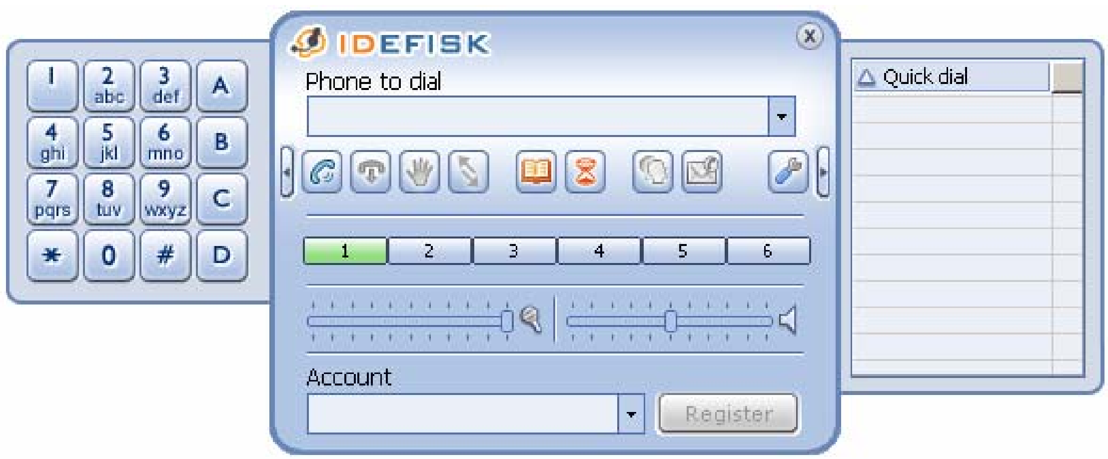
asteriskguru.com

IDEFISK



USER GUIDE

Edition Version 2.0 Beta for Windows®

Date of issue: January 2007

0. Notice

0.1 Liability

- This *User Guide* is intended to demonstrate typical uses and features of **Asteriskguru**'s SIP/IAX2 soft phone: **IDEFISK Free Edition V.2.0**. It is up to the user of this manual to decide whether the information mentioned herein is applicable to the particular IP-based network (s)he wants to run this soft phone application on.

- **Asteriskguru** and persons involved in the composition of this *User Guide* will in no case be held responsible for any incidental, indirect or otherwise consequential damage or loss that may result from using **IDEFISK Free Edition V.2.0**.

0.2 Conventions used in this document

The following typographical conventions are used in this document:

- Hyperlinks to sections inside this document, email and the Internet are underlined and blue.
- The names of windows are marked in ***Bookman Old Style, bold and italics***.
- The names of files, directories and syntax of commands are written in *italics.*
- Parameters of commands are written as follows: <*italics*>
- Buttons are marked in **Bold**. They start with a capital letter.
- Sequential clicking on buttons and names of (sub)directories are separated by an arrow pointing to the right: **>**
- Sequential steps in a process are preceded with numbers: 1,2,3...

CONTENTS

1. IDEFISK Editions

1.1 What's new?

Compared to the previous version of IDEFISK, the new release contains:

- SIP support
- STUN support
- STUN server per account
- New and enhanced Graphic User Interface
- Codec per account
- Changeable ports
- Context for account
- Echo cancellation
- Call recording to WAV file
- Mixed call recording input and output

1.2 Introduction

This document will guide you step by step to your actual aim: reducing the costs of your phone calls by using the VoIP (Voice over Internet Protocol) technology of *Asterisk*® and the latest version of **Asteriskguru**'s widely used soft phone **IDEFISK**.

As always, **Asteriskguru** took your many suggestions into account in order to uplift our soft phone to an even more enhanced level of user-friendliness. We therefore sincerely hope you will have at least as much fun using **IDEFISK** as we had in developing it.

In the rest of the first chapter of this *User Guide* you can find more details about the **IDEFISK Free Edition**. If you are interested in having your "own" phone but don't have the resources to have it developed, **Asteriskguru** offers you the possibility to obtain a re-branded **OEM IDEFISK**.

Do not hesitate to send a mail to idefisk@asteriskguru.com for more information and offers. Our team of experienced professionals is most willing to answer all your questions.

At the same time, our developers are looking forward to implement your specific desires in future versions of **IDEFISK**. You can contact them at support@asteriskguru.com

Together, we make calling more comfortable.

1.3 IDEFISK Free

IDEFISK Free **Edition V.2.0 Beta** has many features to provide you with a very comfortable calling experience:

- availability of up to 6 lines
- a user-friendly interface
- call recording
- easy access to and managing of your contact persons
- caller identification
- easy switching between users
- a clear overview of your phone calls

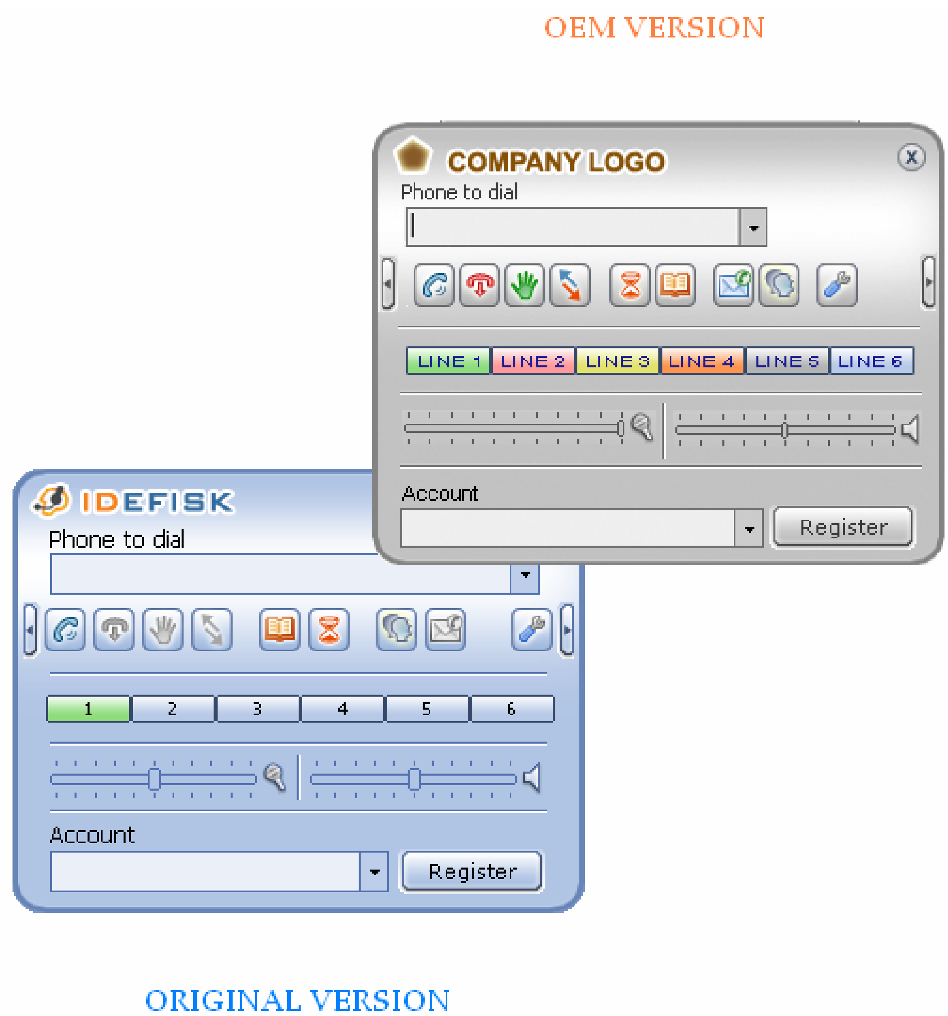
1.4 Personalised OEM IDEFISK editions

- Do you want your employees to use state-of-the-art VoIP technology with your own company's logo and colors clearly visible on it?
- Do you want to surprise your clients with your "own" personal telephone?
- But you don't have the resources to develop your own soft phone...

Asteriskguru offers you the possibility to obtain a personalised verion of IDEFISK Free **Edition**. This rebranding includes:

- changing the name "IDEFISK" to your choice.
- changing the color scheme to your choice.
- changing the logo to the logo of your choice.

Below, you see an example of an IDEFISK **OEM** edition next to an original one:



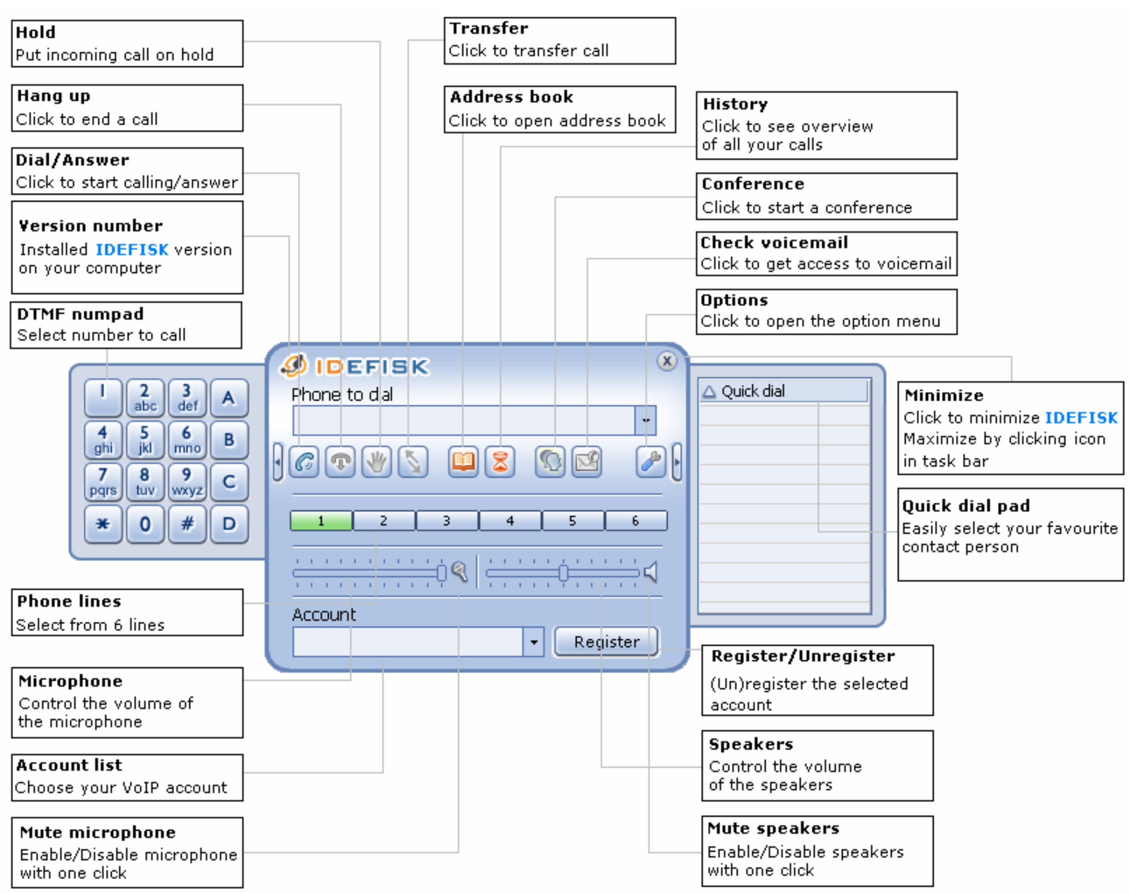
2. Getting started with IDEFISK

2.1 System requirements

In order to use IDEFISK properly, you need at least following system configuration:

- Processor: minimum *Pentium* II 300
- Memory: minimum 256 MB RAM
- Operating systems: *Windows*® 2000, XP and later
- Sound card: 16 bit sound card (SoundBlaster or equivalent)
- Internet connection: wired or wireless broadband

2.2 A glance at IDEFISK



You can see the function of the dial buttons when you hold your mouse over the button.

2.3 Download

You can download **IDEFISK Free Edition** here.

2.4 Installing IDEFISK

To install **IDEFISK**, follow the instructions below:

1. Click on the icon of the downloaded *.exe* file: 

2. The ***InstallShield Wizard*** now opens. Click **Next > Next**.
3. You can choose the directory where you want to install **IDEFISK** in, by clicking on the **Change** button.
4. Click on **Next > Finish**.

2.5 Uninstalling IDEFISK

To remove **IDEFISK** from your computer, you can choose one of the following:

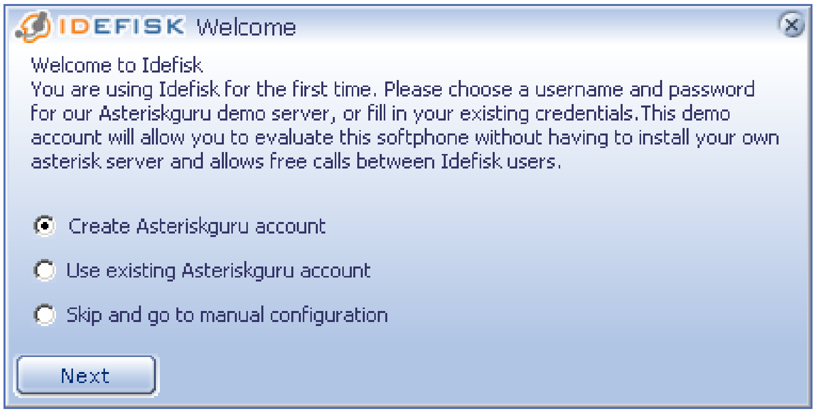
- Uninstall **IDEFISK** via the Control Panel>Add/Remove Programs. After you find **IDEFISK** from the list of applications, click on the **Remove** button.
- Follow these steps:

    1. Click on the icon *idefisk123installer.exe.*
    2. The ***InstallShield Wizard*** now opens. Click **Next**.
    3. Tick the third option *Remove* and click on **Next**.

    4. Click on the **Remove** button.
    5. Click on Finish.

2.6 Starting IDEFISK for the first time

1. To open IDEFISK click **Start > Programs > Asteriskguru > Idefisk**.

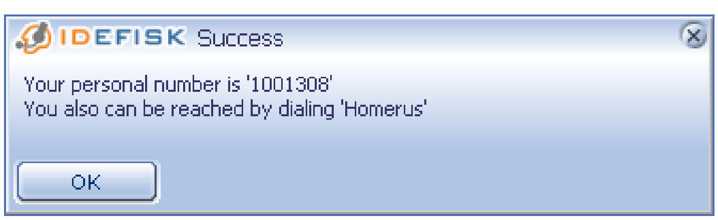


2. The ***Welcome*** screen will appear:
3. Tick one of the options and continue by clicking on **Next**.

2.6.1 Create Asteriksguru account

1. This option will lead you to the following screen:

2. After having filled in succesfully all fields and clicking on **Next**, the ***Success*** window will confirm your registration as in the example below. Click **OK**.



2.6.2 Use existing Asteriskguru account

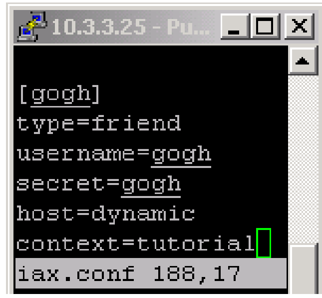
1. Fill in the *Username* and *Password* as in the example below:

2. Click on **Next.**

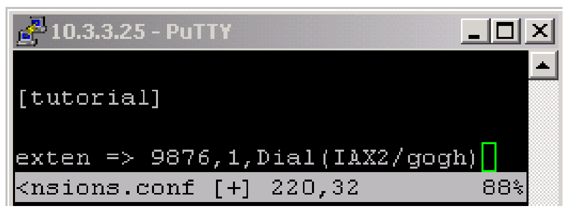
2.7 Creating users in Asterisk®

1. A first thing to do is create users in the iax.conf file. You have to provide values for following fields:

Field	*Description*	*Example*
type	You can choose from 3 different *types:* *friend:* make and receive calls *user:* can only make calls *peer:* can only be called	"friend"
username	Used for authentification.	"gogh"
secret	The password of this user.	"gogh"
host	Dynamic or static.	"dynamic"
context	The environment in which the user can dial himself and be called.	"tutorial"



2. In the extensions.conf file, you link the created users to an *extension* and a *context*. In the example below, the user "Gogh" can call and be called in the context "tutorial" with the Dial application. His extenion is "9876".



3. Reload Asterisk from CLI.

For more information about how to configure the **Asterisk**® configuration files, please read our tutorial about Configuring IP Phones for Asterisk.

3. Calling with IDEFISK

3.1 Selecting an account

- You can select the account you want to call with, by choosing the desired account from the dropdown menu.

- To register an account:
  - Select a user from the list
  - Click on **Register**

- To unregister an account:
  - Select a user from the list.
  - Click on **Unregister**

- All your registered accounts can be dialled to.

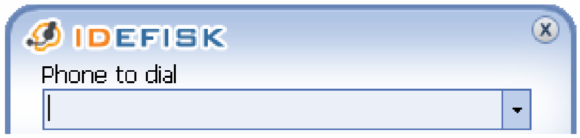
Place the cursor in the *Phone to dial* field and type in the name and the IP address/servername of the callee and click on the **Dial** button. You do not need to have a selected account for direct calling.

3.2 Outgoing calls

3.2.1 Selecting a contact person

There are different ways to select the user you want to call:

- place the cursor in the *Phone to dial* field and type the extension on your keyboard.



- open the Address book. Double click on the person you want to call.

- open the ***DTMF numpad*** by clicking on the small arrow pointing to the left. 

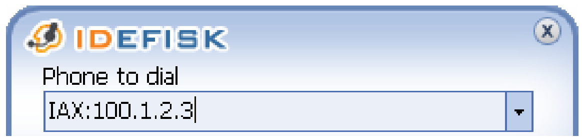
  You can now compose the dial extension with your mouse by clicking on the DTMF buttons.

- enter a dial string in the field *Phone to dial,* as in the example below. The dial string is composed as follows:
  - SIP

  sip: <user>:<password>@<servername>/<extension> @<realm>
  - IAX

  iax: <user>:<password>@<servername>/<extension> @<context>

- Place the cursor in the *Phone to dial* field and type in the name and the IP address/servername of the callee and click on the **Dial/Answer** button. You do not need to have a selected account for direct calling.



- click on the arrow of the *Phone to dial* drop-down list. Here you can find the persons you have called in chronological descending order.

- click on the small arrow pointing to the right The ***Quick dial pad*** opens. If you double click on a contact person, you will start calling him.

Have a look at [this paragraph](#) to find out how to add contact persons to the ***Quick dial pad***.

3.2.2 Selecting a phone line

- You can make use of 6 phone lines for both incoming and outgoing conversations.
- You can easily switch from lines by left clicking on the **Line** button you want to use for the phone call.
- The selected line by default is **Line 1** and its colour is green. When you select any other idle line (e.g any blue line) it becomes selected and green.
- A line in a middle of an established conversation is coloured in yellow. If you select it, its colour changes to green.
- A line, receiving an incoming call becomes pink in colour. Note that an incoming call is also notified with an ***Info Pad***, and an optional pop up window.

3.2.3 Calling a contact person

- If you have selected the person you want to call and which line to use, you can call your contact person by clicking on the **Dial/Answer** button. The **Hang up** button will turn red now.

3.3 Ending a phone call

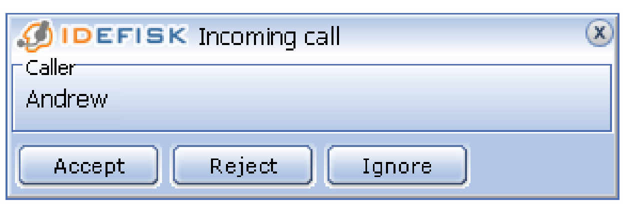
To end a conversation, simply click on the **Hang Up** button 

3.4 Incoming calls

When you have an incoming call, IDEFISK notifies you in the following ways:

- You hear a ringtone in your headphones/speakers.
- An *Info pad* appears on top of IDEFISK. To learn more about the *Info pad*, click here.
- The first idle line will become pink.

- If you want to, you can also choose for a *Pop-up menu* with the *callerID* of the caller, as shown below. With this pop-up window, you have 3 options.

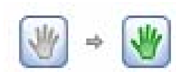


  - You can choose to **Accept** the call. You can also accept a call by clicking on the **Dial/Answer** button.
  - You can choose to **Reject** the call. You can also reject a call by clicking on the **Hangup** button.
  - Clicking on **Ignore** will not end the call. It will just ignore the pop-up screen. The *Info pad* will remain on top of IDEFISK and the ringtone will go on.

Check the Call events options to learn how to receive a pop-up window for incoming calls.

3.5 Hold

- When an incoming or outgoing call becomes accepted, the **Hold** button will become enabled.



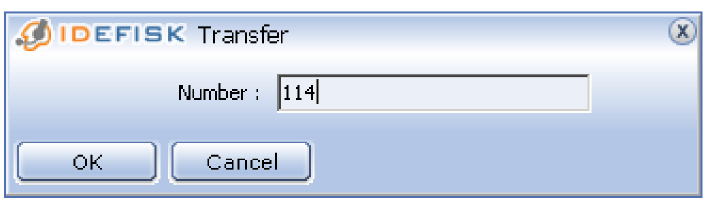
- You can configure Asterisk so that the caller who is put on hold listens to music while waiting on the phone.

3.6 Transferring calls

3.6.1 Transfer

1. When an incoming or outgoing call becomes accepted, the **Transfer** button will become enabled. To transfer a call (no matter incoming or outgoing), click on the **Transfer** button.

2. The ***Transfer*** pop-up window will appear.



2. Use the *Number* field to enter the number to which you would like to transfer the call.

3.6.2 Attended transfer

To make an Attended transfer with IDEFISK:

1. Change the Asterisk configuration file features.conf to the desired DTMF sequence that will be used for Attended transfer.

2. Find in the configuration file the line shown below, to uncomment and set them to the desired values.

- Blind transfer will be set to be executed after you press '#' in a call.
- Attended transfer will be executed on pressing '*'.
- To use the '#' sequence for attended transfer a different sequence should be set for blind transfer sequence.

  - ***Example:***

features.conf

[featuremap]

blindxfer => # ; Blind transfer

*atxfer => * ; Attended transfer*

To transfer calls using DTMF sequences the 't' and 'T' options should be set into the Dial() application parameters in extensions.conf. This allows the called and the caller party to transfer the call by sending the DTMF sequence defined in features.conf.

  - ***Example:***

extensions.conf

----

exten => 555010203,1,Dial(IAX2/some-user,t,)

To make an attended transfer enter the sequence defined in the atxfer field. To finish successfully the attended transfer the transferring side should hang up.

3.7 Missed calls

- If you have a missed call, the **History** button will be flashing red. If you move the mouse over the button, the number of missed calls will be shown.

- To see more details about the missed call in the History, click on the **History** button.

3.8 Conference & Voice mail

Conference and Voice mail are options for IDEFISK BIZ. These functions are not available with IDEFISK FREE. Therefore, both buttons are disabled:  

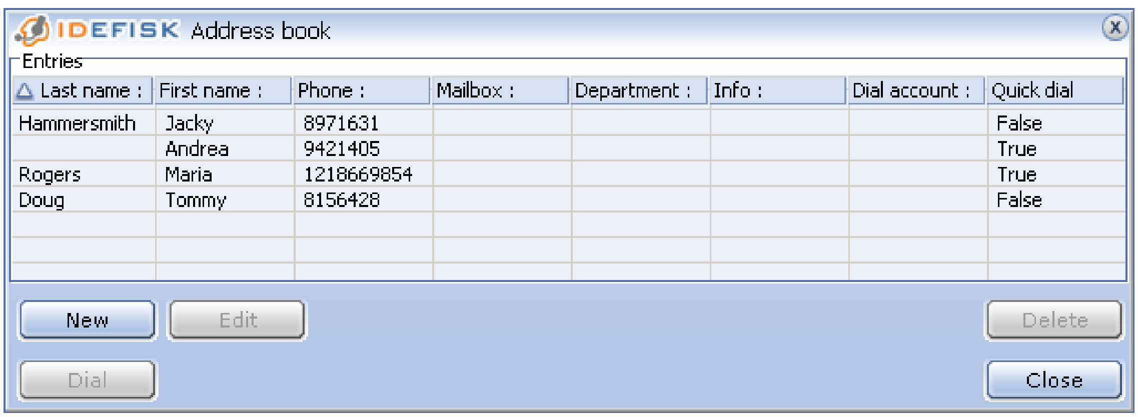
4. Address book

4.1 Accessing the address book

Open your ***Address book*** by clicking on the **Address book** button.

4.2 Contact information

- You can enter useful information about each contact person in 7 different fields: *Last name, First name, Phone, Mailbox, Department, Info* and *Dial account.*



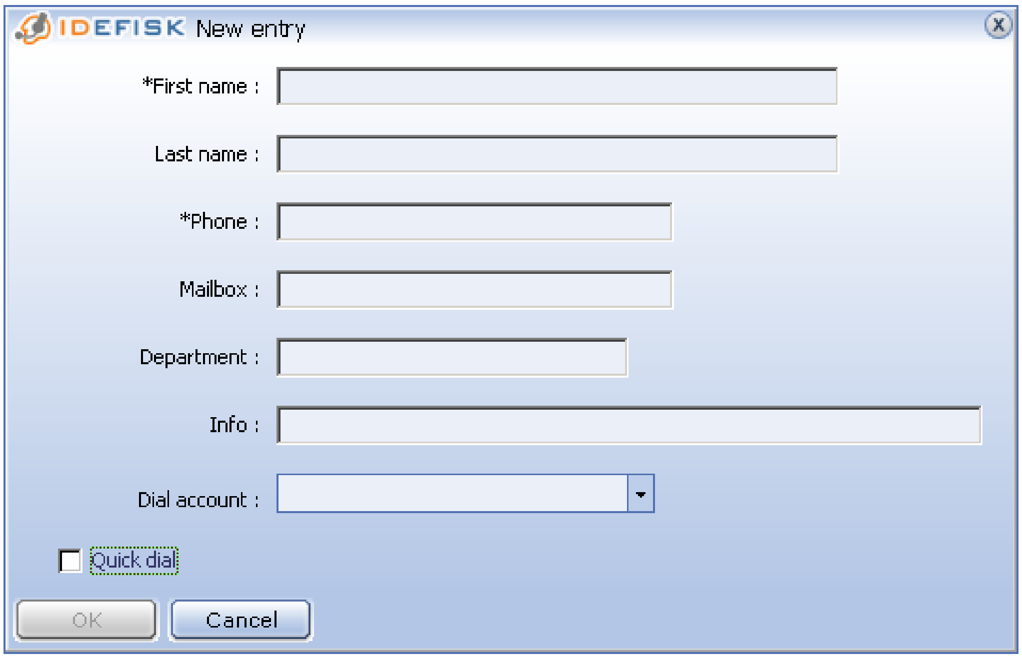
- Provide info for at least *First name* and *Phone.* The other 5 fields can be left empty if you want to.

> To save time, it is best to add the persons you call most often to the ***Quick dial pad***. You can do this by ticking the *Quick dial* check box when you add a new entry to your ***Address book***.

4.3 Adding and deleting entries

4.3.1 New entry

1. Click on the **New** button.
2. Enter information in the ***New entry*** window for at least following fields: **First name* and **Phone extension*. The other fields are added for your convenience and are non-mandatory.



3. Save the entry by clicking on the enabled button **OK**.

4.3.2 Deleting entries

1. Click on the entry you want to delete from your ***Address book***.
2. Click on the **Delete** button.

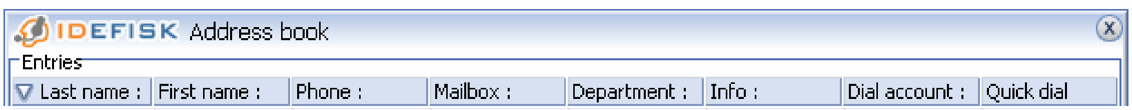
4.4 Editing entries

1. Click on the entry you want to change.
2. Click on the **Edit** button.
3. Edit the information for this entry.
4. Store your changes by clicking on **OK**.

4.5 Sorting entries

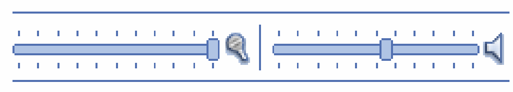
The contact persons can be sorted on the 8 different fields: *Last name, First name, Phone, Mailbox, Department, Info, Dial account* and *Quick dial.*

1. Click on a field and an arrow will appear like in the example below: *Last name*.
2. Click again on the arrow to sort the list in ascending or descending order.



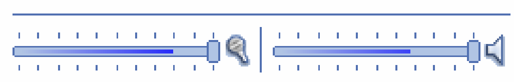
5. Volume control

- You can easily control the incoming and outgoing volume of IDEFISK. Adjust the volume by moving the sliders to the left (less sound) or the right (more volume).



- If you want to disable/enable the incoming/outgoing sound, click on the icon of the **Speaker**, respectively the **Microphone**. The icons will turn red when the sound is mute.

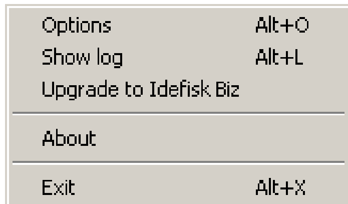
- Also note the Audio level indicators within the sliders. They indicate the outgoing and incoming sound respectively.



6. Log

To open the ***Log***, you must either:

- right-click on **IDEFISK** with your mouse and choose *Show Log* from the popup menu.



- press Alt+L on your keyboard.

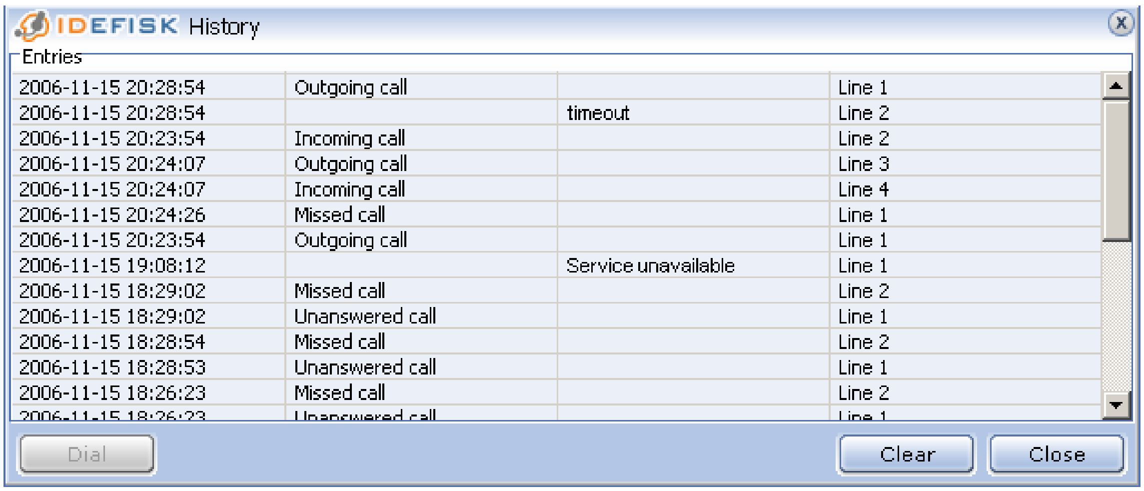
The ***Log*** window contains useful detailed information about the sequence of events, and the time they took place. This information refers to internal **IDEFISK** processes as well, which makes it very useful for more advanced users.

7. History

7.1 Accessing the History

You can open the ***History*** by clicking on the **History** button.

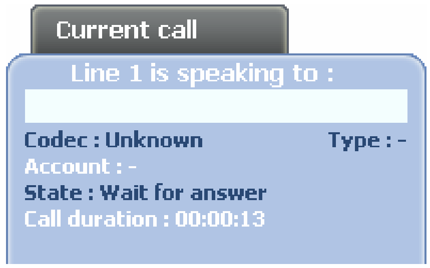
- When in the ***History*** window you click on an entry of a *Missed call*, you can call your contact person by clicking on the enabled **Dial** button. You can also scroll up and down to see the chronological sequence of events.
- You can also choose to **Clear** the content or **Close** the ***History*** window.



- For each phone call the ***History*** contains the following information:
  - Date and time of calling.
  - Status of the call. This is one of the following:
    - missed
    - unanswered
    - new voice mail messages
    - incoming
    - outgoing
  - Name/Number of called/calling person.
  - Phone line used for the call.

7.2 Call information

Every time you dial or receive a call, an *Info pad* pops on top of the **IDEFISK** window.



The *Info pad* is showing information about:

- The callerID of the caller party
- The codec the caller is using
- The type of protocol used for the call
- The account that receives the call
- The state the current line is in. This could be one of the following:
    - Up – when you or the others side picks up the call
    - Down – when you or the other side hangs up the call
    - Ringing – when you or the other side is still ringing
    - Wait for Answer – when you have dialled and wait for the other side to respond
    - Active – the line is in active state when early media is detected
    - Dialling – when you are in a state of dialling before being connected
    - Resolving Port – the line is in a state of resolving 1the port for STUN
- Call duration – the time elapsed since the Info pad appears

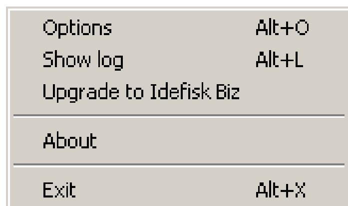
8. Options

8.1 Accessing the options

8.1.1 Options menu

You can access the **Options Menu** of **IDEFISK** in three ways:

- right click on the **IDEFISK** icon in your system tray or on the phone itself and the following pop-up menu appears



- press Alt+O on your keyborad
- click on the options button

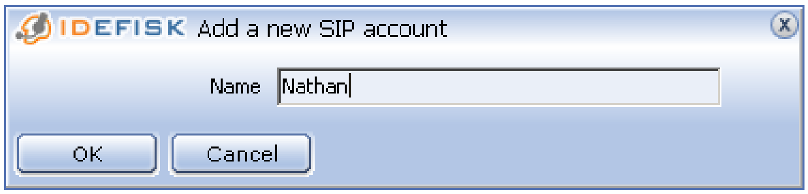
In all cases the following screen will pop up:

Click on the option in the *Option Tree* you want to change the settings for.

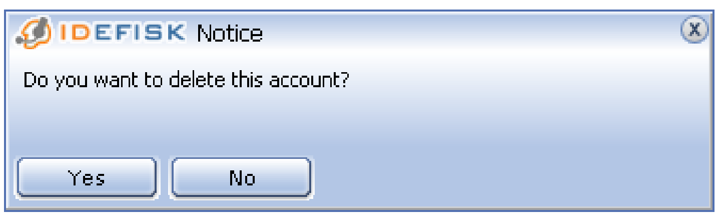
8.2 SIP accounts

8.2.1 Add a new SIP account:

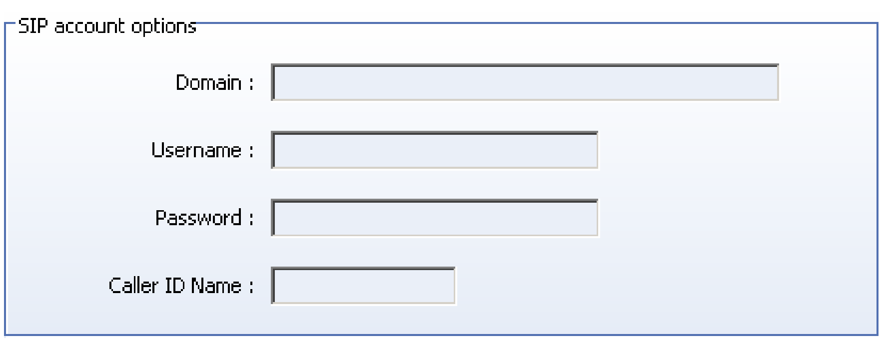
Enter an account name and press the **OK** button.



Each new account is added under the **SIP accounts** options in the *Options Tree*.To delete an existing SIP account, click on the account name in *Options Tree* and press "Del" on your keyboard. A confirmation popup will appear:



After entering an account name, press the **OK** button. Now you can access the **SIP account options**. The ***SIP account options*** look as follows:



Domain

Enter the host name, specified by your VoIP service provider.

Username

Enter the username given to you for registration/authorization.

Password

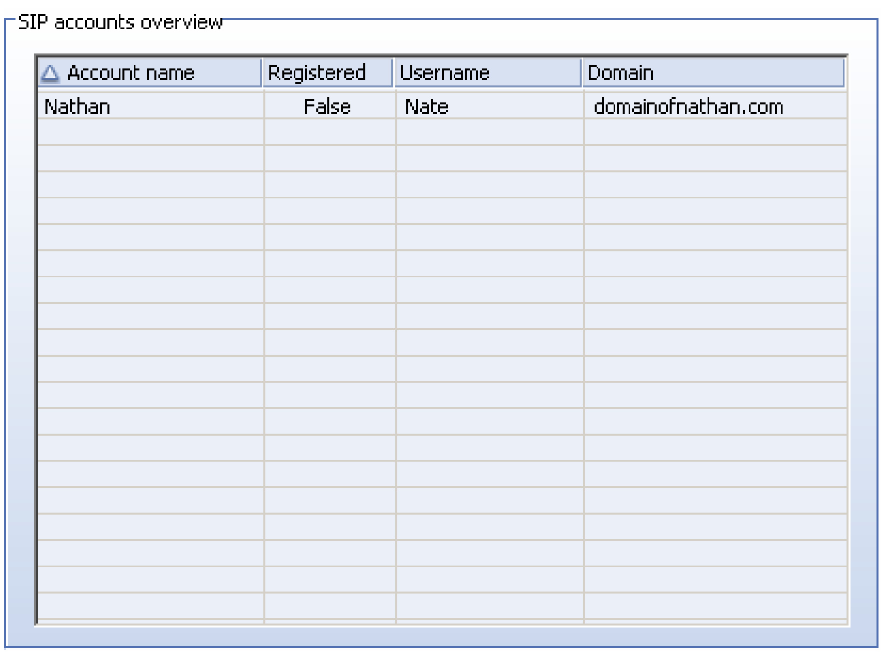
Enter the password given to you for registration/authorization.

Caller ID Name

Enter your Caller ID name. The callee side, if capable, will be seeing this Caller ID name whenever you call.

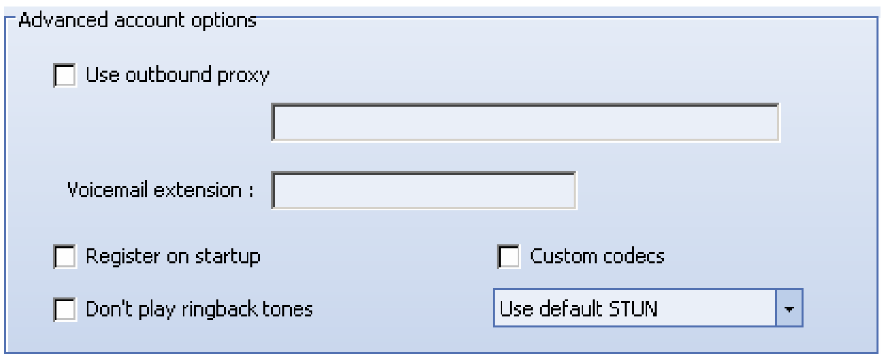
Do not forget to apply all changes by clicking on the **Apply** button.

Please mind that all added ***SIP accounts*** are listed in the ***SIP accounts overview*** table. As clearly visible from the picture below, it displays the *Account name*, *Username* and *Domain*, as well as whether the chosen account is *Registered* (True) or not (False). The SIP accounts overview is accessible by clicking on SIP accounts in the Options Tree. Note that all fields (except for Registered) could be sorted alphabetically or counter-alphabetically. The Registered accounts may be sorted by True or False.



8.2.2 SIP Advanced account options

You can enable/disable the Advanced account options by ticking/unticking the **Advanced options** checkbox at the bottom of the ***Options*** window.



Use outbound proxy

This option is for outgoing calls through a proxy server. Enter the host name or the IP address of the desired outbound proxy in the field below.

Voicemail extension

Enter the extension at which you would check your voicemail messages.

Register on startup

This option is for automatically registering the current account each time **IDEFISK** starts up.

Don't play ringbacktones

Tick this checkbox to mute all ringbacktones.

Custom codecs

You can choose to save the customised selection and order of codecs for this account. The Audio codecs for the chosen account are handled quite like in the general Audio codecs.

The dropdown menu at the bottom reveals the following options:

Use default STUN

Choose the STUN server, set in the Protocol options. To set the default STUN server, go to the STUN options in the Advanced Protocol options.

Use Custom STUN

Choose a custom STUN server. The **STUN options** for the chosen account are handled quite like in the general STUN options.

Don't use STUN

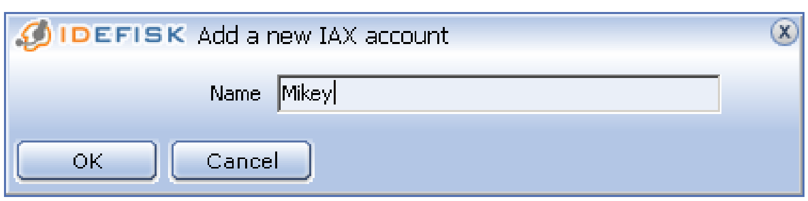
This option is for the case in which you do not need a STUN server for this account.

Do not forget to apply all changes by clicking on the **Apply** button. After applying the changes, the **Audio codecs** options and the **STUN options** for the current account appear underneath its name in the *Options Tree*.

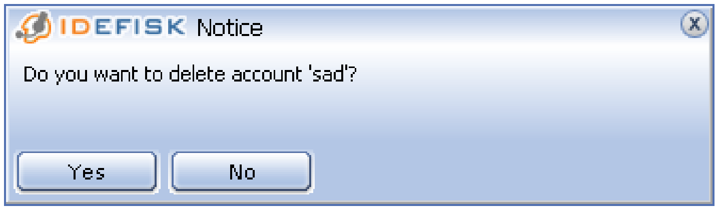
8.3 IAX accounts

8.3.1 Add a new IAX account

Enter an account name and press the **OK** button.

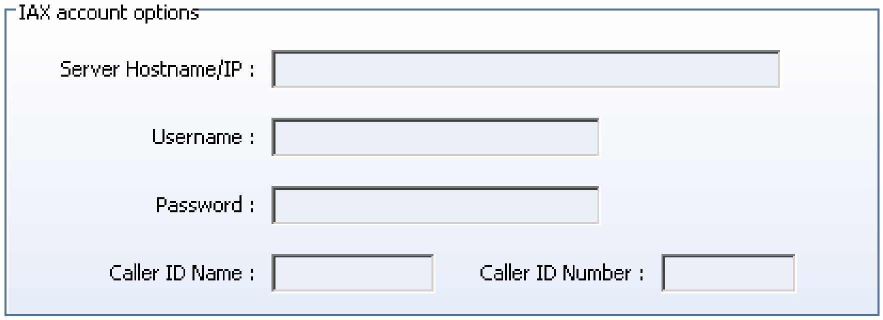


> Each new account is added under the **SIP accounts** options in the *Options Tree*.To delete an existing SIP account, click on the account name in *Options Tree* and press "Del" on your keyboard. A confirmation popup will appear:



After entering an account name, press the **OK** button. Now you can access the **IAX account options**.

The ***IAX account options*** look as follows:



Server Hostname/IP

Enter the IP address of your VoIP PBX or the IP address given to you by your VoIP Service Provider.

Username

Enter the username given to you for registration/authorization.

Password

Enter the password given to you for registration/authorization.

Caller ID Name

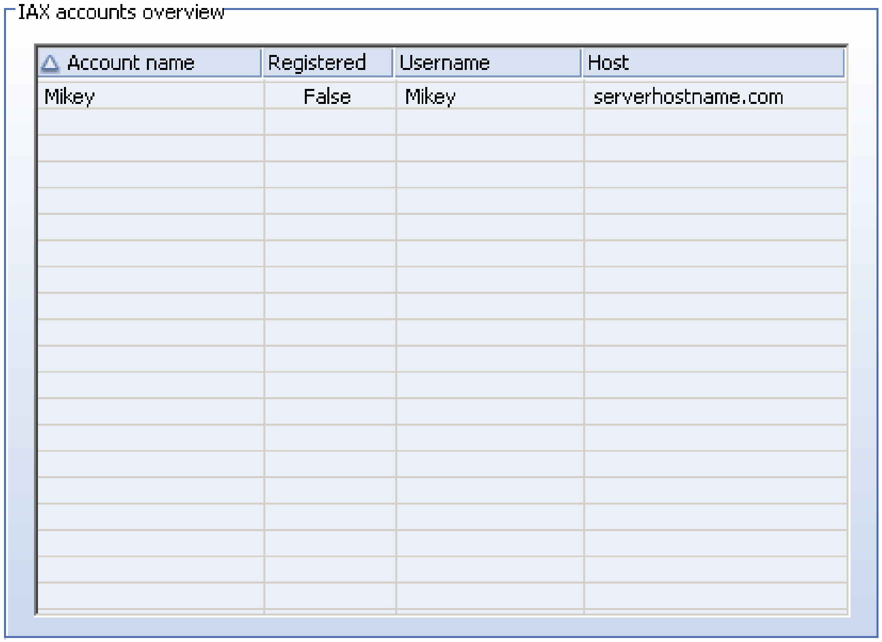
Enter your Caller ID name. The callee side, if capable, will be seeing this Caller ID name whenever you call.

Caller ID Number

Enter your Caller ID number. The callee side, if capable, will be seeing this Caller ID number whenever you call.

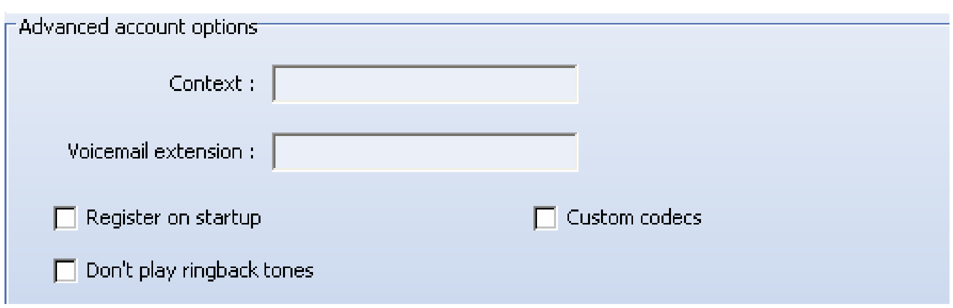
Do not forget to apply all changes by clicking on the **Apply** button.

Please mind that all added ***IAX accounts*** are listed in the ***IAX accounts overview*** table. As clearly visible from the picture below, it displays the *Account name*, *Username* and *Host*, as well as whether the chosen account is *Registered* (True) or not (False). The SIP accounts overview is accessible by clicking on SIP accounts in the Options Tree. Note that all fields (except for Registered) could be sorted alphabetically or counter-alphabetically. The Registered accounts may be sorted by True or False.



<u>8.3.2 IAX Advanced account options</u>

You can enable/disable the ***Advanced account options*** by ticking/unticking the **Advanced options** checkbox at the bottom of the ***Options*** window.



Context

Contexts play an organizational role within an Asterisk dialplan and also define scope. You can view contexts as a way to keep different parts of the dialplan separate. This comes handy for providing different reception destinations for different companies that share the same Asterisk server. Any call that Asterisk handles will begin in a certain context. The instructions defined in this context will determine what things may happen to the call. With this option you can change the context at which your IAX account is working.

Voicemail extension

Enter the extension at which you would like to check for new voicemail messages.

Register on startup

This option is for automatically registering the current account each time **IDEFISK** starts up.

Don't play ringbacktones

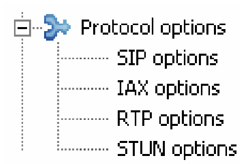
Tick this checkbox to mute all ringbacktones.

Custom codecs: You can choose to save the customised selection and order of codecs for this account. The Audio codecs for the chosen account are handled quite like in the general Audio codecs.

Do not forget to apply all changes by clicking on the **Apply** button. After applying the changes, the **Audio codecs** options the current account appear underneath its name in the *Options Tree*.

8.4 Protocol options

You can enable/disable the Advanced Protocol options by ticking/unticking the **Advanced options** checkbox at the bottom of the ***Options*** window.



8.4.1 SIP options

Port

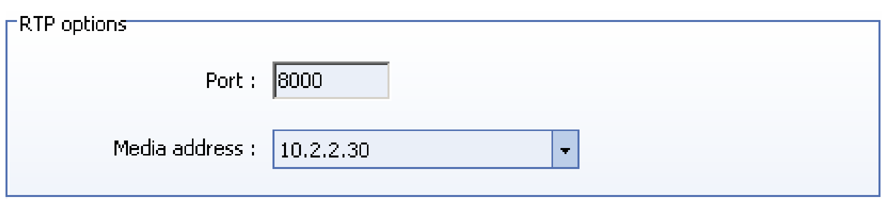
You can change the default port that SIP is using. The default port for SIP is 5060 as shown in the example above.

8.4.2 IAX options

Port:

You can change the default port that IAX is using. The default port for IAX is 4569 as shown in the example above.

8.4.3 RTP options



Port

You can change the default port that RTP is using. The port number could range from 8000 (default) to 8100.

Media address

The **Media address** is negotiated by the SIP in order for the RTP to follow the correct address. The default **Media address** is the external IP of your network.

On startup IDEFISK tries to select external IP. In case no external IP is present, IDEFISK selects the internal IP.

Do not forget to apply all changes by clicking on the **Apply** button.

8.4.3.1 Advanced RTP options

You can enable/disable the ***Advanced RTP options*** by ticking/unticking the **Advanced options** checkbox at the bottom of the ***Options*** window.

Session name

Enter a **Session name** for all the RTP sessions.

User name

Enter your preferable **User name**.

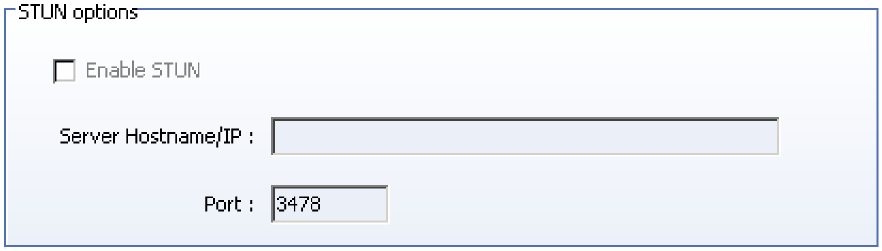
URL

Enter your **URL**.

E-mail

Enter your **E-mail address**.

8.4.4 STUN Options



Enable STUN

Tick this checkbox if you have a STUN server.

Server Hostname/IP

Enter the IP address of your STUN server or the IP address given to you by your VoIP Service Provider.

Port

You can change the default port that STUN is using. The default port for STUN is 3478 as shown in the example above.

8.4.4.1 Advanced STUN options

You can enable/disable the ***Advanced RTP options*** by ticking/unticking the **Advanced options** checkbox at the bottom of the ***Options*** window.

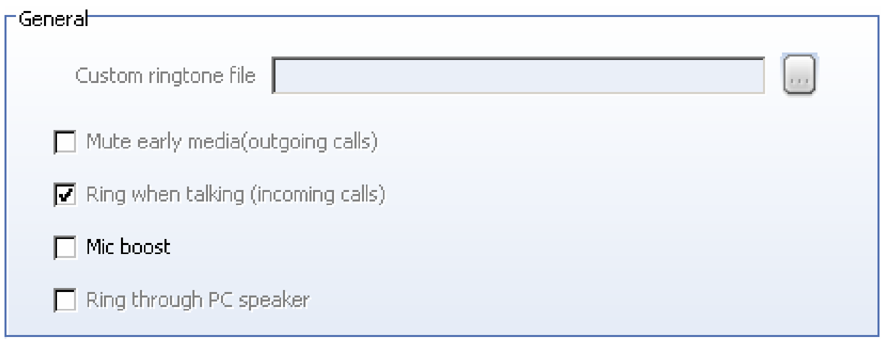
Refresh period

You can change the refresh period (in seconds) for the STUN server. The initially set value is 30 seconds as in the example above.

8.5 Audio options

8.5.1 General

You can adjust the **General audio options**. Always confirm your settings by clicking **Apply**.



Custom ringtone file

You can browse to a preferred ring tone or enter the file path. A ring tone file must be an 8 kHz 16-bit Mono wave file (.wav). Available in **IDEFISK BIZ** only.

Mute early media(outgoing calls)

You can choose to enable/disable hearing ringtone when you dial a call. Also blocks early media. Available in **IDEFISK BIZ** only.

Ring when talking (incoming calls)

You can choose to enable/disable the ringtone, while in a middle of a conversation. The option is available in **IDEFISK BIZ** only and is enabled in **IDEFISK FREE**.

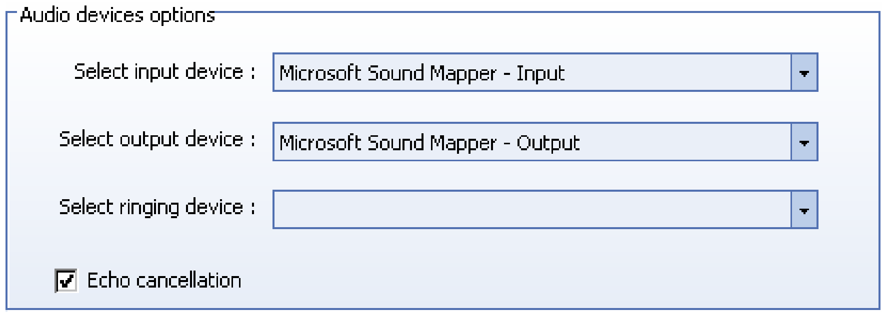
Mic boost

To turn up the volume of your microphone, tick this option. Please mind that this might negatively affect sound quality.

Ring through PC speaker

Tick this option if you want to use a beep from your PC speaker. Please mind that this does not mute the ringing through your headphones or speakers. Available in IDEFISK BIZ only.

8.5.2. Audio devices



Select input/output/ringing device

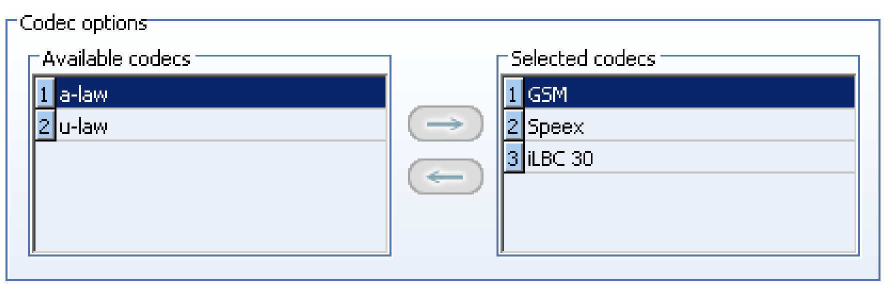
You can select your audio input/output/ringing device(headphones) from each corresponding dropdown menu. Please mind that the drivers for these devices' must be properly installed and recognized by Windows.

Echo cancellation

Tick this option in case of an echo tail to the speech.

8.5.3. Audio codecs

The **Codec options** look as follows:



You can choose among the following codecs:

- GSM
- Raw µ-law (G.711)
- Raw A-Law data (G.711)

- Speex Audio
- iLBC

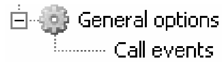
These are the default settings for the audio codecs. They will be used by all accounts unless the custom codecs per account are used.

If you want to use any of the *Available codecs* you have to select it and then press the right direction arrow.

If you want to use stop using any of the *Selected codecs* you have to select it and then press the left direction arrow.

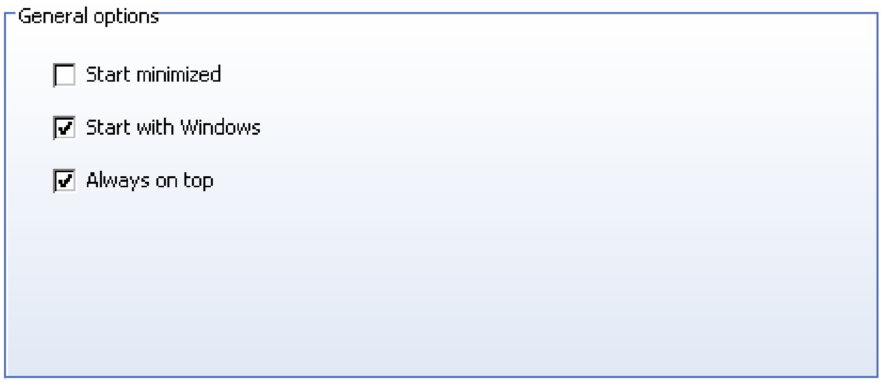
Arrange the codec priority by dragging the blue numbers of the Selected codecs up and down. The codec with the lowest number has the highest priority.

8.6 General options



8.6.1 General

The window of the ***General options*** look as follows:



Start minimized

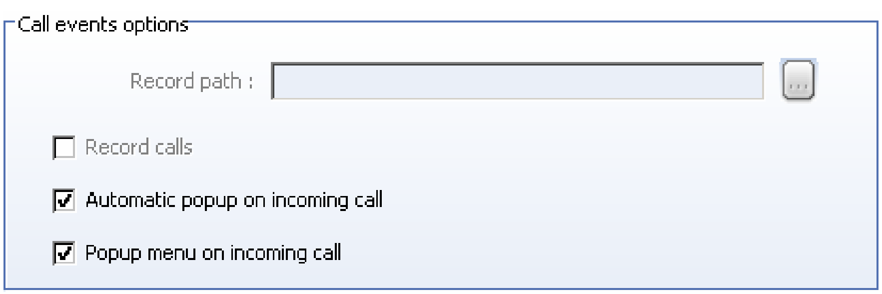
Tick this option in order for **IDEFISK** to start up minimised in your system tray.

Start with Windows

Tick this option in order for **IDEFISK** to automatically start up when opening *Windows*.

8.6.2 Call events

The **Call events** options look as follows:



Record path

Here you can choose the directory in which you would like the calls to be saved. The calls are saved through the *Record calls* option in 16 bit 8 kHz wave files (.wav). Please mind that you need free disk space in order for **IDEFISK** to record all your calls.

Record calls

To record all the phone calls made with **IDEFISK**, tick the check box for this option.

Automatic popup on incoming call

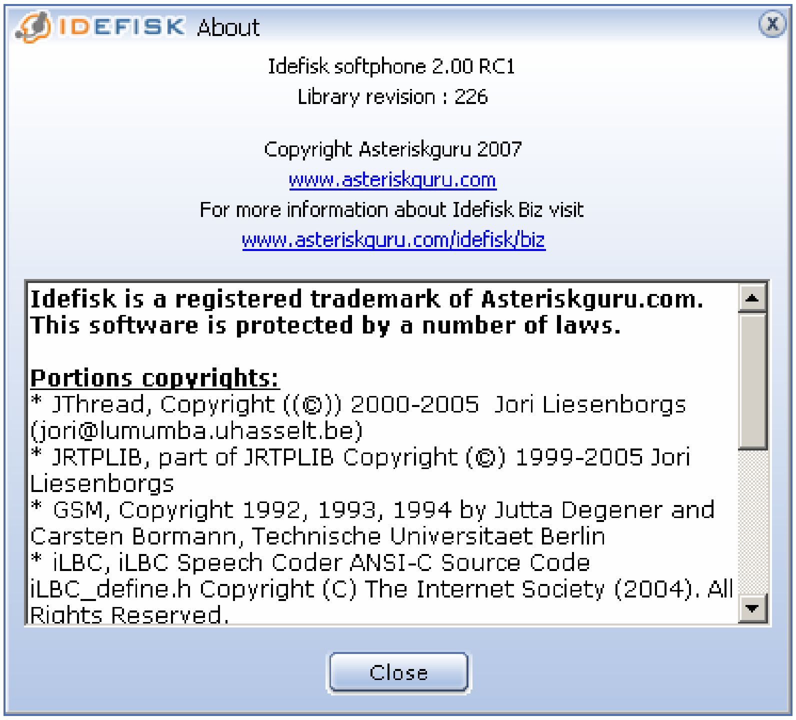
Choose this option if you want **IDEFISK** to pop up when there is an incoming call.

Popup menu on incoming call

When this option is checked, the following pop-up window appears every time you have an incoming call:

8.7 About

On the screen ***About*** you can find more information about the version of **IDEFISK** installed on your computer. To access it, right-click on **IDEFISK** and choose he **About** option at the bottom.



8.8 Upgrading to IDEFISK BIZ

If you choose to switch to **IDEFISK BIZ**, simply right-click on **IDEFISK** for the pop-up to appear and choose [Upgrading to IDEFISK BIZ].