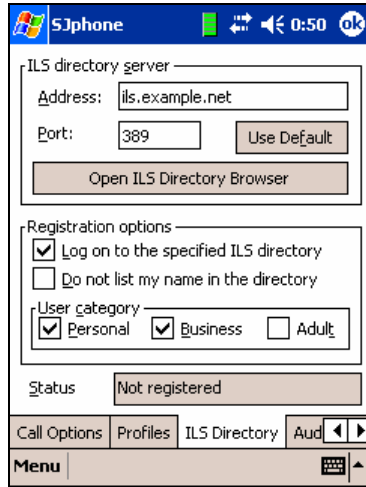
Advanced compression settings such as **Compression settings** or **SJ Labs Extensions** are explained in detail in the **Advanced Audio and Codecs Settings** section. Do not change them unless you completely understand what you are doing, or you may severely degrade the **SJphone™** performance.

3.4. ILS Directory Options

(available in the Advanced Mode interface only and may be unavailable for some service profiles)

ILS (Internet user Location Service) is an Internet service that helps people on the Internet to find and call each other. It is maintained on specially dedicated servers. You may obtain the list of such servers from your system administrator.

ILS Directory options can be set on the **ILS Directory** tab.



ILS Directory Server

Address:

Field for the **ILS Directory Server** address

Port:

Field for the **ILS Directory Server** port number

Registration options

Log on to the specified ILS directory

Select this box to log on to the **ILS Directory**

Do not list my name in the directory

Select this box if you do not want to be listed in the **ILS Directory** list. If you are logged on the directory, you will still be available to others who know your logon name.

User category

Select a box for the category you want to log on

Status:

Shows your status on the directory

Buttons

Use Default:

Tap this button to switch to the **ILS Directory Server** default port (389)

Open ILS Directory Browser

Tap this button to open the **ILS Directory Browser**.

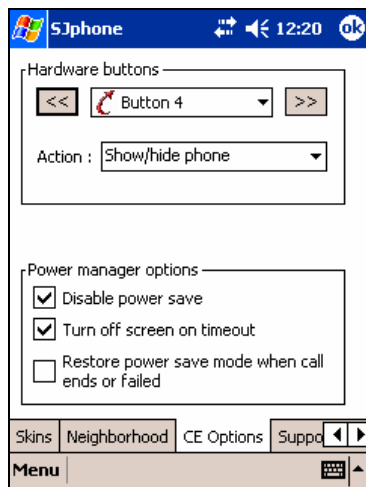
You may use ILS Directory Browser to locate people on a particular ILS server. Go to the **ILS Directory Browser** section for details.

Registration Options

To make yourself available to others you need to register on an ILS server. Using an ILS Directory, people call each other by their aliases. **SJphone™** uses the E-Mail field on the **User Information** tab as your alias on the ILS Directory. It also lists your first and last names from the **User Information** tab when you log on the specified ILS Directory. In addition it lists your IP address, location, comments, user category, and Audio/Video capability flag. If you do not want to be listed on the ILS Directory, select **Do not list my name in the directory**. Others can still call you by using your ILS alias.

SJphone™ logs on to the ILS directory when you tap the **OK** button on the **ILS Directory** tab or start **SJphone™** with selected **Log on to the specified ILS directory**. You need to be on-line at that time.

3.5. CE Options



Hardware Buttons

Select a button you want to use from the list and select an action you want to assign.

Power manager options (PocketPC 2003 (SE) only)

You may set options that allow **SJphone™** to receive incoming calls when your device is in a power save mode.

Disable power save

Select this option if you want **SJphone™** to receive incoming calls when your device is automatically turned off.

Turn off screen on timeout

Note! This option shortens the time your device can live on its battery.

Select this option if you want your device to turn its screen off in a timeout specified in the device's power settings.

Restore power save mode when call ends or failed

Select this option if you want **SJphone™** to return to its power save mode when a call ends or fails.

You may assign device buttons to perform various **SJphone™** actions such as showing/hiding **SJphone™**, receiving calls, etc. For PocketPC 2003(SE) devices, you may also use power management options that help your device conserve battery power when **SJphone™** waits for an incoming call.

3.6. Neighborhood Options

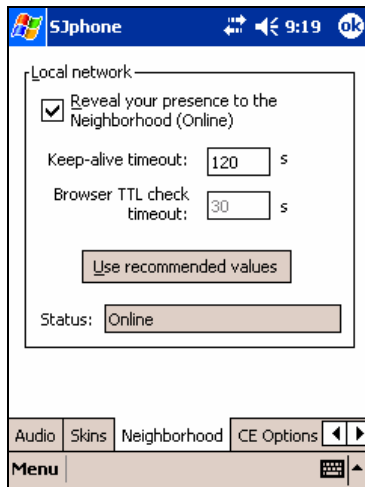
(available in the Advanced Mode interface only and may be unavailable for some service profiles)

You may set your neighborhood parameters on this tab.

Attention! This feature works only on local area networks!

When running, **SJphone™** constantly sends broadcast messages to the local network. Other **SJphones** running on computers on this network receive these messages and add **SJphone™** to their **Neighborhood** list. The users of those computers may find your presence and call you. You may enable or disable this feature.

You may use recommended values for this procedure. Tap the **Use recommended values** button.



Local network

Reveal your presence to the Neighborhood (Online):

Keep-alive timeout:

Status:

Buttons

Use recommended values

Check this box to reveal your presence to your network neighbors

Field for the timeout in which **SJphone™** sends messages about its presence to other **SJphones** on your local network.

Shows your neighborhood status.

Tap this button to set the values to their recommended values

IV. How to:

This chapter explains how to perform basic **SJphone™** operations.

4.1. Place a Call

You can place a call to any computer that has H.323 or SIP compatible software running, an H.323 or SIP compatible Internet phone, or a regular phone through a Gateway or a Gatekeeper.

To place a call, select an appropriate profile and type in the Call to: field on the Main panel one of the following:

- **An IP address or Computer Name for a direct PC-to-PC call:**

Depending on the VoIP protocol you want to use, you need to select either an H.323 direct or SIP direct profile. Go to the **Use profiles** section for details.

If your device is on a private network, you need to place

an H.323 call through a **Gatekeeper** using an appropriate profile. If there are no such profiles, you need to create such profile by yourself. Go to the **Profiles** help page for details.

a direct SIP PC-to-PC call using an available NAT traversal technology. Go to the **NAT Traversal and SIP** section for details.

Most computers connected to the Internet through dial-up lines have IP addresses dynamically assigned, so the only solution here is to use a Proxy or Gatekeeper account.

- **An Alias (User Name) on a Gatekeeper**

You need to select a profile for calls through gatekeepers. If there is no such profiles available, you need to create such profile by yourself. Go to the **Profiles** section for details.

- **A Caller ID for a call through a SIP Proxy**

You need to select a profile for calls through a SIP Proxy. If there is no such profiles available, you need to create such profile by yourself. Go to the **Profiles** section for details.

If your device is on a private network, you need to place the call using an available NAT traversal technology. Go to the **NAT Traversal and SIP** section for details.

- **A Regular Telephone Number**

You need to select a profile for calls through a proper Gateway if you want to call a regular telephone number. If there is no such profiles available, you need to create such profile by yourself. Go to the **Profiles** section for details.

You may also need to add a "prefix" (a special digit) to the phone number. Such prefixes are used to route calls to phones through required gateways. Ask the gateway administrator for details.

You may call the numbers stored in **Contacts**. Go to the **Contacts** section for details.

You may also use a Gatekeeper to make a call to a regular telephone number. Ask your gatekeeper administrator for details.

- **A Nickname in the Phonebook**

You need to specify all necessary information in the **Phonebook** about the nickname you want to type in. This can be done on the **Contacts** panel.

- **An H.323 Address**

SJphone™ supports the **advanced H.323 address syntax**. No service profiles are required for this call.

H.323 Address Examples

If you want to call

- A host `jd.bigcom.com (192.168.1.12)`
Type in the **Call To:** field: `H323:IP:jd.bigcom.com` or `H323:IP:192.168.1.12`
- A phone number `135-7975` through the H.323 Gateway `h323gateway.com`
Type in the **Call To:** field: `H323:GW:h323gateway.com:1357975`
- A user with the nickname `doe` registered with the H.323 Gatekeeper

- Type in the **Call To:** field: H323:GK:H323ID:doe
- A phone number 135 registered with the H.323 Gatekeeper
Type in the **Call To:** field: H323:GK:E164:135
- An external phone number 9-1357975 through the H.323 Gatekeeper
Type in the **Call To:** field: H323:GK:E164:91357975
- **A SIP URL**
SJphone™ supports the **advanced SIP URL syntax**. No service profiles are required for this call.

SIP URL Examples

If you want to call

- A host `jd.bigcom.com` (192.168.1.12)
Type in the **Call To:** field: `sip:jd.bigcom.com` or `sip:192.168.1.12`
- A user `johndoe` at `jd.bigcom.com`
Type in the **Call To:** field: `sip:johndoe@jd.bigcom.com`
- A phone number 135-7975 through a SIP gateway `sipgateway.com`
Type in the **Call To:** field: `sip:135-7975@sipgateway.com;user=phone`
- **An ILS Contact**

SJphone™ supports advanced ILS contact syntax. No service profiles are required for this call.

ILS Directory Contact Syntax:

An ILS Directory contact can be represented as the following string:

```
ils:<Alias>[:<Server>[:<Port>]]
```

where `<Alias>` = User alias with which it registers on the ILS Directory

`<Server>` = ILS Directory server IP address or dns name

`<Port>` = Port number (default is 389)

If you do not specify an ILS Directory server directly in the **Call To** string, **SJphone™** will take it from the **ILS Directory Options**.

ILS Contact Examples

If you want to call

- A person registered as `johndoe@bigcom.com` at a default ILS directory
Type in the **Call To:** field: `ils:johndoe@bigcom.com`
- A person registered as `johndoe@bigcom.com` at an ILS directory at the `ilsserver.com` server
Type in the **Call To:** field: `ils:johndoe@bigcom.com:ilsserver.com`

You may also use an **ILS Directory Browser** to log in to a particular ILS directory server. Go to the **Contacts** section for details.

And tap the Dial button on the SJphone™ Main panel.

Alternatively, you can use **Neighborhood Browser**, **Speed Dial** or **Contacts** to place a call.

SJphone™ stores a list of recently called addresses, names, nicknames, and telephone numbers. To access this list, tap the arrow next to the **Call To:** field or use the **Up** and **Down** arrow keys.

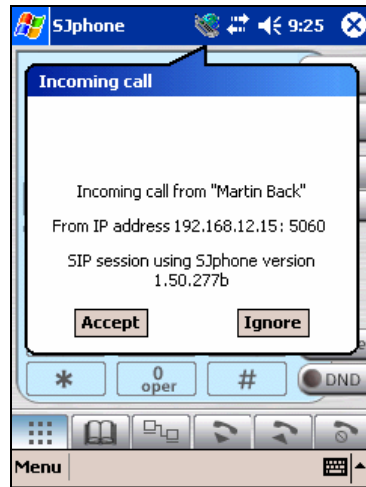
If there is no H.323 or SIP compatible software running on the computer you are calling, you will see the *Connection failed* message in the status line.

4.2. Answer a Call

You need to have **SJphone™** running to answer an incoming call.

When your device receives a call, you will hear a ring sound. Simultaneously, an **Incoming Call** message will appear.

- **To answer** the call, tap the **Accept** button.
- **To reject** the call, tap the **Ignore** button.



You can configure your **SJphone™** to automatically accept or reject all incoming calls. Go to the **Call Options** section for details.

Attention! When any of the **Option** tabs is active, **SJphone™** automatically rejects all incoming calls.

4.3. Control an Active Call

You may put your active call on hold or transfer it to another respondent.

Works not for all call types. If **SJphone™** cannot control the call, the **Hold** and **Transfer** buttons are disabled.

To put an active call on hold,

- Select a call you want to put on hold and tap the **Hold** button
- or
- Tap and hold a call you want to put on hold and select **Hold** in the pop-up menu

To resume a call on hold,

- Select a call you want to resume and tap the **Hold** button
- or
- Tap and hold a call you want to resume and select **Resume** in the pop-up menu

To transfer an active call to another respondent,

Blind transfer:

- Select a call you want to transfer and tap the **Transfer** button
- Select **Enter number** and enter the respondent address in the **Call To** field
- Tap the **Transfer** button once again

Attended transfer:

- Establish a call to a respondent to whom you want to transfer the call
- Select a call you want to transfer and tap the **Transfer** button or Tap and hold the respondent and select **Transfer** in the pop-up menu.
- Select the call with the respondent in the pop-up menu

4.4. Switch between Respondents

On the **Main** panel, tap the respondent with whom you want to talk.

4.5. Start a Conference

Although **SJphone™** supports connections with multiple respondents, you may usually talk only to one respondent at a time.

If you want to talk to several respondents simultaneously,

- If there are other calls with parties you do not want to include in the conference, put them **on hold**.
- Tap **Menu** and select **Conference**.

All active respondents will be included into the conference. Now you may talk to several respondents simultaneously. They also can talk to each other.

If you want to exclude someone from the conference, select the party and put her/him **on hold**. To include someone back to the conference, select her/him and **resume** the call.

4.6. Find Other SJphone™ Users on the Local Network

When running, **SJphone™** constantly sends broadcast messages to the local network. Other **SJphones** running on computers on this network receive these messages and add the **SJphone** to their **Neighborhood** list. You may enable or disable this feature. Go to the **Neighborhood Options** section for details.

To automatically find other SJphone™ users on your local network,

- Tap the **Contacts** button and select the **Neighbors** tab.

Go to the **Neighbors** section for details.

4.7. Receive a Voice Mail

Note: Your service should support this feature

If you have voice mail waiting, you will see a message in the respondent control showing how many messages are waiting.

To listen to the messages, make a call to your voice mail box and follow the instructions.

If you use this feature often, you may create a special record in the Phonebook for the voice mail box address/number.

4.8. Send a DTMF Signal

SJphone™ supports **DTMF** signal sending both for SIP and H.323 protocols.

To send a DTMF signal,

- Tap a digit button on the **Main** panel
- or
- Use a keyboard on your device.

The **DTMF** tone will be sent to the respondent's device and played on the host system through the playback sound device selected on the **Audio** tab.

If **SJphone™** plays back **DTMF** signals too loudly on the host system, you may disable local DTMF playback by selecting the **Disable local DTMF playback** option on the **Advanced Settings** panel.

Depending on the VoIP protocol and service, signals may be sent using various methods. The **DTMF** method is specified in the service profile.

4.9. Use Service Profiles

SJphone™ uses predefined sets of options for specific types of calls or for calls through different Internet Telephony Service Providers (ITSP), called *service profiles*. You can download profile from a website or create your own, although that requires some knowledge of VoIP technology.

If you have several profiles, you may easily switch between them.

To select a profile,

- Tap **Menu**, select **Services**, and then the required profile

or

- Tap the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab.
- Select the required profile and tap the **Use** button.

Some profiles require initializing, that is, they require the user to enter some account-specific information, such as a phone number, access password, etc. A **Service Properties** message will appear the first time you select such profile. You may also re-initialize the profile, if its account-specific information changes.

To initialize or re-initialize a profile,

- Tap **Menu**, select **Services**, and then the required profile
- Tap **Menu** once again and select **Re-initialize** profile.

The **Service Properties** message will appear

- Enter the required information and tap the **OK** button

or

- Tap the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab.
- Select the required profile and tap the **Initialize** button.

The **Service Properties** message will appear

- Enter the required information and tap the **OK** button

4.10. End a Call

To end a call,

- Select the respondent and tap the **Hangup** button on the **SJphone™ Main** panel,

or

Tap and hold the respondent and select the **Hangup** item in the pop-up menu.

4.11. Restart SJphone™

To restart SJphone™,

- Tap **Menu** and select **Restart**.

4.12. Exit SJphone™

To exit SJphone™,

- Tap **Menu** and select **Shut down**.

Note: When you exit **SJphone™** you will be unable to receive incoming calls.

V. Advanced Features

The following features are for advanced users. You need to be familiar with the details of the ITU-T H.323 and SIP protocols, audio codecs, and other technical details in order to use them correctly.

Attention! When any of the **Option** tabs is active, **SJphone™** automatically rejects all incoming calls.

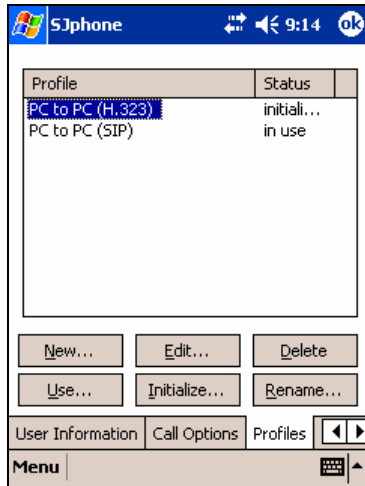
5.1. Profiles

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

SJphone™ may use predefined sets of options for specific types of calls or for calls through different Internet Telephony Service providers (ITSP) called *service profiles*.

An example of a service profile is that for the direct call to a host using the SIP VoIP protocol. You do not need to enter a complex dial string like sip:johndoe@jd.bigcom.com. Instead, you may select the SIP direct profile and type the URL in the **Call To:** field. Or, if you use several ITSPs, you may easily switch between them just selecting a required profile, usually supplied by the service provider.

You may even create your own profile for your IP-PBX, ITSP, or Gateway with your own set of options, although that requires some knowledge of VoIP technology.



New	Tap this button to create a new profile
Edit	Tap this button to edit a selected profile
Delete	Tap this button to delete a selected profile
Use	Tap this button to switch to a selected profile
Initialize	Tap this button to initialize a selected profile
Rename	Tap this button to rename a selected profile

To select a profile,

- Tap **Menu**, select **Services**, and then the required profile
- or
- Tap **Menu**, select **Options**, and go to the **Profiles** tab
- Select the required profile and tap the **Use** button.

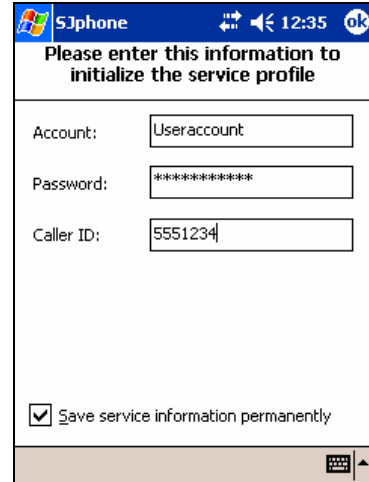
Some profiles require initializing, that is, they require the user to enter some account-specific information, such as a phone number, access password, etc. A **Service Properties** message will appear the first time you select such profile. You may also re-initialize the profile, if its account-specific information has changed.

To edit a profile,

- Tap **Menu**, select **Options**, and go to the **Profiles** tab
- Select the required profile and tap the **Edit** button
- Change the required parameters and tap the **OK** button

To initialize or re-initialize a profile,

- Tap **Menu**, select **Services**, and then the required profile
 - Tap **Menu** once again and select **Re-initialize** profile.
The **Service Properties** message will appear
 - Enter the required information and tap the **OK** button
- or
- Tap **Menu**, select **Options**, and go to the **Profiles** tab
 - Select the required profile and tap the **Initialize** button.
The **Service Properties** message will appear
 - Enter the required information and tap the **OK** button



To remove a profile,

- Tap **Menu**, select **Options**, and go to the **Profiles** tab
- Select the required profile and tap the **Remove** button
A **Delete Profiles** message will appear



Tap the **Yes** button to confirm profile removal.

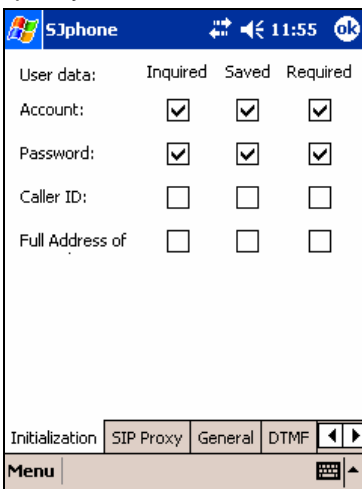
Attention! If you accidentally remove a profile, the only way to restore the profile is to create it once again.

5.2. Creating your own service profile

SJphone™ allows you to create your own profile for your ITSP (Internet Telephony Service Provider) or your own SIP proxy, H.323 gatekeeper or gateway. Please keep in mind that creating a profile requires some knowledge of at least basic details of the SIP and ITU-T H.323 VoIP protocols. If you experience problems, your best friend(s) is your ITSP support or/and network administrator. Always consult them when in doubt.

Initialization

Some profiles require initializing, that is, they require the user to enter some account-specific information, such as a phone number, access password, etc. You will enter this information during profile initialization. Initialization information may be either permanently stored, or you will have to enter every time you switch to this profile. You will specify on the **Initialization** tab which information is required for initialization and how it will be stored.



Initialization options:

- Inquired** If selected, this parameter will be asked during profile initialization
- Saved** If selected, this parameter will be permanently saved
- Required** If selected, this parameter is required for this profile

SIP Profiles

Calls through a SIP Proxy

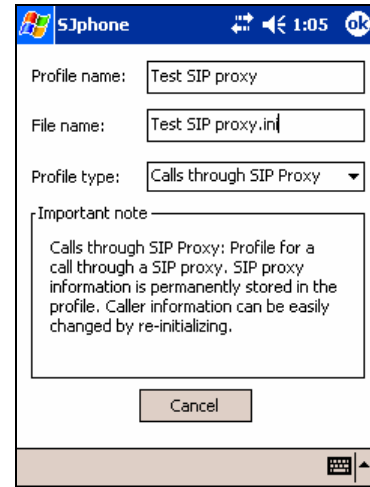
This profile is used to make calls to another computer using a registration with a SIP proxy.

To create a new profile for calls through a SIP proxy,

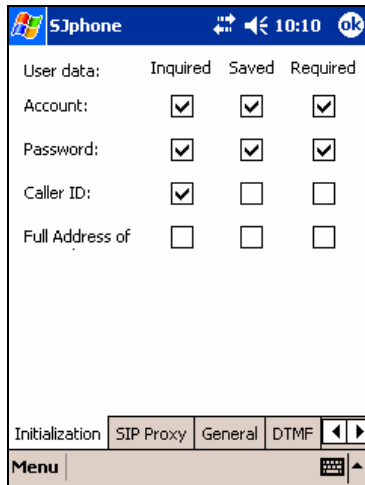
- Tap the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
- Tap the **New** button

A **Create New Profile** window will appear

- Select **Calls through SIP Proxy** in **Profile**, specify a name for the profile, and tap the **OK** button.
- Specify the required profile options and tap the **OK** button.



Initialization



Accounts	Your account (login) at the proxy
Password	Your password at the proxy
Caller ID	Your Caller ID at the proxy. Used to form your SIP Address of Record
Full Address of Record	Your Full Address of Record. Used to form your SIP Address of Record

Either Caller ID or Full Address of Record can be specified.

SJphone™ forms your **SIP Address of Record** in the following way:

If **Full Address of Record** is specified explicitly, it is used as the **SIP Address of Record** regardless of other parameters.

Otherwise, the **Address of Record** is a string of the following form:

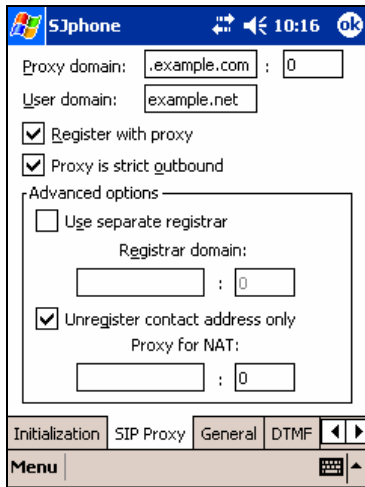
sip:CallerID@UserDomain.

Where

CallerID is selected on the **Initialization** tab and specified in the **Caller ID** option during initialization. If this option is not selected, **Account** will be used.

UserDomain is specified in the **User domain:** option on the **SIP Proxy** tab. If this option is not specified, **Proxy domain** on the same tab will be used.

SIP Proxy



Proxy domain The address and port for the SIP proxy. May be either a DNS name or IP address. This option is used to form a user's address-of-record (if its Full Address of Record is not explicitly specified) and the user's registrar address and the REGISTER queries (if the Use separate registrar option is not selected and user's Registrar domain is not explicitly specified). It is also used to form a full SIP URL for outbound calls if incomplete information, such as a phone number or a user name, is entered. Enter 0 if you do not want to specify an exact port number.

User domain The address and port for the user domain. May be either a DNS name or IP address. If the User domain is specified, it will be used to form the user's address-of-record and incomplete SIP URLs for outbound calls instead of the Proxy domain option. Enter 0 if you do not want to not specify the port number.

Register with proxy If this option is selected, **SJphone™** will register with the proxy. Always select this option unless instructed otherwise. Recommended status: **Selected**.

Proxy is strict outbound

If this option is selected, all queries (including REGISTER) will be sent through this proxy, regardless of the SIP URI obtained or entered during the query forming. If this option is not selected, the queries are sent to the address obtained using the resolution procedure from RFC 3263. Recommended status: **Selected**.

Advanced options

Do not change these options unless you have reasons to do so.

Always consult your ITSP support or/and network administrator when in doubt.

Use separate registrar

A very rarely used option. If this option is selected, **SJphone™** will register with a separate SIP entity rather than with the proxy specified in the Proxy domain option. Recommended status: **Cleared**.

Registrar domain

The address and port for the separate SIP registrar. May be either a DNS name or IP address. Enter 0 if you do not want to specify an exact port number. Used if the Use separate registrar option is selected.

Unregister contact address only

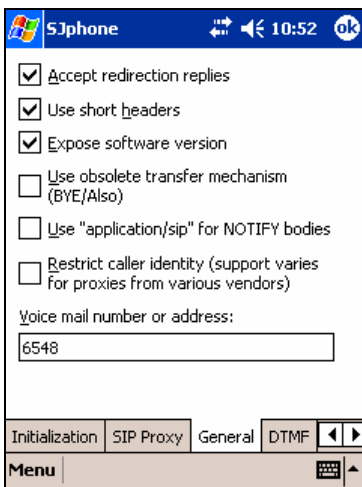
If this option is selected, during the process of unregistering, **SJphone™** will unregister only its temporary SIP URL associated with the current running **SJphone™** instance. If this option is cleared, **SJphone™** will unregister all temporary SIP URLs associated with the permanent user address-of-record. Recommended status: **Selected**.

Proxy for NAT

The address and port for the SIP proxy used to traverse NAT. May be either a DNS name or IP address. If **SJphone™** detects, using the STUN procedure, that it is behind NAT, this address will be used instead of the Proxy domain address. Enter 0 if you do not want to not specify the port number.

SJphone™ resolves addresses for the REGISTER and INVITE (the first one in a session) queries using the procedure described in RFC 3263.

General



Accept redirection replies If this option is selected, **SJphone™** will automatically call the addresses returned with redirection replies. (Valid only for SIP URI). Recommended status: **Selected**.

Use short headers If this option is selected, **SJphone™** will use short header names in SIP messages. Clear this option only if your system does not support short headers. Recommended status: **Selected**.

Expose software version If this option is selected, **SJphone™** will send information about itself in the Server and User-Agent headers. Recommended status: **Selected**.

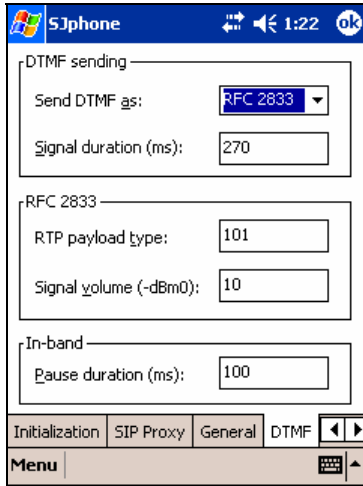
Use obsolete transfer mechanism (BYE/Also) If this option is selected, **SJphone™** will use the BYE/Also mechanism for blind transfers. In fact, this mechanism is deprecated. Recommended status: **Cleared**.

Restrict caller identity (support varies for proxies from different vendors) Use this option only if your SIP proxy supports this feature. Recommended status: **Cleared**.

Voice mail number or address

If the SIP proxy supports this feature, enter a voicemail number or address.

DTMF



DTMF sending

SJphone supports 3 methods:

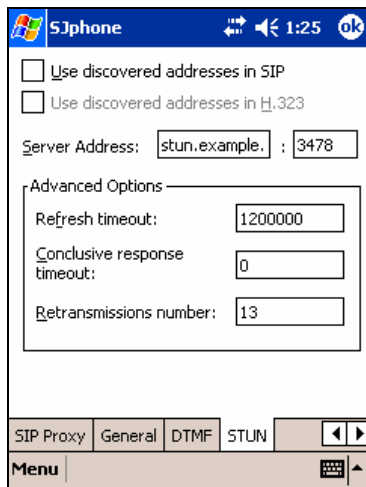
1. **In-band:** DTMF signals are sent as digitized sound in the RTP channel.
2. **RFC2833:** DTMF signals are sent as special packets controlled in the RTP channel.
3. **SIP INFO:** DTMF signals are sent in the **INFO** message in the SIP session control channel.

If the **SIP INFO** method is selected, the **DTMF** signals are always sent in the **INFO** message. If the **RFC2833** method is selected and the remote system does not support this method, the **In-band** method will be used.

For a direct call, the remote user does not hear **DTMF** signals when the **SIP INFO** or **RFC2833** methods are used. For a call through a SIP proxy, this depends on the proxy. Some convert **DTMF** signals into sound, some do not. The remote user will hear **DTMF** signals when the **In-band** method is used.

Depending on your local DTMF standard, you may need to adjust some DTMF parameters. If necessary, consult your ITSP support or/and network administrator.

STUN



Use discovered addresses in SIP

If this option is selected, **SJphone™** will use addresses discovered by the STUN procedure in SIP packets. Recommended status: **Cleared**.

Server Address

The address and port for the STUN server. May be either a DNS name or IP address. Unless instructed otherwise, do not change this option.

Advanced Options

Use this option if the STUN discovery procedure works unstably or does not work at all. Always consult your ITSP support or/and network administrator when in doubt. Do not change them unless you know what you are doing.

Refresh timeout

A time interval in milliseconds after which a status of each network interface is determined. Should be at least 100 msec.

Conclusive response timeout

A time interval in milliseconds during which the STUN client waits for re-transmitted answers after the first answer to the STUN query has been received. Should be at least 100 msec. RFC3489 recommends 10000. The special value 0 means that the STUN client will not wait for the re-sent answers.

Retransmission number

The maximum number of re-transmitted first STUN queries when the network interface status is being determined. Should be less than, or equal to, 13.

Initialization:**To initialize or re-initialize this profile,**

- Tap **Menu**, select **Services**, and then the required profile
 - Tap **Menu** once again and select **Re-initialize**
 - Tap **Menu**, select **Options**, and go to the **Profiles** tab
 - Select the required profile and tap the **Initialize** button.
- The **Service** message will appear
- Enter the required information and tap the **OK** button
 - Select Save service information permanently if you do not want to enter this information every time you select this profile.

NAT traversal and SIP**Symptoms that you have problems with NAT:**

- Audio goes only one way (either inbound or outbound).
- Incoming calls cannot reach you.

Note: If you experience problems with NAT your best friend(s) is your ITSP support or/and network administrator. Always consult them when in doubt.

Most likely you are behind a NAT if you use an Internet gateway connecting your local network to a broadband network via a DSL or cable modem, use Internet Connection Sharing in Windows XP, or are on a private network (your computer has an IP address like 10.x.x.x, 192.168.x.x, 172.16.0.0 - 172.31.255.255).

If your service profile allows you to do this, you may check if **SJphone™** is behind NAT. To do so, switch to **Advanced Mode**.

You **are NOT** behind NAT if you see the following:

- NAT/Firewall: Open Internet

You **are** behind NAT if you see the following:

- NAT/Firewall: Full Cone NAT
- NAT/Firewall: Restricted Cone NAT
- NAT/Firewall: Port Restricted Cone NAT
- NAT/Firewall: Symmetric NAT
- NAT/Firewall: NAT of Unknown Type
- NAT/Firewall: Symmetric Firewall (Strictly speaking, a symmetric firewall is not a NAT device but **SJphone™** works behind a symmetric firewall as it were behind NAT.)

It is **unclear** if you are or are not behind NAT if you see the following:

- NAT/Firewall: Unknown
- NAT/Firewall: Blocked

SJphone and NAT

NAT (stands for Network Address Translation) is a widely-used solution that allows multiple PCs on a private network to share a single, globally accessible IP address. A main reason to use NAT is the fact that IP addresses rapidly become scarce. In addition, NAT provides an extra layer for network security by obscuring a private network from an outsider. Please note that NAT is not the same thing as a firewall or proxy server.

Unfortunately, NAT makes communications for **SJphone™** and other Internet phones much more difficult. The situation is even worsened by the fact that there are several types of NAT, each imposing different restrictions, and different methods are required to traverse a NAT. **SJphone™** provides most of them.

NAT traversal methods

Below are NAT traversal methods that **SJphone™** supports. Which is to be used, depends on your ITSP, on your network configuration, etc. Always consult your ITSP support or/and network administrator when in doubt.

1. Symmetric media streams: **SJphone™** is preconfigured to use it. No additional configuring is required.
2. STUN protocol
3. rport SIP extension
4. OPTIONS request registration refresh
5. Outgoing INVITE transaction refresh

Settings in the *Call through SIP Proxy* profile type for the NAT traversal

Attention! You must completely understand what you are doing while changing these settings. Always consult your ITSP support or/and network administrator when in doubt.

Consult your ITSP or SIP Proxy administrator on which method to use:

1. A second proxy is used for NAT traversal:
 - Enter the proxy's DNS name or IP address to the Proxy for NAT field on the **SIP Proxy** tab. Enter the port number if it is not a default value (5060) and there is no this information in the NAPTR/SRV DNS records.

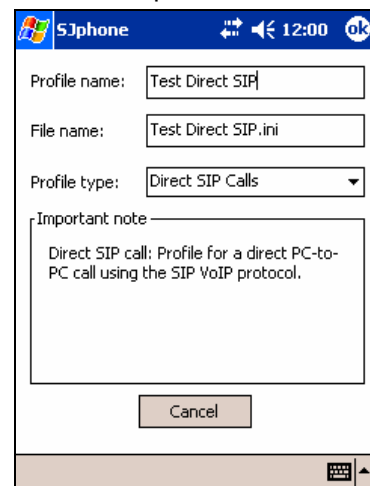
Note: If a second proxy for NAT is used, SIP messages do not include the STUN information.
2. One proxy supports NAT traversal or rport.
 - Disable STUN usage by clearing the Use discovered addresses in SIP option on the **STUN** tab.
3. One proxy does not support NAT traversal
 - Use discovered addresses in SIP option on the **STUN** tab. Enter STUN settings on the STUN tab. For the most cases the default settings are correct.
 - This is the worst case and you may experience problems with calls. Consult your ITSP (Internet Telephony Service Provider) support or/and network administrator in this case.

Direct SIP PC-to-PC call

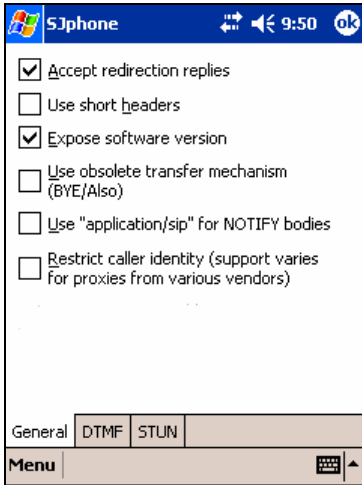
This profile is used to make direct calls to another computer using the SIP VoIP protocol.

To create a new profile for direct SIP PC-to-PC calls,

- Tap the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
 - Tap the **New** button
- A **Create New Profile** window will appear
- Select **Direct SIP Calls** in **Profile**, specify a name for the profile, and tap the **OK** button.
 - Specify the required profile options and tap the **OK** button.



General



Accept redirection replies

If this option is selected, **SJphone™** will automatically call the addresses returned with redirection replies. (Valid only for SIP URI). Recommended status: **Selected**.

Use short headers

If this option is selected, **SJphone™** will use short header names in SIP messages. Clear this option only if your system does not support short headers. Recommended status: **Selected**.

Expose software version

If this option is selected, **SJphone™** will send information about itself in the Server and User-Agent headers. Recommended status: **Selected**.

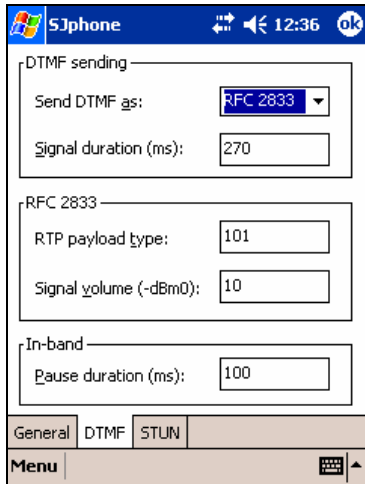
Use obsolete transfer mechanism (BYE/Also)

If this option is selected, **SJphone™** will use the BYE/Also mechanism for blind transfers. In fact, this mechanism is deprecated. Recommended status: **Cleared**.

Restrict caller identity (support varies for proxies from different vendors)

Use this option only if your SIP proxy supports this feature. Recommended status: **Cleared**.

DTMF



DTMF sending

SJphone supports 3 methods:

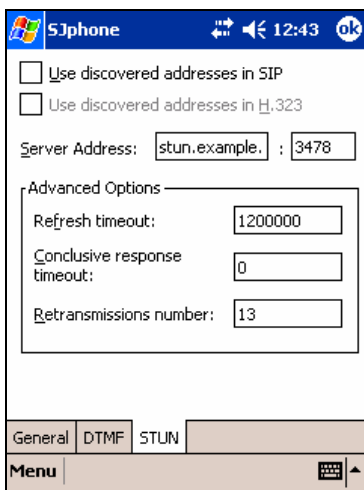
1. **In-band:** DTMF signals are sent as digitized sound in the RTP channel.
2. **RFC2833:** DTMF signals are sent as special packets controlled in the RTP channel.
3. **SIP INFO:** DTMF signals are sent in the **INFO** message in the SIP session control channel.

If the **SIP INFO** method is selected, the **DTMF** signals are always sent in the **INFO** message. If the **RFC2833** method is selected and the remote system does not support this method, the **In-band** method will be used.

For a direct call, the remote user does not hear **DTMF** signals when the **SIP INFO** or **RFC2833** methods are used. For a call through a SIP proxy, this depends on the proxy. Some convert **DTMF** signals into sound, some do not. The remote user will hear **DTMF** signals when the **In-band** method is used.

Depending on your local DTMF standard, you may need to adjust some DTMF parameters. If necessary, consult your ITSP support or/and network administrator.

STUN



Use discovered addresses in SIP

If this option is selected, **SJphone™** will use addresses discovered by the STUN procedure in SIP packets. Recommended status: **Cleared**.

Server Address

The address and port for the STUN server. May be either a DNS name or IP address. Unless instructed otherwise, do not change this option.

Advanced Options

Use this option if the STUN discovery procedure works unstably or does not work at all. Always consult your ITSP support or/and network administrator when in doubt. Do not change them unless you know what you are doing.

Refresh timeout

A time interval in milliseconds after which a status of each network interface is determined. Should be at least 100 msec.

Conclusive response timeout

A time interval in milliseconds during which the STUN client waits for re-transmitted answers after the first answer to the STUN query has been received. Should be at least 100 msec. RFC3489 recommends 10000. The special value 0 means that the STUN client will not wait for the re-sent answers.

Retransmission number

The maximum number of re-transmitted first STUN queries when the network interface status is being determined. Should be less than, or equal to, 13.

Initialization: This profile does not require initialization.

H.323 Profiles

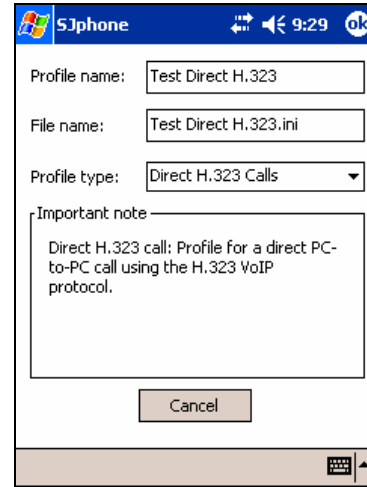
Direct H.323 PC-to-PC call

To create a new profile for direct H.323 PC-to-PC calls,

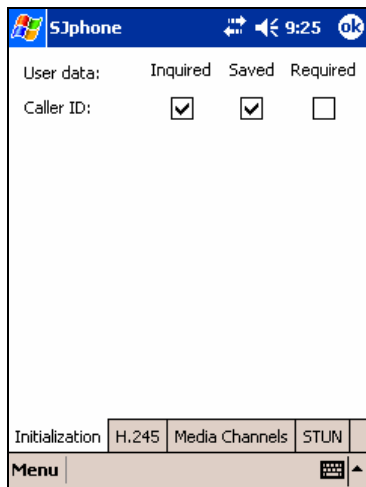
- Tap the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
- Tap the **New** button

A **Create New Profile** window will appear

- Select **Direct H.323 Calls** in **Profile**, specify a name for the profile, and tap the **OK** button.
- Specify the required profile options and tap the **OK** button.



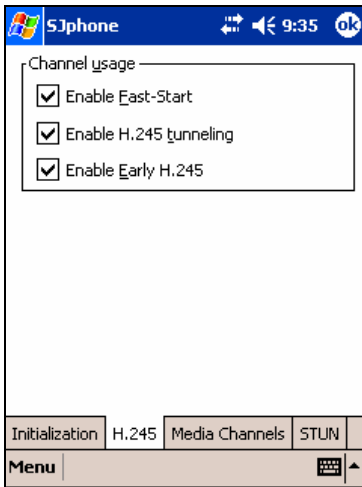
Initialization



Caller ID

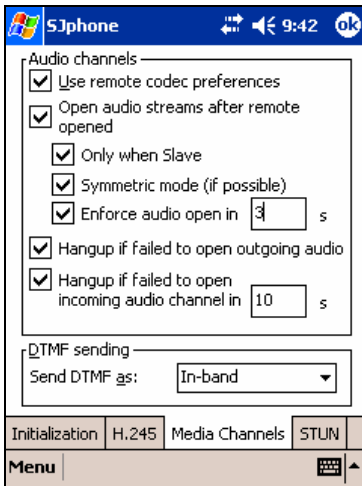
This number may be used by internet or telephone network providers for call identification and billing. Contact your system administrator or VoIP service provider for details

H.245



Channel usage These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by your system administrator.

Media Channels



Audio channels These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by your system administrator.

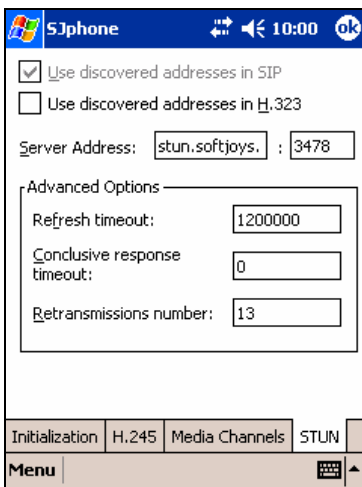
DTMF sending

SJphone supports 4 methods:

1. **In-band: DTMF** signals are sent as digitized sound. This is the choice if you want to control an external telephone device such as a fax machine, answering machine, etc.
2. **Q.931 keypad: DTMF** signals are sent through the Q.931 Keypad/Facility Information Element of Q.931 messages.
3. **H.245 alphanumeric: DTMF** signals are sent as one of the out-of-band method in H.245 protocol messages.
4. **H.245 signal: DTMF** signals are sent as one of the out-of-band method in H.245 protocol messages.

For a direct call, the remote user does not hear **DTMF** signals when the **Q.931 keypad**, **H.245 alphanumeric**, or **H.245 signal** methods are used. The remote user will hear **DTMF** signals when the **In-band** method is used.

STUN



Use discovered addresses in H.323 If this option is selected, SJphone™ will use addresses discovered by the STUN procedure in H.323 packets. Recommended status: **Cleared**.

Server Address The address and port for the STUN server. May be either a DNS name or IP address. Unless instructed otherwise, do not change this option.

Advanced Options Use this option if the STUN discovery procedure works unstably or does not work at all. Always consult your ITSP support or/and network administrator when in doubt. Do not change them unless you know what you are doing.

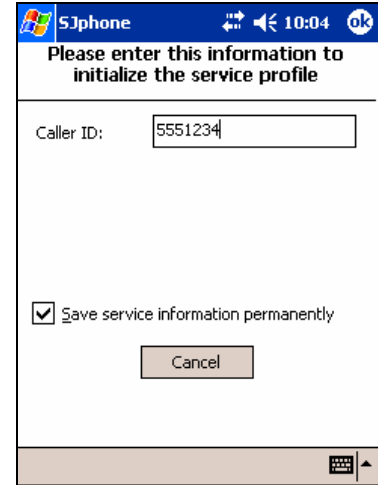
Refresh timeout A time interval in milliseconds after which a status of each network interface is determined. Should be at least 100 msec.

Conclusive response timeout	A time interval in milliseconds during which the STUN client waits for re-transmitted answers after the first answer to the STUN query has been received. Should be at least 100 msec. RFC3489 recommends 10000. The special value 0 means that the STUN client will not wait for the re-sent answers.
Retransmission number	The maximum number of re-transmitted first STUN queries when the network interface status is being determined. Should be less than, or equal to, 13.

Initialization:

To initialize or re-initialize this profile,

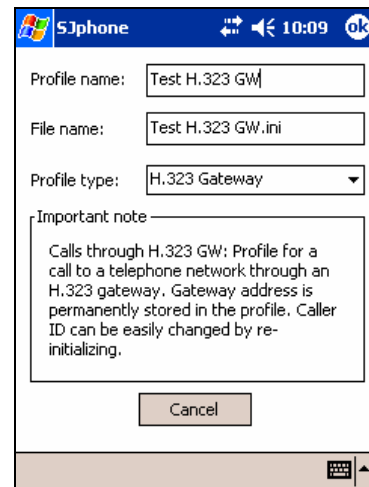
- Tap **Menu**, select **Services**, and then the required profile
 - Tap **Menu** once again and select **Re-initialize** profile.
- or
- Tap **Menu**, select **Options**, and go to the **Profiles** tab
 - Select the required profile and tap the **Initialize** button.
- The **Service** message will appear
- Enter the required information and tap the **OK** button
 - Select Save service information permanently if you do not want to enter this information every time you select this profile.



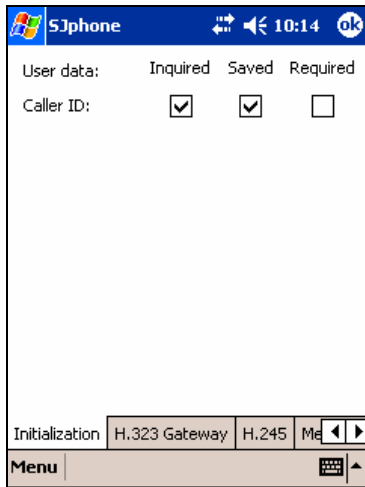
Calls through an H.323 Gateway

To create a new profile for calls through an H.323 gateway,

- Tap the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
 - Tap the **New** button
- A **Create New Profile** window will appear
- Select **Calls through H.323 Gateway** in **Profile**, specify a name for the profile, and tap the **OK** button.
 - Specify the required profile options and tap the **OK** button.

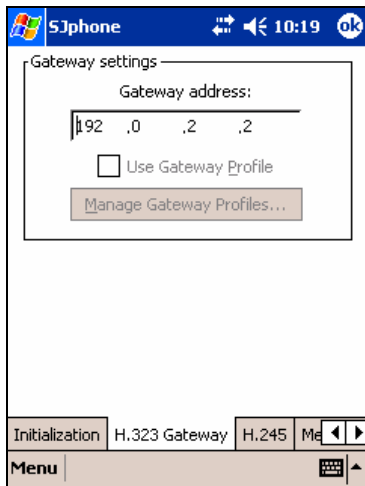


Initialization



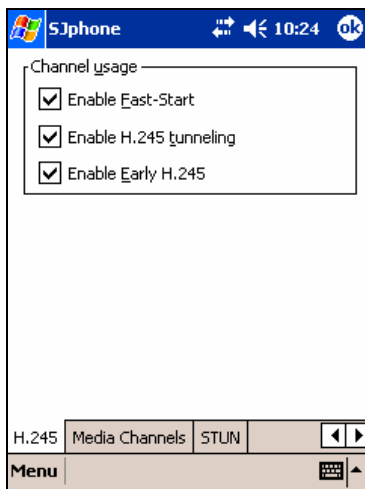
Caller ID This number may be used by the gateway for call identification and billing. Contact your system administrator or VoIP service provider for details

H.323 Gateway



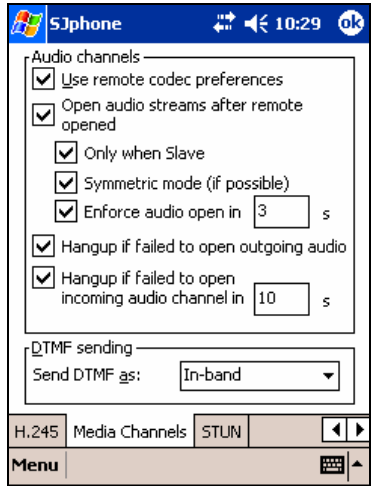
Gateway settings
Gateway address The Gateway IP address

H.245



Channel usage These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by your system administrator.

Media Channels



Audio channels These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by your system administrator.

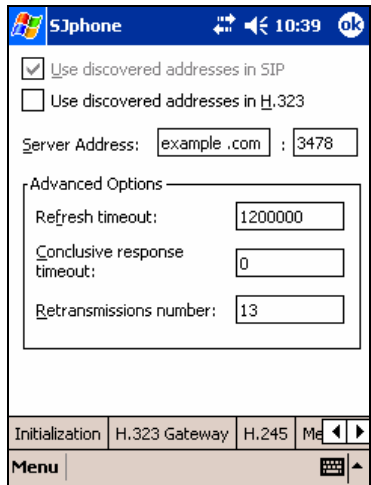
DTMF sending

SJphone supports 4 methods:

1. **In-band:** DTMF signals are sent as digitized sound. This is the choice if you want to control an external telephone device such as a fax machine, answering machine, etc.
2. **Q.931 keypad:** DTMF signals are sent through the Q.931 KeypadFacility Information Element of Q.931 messages.
3. **H.245 alphanumeric:** DTMF signals are sent as one of the out-of-band method in H.245 protocol messages.
4. **H.245 signal:** DTMF signals are sent as one of the out-of-band method in H.245 protocol messages.

For a direct call, the remote user does not hear **DTMF** signals when the **Q.931 keypad**, **H.245 alphanumeric**, or **H.245 signal** methods are used. For a call through an H.323 gateway, this depends on the gateway. Some convert **DTMF** signals into sound, some do not. The remote user will hear **DTMF** signals when the **In-band** method is used.

STUN



Use discovered addresses in H.323 If this option is selected, **SJphone™** will use addresses discovered by the STUN procedure in H.323 packets. Recommended status: **Cleared**.

Server Address The address and port for the STUN server. May be either a DNS name or IP address. Unless instructed otherwise, do not change this option.

Advanced Options Use this option if the STUN discovery procedure works unstably or does not work at all. Always consult your ITSP support or/and network administrator when in doubt. Do not change them unless you know what you are doing.

Refresh timeout A time interval in milliseconds after which a status of each network interface is determined. Should be at least 100 msec.

Conclusive response timeout

A time interval in milliseconds during which the STUN client waits for re-transmitted answers after the first answer to the STUN query has been received. Should be at least 100 msec. RFC3489 recommends 10000. The special value 0 means that the STUN client will not wait for the re-sent answers.

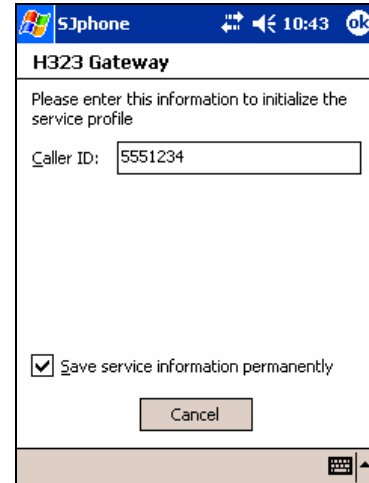
Retransmission number

The maximum number of re-transmitted first STUN queries when the network interface status is being determined. Should be less than, or equal to, 13.

Initialization:

To initialize or re-initialize this profile,

- Tap **Menu**, select **Services**, and then the required profile
 - Tap **Menu** once again and select **Re-initialize** profile.
- or
- Tap **Menu**, select **Options**, and go to the **Profiles** tab
 - Select the required profile and tap the **Initialize** button.
- The **Service** message will appear
- Enter the required information and tap the **OK** button
 - Select Save service information permanently if you do not want to enter this information every time you select this profile.

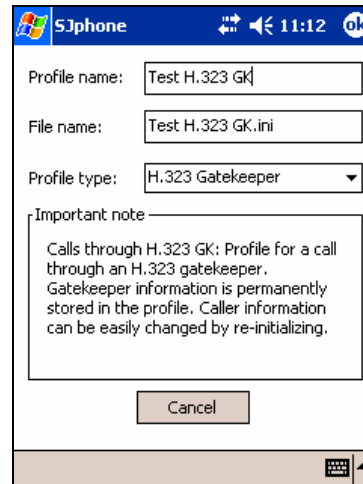


Calls through an H.323 Gatekeeper

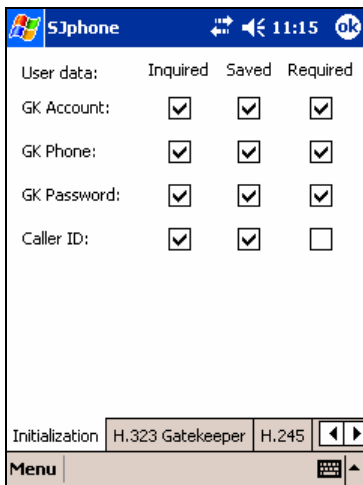
When you place a call to a regular phone through a gatekeeper, you may need to add a "prefix" (a special digit) to the phone number. Such prefixes are used to route calls to phones through required gateways. Ask the gatekeeper administrator for details.

To create a new profile for calls through an H.323 gatekeeper,

- Tap the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
 - Tap the **New** button
- A **Create New Profile** window will appear
- Select **Calls through H.323 Gatekeeper** in **Profile**, specify a name for the profile, and tap the **OK** button.
 - Specify the required profile options and tap the **OK** button.



Initialization



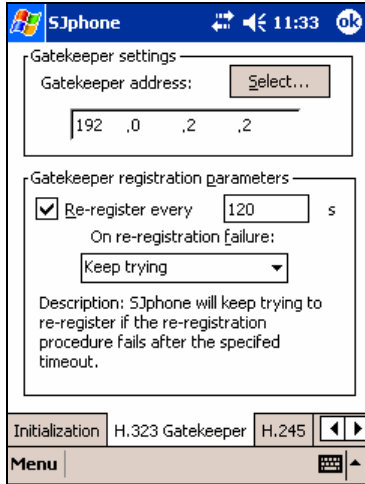
Gatekeeper Account
Gatekeeper Phone

Gatekeeper Password
Caller ID

A user account on the gatekeeper
You may register with a gatekeeper using only a phone number. If someone calls this number, the gatekeeper will direct the call to you. Contact the gatekeeper administrator for details.

A user password on the gatekeeper
This number may be used by the gatekeeper for call identification and billing. Contact your gatekeeper administrator or VoIP service provider for details

H.323 Gatekeeper



Gatekeeper settings

Gatekeeper address The Gatekeeper IP address.

Gatekeeper registration parameters

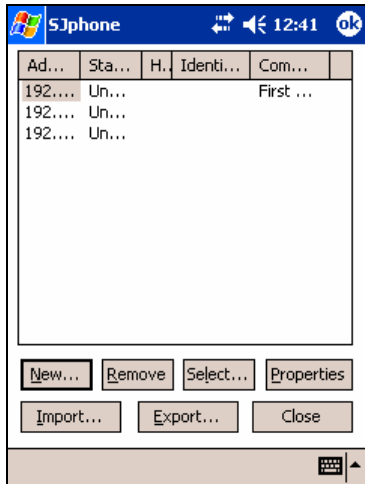
Reregister every...seconds Some gatekeepers require their users to re-register after a certain time interval. Time interval between re-registration with a gatekeeper. If you want to disable this feature, clear this option.

On re-registration failure This option specifies what **SJphone™** will do if it fails to re-register with a gatekeeper. Available options are **Keep trying** to re-register, or **Continue unregistered**. In the latter case, it will stop attempts to re-register, alert the user, and disable gatekeeper usage.

List of known gatekeepers

Your system administrator or VoIP service provider may give you a file with a list of known gatekeepers. Such files can be used for **SJphone™** versions for all OS

Tap the **Select...** button to activate the **Known Gatekeepers** panel.



Address s Shows the gatekeeper address

Status Shows the gatekeeper status. The status may be: Indeterminate, In progress, Available, Rejected, Not responding

H.323 Shows the version of the H.323 protocol that the gatekeeper supports

Identifier Shows the gatekeeper identifier

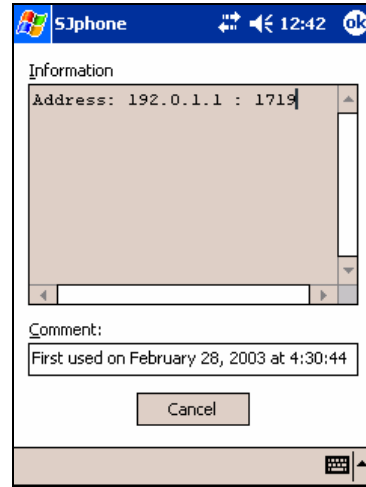
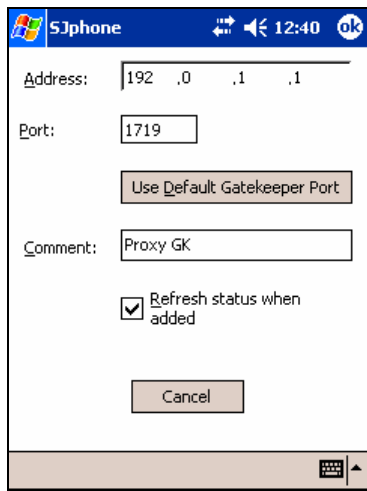
Comment Shows gatekeeper comments.

Buttons

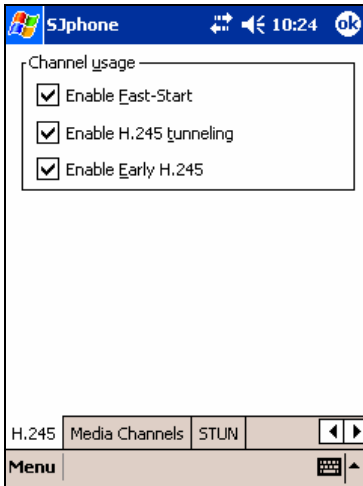
- New Tap this button to add a new gatekeeper to the list
- Remove Tap this button to remove the selected gatekeeper
- Properties Tap this button to see the properties of the selected gatekeeper
- Select Tap this button to select all gatekeepers with the same status
- Close Tap this button to close the list of known gatekeepers
- Import Tap this button to import a file with a list of gatekeepers. The default file extension is *.gk1.
- Export Tap this button to export the list of gatekeepers into a file. The default file extension is *.gk1.

You may add gatekeepers to the list. Tap the **New** button and an **Add Gatekeeper** window will appear. Enter the required information about the gatekeeper and tap the **OK** button.

You may see the properties of each gatekeeper. Tap the **Properties** button and the **Gatekeeper Properties** window will appear.



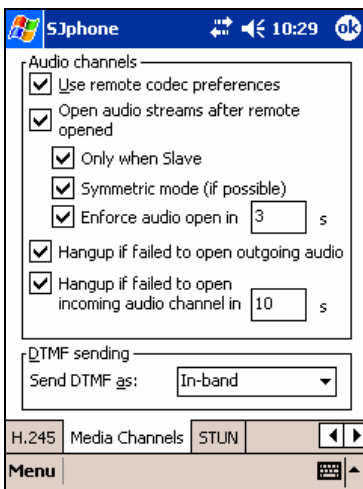
H.245



Channel usage

These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by your system administrator.

Media Channels



Audio channels

These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by your system administrator.

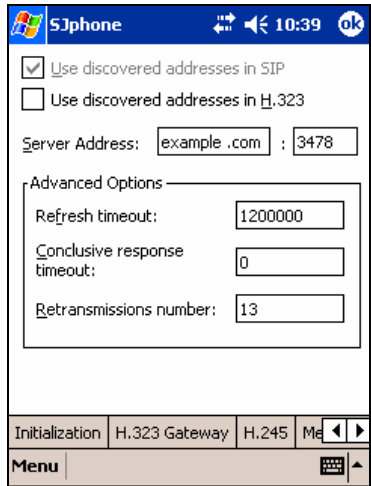
DTMF sending

SJphone supports 4 methods:

1. **In-band:** DTMF signals are sent as digitized sound. This is the choice if you want to control an external telephone device such as a fax machine, answering machine, etc.
2. **Q.931 keypad:** DTMF signals are sent through the Q.931 KeypadFacility Information Element of Q.931 messages.
3. **H.245 alphanumeric:** DTMF signals are sent as one of the out-of-band method in H.245 protocol messages.
4. **H.245 signal:** DTMF signals are sent as one of the out-of-band method in H.245 protocol messages.

For a direct call, the remote user does not hear **DTMF** signals when the **Q.931 keypad**, **H.245 alphanumeric**, or **H.245 signal** methods are used. For a call through an H.323 gatekeeper, this depends on the gatekeeper. Some convert **DTMF** signals into sound, some do not. The remote user will hear **DTMF** signals when the **In-band** method is used.

STUN



Use discovered addresses in H.323

If this option is selected, **SJphone™** will use addresses discovered by the STUN procedure in H.323 packets. Recommended status: **Cleared**.

Server Address

The address and port for the STUN server. May be either a DNS name or IP address. Unless instructed otherwise, do not change this option.

Advanced Options

Use this option if the STUN discovery procedure works unstably or does not work at all. Always consult your ITSP support or/and network administrator when in doubt. Do not change them unless you know what you are doing.

Refresh timeout

A time interval in milliseconds after which a status of each network interface is determined. Should be at least 100 msec.

Conclusive response timeout

A time interval in milliseconds during which the STUN client waits for re-transmitted answers after the first answer to the STUN query has been received. Should be at least 100 msec. RFC3489 recommends 10000. The special value 0 means that the STUN client will not wait for the re-sent answers.

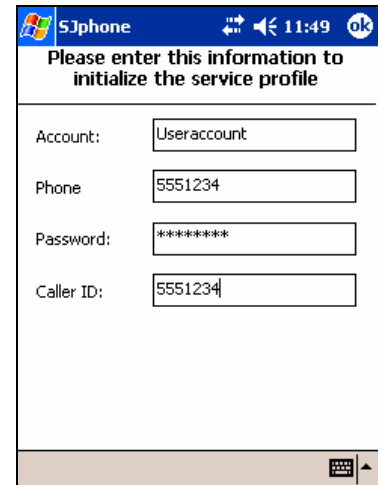
Retransmission number

The maximum number of re-transmitted first STUN queries when the network interface status is being determined. Should be less than, or equal to, 13.

Initialization:

To initialize or re-initialize this profile,

- Tap **Menu**, select **Services**, and then the required profile
 - Tap **Menu** once again and select **Re-initialize** profile.
- or
- Tap **Menu**, select **Options**, and go to the **Profiles** tab
 - Select the required profile and tap the **Initialize** button.
- The **Service** message will appear
- Enter the required information and tap the **OK** button
 - Select Save service information permanently if you do not want to enter this information every time you select this profile.



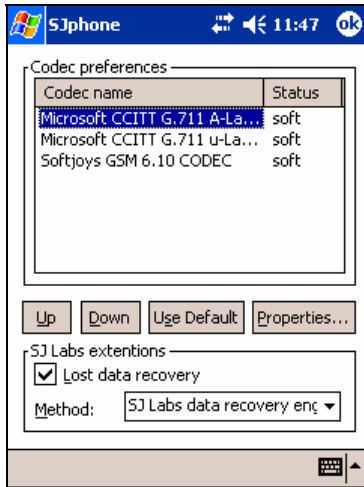
5.3. Advanced Audio and Codecs Settings

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

To change the advanced compression settings, tap the **Codecs** button on the **Audio** tab.

Attention! You must completely understand what you are doing while changing these settings, or you can severely degrade **SJphone™** performance.

When two VoIP systems are establishing a call, they negotiate an audio compression codec they are going to use. Which codec to choose depends on many factors: what codecs are installed on both systems, bandwidth limitations, desired sound quality, etc. You may adjust **SJphone™** codec preferences on the **Advanced Compression Settings** window. During the negotiation, **SJphone™** offers to the remote system the first codec from the list. If the remote system rejects the codec, **SJphone™** offers the next one from top to bottom until they both accept the codec. You may arrange the codecs in the desired order using the **Move Up** and **Move Down** buttons. The **Use Default** button selects the default codec preference.



Codec preferences

Codec name

Shows codec name

Status

Shows codec status. May be either:

- Soft: the codec is implemented as a software driver
- Hard: the codec is implemented as a hardware device
- Disabled: the codec is disabled.

A codec may be enabled or disabled on the Audio Codec Properties window.

Buttons

Up:

Tap this button to move the selected codec up

Down:

Tap this button to move the selected codec down

Use Default

Tap this button to use a default codec preference

Properties:

Tap this button to activate the Audio Codec Properties window

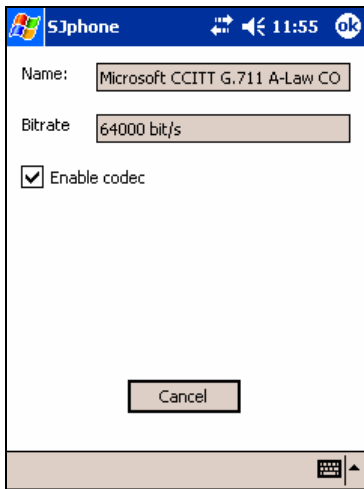
SJ Labs Extensions

Lost data recovery

Check this box to enable the advanced data recovery system

Method

Select the method of the advance data recovery from the list of available methods



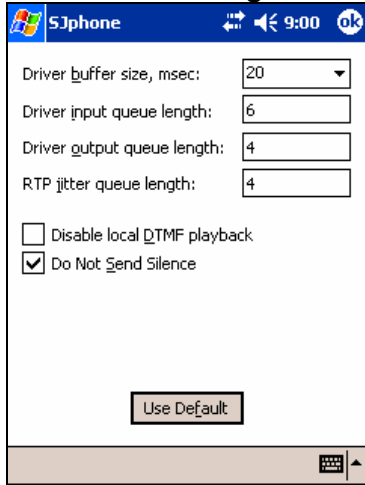
If you select a codec and tap the **Properties** button, or double-tap the codec, its **Audio Codec Properties** window will appear: This window shows some of the codec's basic properties. You may also enable or disable the codec on this message.

SJ Labs Extensions

Lost data recovery allows you to obtain good sound quality over poor Internet connections. The current **SJphone™** version supports **SJ Labs data recovery engine** .

Attention! SJ Labs data recovery engine is not fully H.323-compatible. Use it only while communicating with other **SJphone™**-compatible software.

SJ Labs data recovery engine sends additional IP packets. This doubles bandwidth requirements but completely restores up to 50% of lost packets. This feature is useful for wideband but unstable connections such as Mobile Internet, WaveLan, or Radio Ethernet.

Advanced Audio Settings

These settings allow you to change sizes of various internal **SJphone™** sound buffers and queues if you experience **problems with sound**.

Do not change them unless you completely understand what you are doing, or you may severely degrade the **SJphone™** performance.

If **SJphone™** plays back DTMF signals too loudly, you may disable local DTMF playback by selecting the **Disable local DTMF playback** option.

Do not Send Silence: If this option is selected, **SJphone™** does not send IP packets when you are silent, or you mute **SJphone™**. That reduces network load.

Recommended advanced audio settings

	Minimum	Recommended
Input queue length	4	6
Output queue length	2	4
RTP jitter queue length	4	4
Driver buffer size	20	20

5.4. H.323 Address Syntax

An extended H.323 address can be represented as the following string:

```
h323:<AddressType>:<Address1>[:<Address2>]
  where <AddressType> = ip
        <Address1> = <ip address> | <dns name>
<Address type> = gw
        <Address1> = <GW ip address> | <GW dns name>
        <Address2> = <phone number to dial>
<Address type> = gwhunt
        <Address1> = <phone number to dial>
        <Address2> = <File with GW profile>
<AddressType> = gk
  a) <Address1> = ip
        <Address2> = <ip address> | <dns name>
  b) <Address1> = h323id
        <Address2> = <registered user alias with GK>
  c) <Address1> = e164
        <Address2> = <user's phone number registered with GK or
        respondent's phone number in PSTN>
```

All parameters are not case-sensitive. You may use "+", "(", ")", "-", and space in telephone numbers.

H.323 Address Examples

If you want to call

- A host `jd.bigcom.com` (192.168.1.12)
Type in the **Call To:** field: `H323:IP:jd.bigcom.com` or `H323:IP:192.168.1.12`
- A phone number 135-7975 through the H.323 Gateway `h323gateway.com`
Type in the **Call To:** field: `H323:GW:h323gateway.com:1357975`
- A user with the nickname `doe` registered with the H.323 Gatekeeper
Type in the **Call To:** field: `H323:GK:H323ID:doe`
- A phone number 135 registered with the H.323 Gatekeeper
Type in the **Call To:** field: `H323:GK:E164:135`

- An external phone number 9-1357975 through the H.323 Gatekeeper
Type in the **Call To:** field: H323:GK:E164:91357975

5.5. SIP URL Syntax

A SIP URL can be represented as the following string:

```

sip:[<user [:password] | phone-number @>]<DNSName |
  ipaddress>[ ;<parameter1=paramvalue1>[ ;parameter2=paramvalue2]...]
  where <parameter1> = user
        <paramvalue1> = <ip> | <phone>
        <parameter2> = transport
        <paramvalue2> = <tcp> | <udp>

```

All parameters are case-sensitive.

Comprehensive SIP URL syntax is described in RFC3261.

SIP URL Examples

If you want to call

- A host `jd.bigcom.com` (192.168.1.12)
Type in the **Call To:** field: `sip:jd.bigcom.com` or `sip:192.168.1.12`
- A user `johndoe` at `jd.bigcom.com`
Type in the **Call To:** field: `sip:johndoe@jd.bigcom.com`
- A phone number 135-7975 through a SIP gateway `sipgateway.com`
Type in the **Call To:** field: `sip:135-7975@sipgateway.com;user=phone`

5.6. Used Ports

SJphone™ uses the following ports:

Port	Protocol	Description
1719	UDP	H.323 Gatekeeper RAS port
1720	TCP	H.323 Call Signaling
1812	UDP	Radius server in SJphone™
3478	UDP	STUN service
3479	UDP	STUN service
5002	TCP	MLP protocol server
5003	UDP	Neighborhood service
5060	UDP	SIP UAS
5060	TCP	SIP UAS
49152-65535	UDP	RTP, RTCP multimedia streaming

VI. Troubleshooting

6.1. Sound System Issues

Sound quality can vary significantly depending on a sound system and microphone in your device, and Internet connection. It is a good idea to use earphones to avoid sound feedback.

You can adjust the **SJphone™** sound characteristics using standard **PocketPC** audio controls.

You can use the automatic microphone gain feature in your sound system to automatically adjust the input sound level. Consult your device documentation for details.

If you experience a bad sound quality, turn off the Automatic Gain Control feature on the **PocketPC** sound control panel. Go to **Settings**, then to **System**, then to **Audio/Microphone**.

If there is a strong background sound when the **Automatically adjust silence detection level** on the **Audio** tab is activated, **SJphone™** may not detect silence correctly, and fragments of your speech may be lost. In this case, turn this feature off.

If your sound system supports DirectSound, select the **Enable Direct Sound for improved audio performance** box on the **Audio** tab. This will decrease sound latency time.

If you experience bad sound quality, clicks, long sound delay, or dropped sound fragments, you may try to adjust **Advanced Audio Settings**.

Do not change them unless you completely understand what you are doing, or you may severely degrade the **SJphone** performance. You may always return to the default values by tapping the **Use Default**.

Audio channel latency time

If you want to decrease channel latency time, decrease either **Driver buffer size**, **Driver output queue length**, **RTP jitter queue length**, or all of those parameters. Please, note that **Driver input queue length** does not change the latency time.

Recommended **Driver output queue length** is 2 if DirectX is installed and used, and 4 without DirectX. You may check DirectX status on the **Audio** tab.

Output audio channel (a microphone) latency is proportional to the Driver buffer size.

Input audio channel (speakers) latency is proportional to the Driver buffer size*(Driver output queue length + RTP jitter queue length). The Driver input queue length does not affect this latency.

To minimize the latency,

1. On the **Advanced Settings** set the parameters to their default values.
2. Select any direct profile
3. Decrease the **Driver buffer size**, **Output queue length**, and **Input queue length** until sound distortions appear during a call to `localhost`. If sound distortions appear for the default values, increase them until distortions disappear.
4. Increase the **Input queue length** by two times.
5. Select a profile that you plan to use. Make test calls (over a network) and select an optimal value for the **RTP jitter queue length**.

Distorted sound and clicks

If you hear clicks or sound is distorted:

- on the remote side, increase **Driver input queue length**.
- on your side: increase **Driver output queue length**. If this does not help: increase **Driver buffer size**.

Sound drops

If you experience sound drops,

1. Check the number of lost and out-of-order packets on the **Main** panel (without skins). Go to the **Connection Issues** section for details.
2. Try to increase **RTP jitter queue length**.

If the problem persists, contact SJ Labs technical support.

6.2. Connection Issues

If you want to call a regular telephone, you need to use a Gateway. Different phone numbers may require different gateways.

If your host or the host being called is on a private network, VoIP communication is possible only through a Gatekeeper. This is a feature of the H.323 protocol.

Lost packets should not exceed 10-15% for good communication quality. If you loose speech fragments due to a large number of lost packets, try to adjust codecs or turn on the **SJ Labs data recovery engine**. Go to the **Compression and Advanced Audio settings** section for details, or contact your Internet provider or system administrator for assistance.

If the number of out-of-order packets is large, your network is overloaded or unstable. Congestion in packet switched networks can cause packets to take different routes to reach the same destination. Packets may arrive in different order resulting in garbled speech. To prevent this, **SJphone™** stores received packets in a special buffer and reorders them. That is why even large number of out-of-order packets does not always deteriorate sound quality if the packets arrive within a certain time. If a packet comes too late, it gets lost.

If you have any questions or comments, please write to: sjphone@sjlabs.com

6.3. NAT Issues

Symptoms that you have problems with NAT:

- Audio goes only one way (either inbound or outbound).
- Incoming calls cannot reach you.

Read the **NAT traversal and SIP** section for details

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