SJphone™

Windows XP version

User's Guide

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I.   Welcome to SJphone™

SJphone™ is a softphone that allows you to speak over Internet with any desktops, notebooks, PDAs, stand-alone IP-phones and even with any conventional landline or mobile phones. It supports both SIP and H.323 industry standards, and is fully inter-operable with most major Internet Telephony Service Providers (ITSP) and VoIP software and hardware.

For calling over the world you need to open an account with an IP-Telephony service and to get their service profile, or to make their **SJphone™** profile yourself.

Advanced users may build their own Voice over IP networks using either an H.323 Gatekeeper, SIP proxy, IP-PBX, ILS Directory server, and other components. They can also connect to conventional telephones using H.323/SIP **Gateways** (such as Cisco 26xx/36xx/37xx/53xx).

1.1. SJphone™ Features

SIP Compatibility allows you to speak over Internet
- directly with most other SIP-softphones and stand-alone SIP-phones (such as Cisco 7940)
- with other people in your company through your SIP IP-PBX system or Proxies (such as Asterisk, Broadsoft and others)
- with all other people around the world using most SIP ITSP accounts such as FWD (www.fwd.pulver.com)
- a regular telephone using **most SIP Gateways**

H.323 Compatibility allows you to speak over Internet
- directly with any other H.323-softphones, and stand-alone H.323-phones.
- with other people in your company through your H.323 IP-PBX system, such as Cisco Call Manager
- with all other people around the world through any H.323 ITSP accounts
- with a computer or telephone on a private network through a **Gatekeeper**
- a regular telephone using most **H.323 Gateways**

Features that make SJphone™ Easy to Use:
- Skins allow you to change the appearance of the **SJphone™ Main** panel
- Call any host by its name or IP address
- Proven Interoperability with most IT service providers (you can download profiles for some of them)
- Fast Service switching between ITSP, IP-PBX, Gatekeeper, Gateway, or SIP Proxy settings and GUI
- Contacts, call logs, and speed dialing
- Call a computer or telephone on a private network through a **Gatekeeper**
- Call transfer, hold, and mute
- Can automatically accept incoming calls, or reject them immediately (the "**Do Not Disturb**" mode) or after a specified time
- Automatic discovery of neighbor **SJphones™**
- Respondent list allows you to maintain and manage multiple calls
- Dial pad to type in numbers and send DTMF signals, sent using the either as audio, INFO message, or out-of-band methods.
- Input and output sound level indication and control
- Automatically adjustable microphone volume
- Automatically adjustable silence detection level
- Support for most USB phones (several drivers are downloadable)
- Support for most popular codecs: G711, GSM, iLBC, (G729 available only in commercial versions)
- Automatic NAT detection
- Calls from/to a private IP-network using the NAT traversal technology
- Message waiting indication

Features for Advanced Users:
- Customized service profiles allow you to create your own profiles for calls through your H.323 Gatekeeper, Gateway, or SIP Proxy
- Multisession calls
- 3-way conferencing using different VoIP protocols
- Manual codec selection
- DNS support in the SIP stack
- ILS (Internet user Location Service) directory
- Support for extended H.323 address syntax
- Support for advanced SIP URL syntax
- Support for ILS contact syntax
- Selectable SIP transport protocol (TCP/UDP)
- Lost and out-of-order packet indication
- Log window for operation monitoring
- Advanced lost packet recovery, offering better sound quality over a poor connection.
- CallTo URL support
- Installable Advanced Help
- Remote support console
- Open API for developing third-party VoIP applications

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

- Video
- Open API allows you to develop your own VoIP application
- Full uPnP support
- Instant Messenger and Presence Support
- Hi-Quality Voice

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

A PC running **Windows 98SE/ME/2000sp4/XPSP2:**

Attention Windows 2000 users! You need to install **Microsoft Installer** 2.0 before installing **SJphone™ from www.microsoft.com**.

Hardware requirement:

At least Intel Celeron 300 CPU with 32MB for Windows 98SE, 128MB for Windows 2000, 256 for Windows XP.

A full-duplex sound card with a microphone and speakers, or any USB phones. Installed latest **DirectX** (7.0 at least, current version is 9.0) is strongly recommended; you can check it by `dxdiag.exe`. Please use latest drivers for soundcards and USB phones.

Any kind of broadband TCP/IP connection like wired or wireless (802.11x, 802.16x) Ethernet or wide area connections (WAN) like DSL, E1/T1, etc are recommended. You will get a better sound quality with a wideband connection but a v.90 analog modem is enough if you are using voice compression with an appropriate codec.

If you need to place a call to/from PSTN, you must have an ITSP account or access to an H.323 or SIP Gateway like CISCO 26xx/36xx/3700/53xx, etc. Refer to your gateway manual to check whether your gateway is configured correctly.

1.4. Installing SJphone™

Attention Win2000/XP users! You must have administrative privileges to install SJphone™. If you are not sure whether you have such privileges, you almost certainly do not have them. Contact your system administrator for assistance.

To install SJphone™

- Download and run the installation file.
- Follow the on-screen instructions. .

To uninstall SJphone™, go to **Control Panel**->**Add/Remove Programs**->**SJphone**->**Install/Uninstall**.

To start SJphone™, go to **Start**->**SJ Labs** program group->**SJphone**->**SJphone**.

1.5. Registering SJphone™

You may register your **SJphone™**. The registration is free and removes the advertising pop-up window. You will also be able to receive information about new **SJphone™** releases.

To register your SJphone™ on-line,

- Click the **Options** button on the **SJphone™ Main** panel. The **Options** panel will appear.
- Select the **Support** tab, then click the **Register** button.

An **SJphone registration** message will appear.

- Enter your activation key and click 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.
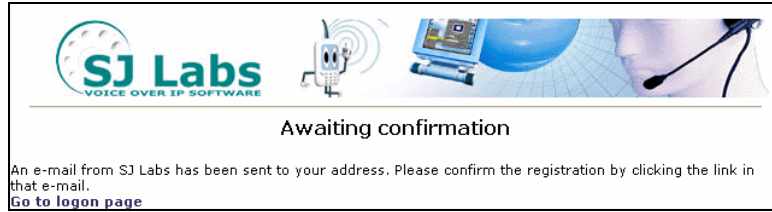
If you do not have the activation key yet,
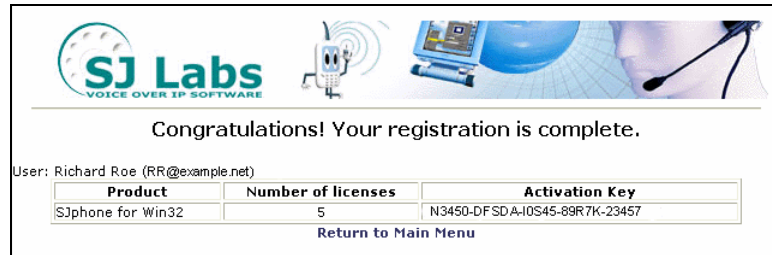
- Click the **Get Key** button.

The **Product Registration** page will appear in your default web browser.

- Enter the necessary information and click the **Register** button.

A confirmation e-mail will be sent to your E-mail and the **Awaiting Confirmation** page will appear in your default web browser.

- When the mail arrives, click a link in the mail.

The **Congratulations! Your registration is complete.** page will appear in your default web browser.

- Copy and paste the activation key to the **SJphone registration** message and click the **OK** button.

If you want to change your personal information

- Click the **Options** button on the **SJphone™ Main** panel. The **Options** panel will appear.
- Select the **Support** tab, then click the **Register** button.

An **SJphone registration** message will appear.

- Click the **Get Key** button.

The **Product Registration** page will appear in your default web browser.
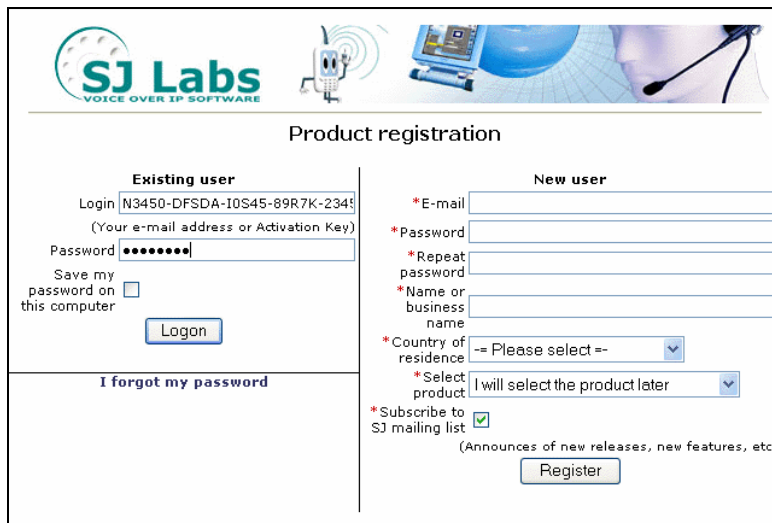
- Enter your Login and Password and click the **Logon** button. The Login is either an activation key of an **SJ Labs** registered product or your e-mail address used for the registration. If you come to this page from an already registered product, its activation key will be entered into Login automatically.

The **Main Menu** page will appear in your default web browser.

- Click **Change personal information**.
The **Change personal information** page will appear in your default web browser.

- Change required information and click the **Submit changes** button.

If you forgot your password,

- Click the **Options** button on the **SJphone™ Main** panel. The **Options** panel will appear.
- Select the **Support** tab, then click the **Register** button.
An **SJphone registration** message will appear.

- Click the **Get Key** button.
The **Product Registration** page will appear in your default web browser.

- Click **I forgot my password**.
An e-mail with the password will be sent to the e-mail address used for registration and the **Your password has been sent to your e-mail address** page will appear in your default web browser.
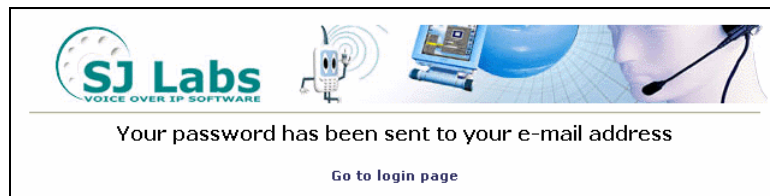
1.6. Contact Information and Technical Support

Email: sjphone@sjlabs.com

http://www.sjlabs.com

or

Click the **Option** button on the **Main** panel, select the **Support** tab, and click the **Web** button.

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

- Right-click the **SJphone** system tray icon or skin, select **Bug Reporting**, and then **Start Recording**.
- Reproduce the problem

- Right-click the **SJphone** system tray icon or skin 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.
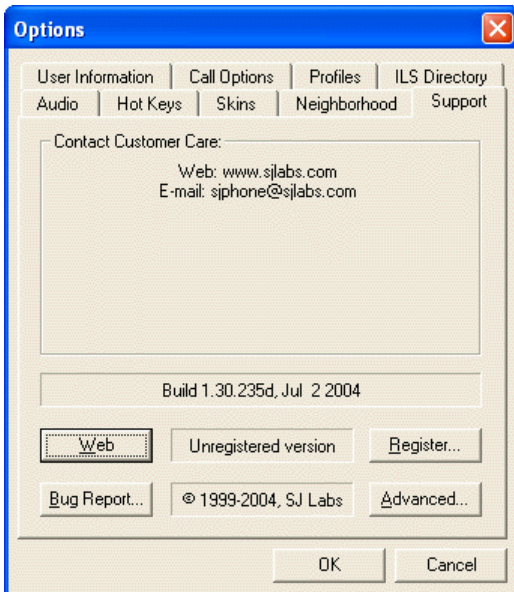
<table>
<tr><td></td><td>Web</td><td>Click this button to visit **SJ Labs** home page</td></tr>
<tr><td></td><td>Bug Report</td><td>Click this button to generate an e-mail bug report.</td></tr>
<tr><td></td><td>Register</td><td>Click this button to register your copy of **SJphone™**</td></tr>
<tr><td></td><td>Advanced</td><td>Click this button to activate **Remote Console**</td></tr>
</table>
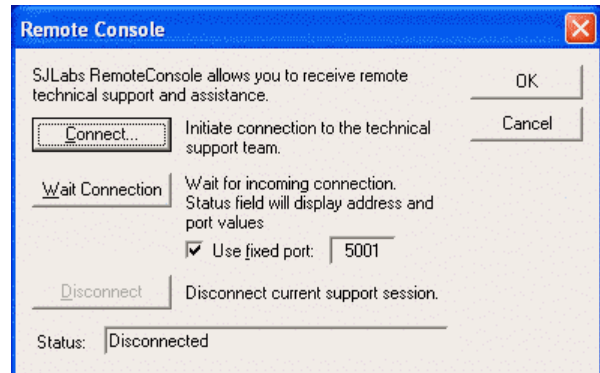
II.   SJphone™ Panels and User Features
2.1. Controls on the Main Panel

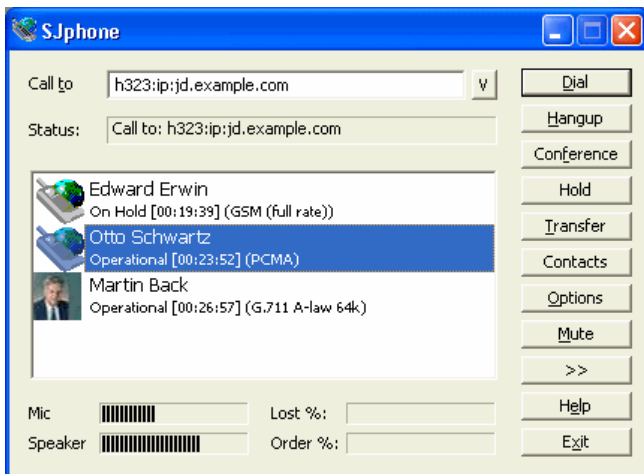**Attention!** Some **SJphone™** skins do not have the **Options** button. In this case, you can go to **Options** by right-clicking the **SJphone™** system tray icon or skin and selecting **Options**.

The standard **Main** panel has two screens. Some skins may have more.

Call to:	Type here the IP address, host name, nickname, or telephone number you want to call
Status:	Displays the call status
Mic	Indicates input sound level for the current session
Speaker	Indicates output sound level for the current session
Lost %:	Indicates how many packets are lost for the current session
Order %:	Indicates how many packets are out of order for the current session

Buttons

| Dial	Click this button to dial the address typed in the **Call to**: field *Shortcut key:* **Alt + d** or **Enter**
Hangup	Click this button to end the call *Shortcut key:* **Alt + h** or **Enter** when in a call
Conference	Click this button to join selected respondents to **a conference talk**
Hold	Click this button to put the call on hold
Transfer	Click this button to transfer the call to the address specified in the **Call to**: field
Contacts	Click this button to activate the **SJphone™ Contacts**
Voice Mail	Click this button to receive your **voice mail**.
Options	Click this button to activate the **SJphone™ Options** panel. Using this panel, you can configure **SJphone™**. *Shortcut key:* **Alt + o**
Mute Speak	Click the **Mute** button to mute your voice. The **Speak** button will appear. Click the **Speak** button to resume the conversation. The **Mute** button will appear again. *Shortcut key:* **Alt + m** and **Alt + s**
>> <<	Click this button to go to the next screen of **SJphone™ Main** panel
Exit	Click this button to close **SJphone™** into the system tray *Shortcut key:* **Alt + x**
Help	Click this button to start **SJphone™**'s **Help** *Shortcut key:* **Alt + e**

Sliders

Mic	Use this slider to adjust microphone volume
Speaker	Use this slider to adjust output sound volume
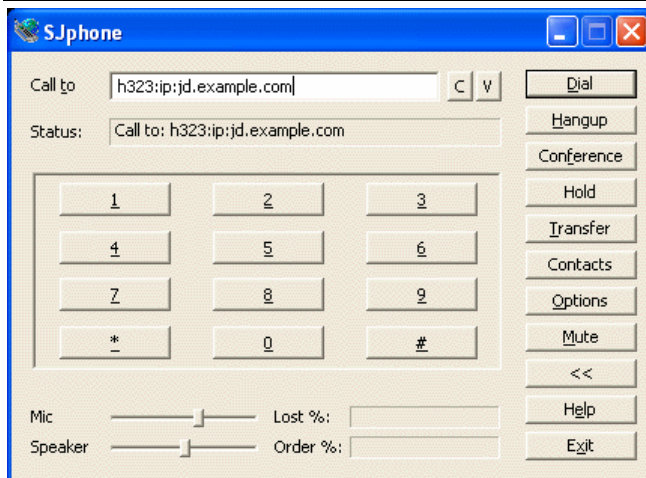
If you want to use **Speed Dial**, or send DTMF signals, you need to activate the dial pad screen by pressing the **>>** button. Use the ***** button to enter the dot. Use the **C** button to enter backspace.

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

You may look at the respondent properties. Right-click the respondent and select the **Properties** item in the pop-up menu. The **Session Properties** message will appear.
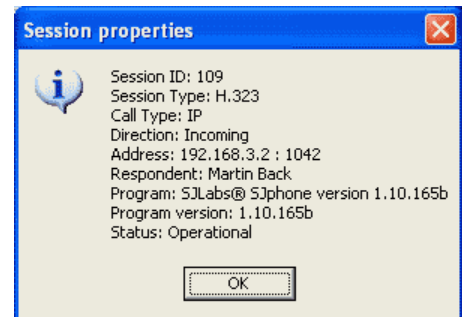
Lost and Out-of-Order Packets:

For quality communication, the number of lost packets should not exceed 10-15%. If the number of lost packets is too high, go to the **Connection issues** section for troubleshooting.

A large number of out-of order packets means that 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 the wrong order resulting in garbled speech. To prevent this, **SJphone™** stores received packets in a special buffer and reorders them. That is why even a large number of out-of-order packets does not always deteriorate the sound quality if the packets arrive within a certain time. If a packet comes too late, it gets lost.

Sound Adjustments:

Generally, you can adjust the **SJphone™** sound characteristics using standard **Windows** audio controls or controls on the **Main** panel.

The **Mic** and **Speaker** controls on the **Main** panel are connected with the standard **Windows** audio controls. Any changes in their settings automatically change the corresponding settings on the **Windows** volume control panel.

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. They also have **ILS Directory** and **Neighborhood** browsers, and **logs** for missed, received, and dialed calls.

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.
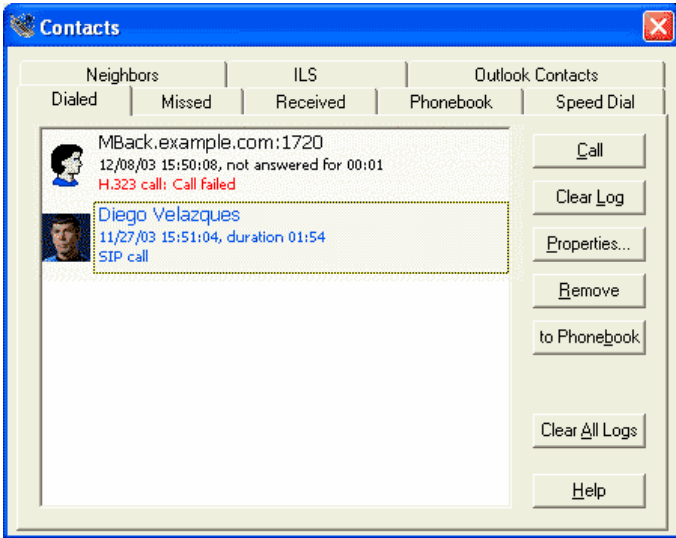
Call   Click this button to call the selected respondent

Clear Log   Click this button to remove all records in this log.

Properties   Click this button to view the information about a selected respondent

Remove   Click this button to remove the selected record from this log

to Phonebook   Click this button to add a selected respondent to **Phonebook**

Clear All Logs   Click this button to clear all logs.

Help   Click this button to activate **SJphone™ Help**

You may view the information about a call

To do this, double-click the call, or click the **Properties** button.

The **Call Record** message will appear.
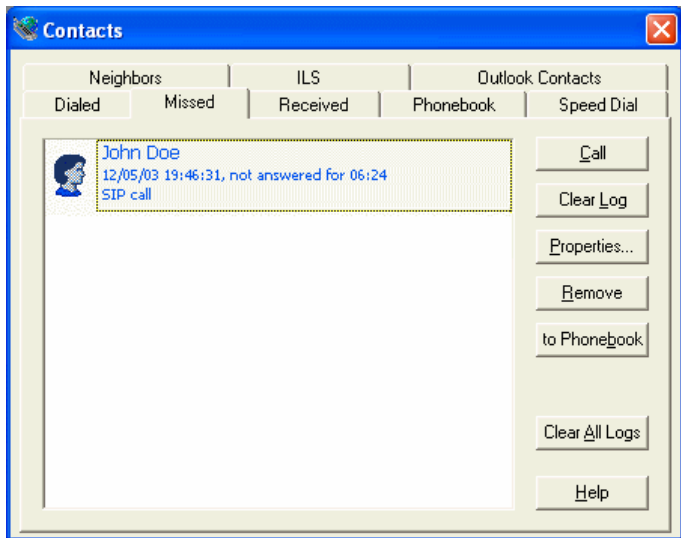
Call Record message
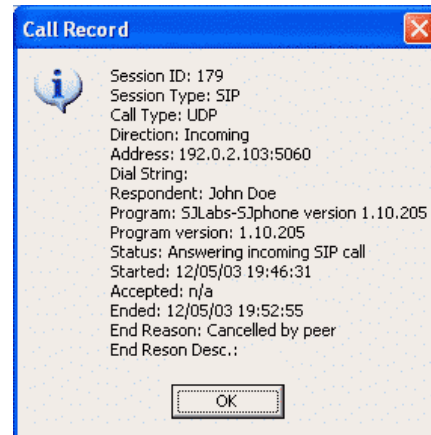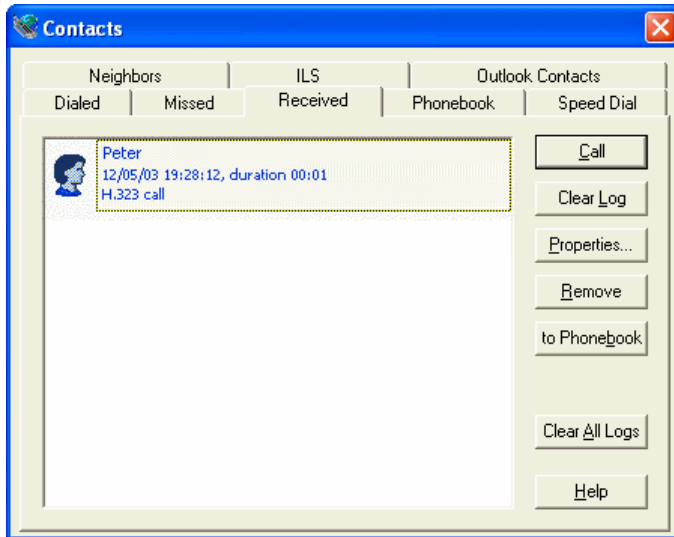
Call Record message
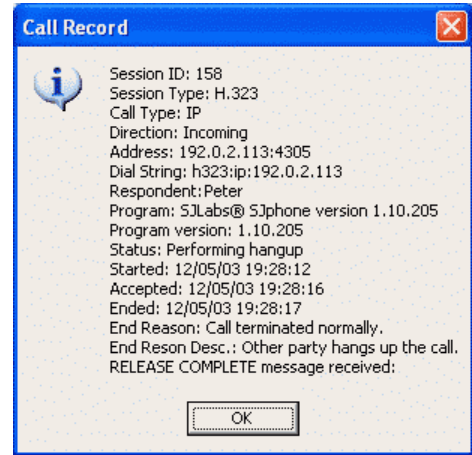
Phonebook

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.
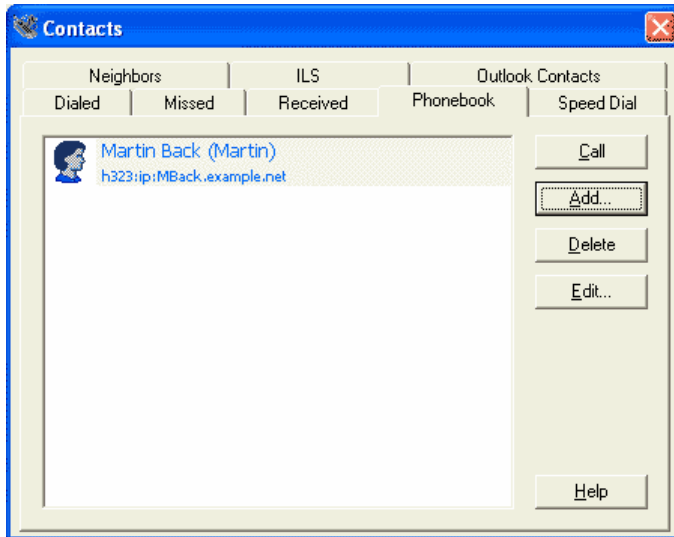
Call	Click this button to call the selected respondent
Add	Click this button to add a new record to **Phonebook**
Delete	Click this button to remove the selected record from **Phonebook**
Edit	Click this button to edit the selected record
Help	Click this button to activate **SJphone™ Help**
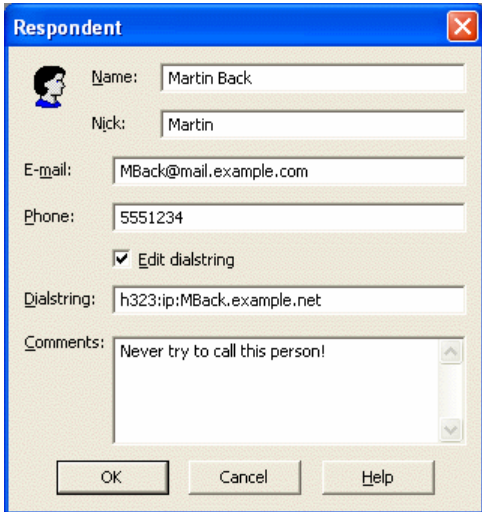
Full name:	Field for the respondent's full name
Nick name:	Field for the respondent's nickname
E-mail:	Field for the respondent's e-mail
Phone:	Field for the respondent's phone number
Edit Dialstring	Select this box if you want to use an advanced dial string. Go to the **Place a Call** section for details
Dialstring:	Field for the advanced dialstring for the respondent.
Comments:	Field for comments

Speed Dial

Click the necessary button on the main panel, or press the digital button on your keyboard. **SJphone™** will place the information in the **Call To**: field.

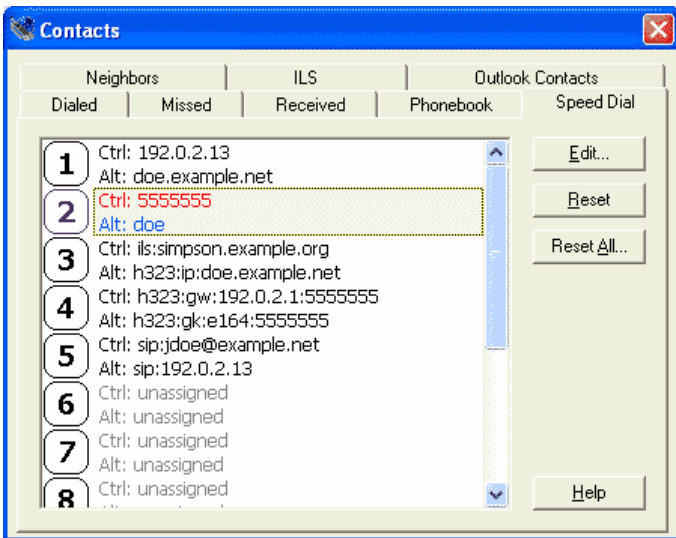| Edit	Click this button to edit the selected record. The **Speed Dial Key** window will appear.
Reset	Click this button to delete the information in the selected record
Reset All	Click this button to delete the information in all records
Help	Click this button to activate **SJphone™ Help**

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! If you use a personal firewall on your computer, make sure that it does not block UDP port 5003.
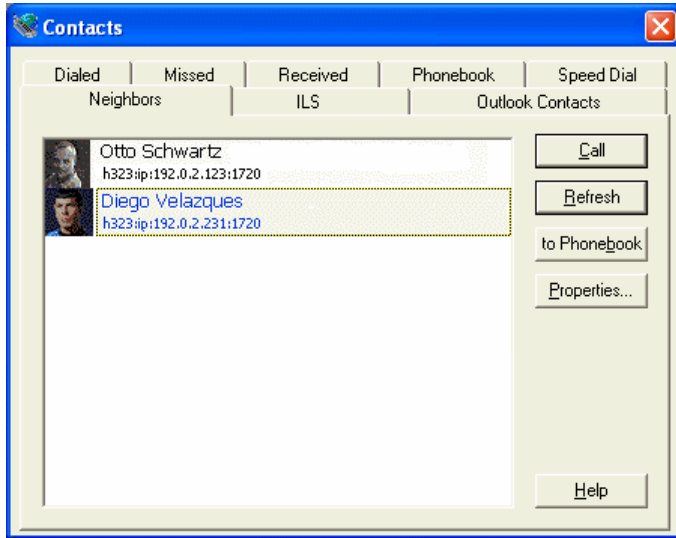
Call	Click this button to call a selected neighbor
Refresh	Click this button to refresh the list of neighbors
to Phonebook	Click this button to add a respondent from the **Neighbors** list to **Phonebook**
Properties	Click this button to view the information about a selected neighbor
Help	Click this button to activate **SJphone™ Help**
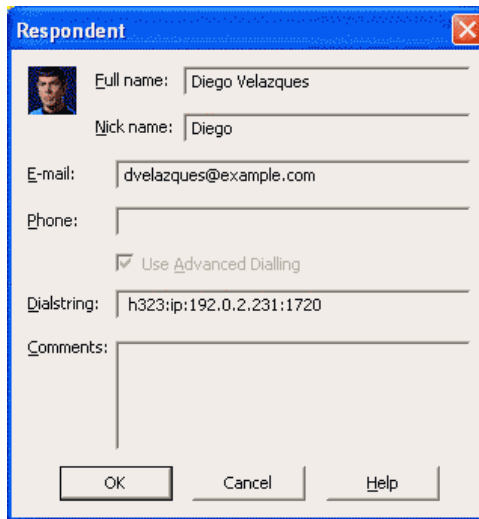
ILS Directory Browser

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. Click the **from Options** button.
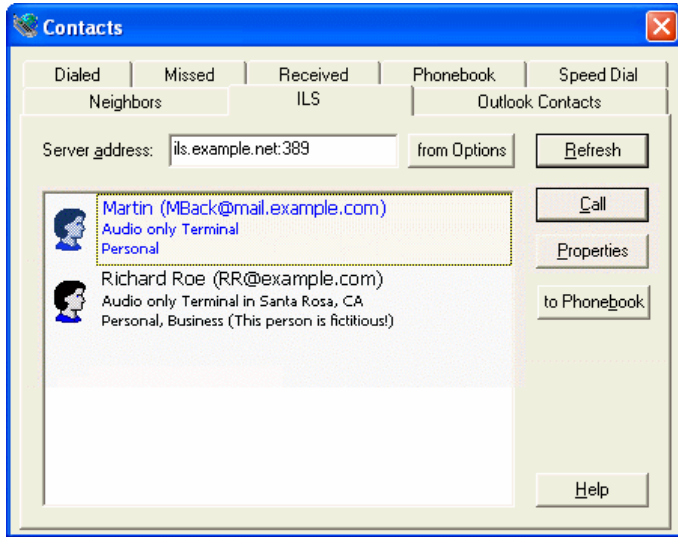
Refresh	Click this button to refresh the list of neighbors
Call	Click this button to call a selected neighbor
to Phonebook	Click this button to add a respondent from the **Neighbors** list to **Phonebook**
Properties	Click this button to view the information about a selected neighbor
Help	Click this button to activate **SJphone™ Help**

Outlook Contacts

Shows the contacts in the **Microsoft Outlook** or **Outlook Express** with phone numbers. It appears only if **Microsoft Outlook** or **Outlook Express** is installed.

To call the phone number of a selected contact, you need to select a profile for telephone calls.
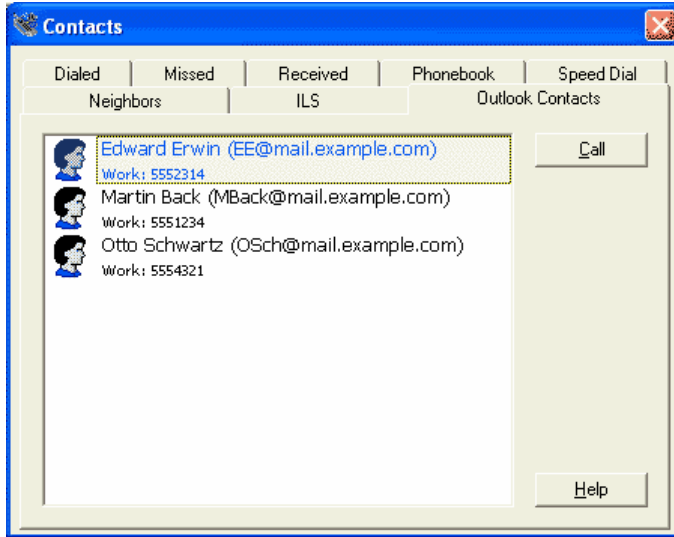
Call   Click this button to call a selected neighbor

Help   Click this button to activate **SJphone™ Help**

2.3. Skins

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

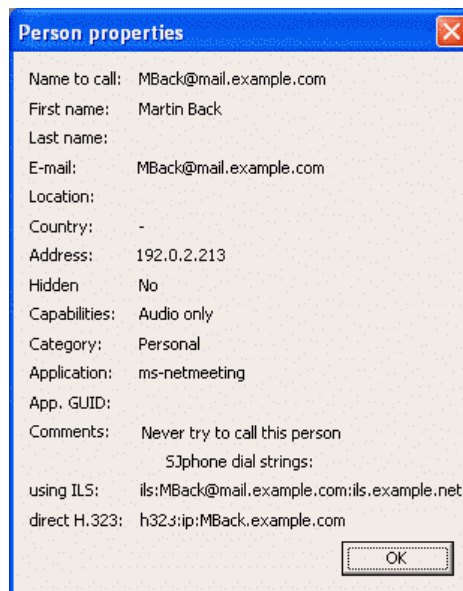**Attention!** Some skins may have no **Options** button. To access the **Options** panel, right-click the **SJphone™** system tray icon or skin and select **Options** on the system tray menu.
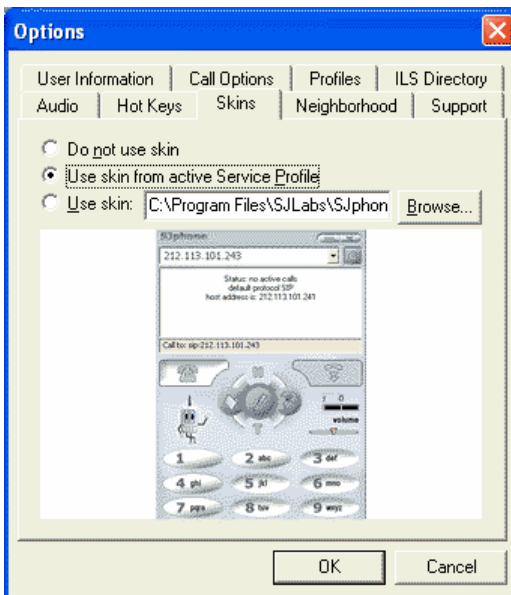
Do not use skin	Select this option if you do not want to use skins
Use skin from active Service Profile	Select this option if you want to use a skin specified in the active service profiles
Use skin	Select this option if you want to use a skin that you have selected
Browse	Click this button to browse for a skin

2.4. Hot Keys

You may assign several hot keys to perform common operations on the **Hot Keys** tab.
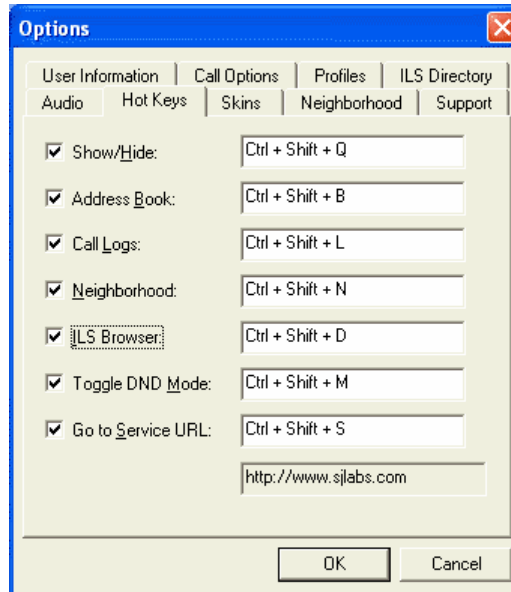
Attention! When any of the **Option** tabs is active, the hot keys are disabled.

III.   Configuring SJphone™

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

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

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

Interface mode switch:
- Right-click the **SJphone™** system tray icon or skin,
- 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 clicking the **Options** button on the **Main** panel, or by right-clicking the **SJphone™** system tray icon or skin and selecting **Options** in the tray menu.

Attention! Some **SJphone™** skins do not have the **Options** button. In this case, right-click the **SJphone™** system tray icon or skin.

3.1. Entering Your Personal Information

You need to personalize your **SJphone™** copy when you start your **SJphone™** for the first time. You also may change this information any time you like.

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	Click this button to browse for your picture

During the setup, **SJphone™** writes your Windows logon name in the **Name** field. If these fields are left blank, **SJphone™** uses "**SJ Labs(R) SJphone(TM) User**" as the username for H.323 calls and your IP address and Port 5060 for SIP calls.

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.
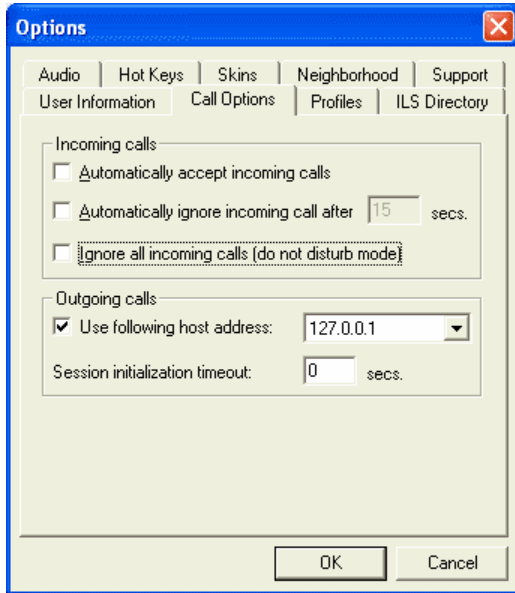
Incoming Calls:

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** checkboxes can be selected at a time.

Alternatively, you can turn the **Ignore all incoming calls (Do not disturb mode)** on and off on the **SJphone™** system tray menu. Right-click the system tray icon or skin to access this menu.

Outgoing Calls:

If your computer 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.

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 computer has several IP addresses and select one from the list of available addresses
Session initialization timeout	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

3.3. Audio Options

Generally, you can adjust the **SJphone™** sound characteristics using standard **Windows** audio controls or controls on the **Main** panel. Some additional audio options may be set on the **Audio** tab.
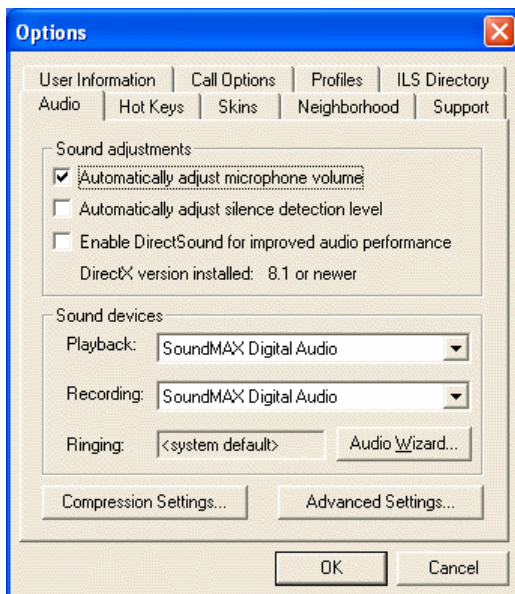
Sound adjustments	
Automatically adjust microphone volume:	Select this box to let **SJphone™** adjust the microphone volume automatically
Automatically adjust silence detection level:	Select this box to enable the automatic adjustment of silence detection level
Enable DirectSound for improved audio performance:	Select this box if your sound card supports **DirectSound**
DirectX version installed:	Check **DirectX** version installed in your system
Sound devices	
Playback:	Shows what audio device **SJphone™** uses to playback audio.
Recording:	Shows what audio device **SJphone™** uses to record audio.

Compression Settings:	Click this button to set **SJphone™** compression settings.
Advanced Settings:	Click this button to set **SJphone™** advanced audio settings.

Automatically adjust microphone volume

Some sound cards 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 compress silence reducing processor 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.

Enable Direct Sound for improved audio performance

Select **Enable Direct Sound for improved audio performance** if your sound card supports **DirectSound**. This will decrease sound latency time. Most sound cards from Turtle Beach, Yamaha, Creative Labs (excluding the Ensoniq-based AudioPCI types), Diamond, Crystal, or Microsoft USB speakers support this feature.

SJphone™ supports DirectX version 7.0 and higher.

Sound Devices

If your computer has several sound cards, you may select which one will be used. **SJphone™** uses a system default audio device to play audio during a call unless you manually select a specific audio device in this option. If the system does not find this device, **SJphone™** will use the system default audio device. When the system finds the specified audio device, **SJphone™** will use it again.

SJphone™ always uses the system default audio device to play incoming ring sound.

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

Advanced audio and compression settings such as **Compression settings** or **SJ Labs Extensions** are explained in detail in the **Compression and Advanced Audio 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.
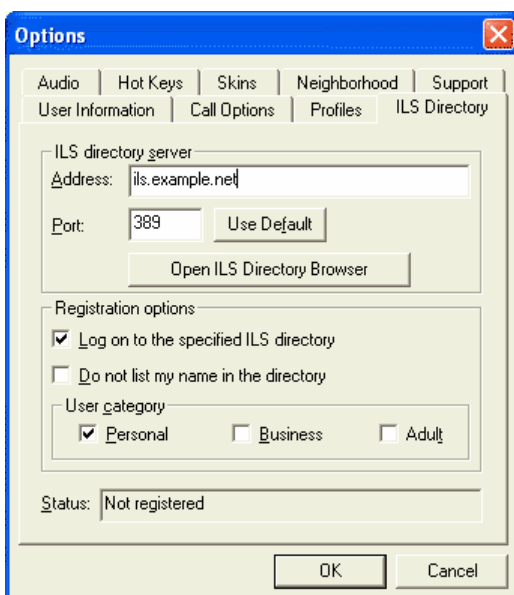
ILS Directory Server	
Address:	Field for the **ILS Directory Server** address
Port:	Field for the **ILS Directory Server** port number
Registration options	
Log on to the specified ILS directory	Select this box to log on to the **ILS Directory**
Do not list my name in the directory	Select this box if 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:	Click this button to switch to the **ILS Directory Server** default port (389)
Open ILS Directory Browser	Click 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 click 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. 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! If you use a personal firewall on your computer, make sure that it does not block UDP port 5003.

When running, **SJphone™** constantly sends broadcast messages to the local network. Other **SJphone**s 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.
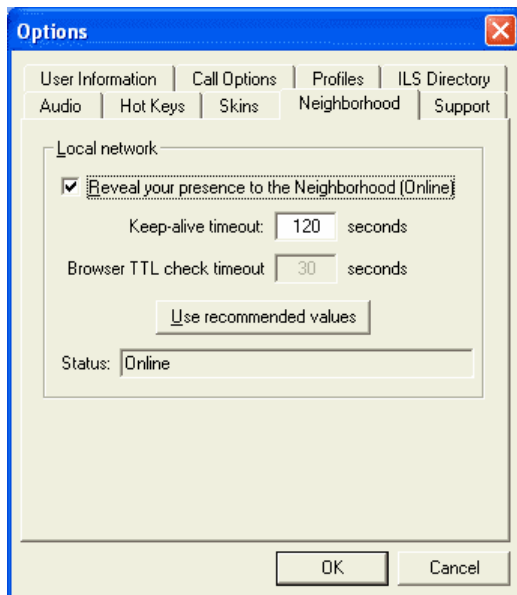
Local network

Reveal your presence to the Neighborhood (Online):	Check this box to reveal your presence to your network neighbors
Keep-alive timeout:	Field for the timeout in which **SJphone™** sends messages about its presence to other **SJphone**s on your local network.
Status:	Shows your neighborhood status.
Buttons	
Use recommended values	Click/tap this button to set the values to their recommended values

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

3.6. Customized Incoming Call Sound

You may use an incoming call sound of your own choice if you rename the sound file into `ring.wav` and place it in the **SJphone™** folder, default is …`\Program Files\SJLabs\SJphone`. You may also use another sound file named `beep.wav` as a second incoming call notification (it will be played if you have an active session). Place it in the same folder.

3.7. Personal Firewall Settings

If you have a personal firewall installed on your computer, a number of ports should be opened for the `SJphone.exe` application both for inbound and outbound connections. Go to the **Used Port** section for details.

Refer to your personal firewall documentation for instructions how to make necessary adjustments.

At some firewalls, you may set `SJphone.exe` as a trusted application, that is, an application which is allowed to make any connection by default. Refer to your personal firewall documentation for instructions.

3.8. Windows XP Service Pack 2 Settings

Generally, **SJphone™** should work fine with minimum adjustments in **Windows Security Center**.

If you turn off the **Windows firewall** in **Security Center**, no settings are required to change for **Windows** or **SJphone™**.

If **XP Service Pack 2** is installed on your computer with its default-enabled **Windows firewall**, the **Windows Security Alert** message will appear the first time you start **SJphone™**. This message will inform you that it has blocked **SJphone™** from accepting connections from the Internet or a network.
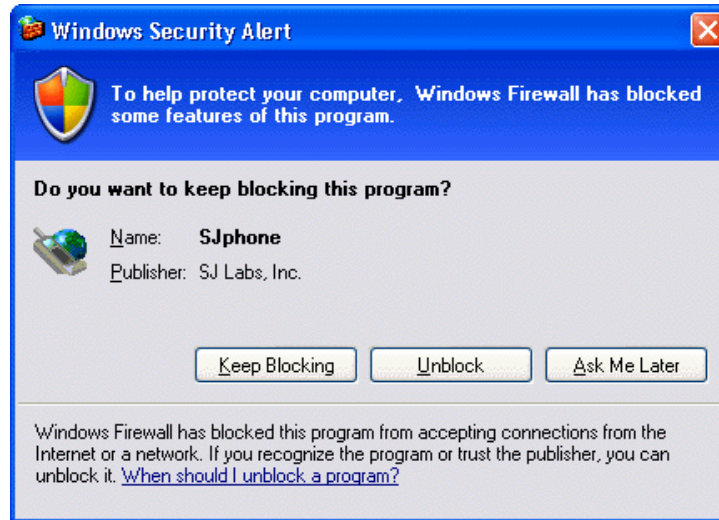
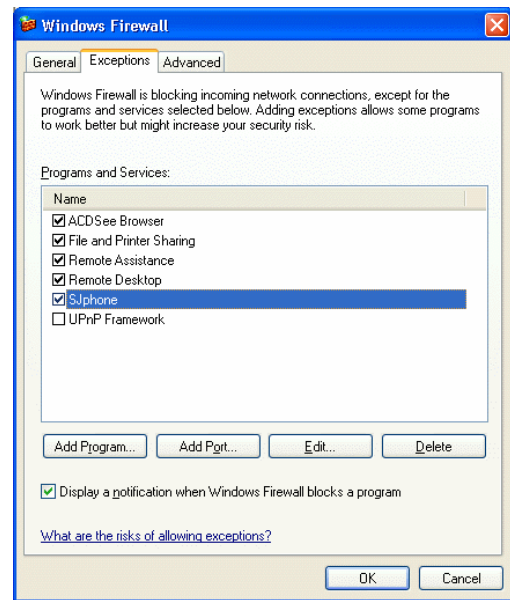

To allow SJphone™ receive incoming calls,

- click the **Unblock** button on the message.

Now **SJphone™** will be able to receive incoming calls. No other additional settings are required for **Windows** or **SJphone™**.

If you have clicked the **Keep Blocking** button, **SJphone™** will not be able to receive incoming calls.

To allow SJphone™ receive incoming calls,

- Check that **SJphone™** is running.
- Go to **Security Center** and click **Windows Firewall**.

- Check that **Don't allow exceptions** is not selected on the **General** tab.
- Go to the **Exceptions** tab and select the `SJphone.exe` checkbox.

- Click the **Edit** and then **Change scope** buttons and check that the **Any computers (including those on the Internet)** option is selected.
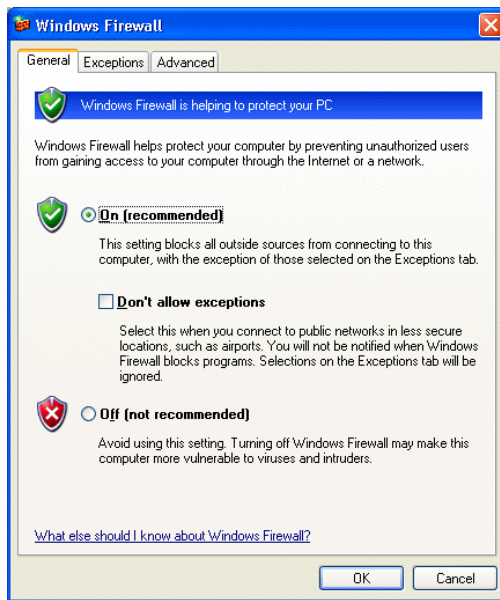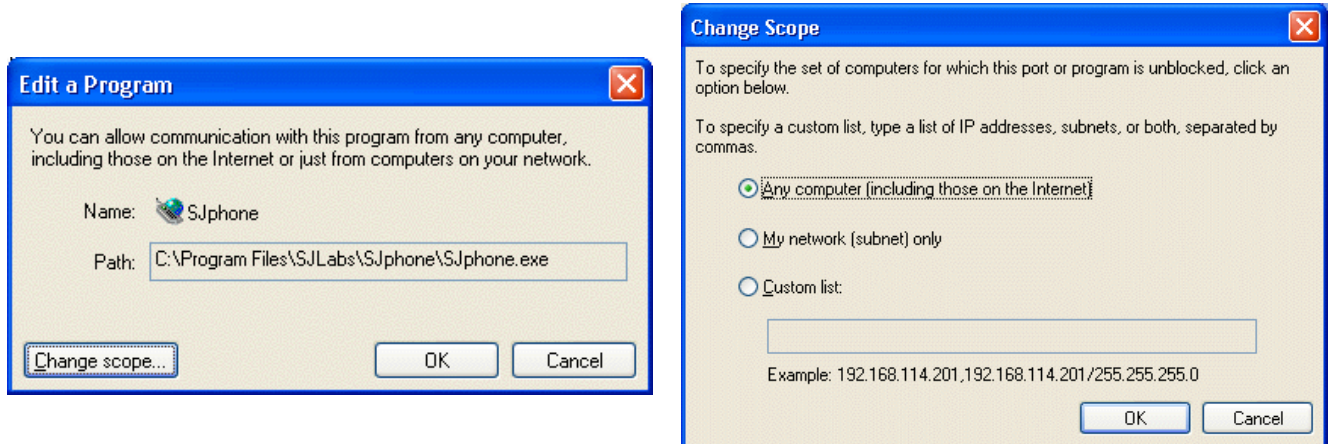

- Click the **OK** button.

Now **SJphone™** will be able to receive incoming calls. No other additional settings are required for **Windows** or **SJphone™**.

If you want to add manually SJphone™ to the list of programs that can accept incoming connections,

- Go to **Security Center** and click **Windows Firewall**.
- Check that **Don't allow exceptions** is not selected on the **General** tab.
- Go to the **Exceptions** tab and click the **Add Program** button.
- Click the **Browse** button and select `SJphone.exe` in its home folder. Default is
  `…\Program Files\SJLabs\SJphone`.
- Click the **Change scope** button, check that the **Any computers (including those on the Internet)** option is selected, and click the **OK** button.
- Check that `SJphone.exe` is in the **Program and Services** list on the **Exceptions** tab, and its checkbox is selected,
- Click the **OK** button.

Now **SJphone™** will be able to receive incoming calls. No other additional settings are required for **Windows** or **SJphone™**.

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 computer is on a private network, you need to place

  an H.323 call through a **Gatekeeper** using an appropriate profile. If there is 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 computer 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" to the phone number to route calls to phones through required gateways. Ask the gateway administrator for details.

  If you have **Microsoft Outlook** or **Outlook Express** installed on your computer, you may call the numbers stored in its contacts. Go to the **Contacts** section for details.

  You may also use a Gateway 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 form 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 click the Dial button on the SJphone™ Main panel, or press Alt + d or Enter.

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, click 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 computer receives a call, you will hear a ring sound. Simultaneously, an **Incoming Call** message will appear.
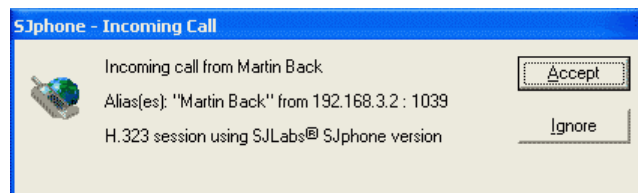
- **To answer** the call, click the **Accept** button, or press **Alt + a** or **Enter**.
- **To reject** the call, click the **Ignore** button, or press **Alt + i**.

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

The **Do Not Disturb** mode can be also turned on and off on the system tray menu. To access the menu, right-click the system tray icon or skin.

You may customize the sound **SJphone™** plays when a call is incoming. Go to the **Customized Incoming Call Sound** 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,

- Double-click a call you want to put on hold

or

- Select a call you want to put on hold and click the **Hold** button

or

- Right-click a call you want to put on hold and select **Hold** in the pop-up menu

To resume a call on hold,

- Double-click a call you want to resume

or

- Select a call you want to resume and click the **Hold** button

or

- Right-click 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 click the **Transfer** button
- Enter the respondent address in the **Call To** field and
- Click the **Transfer** button once again or press **Enter**

Attended transfer:

- Establish a call to a respondent you want to transfer call to
- Select a call you want to transfer and click the **Transfer** button
- Select the call with the respondent
- Click the **Transfer** button once again

4.4. Switch between Respondents

On the **Main** panel, double-click the respondent with whom you want to talk.

If **Enable Conference Mode** on the **Audio** tab is selected, such double-click adds/removes the respondent to/from the conference. Go to the Start a **Conference** section for details.

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, you need to start a conference. That can be done by clicking the **Conference** button on the **SJphone Main** panel. Now you may talk to several respondents simultaneously.

A double-click adds/removes the respondent to/from the conference. Or you may select several respondents, right-click them, and select **Join to Conference**/**Exclude from Conference** item in the pop-up menu.

To switch back from the conference mode, click the **Conference** button on the **SJphone™ Main** panel once again.

4.6. Find Other SJphone™ Users on the Local Network

When running, **SJphone™** constantly sends broadcast messages to the local network. Other **SJphone**s 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,

- Click 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 showing how many messages are waiting, and the **Voice Mail** button will be active.

To listen the messages, click the **Voice Mail** button and follow instructions from the server.

4.8. Send a DTMF Signal

SJphone™

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

To send a DTMF signal,

- Click a digit button on the **Main** panel

  or

  Press a digit key on a keyboard.

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 Audio 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 make your own, although that requires some knowledge of VoIP technology.

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

To select a Service Profile,

- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile

  or

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

Some profiles require initializing, that is, they require the user to enter some account-specific information, such as your phone number, login, access password, etc. A **Service** message will appear the first time you select such profile. You may also re-initialize the profile, if its account-specific information changes.
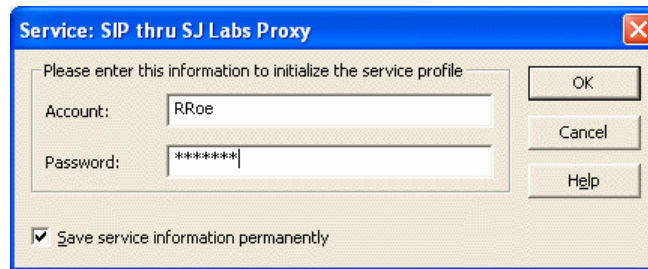
To re-initialize a current profile,
- Double-click the **SJphone** respondent control
  The **Service:** message will appear
- Enter the required information and click the **OK** button

To initialize or re-initialize a 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.
  The **Service Properties** message will appear
- Enter the required information and click the **OK** button

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.
  The **Service Properties** message will appear
- Enter the required information and click the **OK** button

4.10. Create a Desktop Shortcut to a Respondent

You may create a desktop shortcut to a respondent. When it is created you may call this respondent simply by clicking this desktop shortcut.

To create a desktop shortcut,
- Right-click anywhere on the desktop and select **New** -> **Shortcut**
  The **Create Shortcut** wizard will appear
- Type the **callto URL** to the respondent. The **callto URL** is described on the **SJphone™ Command Line Switches** section.

Callto URL examples:

If you want to call
- A host `jd.bigcom.com`
  Type in the **Create Shortcut** wizard callto:jd.bigcom.com
- A host with the IP address `192.168.12.12` using the H.323 VoIP protocol
  Type in the **Create Shortcut** wizard callto:192.168.12.12+type=ip
- A host with the IP address `192.168.12.12` using the SIP VoIP protocol
  Type in the **Create Shortcut** wizard callto:192.168.12.12+type=sip

4.11. End a Call

To end a call,
- Select the respondent and click the **Hangup** button on the **SJphone™ Main** panel, or press **Alt + h** or **Enter**,
or
  Right-click the respondent and select the **Hangup** item in the pop-up menu.

You may end several calls simultaneously. Select the respondents, right-click them, and select **Hangup** in the pop-up menu.

4.12. Restart SJphone™

To restart SJphone™,

- Right-click the **SJphone™** system tray icon or skin and select **Restart** on the system tray menu.

4.13. Exit SJphone™

To exit SJphone™,

- Right-click the **SJphone™** system tray icon or skin and select **Shut down** on the system tray menu.
  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 write your own profile for your IP-PBX, ITSP or Gateway parameters with your own set of options, although that requires some knowledge of VoIP technology.
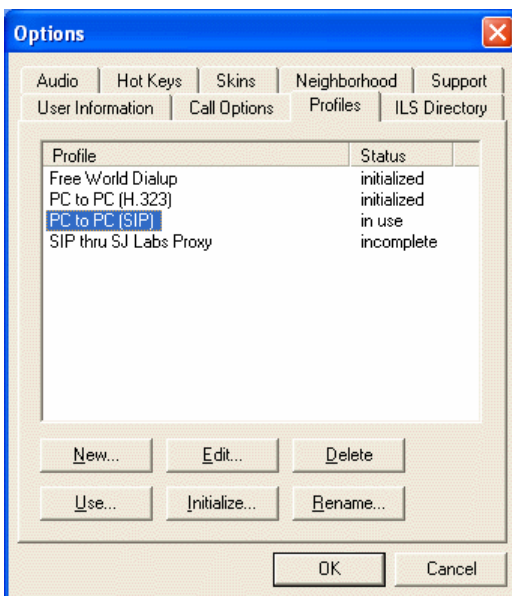
New	Click this button to create a new profile
Edit	Click this button to edit a selected profile
Delete	Click this button to delete a selected profile
Use	Click this button to switch to a selected profile
Initialize	Click this button to initialize a selected profile
Rename	Click this button to rename a selected profile
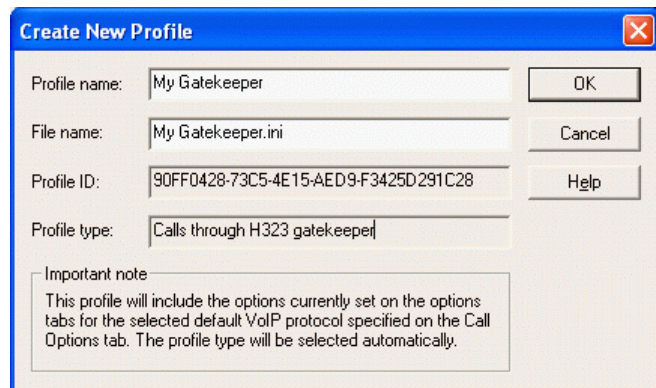
To create a new profile,

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
- Click the **New** button
A **Create New Profile** window will appear
- Select profile type, and specify its name.
- Specify the required parameters and click the **OK** button.
The newly created profile will appear in the profile list.
Note: To enter some parameters like a Gatekeeper, Gateway, or SIP proxy address, caller ID, and password, a profile may require initialization.

To select a profile,

- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile

or

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab.
- Select the required profile and click 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** message will appear the first time you select such profile. You may also re-initialize the profile, if its account-specific information changes.
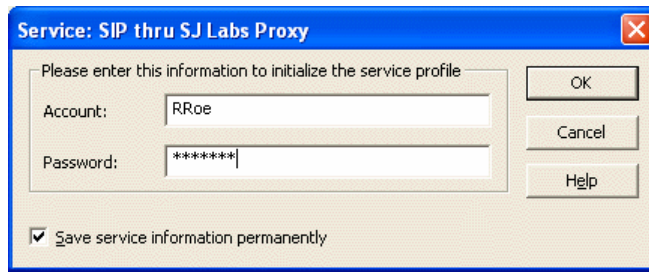
To initialize or re-initialize a profile,

- 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

The **Service** message will appear

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

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

The **Service** message will appear

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

To edit a profile,

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
- Select the required profile and click the **Edit** button
- Change the required parameters and click the **OK** button

To remove a profile,

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
- Select the required profile and click the **Remove** button

A **Delete Profiles** message will appear
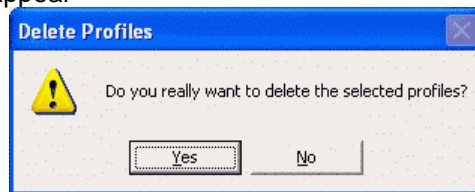
- Click the **Yes** button to confirm profile remove.

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

5.2 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 (Internet Telephony Service Provider) 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 Windows, or are on a private network (your computer has an IP address started with 10.0.x.x, 192.168.x.x, 172.x.x.x).

If your service profile allows you to do this, you may check if SJphone is behind NAT. Click Options and 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

It is **unclea**r if you are or are not behind NAT if you see the following:
- NAT/Firewall: Unknown
- NAT/Firewall: Symmetric Firewall
- 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 (Internet Telephony Service Provider), on your network configuration, etc. Always consult your ITSP (Internet Telephony Service Provider) 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 (Internet Telephony Service Provider) 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 "Enable STUN usage" checkbox on the Compatibility tab.

3. One proxy does not support NAT traversal

Select the Enable STUN usage checkbox on the Compatibility tab. STUN settings are 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) or/and network administrator in this case.

5.3. Compression and Advanced Audio Settings

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

To change the compression or advanced audio settings, click the **Compression** or **Advanced Settings** buttons 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 the systems, bandwidth limitations, desired sound quality, etc. You may adjust **SJphone™** codec preferences on the **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.
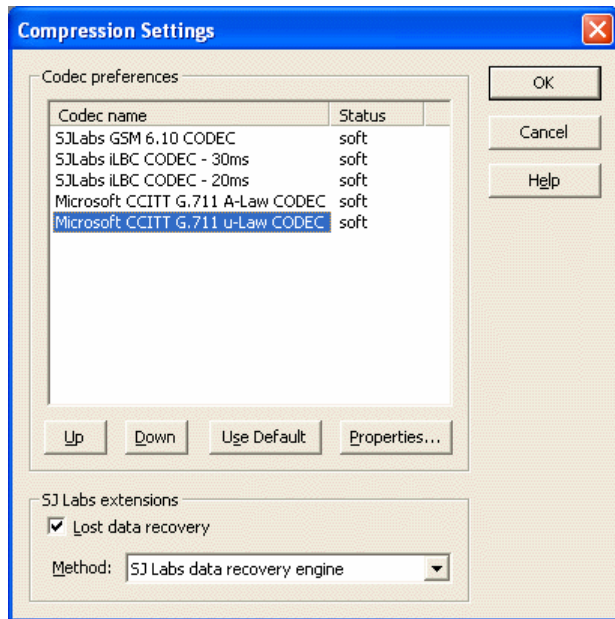
Codec preferences

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:	Click this button to move the selected codec up
Down:	Click this button to move the selected codec down
Use Default	Click this button to use a default codec preference
Properties:	Click this button to activate the Audio Codec Properties window
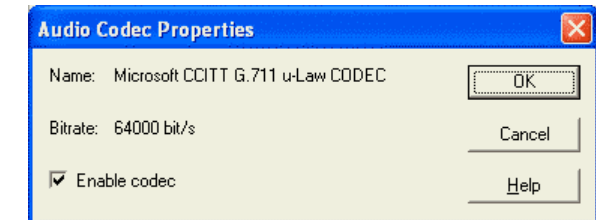
SJ Labs Extensions

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 click the **Properties** button, or double-click 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.

The current **SJphone™** version supports the following codecs:

- G711 A –Law and G711 u –Law codecs. They provide good sound quality but low voice compression. Both codecs require 64 kbit/sec bandwidth and may be recommended for wide bandwidth networks, such as LANs.
- GSM and G.729 codecs may be recommended if network bandwidth is limited, for example for long-distance calls or for dial-up connections. G729 codec provides good voice compression at 8 kbit/sec plus overhead. GSM codec provides an average sound quality and good voice compression at 13 kbit/sec plus overhead.

- iLBC codec may be recommended for networks with a large number of lost packets. It provides good sound quality at 13.3 kbit/sec plus overhead.

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 802.11x.

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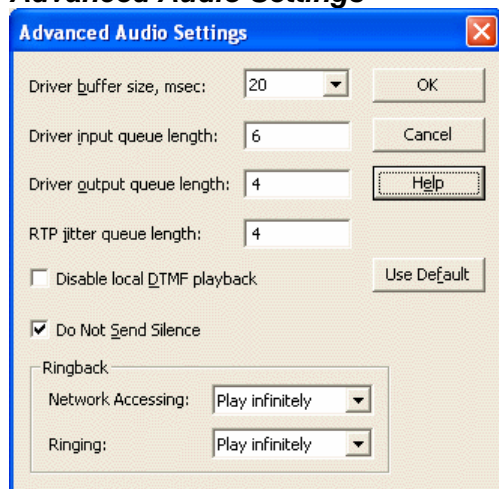
These settings allow you to change sizes of various internal **SJphone™** sound buffers and queues if you experience **problems with sound**.

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.

Ringback: You may select how many times **SJphone™** plays sound when it accesses the other party (**Network Accessing**) or waits for the other party to answer (**Ringing**).

5.4. SJphone™ Command Line Switches

You can run **SJphone™** from a command line.

In order to do this, click the **Start**->**Run**->**Browse** to find the place where `SJphone.exe` is installed, or run it from the Windows Command Prompt: `C:\Program Files\SJ Labs\SJphone.exe`

> Syntax: `SJphone [<DialString>][<keys>]`
>> where:
>>> `<DialString>` is either an Extended H323 Address, SIP URL, or ILS directory person.
>>> `<keys>` - one of the following:
>>>> `/?` or `/h` or `/help` invokes command line switch help.
>>>> `/lf <FileName>` - **SJphone™** writes its log to `FileName`. This key may be used with other keys.
>>>> `/p <PortNumber>` - **SJphone™** starts at port number `<PortNumber>`. Default is 1720.
>>>> `/d` - **SJphone™** hangs up all existing calls.
>>>> `/callto <URL>` - **SJphone™** calls to the specified `URL`, where `URL` is an URL specified as:
>>>>> a host name (example: jd.bigcom.com)
>>>>> an IP address `+type=ip` (example: 192.168.1.12+type=ip)
>>>>> a telephone number`+type=phone` (example: 1357999+type=phone) - a call through a default Gateway or Gatekeeper, whichever is selected by the default profile.
>>>>> a telephone number`+gateway=<gatewayname>+type=phone` (example: 1357999+gateway=h323gateway.com+type=phone) - a call through a specified Gateway
>>>>> a sip URL `+type=sip` (example: jd@bigcom.com+type=sip) - a call to the user jd at the host bigcom.com using the SIP protocol.
>>>>> any string `+type=hangup` (example: hangup+type=hangup) - all active calls will be terminated
>>>>> `<DialString>+type=dialstring` (example: h323:gk:h323id:Gedy+type=dialstring) -a call to user Gedy registered with the current gatekeeper.

The parameter `existingcalls` may be added to any `callto` string. This parameter specifies what to do with active calls.

+`existingcalls=leave` (default) leaves all calls in their current state.

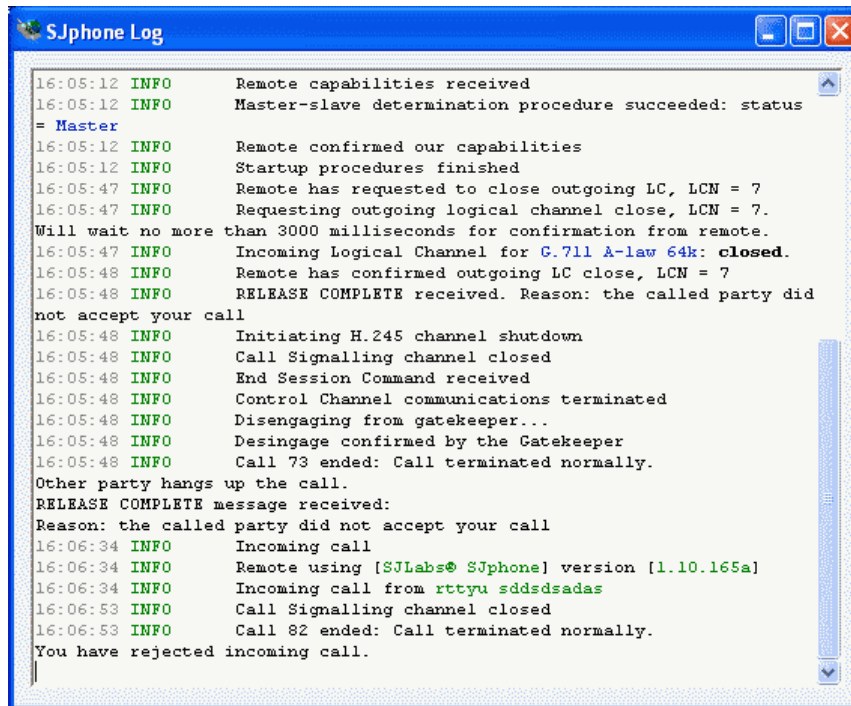+`existingcalls=hangup` terminates all active calls before a new call.

Callto *example:* callto:h323:gk:h323id:Boss+type=dialstring+existingcalls=hangup - all active calls will be terminated and **SJphone™** will start a call to user Boss registered with a current **SJphone™** gatekeeper.

5.5. Log Window

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

The log window shows you technical information about **SJphone™** operation and call progress.

To understand this information, you need to be familiar in depth with the ITU-T H.323 protocol details.

To activate the SJphone™ Log window, right-click the **SJphone™** system tray icon or skin and select **Show Log** on the system tray menu. The **SJphone Log** window will appear.

To end the SJphone™ Log window, right-click the **SJphone™** system tray icon or skin and clear Show Log.

Alternatively, you can start **SJphone™** with the following key:

        SJphone.exe [<IPAddress>] [<keys>]

    Where

    `<IPAddress>` is the IP address of the system to be called,

    `<keys>` – one or more of the following

     `/lf <Filename>` - log output to the file <Filename>

SJphone™ will write its log to the specified log file.

5.6. 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

5.7. 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

5.8. DTMF Signal Sending

Depending on the VoIP protocol and service, **DTMF** signals may be sent using various methods. Which method is to choose depends on a selected service. The method is specified in the service profile.
SIP

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

H.323

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**:Q.931 keypad: **DTMF** signals are sent through the Q.931 KeypadFacility Information Element of Q.931 messages.
3. **H.245 alphanumeric**:H.245 alphanumeric: **DTMF** signals are sent as one of the out-of-band method in H.245 protocol messages.
4. **H.245 signal**:H.245 signal: **DTMF** signals are sent as one of the out-of-band method in H.245 protocol messages.

The remote user will hear **DTMF** signals when the **In-band** method is used.

5.9. 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

Check your CPU usage. It should be less than 50%, otherwise you might consider upgrading your CPU.

6.1. Start-up Troubles

If **SJphone™** does not start up properly, try the following:

- Check that other audio recording or playing software is not running.
- Check that other H.323 or SIP software, such as NetMeeting or Windows Messenger, is not running.

If the problem persists, contact SJ Labs technical support.

6.2. Sound System Issues

Sound quality can vary significantly depending on your sound card, microphone, speakers, and Internet connection. Make sure that you have a good microphone and speakers, and the latest driver for the soundcard.

You can adjust the **SJphone™** sound characteristics using standard Windows audio controls or controls on the **Main** panel. The **Mic** and **Spk** controls are connected with the standard Windows audio controls. Any changes in their settings automatically change the corresponding settings at the Windows volume control panel.

SJphone™ requires a full-duplex sound card. You can check whether your sound card is full duplex. Try to record sound from the microphone with the **Sound Recorder** while another copy of the **Sound Recorder** plays back any sound file. If you can do this, your sound card is full duplex. If you cannot, your sound card driver may be not full duplex. Obtain a full duplex driver from the sound card manufacturer.

You can use the automatic microphone gain feature in your sound card to automatically adjust the input sound level. Consult your sound card documentation for details. Some sound cards do not have the automatic microphone gain feature. You can select the **Automatically adjust microphone volume** box on the **Audio** tab to let **SJphone™** automatically adjust the microphone sound level.

If you experience a bad sound quality, turn off the Automatic Gain Control feature on the Windows sound control panel. Go to **Control Panel**, then to **Sounds, Speech, and Audio Devices**, then to **Sounds and Audio Devices**, then to **Volume**, and to **Advanced** for **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 card supports DirectSound, select the **Enable Direct Sound for improved audio performance** box on the **Audio** tab. This will decrease sound latency time. Most sound cards from Turtle Beach, Yamaha, Creative Labs (excluding the Ensoniq-based AudioPCI types), Diamond, Crystal, or Microsoft USB speakers support this feature.

If you are using DirectX option, make sure that you have DirectX 7.0+ installed. You can check it with `dxdiag.exe`.

If you hear hissing noise from your speakers while talking, try the following:

- Move the microphone farther from the speakers.
- Decrease the microphone or speaker sound level.

If the problem persists, contact SJ Labs technical support.

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 clicking 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.

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.

6.3. 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.

If you experience problems in connecting with a remote host, try to check the connection using the **Ping** command in the DOS prompt. Refer to **Windows** documentation for details.

The **Ping** command can also be used to check communication delay with the host being called. Reasonable roundtrip time should be 100-200 ms for a dedicated and 200-300 ms for dial-up connection. The delay should vary within 10% of average connection delay. If the delays are unexpectedly long, contact your Internet provider or system administrator.

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

6.4. 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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