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## Creating your own service profile for SJphone™

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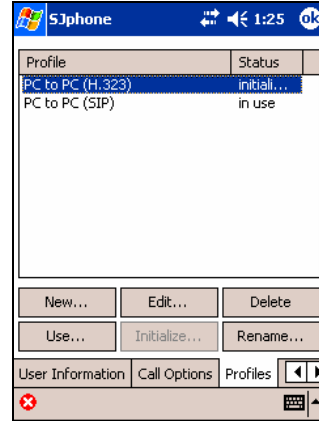
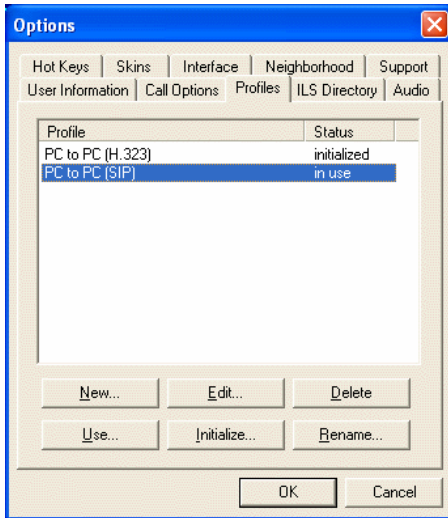
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**SJphone™** allows you to create your own profile for your ITSP (Internet Telephony Service Provider) or your own SIP proxy, H.323 gatekeeper or gateway. Please keep in mind that creating a profile requires some knowledge of at least basic details 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.

You may create your profile on the **Profiles** tab.

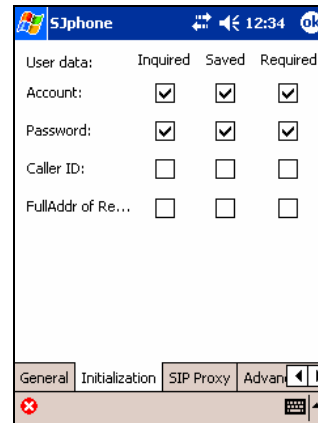
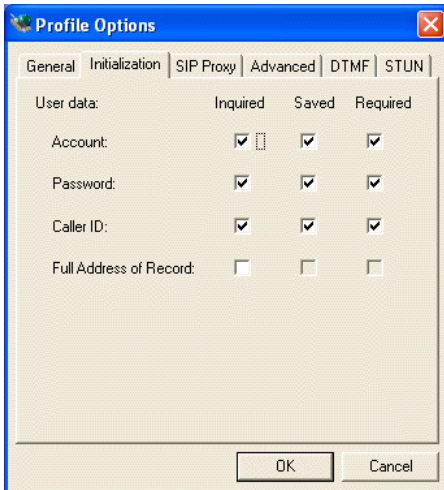
**To go to the Profiles tab,**

- Click the **Options** button on the **SJphone™** **Main** panel and go to the **Profiles** tab
- Tap **Menu**, select **Options**, and go to the **Profiles** tab



## 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

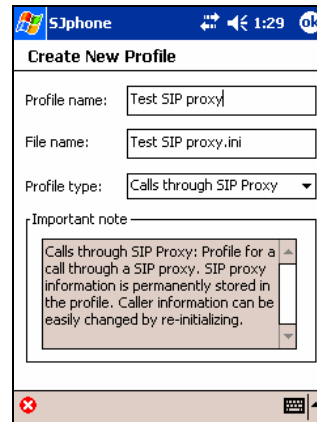
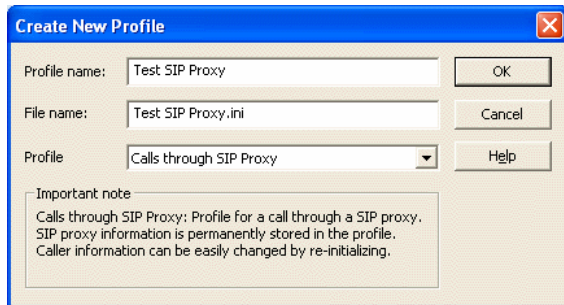
## SIP Profiles

### Calls through a SIP Proxy

**Note:** If your SJphone™ is behind NAT, please read the **NAT traversal and SIP** section. 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,**

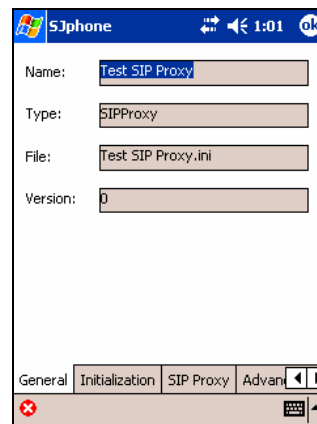
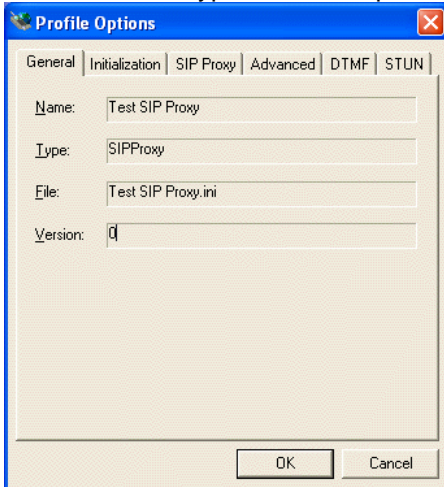
- Click/tap the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
  - Click/tap the **New** button
- A **Create New Profile** window will appear



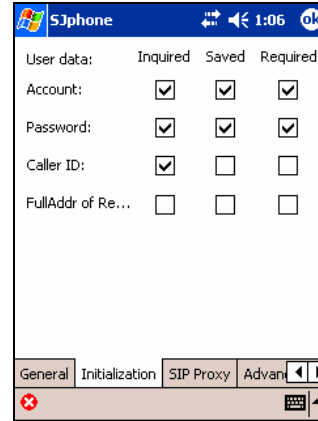
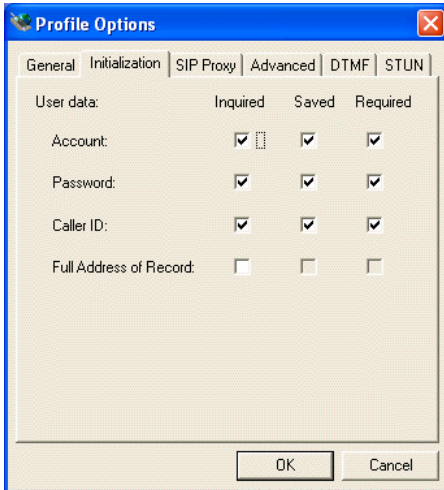
- Select **Calls through SIP Proxy** in **Profile**, specify a name for the profile, and click/tap the **OK** button.
- Specify the required profile options and click/tap the **OK** button.

### General

It shows the name, type, and other profile's properties



Initialization

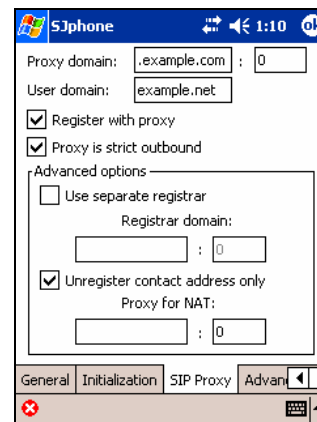
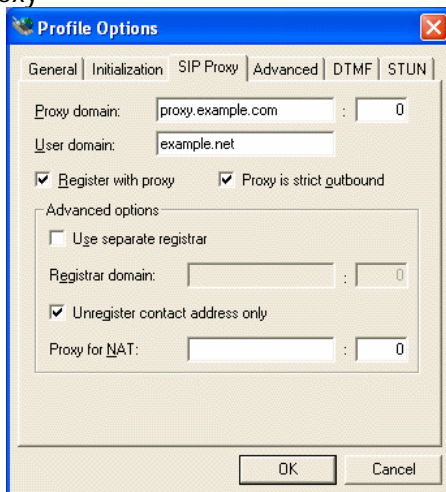


- Accounts Your account (login) at the proxy
  - Password Your password at the proxy
  - Caller ID Your Caller ID at the proxy. Used to form your SIP **Address of Record**
  - Full Address of Record Your Full Address of Record. Used to form your SIP **Address of Record**
- Either Caller ID or Full Address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

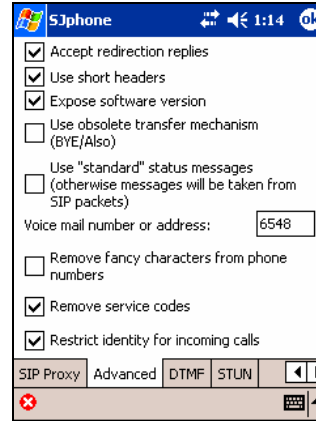
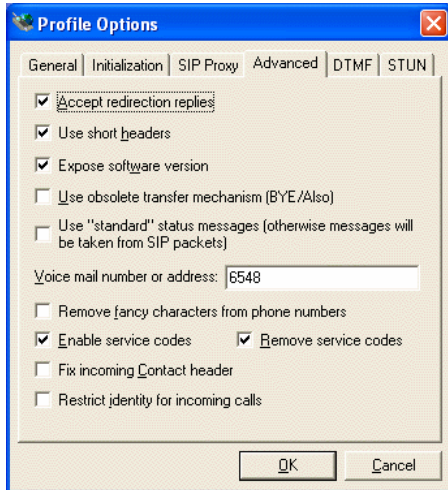
SIP Proxy



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 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.
Register with proxy	If this option is selected, <b>SJphone™</b> will register with the proxy. Always select this option unless instructed otherwise. Recommended status: <b>Selected</b> .
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: <b>Selected</b> .
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, <b>SJphone™</b> will register with a separate SIP entity rather than with the proxy specified in the Proxy domain option. Recommended status: <b>Cleared</b> .
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, <b>SJphone™</b> will unregister only its temporary SIP URL associated with the current running <b>SJphone™</b> instance. If this option is cleared, <b>SJphone™</b> will unregister all temporary SIP URLs associated with the permanent user address-of-record. Recommended status: <b>Selected</b> .
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 <b>SJphone™</b> 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.

Advanced



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**.

Use "standard" status messages

If this option is selected, **SJphone™** will display SIP status messages from its internal SIP message list. If the option is clear, SIP messages will be taken from incoming SIP packets. Select this option if you have a completely localized **SJphone™** version. Recommended status: **Cleared**.

Voice mail number or address

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

Remove fancy characters from phone numbers

If this option is selected, **SJphone™** will remove +, (, ), -, and space from dialed phone numbers. Recommended status: **Cleared**.

Enable service codes

If this option is selected, **SJphone™** will follow issued vertical service codes.

Remove service codes

If this option is selected, **SJphone™** will remove vertical service codes before sending an entered number to the server.

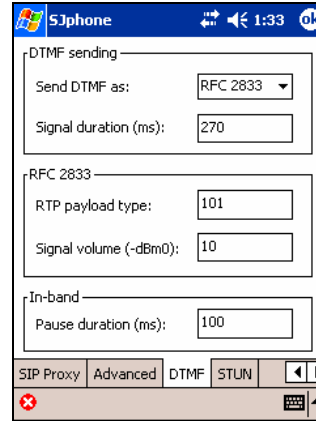
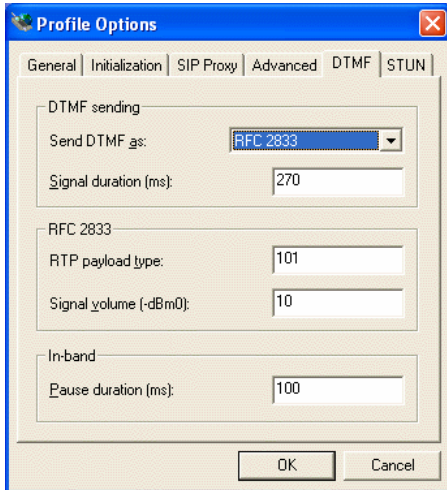
Fix incoming Contact header

If this option is selected, **SJphone™** will change the SIP Contact header trying to compensate for packet address changes made by various NAT systems. Select it if you have problems in making calls between open Internet and private networks. Recommended status: **Cleared**.

Restrict identity for incoming calls

If this option is selected, **SJphone™** will conceal your name, username, and other sensitive information for incoming calls as much as possible. Please note that the policy of accepting/delivering anonymous calls varies for service providers.

DTMF



DTMF sending

SJphone™ supports 3 methods:

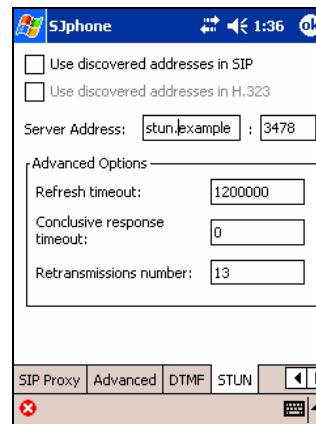
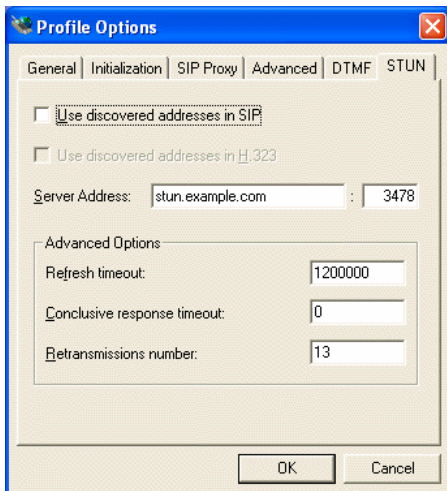
1. **In-band:** DTMF signals are sent as digitized sound in the RTP channel.
2. **RFC2833:** DTMF signals are sent as special packets controlled in the RTP channel.
3. **SIP INFO:** DTMF signals are sent in the **INFO** message in the SIP session control channel.

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:**

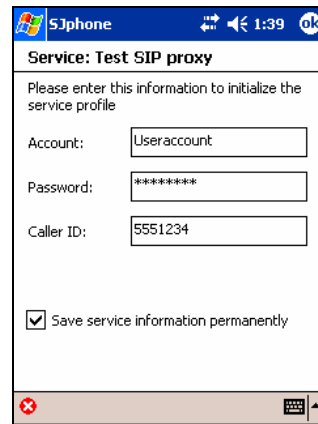
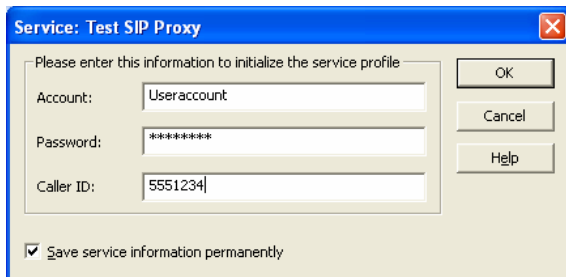
**To initialize or re-initialize this profile,**

- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile
  - Right-click the **SJphone™** system tray icon or skin once again and select **Re-initialize** profile
- or
- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
  - Select the required profile and click the **Initialize** button

- 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

The **Service** message will appear



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

- 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

Select the 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.

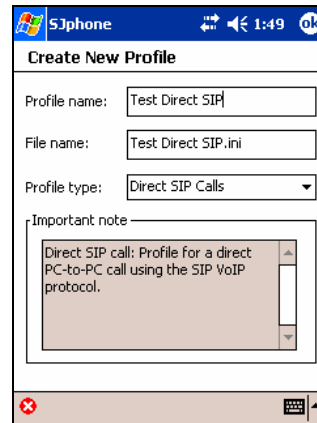
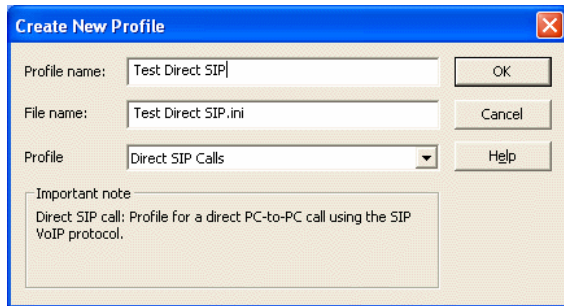
**Direct SIP PC-to-PC call**

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,**

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

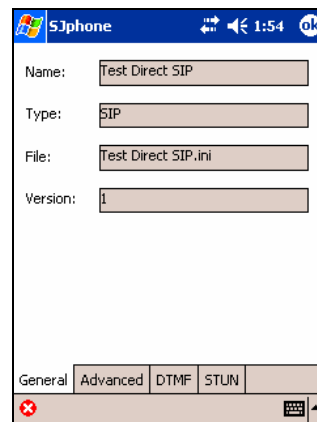
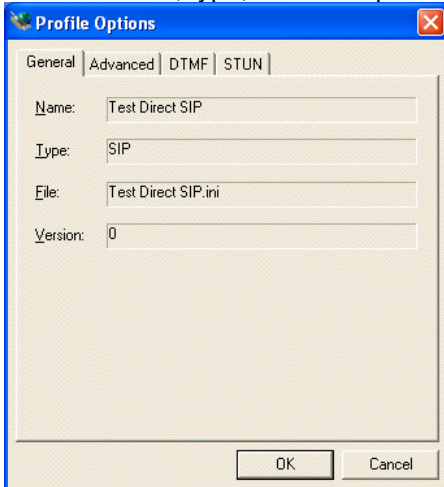
A **Create New Profile** window will appear



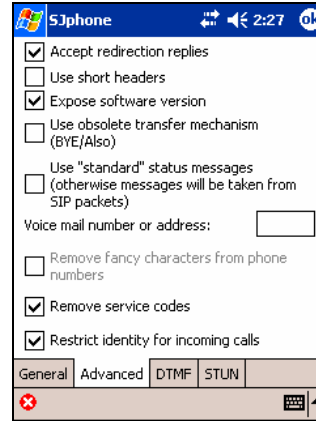
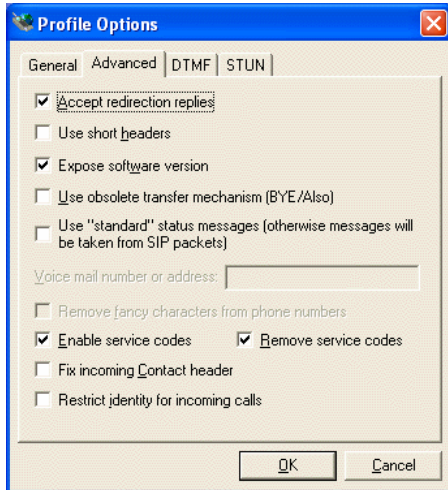
- Select **Direct SIP Calls** in **Profile**, specify a name for the profile, and click/tap the **OK** button.
- Specify the required profile options and click/tap the **OK** button.

General

It shows the name, type, and other profile's properties



Advanced



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)

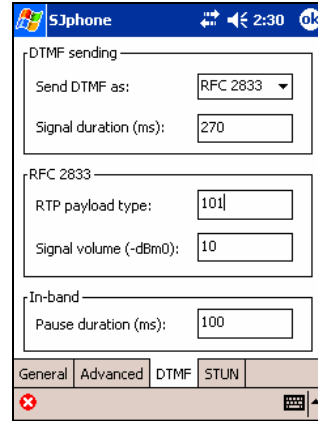
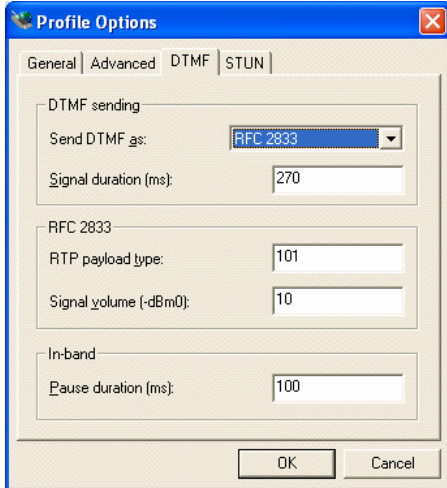
If this option is selected, **SJphone™** will use the BYE/Also mechanism for blind transfers. In fact, this mechanism is deprecated. Recommended status: **Cleared**.

Use "standard" status messages

If this option is selected, **SJphone™** will display SIP status messages from its internal SIP message list. If the option is clear, SIP messages will be taken from incoming SIP packets. Select this option if you have a completely localized **SJphone™** version. Recommended status: **Cleared**.

Enable service codes	If this option is selected, <b>SJphone™</b> will follow issued vertical service codes.
Remove service codes	If this option is selected, <b>SJphone™</b> will remove vertical service codes before sending an entered number to the server.
Fix incoming Contact header	If this option is selected, <b>SJphone™</b> will change the SIP Contact header trying to compensate for packet address changes made by various NAT systems. Select it if you have problems in making calls between open Internet and private networks. Recommended status: <b>Cleared</b> .
Restrict identity for incoming calls	If this option is selected, <b>SJphone™</b> will conceal your name, username, and other sensitive information for incoming calls as much as possible. Please note that the policy of accepting/delivering anonymous calls varies for service providers.

DTMF



**DTMF Sending:**

**SJphone** supports 3 methods:

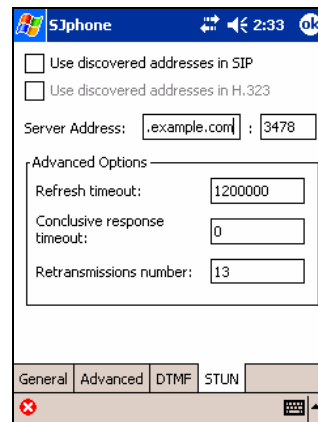
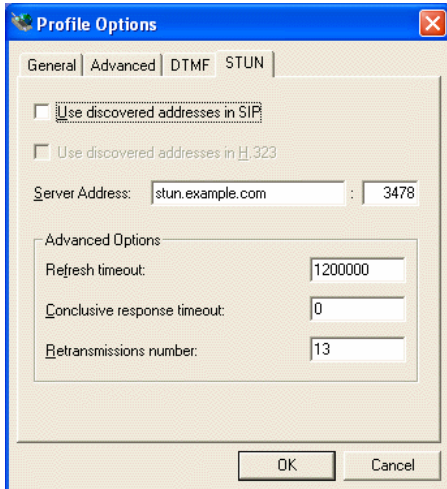
1. **In-band:** DTMF signals are sent as digitized sound in the RTP channel.
2. **RFC2833:** DTMF signals are sent as special packets controlled in the RTP channel.
3. **SIP INFO:** DTMF signals are sent in the **INFO** message in the SIP session control channel.

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. 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 you have reasons for 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.

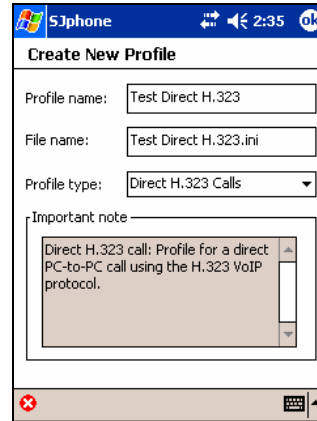
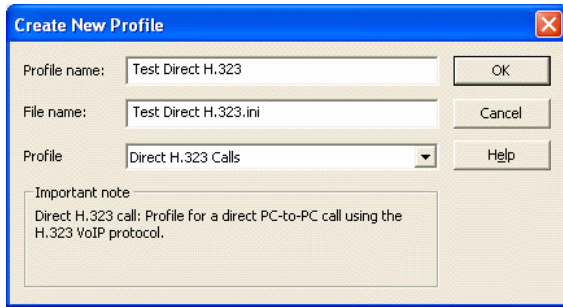
## H.323 Profiles

### Direct H.323 PC-to-PC calls

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

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

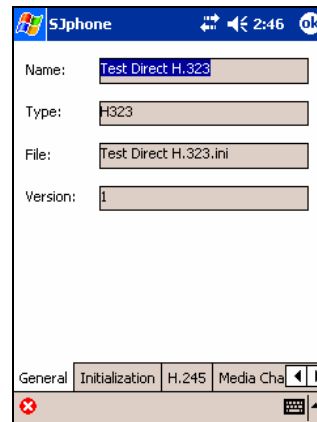
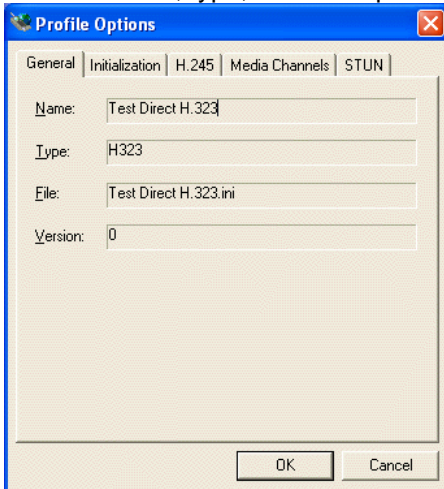
A **Create New Profile** window will appear



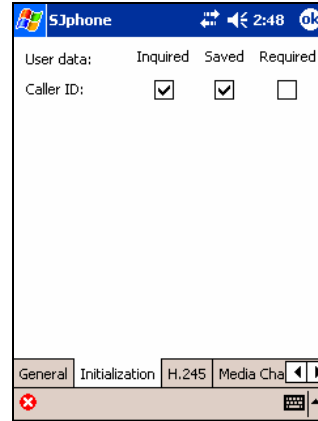
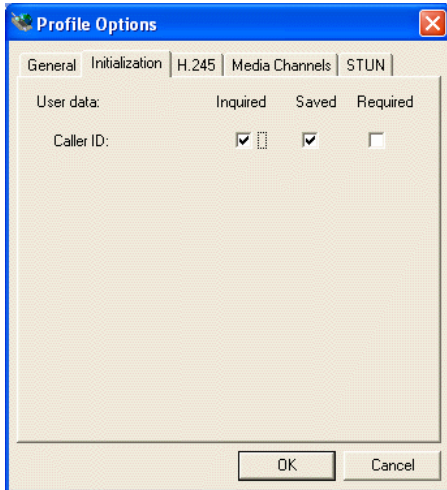
- Select **Direct H.323 Calls** in **Profile**, specify a name for the profile, and click/tap the **OK** button.
- Specify the required profile options and click/tap the **OK** button.

### General

It shows the name, type, and other profile's properties



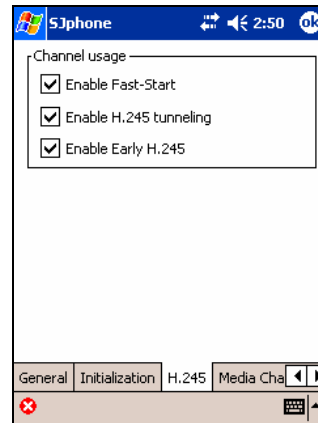
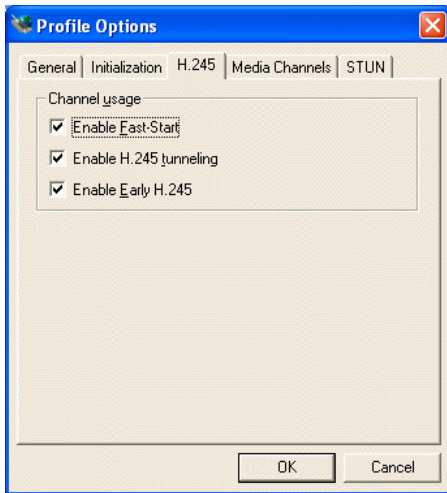
Initialization



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

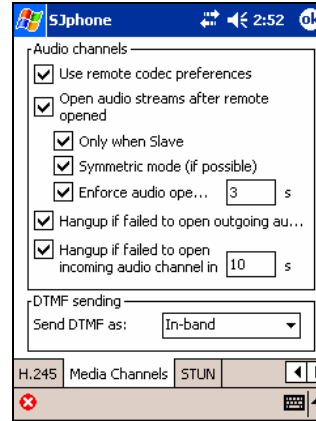
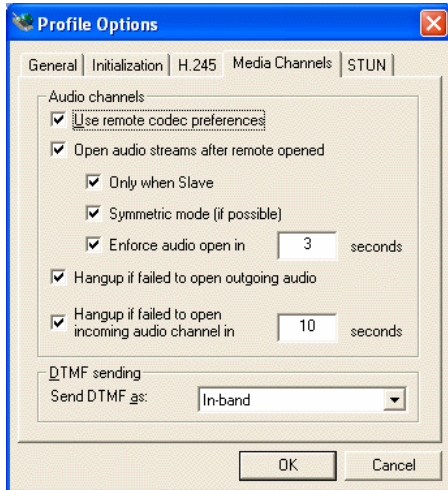
H.245



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.

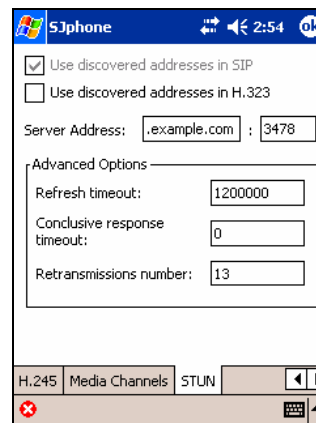
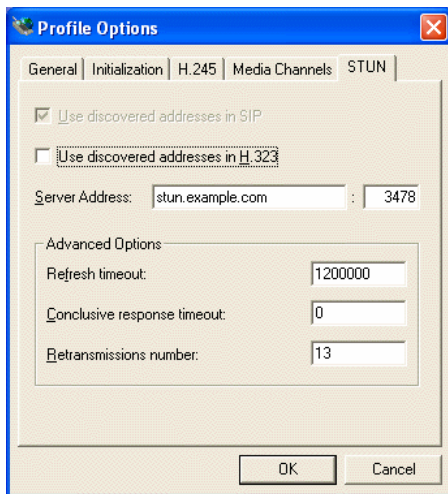
DTMF sending

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 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. 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:**

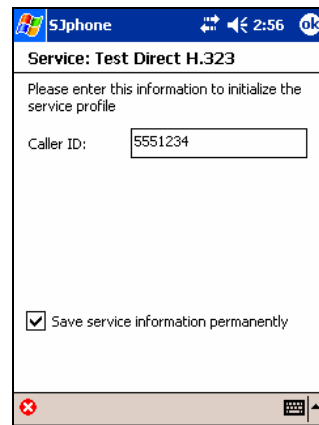
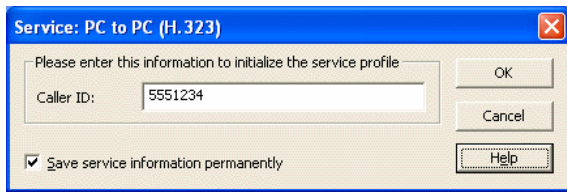
**To initialize or re-initialize this profile,**

- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile
  - Right-click the **SJphone™** system tray icon or skin once again and select **Re-initialize** profile
- or
- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
  - Select the required profile and click the **Initialize** button

- 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

The **Service** message will appear



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

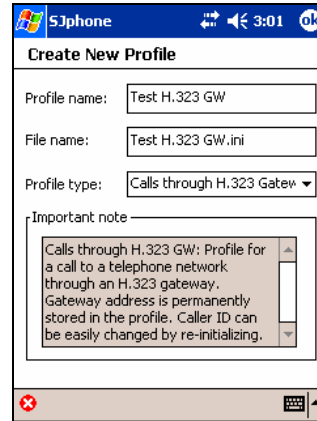
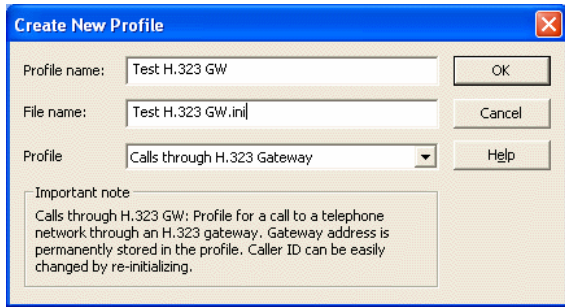
Select Save service information permanently if you do not want to enter this information every time you select this profile.

***Calls through an H.323 Gateway***

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

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

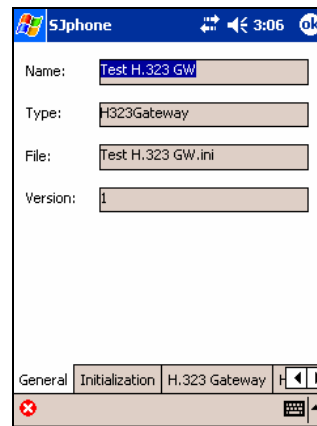
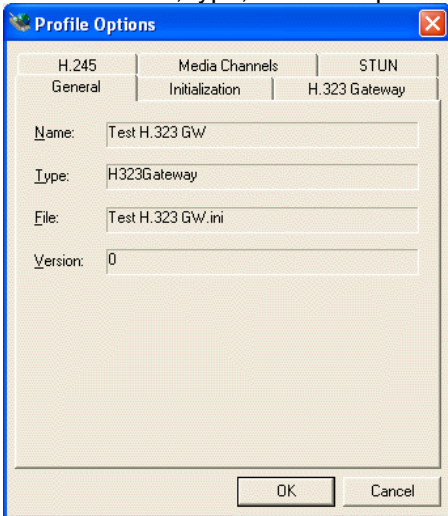
A **Create New Profile** window will appear



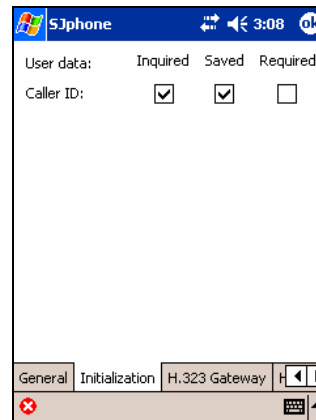
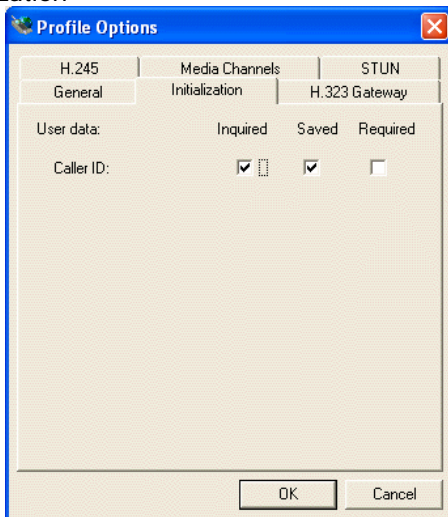
- Select **Calls through H.323 Gateway** in **Profile**, specify a name for the profile, and click/tap the **OK** button.
- Specify the required profile options and click/tap the **OK** button.

General

It shows the name, type, and other profile's properties



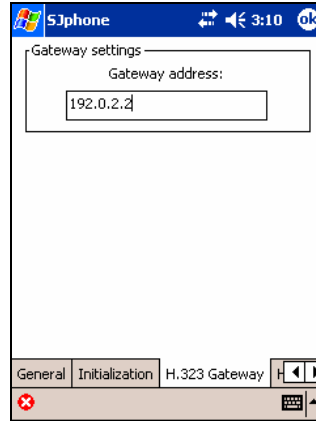
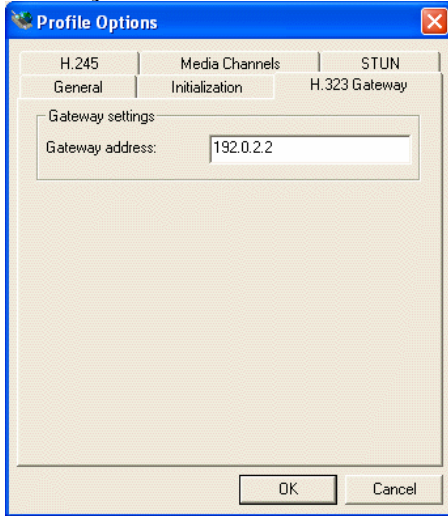
Initialization



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

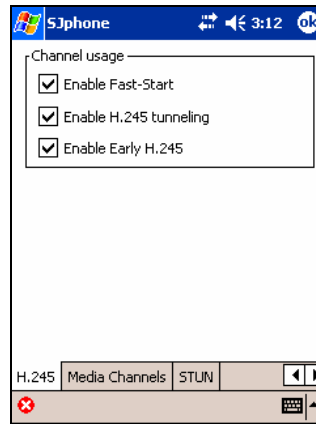
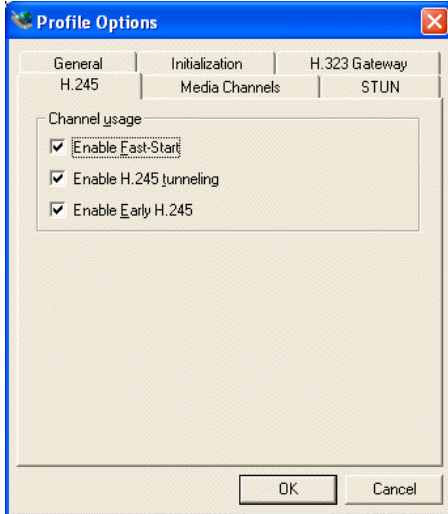
H.323 Gateway



Gateway settings  
Gateway address

The Gateway IP address

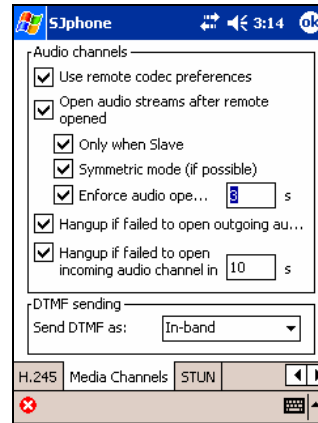
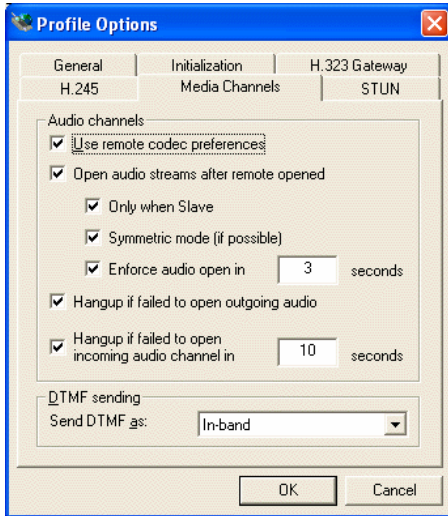
H.245



Channel usage

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

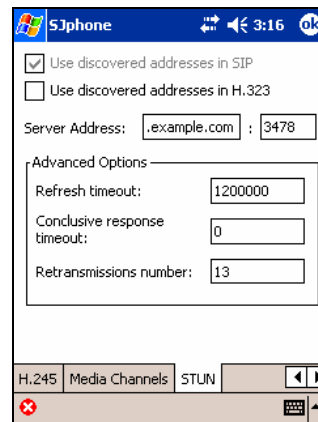
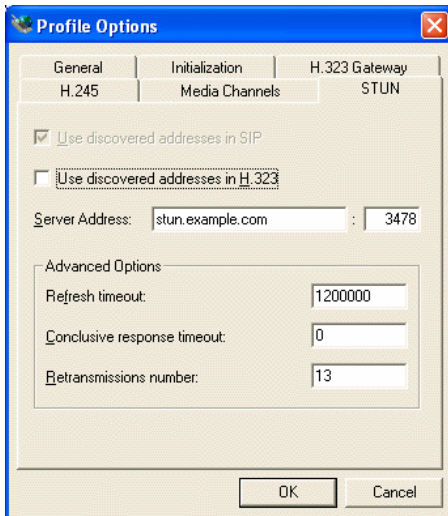
DTMF sending

**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, <b>SJphone™</b> will use addresses discovered by the STUN procedure in H.323 packets. Recommended status: <b>Cleared</b> .
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:**

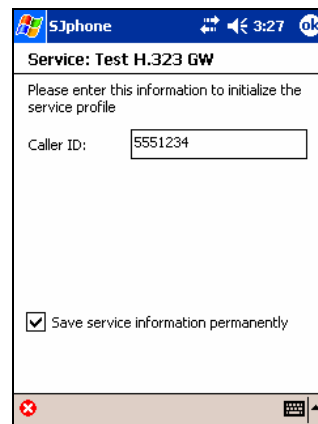
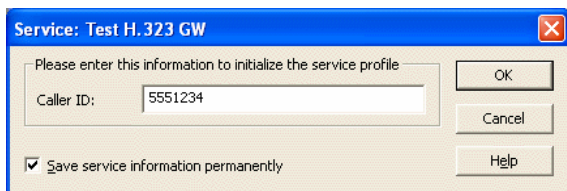
**To initialize or re-initialize this profile,**

- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile
  - Right-click the **SJphone™** system tray icon or skin once again and select **Re-initialize** profile
- or
- Click the **Options** button on the **SJphone™** **Main** panel and go to the **Profiles** tab
  - Select the required profile and click the **Initialize** button

- 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

The **Service** message will appear



- Enter the required information and click the **OK** button
- 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.

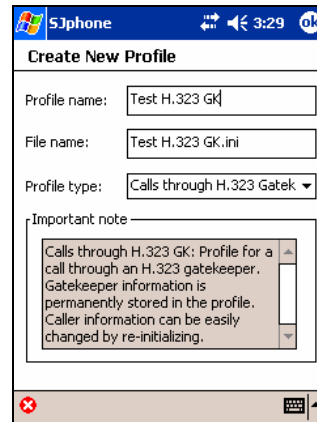
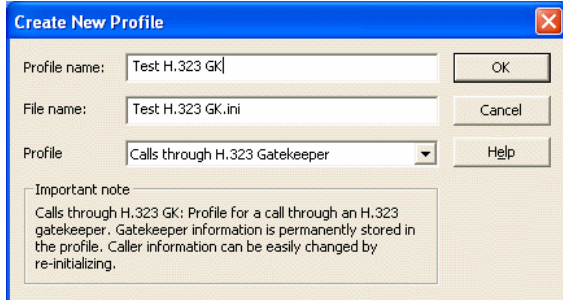
***Calls through an H.323 Gatekeeper***

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,**

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

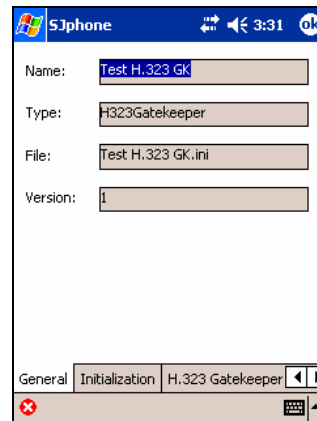
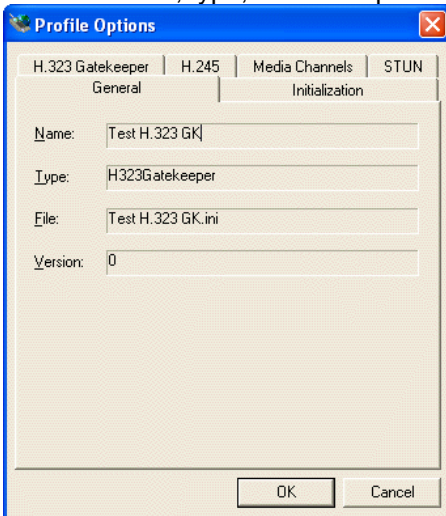
A **Create New Profile** window will appear



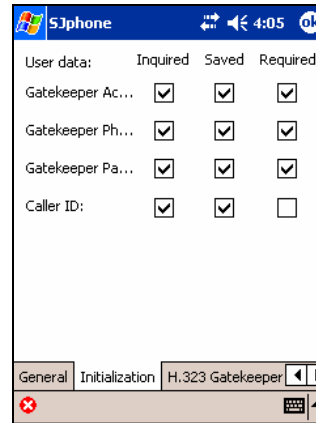
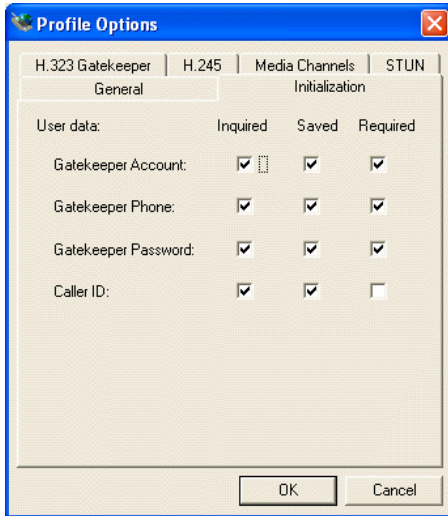
- Select **Calls through H.323 Gatekeeper** in **Profile**, specify a name for the profile, and click/tap the **OK** button.
- Specify the required profile options and click/tap the **OK** button.

**General**

It shows the name, type, and other profile's properties



Initialization



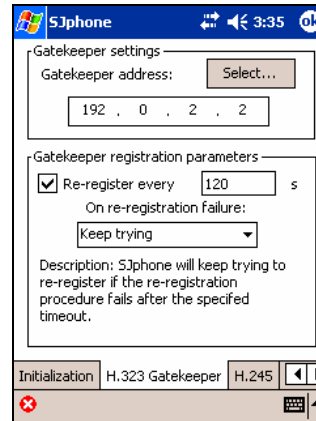
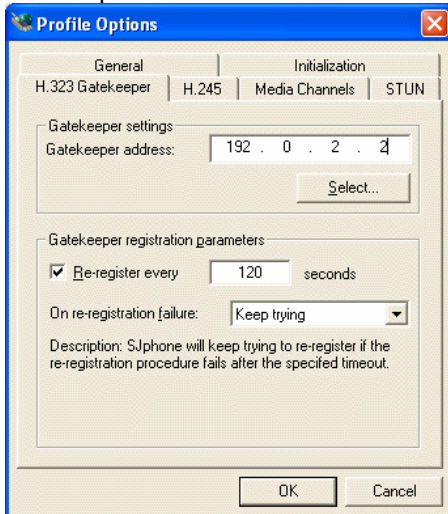
Gatekeeper Account  
Gatekeeper Phone

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.

Gatekeeper Password  
Caller ID

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  
Gatekeeper registration parameters  
Reregister every...seconds

The Gatekeeper IP address.

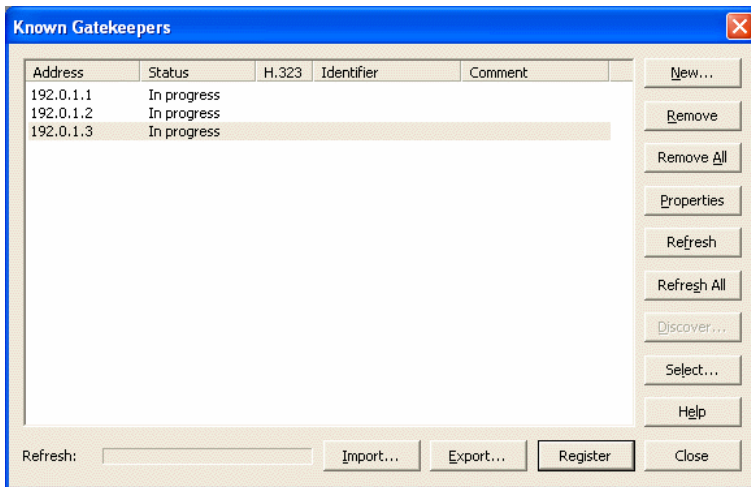
On re-registration failure

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.  
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  
Click/tap the **Select.../Select Gatekeeper** button to activate the **Known Gatekeepers** panel.

- Address 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.
- Windows XP only!**
- Refresh Shows refresh progress



### Buttons

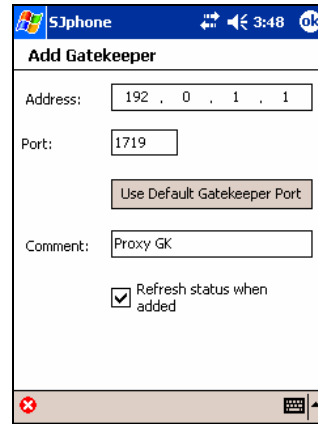
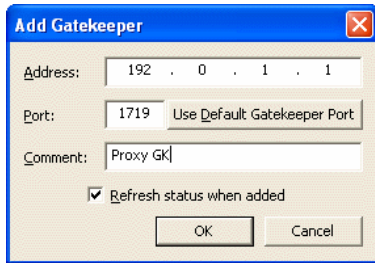
- New Click/tap this button to add a new gatekeeper to the list
- Remove Click/tap this button to remove the selected gatekeeper
- Properties Click/tap this button to see the properties of the selected gatekeeper
- Select Click/tap this button to select all gatekeepers with the same status
- Close Click/tap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

### Windows XP only

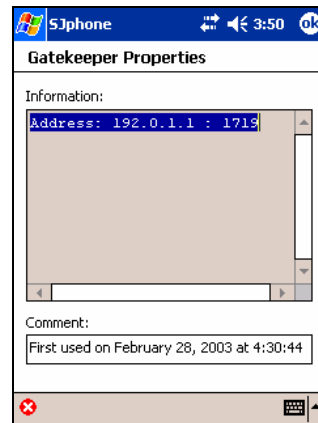
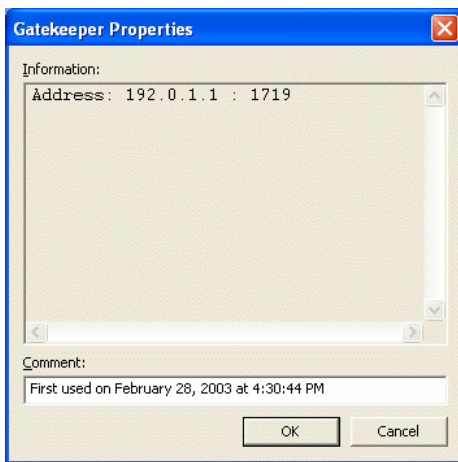
- Remove All Click this button to remove all gatekeepers from the list
- Refresh Click this button to refresh the status of the selected gatekeeper
- Refresh All Click this button to refresh the status of all gatekeepers in the list
- Register Click this button to register with the selected gatekeeper
- Discover Not implemented yet

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

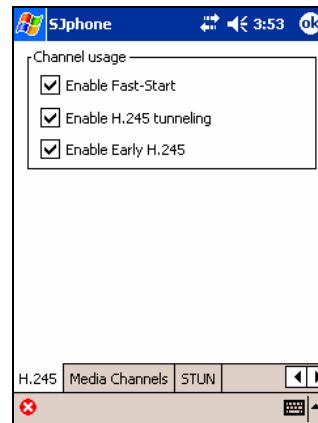
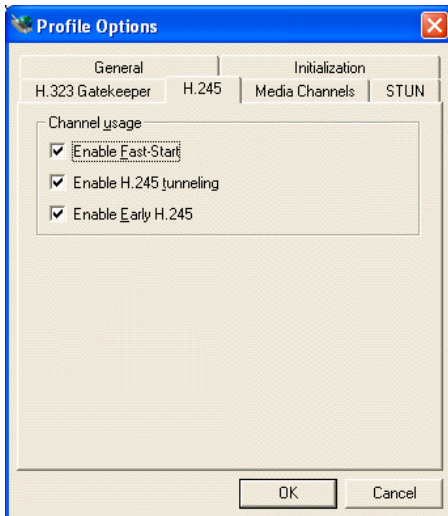




You may see the properties of each gatekeeper. Click/tap the **Properties** button and the **Gatekeeper Properties** window will appear.



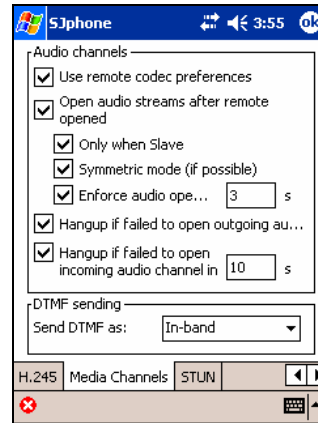
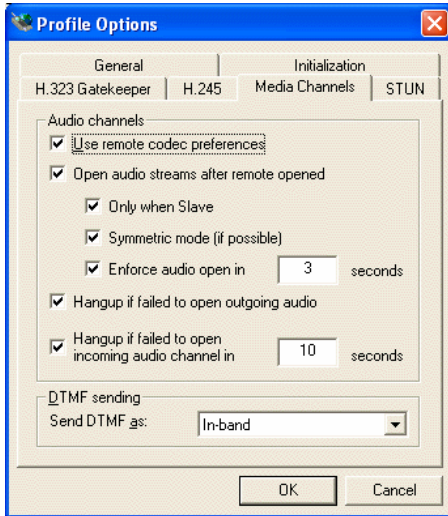
H.245



Channel usage

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

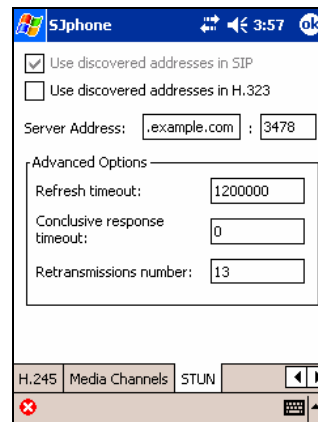
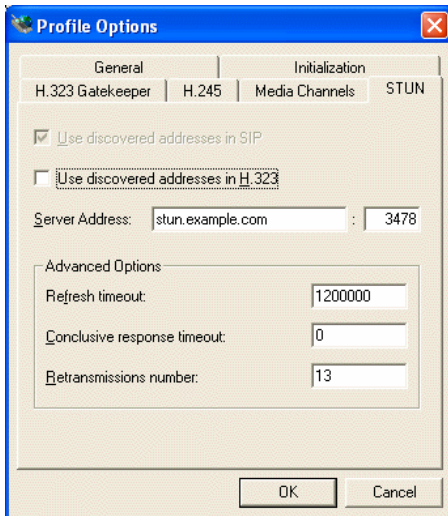
DTMF sending

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, <b>SJphone™</b> will use addresses discovered by the STUN procedure in H.323 packets. Recommended status: <b>Cleared</b> .
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:**

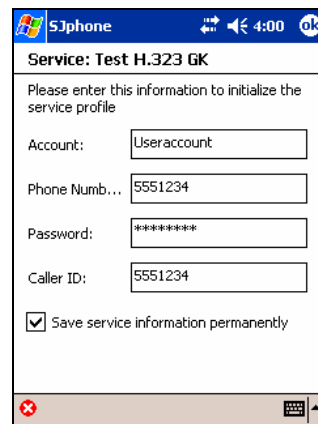
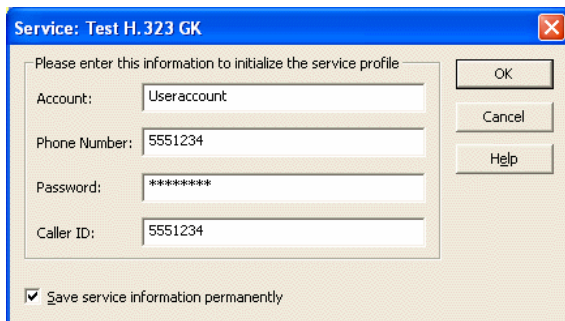
**To initialize or re-initialize this profile,**

- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile
  - Right-click the **SJphone™** system tray icon or skin once again and select **Re-initialize** profile
- or
- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
  - Select the required profile and click the **Initialize** button

- 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

The **Service** message will appear



- Enter the required information and click the **OK** button
- 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.