Acoustica 4

User Guide

Acon Digital Media GmbH

Acoustica User Guide

Copyright © 2003-2006 Acon Digital Media GmbH

Table of Contents

1   Introduction

Acoustica 4 is an ideal solution for audio editing and mastering. The program contains everything you need to create great sounding recordings and audio CDs, including professional tools for recording, analysis, editing and CD burning. The Acoustica user interface was designed with speed, accuracy and ease of use in mind. The support for audio resolutions up to 32 bit and sampling rates up to 192 kHz allows you to record and edit in an amazing audio quality.

A large range of high quality audio tools and effects are already integrated in Acoustica — including tools for dynamic processing, equalizing, numerous effects such as reverb, chorus and flanger, as well time stretching and key transposition tools. Recordings distorted by noise, clicks, crackle, clipping or missing high frequency content can be restored. The support for DirectX and VST plug-ins allows you to use tools and effects from other third party manufacturers directly from Acoustica.

You can monitor the effect of your editing steps visually using the realtime analysis tools such as the FFT-Analysis (frequency spectrum) or the phase correlation meter. The effect chain editor simplifies mastering by allowing you to chain internal tools and effects as well as DirectX or VST plug-ins and store the chain including all the effect settings for later use.

1.1   What is new in Acoustica 4

Acoustica 4 comes with a brand new user interface with customizable menus and toolbars. The greatly improved Cleaning Wizard now allows more detailed editing and makes it even easier to transfer old recordings to CDs. With the integrated digital emulation of a phono preamplifier, there is no longer a need for an external phono preamplifier. The record player can be connected directly to the sound card of your computer.

New Features
- Runs under Windows Vista
- Supports VST effects.
- Real-time analyzers lets you analyze the output audio like a professional, including lever meter with RMS, peak and peak hold, FFT analyzer, phase correlation meter and a big time display.
- The new user interface supports customizable toolbars, menus and window positioning.
- New declipper restores audio recordings that suffer from digital or analog clipping.
- The declicker includes a new decrackle option that eliminates short but frequent clicks (crackle) more efficiently.
- Integrated phono preamplifier emulator eliminates the need for an external phono

preamplifier when connecting record players to the sound card

- Greatly improved Cleaning Wizard guides you through all the steps from connection of the stereo equipment and recording to restoration and CD burning.

1.2   Requirements

Before you install Acoustica, please make sure your computer fulfills the following requirements:

- A Pentium or compatible processor with at clock rate of 800 MHz or more
- Minimum 256 MB RAM
- Minimum 16 bit color depth
- A Windows compatible sound card

In addition, you will need the following software installed on your computer:

- Windows 2000, Windows XP or Windows Vista
- DirectX 8.0 or later

2   Basic Audio Editing

This chapter describes the basic audio editing capabilities of Acoustica such as loading and saving files and editing using the clipboard or drag and drop.

2.1   The Acoustica Workspace

The Acoustica workspace can contain several audio editing windows, CD projects or Cleaning Wizard projects.
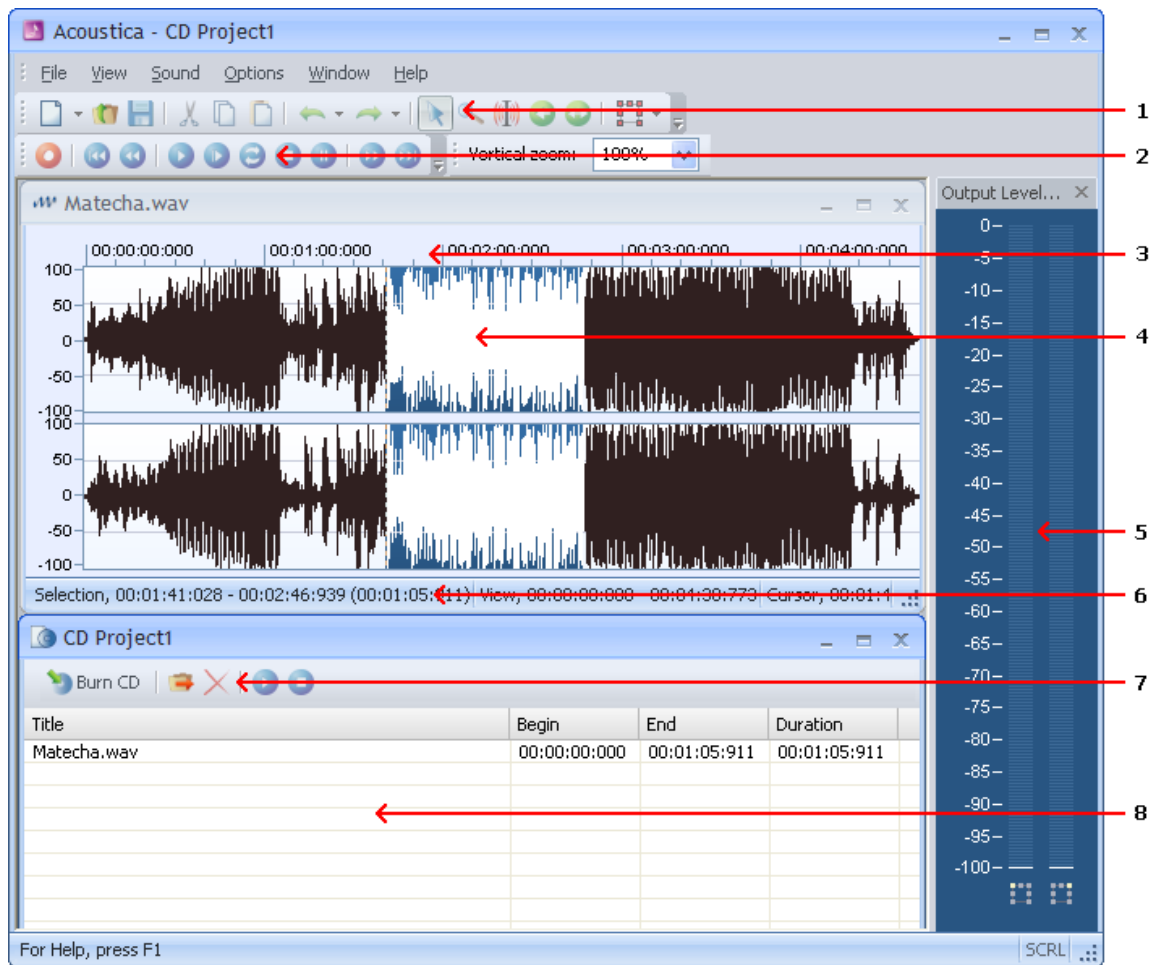
The Acoustica workspace

The figure above shows the Acoustica workspace with an audio editing window and a CD project. The elements indicated with the red arrows are explained below:

1. The main toolbar with short cut icons for commonly used commands.
2. The navigation toolbar for recording, playback and cursor positioning.
3. An audio editing window containing the audio file "Matecha.wav". The audio is visualized with a curve corresponding to the recorded audio (see Time Domain).
4. A selected region indicated with white curve on a blue background (colors depend on your system settings). The current cursor position is indicated by a blinking line.
5. The output level meters show the current output level during audio playback.
6. The status bar indicating the selected region, the viewed region and the cursor position. During processing, the progress is displayed in the status bar.
7. A "CD project" window containing a list of tracks and a toolbar for commands relevant to CD recording.
8. The list of CD tracks that are about to be recorded to an audio CD.

2.2   Loading Audio Files

To open existing audio file,

1. Select from the *File* menu the command *Open...*
2. Choose the folder in which your file is located from the *Look in* drop-down list.
3. Click the audio file you wish to open and click the button labelled *Ok.*

2.3   Saving Audio Files

To save an edited audio file with its original name:

1. Select from the *File* menu the command *Save* or press Ctrl+S.

If you wish to save the content of an audio editing window with a different name, in a different folder or with different settings:

1. Select from the *File* menu the command *Save as...*
2. Choose the folder in which you wish to save the file from the *Look in* drop-down list.
3. Enter a name for your audio file.
4. Choose the file format of your audio file from the *File type* drop-down list.
5. Most export filters offer different settings such as encoding bit rate or number format. To change the settings, click the button labelled *Options... .*

2.4   Playing a Recording

To play the selected region of a recording:

1. Select from the *Sound* menu the command *Play* or press the space bar.
2. You can stop the playback by selecting from the *Sound* menu the command *Stop* or by pressing the space bar.

You can also play the complete recording:

1. Select from the *Sound* menu the command *Play all* or hold the shift key while pressing the space bar.
2. You can stop the playback by selecting from the *Sound* menu the command *Stop* or by pressing the space bar.

To play the selected region as a loop:

1. Select from the *Sound* menu the command *Play looped* or hold the control key while pressing the space bar.
2. You can stop the playback by selecting from the *Sound* menu the command *Stop* or by pressing the space bar.

Alternatively, you can use the navigation toolbar for convenient playback:

The navigation toolbar with the logic similar to consumer audio equipment such as CD players or tape decks.

2.5   Selecting Regions

Acoustica does all processing on the selected region and the selected channels only. The selected region is highlighted. To change the selected region:

1. Click the beginning of the region you wish to select and keep the mouse button down.
2. Move the mouse cursor to the end of the region you wish to select while keeping the mouse button down.
3. Release the mouse button.
4. The newly selected region should now by highlighted.

2.6   Selecting Channels

Acoustica does all processing only on the selected region and the selected channels. To change the selected channel or channels (only possible with stereo recordings):
1. Click the channel selection button in the main toolbar (see <u>The Acoustica Workspace</u>):
2. A drop-down list with the channels appears.
3. Click the channel you wish to activate or deactivate.

2.7   Zooming and Scrolling

You can zoom in to get a more detailed view in an editing window. You can zoom either horizontally to view a smaller part of the recording or vertically to show a smaller amplitude range.

Horizantal zoom
To zoom in horizontally:

- Press the arrow up key

- or -

- Move the mouse wheel upwards

To zoom out for a better overview:

- Press the arrow down key

- or -

- Move the mouse wheel downwards

Note	You can zoom out to view the whole recording by selecting View | View all or by pressing Shift + A.

Scrolling

In those cases where only a part of the recording is shown, a scroll bar is visible below the visualisation of the wave form. You can use the scroll bar to view other parts of the recording.

Vertical zoom

To zoom in vertically:

- Press and hold the control key while pressing the arrow up key

To zoom out vertically:

- Press and hold the control key while pressing the arrow down key

You can also enter the zoom factor in the toolbar and press the enter key:

The vertical zoom bar

2.8   Drag and Drop Editing

You can move or copy regions using so-called drag and drop editing. To move a region to a another location in the same editing window or to another editing window:

1. Select the region you wish to move (see <u>Selecting Regions</u>).

2. Click the somewhere within the highlighted region and keep the mouse button down.
3. Press the *Shift* key while moving the mouse cursor to the new start position and release the mouse button.

To insert a copy of a region using drag and drop:

1. Select the region you wish to copy (see <u>Selecting Regions</u>).
2. Click the somewhere within the highlighted region and keep the mouse button down.
3. Press the *Ctrl* key while moving the mouse cursor to the insert position and release the mouse button.

To mix two regions using drag and drop:

1. Select the first of the regions you wish to mix (see <u>Selecting Regions</u>).
2. Click the somewhere within the highlighted region and keep the mouse button down.
3. Press the *Alt* key while moving the mouse cursor to the start position and release the mouse button.

> **Note**   If you press neither the *Ctrl* key nor the *Shift* key during drag and drop, Acoustica will perform the default action which is a move operation if the drop destination is the same editing window and a copy operation if the drop destination is another editing window.

2.9   Editing using the Clipboard

The clipboard concept provides a common way of editing documents for all Windows applications. You can edit your recordings by copying the selected region to the Acoustica clipboard using the command *Edit | Copy* and paste the region into another location using the command *Edit | Paste Insert*. The *Paste Insert* command is equivalent to the normal paste command common in most Window applications. Acoustica has offers two additional ways of pasting:

- Paste Overwrite, which substitutes a selected region with the content of the clipboard.
- Paste Mix, which mixes the selected region with the content of the clipboard.

The command *Edit | Cut* copies the selected region to the clipboard before deleting it from the source recording. For the purpose of deleting parts of the recording, use one of

the following two commands:

- Delete, which deletes the selected region
- Crop, which deletes everything but the selected region.

2.10   Audio Scrubbing

Sometimes it is hard to find a specific part of a recording using the visualization only. The audio scrub mode simplifies the search by giving aural feedback while you can move the playback position using the mouse. To use the audio scrubbing:

1. Enable the audio scrubbing mode clicking the audio scrubbing symbol in the toolbar ( ) or by pressing A.
2. Click the left mouse button and keep it down while moving the mouse cursor
3. Release the mouse button when you have found the part you were searching for.
4. Enable the selection mode by clicking the selection mode symbol in the toolbar ( ) or by pressing S.

2.11   Labels and Regions

You can simplify the editing process by inserting anchors such as labels and regions to your recording.

Adding Labels
To add a label to the recording:

1. Move the cursor position to where you want the to insert a label
2. Click the right mouse button and select "Add Label..." or press L.
3. A label appears at the cursor position

Adding Region Markers
To add a region marker to the recording:

1. Select the part of the recording where you want a region marker
2. Click the right mouse button and select "Add Region..." or press R.
3. A region is appears at the selection

Renaming, Moving and Deleting Anchors
You can easily rename anchors:

1. Double click the anchor text
2. A properties dialog box appears.
3. Enter the new name of the anchor.
4. Click the button labelled OK.

You can also move the anchors:

1. Click the left mouse button at the upper part of red line visualizing the anchor.
2. Keep the mouse button down while moving the anchor to its new position
3. Release the mouse button.

To delete an anchor:

1. Click the right mouse button at the anchor text
2. A context menu appears.
3. Select "Delete"

The Label and Region List Windows
You can edit and keep track of your anchors by showing the region list and label list windows. To show the label list window, select "Label List" from the View menu. The menu item "Region List" from the same menu shows the region list.

Label List		×	Region List			×
Name	Position		Name	Begin	End	
Hit	00:02:21:120		Intro	00:00:00:000	00:00:27:157	
Ending	00:02:51:671		Part A	00:00:27:157	00:01:05:225	
			Chorus	00:01:05:225	00:02:00:267	

The Region and Label List windows make it easy to edit and keep track of the anchors.

2.12   Using Analyzers

The integrated analyzers allow you to analyze the output audio signal in real time during playback. You can hide or show the analyzers by choosing *View | Analyzers* and selecting one of the analyzers from the sub menu.

2.12.1   Level Meter

The *level meter* lets you analyze the output level in terms of peak, peak hold and RMS values. The peak value is the maximum sample within a short analysis interval and is the value defining the height of the level meter bars. The peak hold value is the

maximum sample level over a longer period of time. It is indicated as a white line above or at the top of the level meter bar. RMS stands for root-mean-square and is calculated by the root of the sum of the squared sample values during the analysis interval.
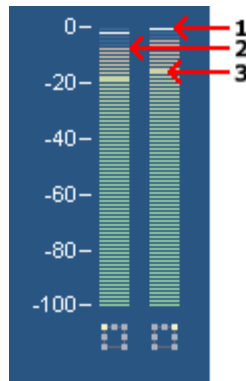
The level meter analyzer showing the peak hold value (1), peak value (2) and RMS value (3).

2.12.2   FFT Analyzer

The *FFT Analyzer* shows the frequency content of short analysis time frames. FFT stands for Fast Fourier Transform. FFT is an efficient way of calculating the frequency domain of a signal. For more information about the frequency domain, please see Frequency Domain.
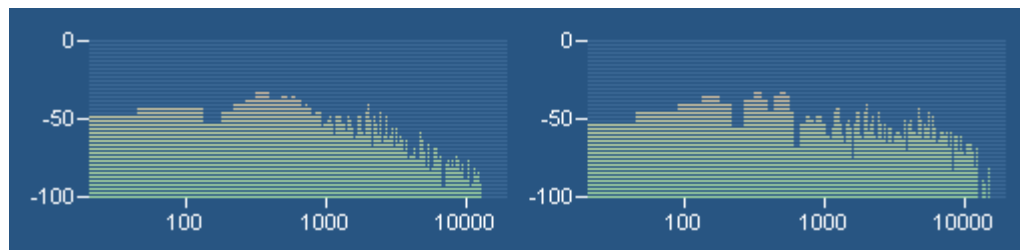
The FFT Analyzer shows the frequency content of the output audio signal.

2.12.3   Phase Correlation Meter

The *phase correlation meter* shows the phase relationship between the left and the right audio channel in a stereo recording and is an important tool when mastering stereo recordings. If both channels contain exactly the same signal, the phase correlation meter will show a vertical line. If one channel is exactly the opposite of the other channel, the phase correlation meter shows a horizontal line. Normal stereo recordings will show a cloud of dots spread out vertically and horizontally (see the picture below). In a properly mastered recording, the cloud of dots should not be wider than it is tall.
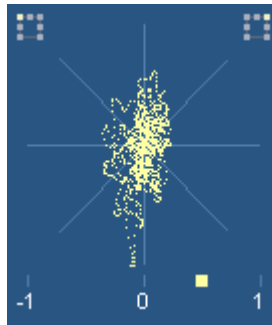
The phase correlation meter shows the relationship between the left and the right channel in a stereo recording.

2.12.4   Big Time Display

The big time display shows the current playback position in a resizable and dockable window.

3   Connecting Your Stereo Equipment

This chapter describes how to connect your tape deck (cassette tape player) or your record player to your computer.
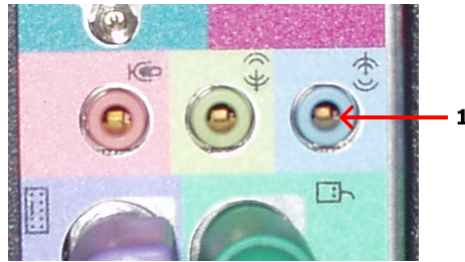
3.1   Connecting a Tape Deck (Cassette Tape)

If you have an external tape deck or a compact stereo system, it will probably have so called "RCA connectors" on the backside labelled *Line Out*, *Tape Out* or *Tape Rec*, as depicted below:
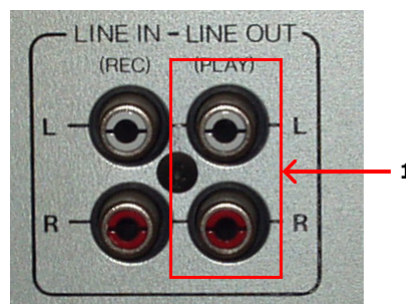
RCA connectors on a tape deck unit. Connect the sound card to the Line Out connector (1).

The *Line Out* of the tape deck or stereo system will now need to be connected to the computer. Integrated sound cards are usually equipped with "mini jack" connectors (also called 3,5 mm jack).

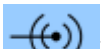*The back pane of a computer with an integrated sound card and a line in mini jack connector (1).*

Connecting to a Sound Card with Mini Jack Connector

If you computer has mini jack connectors, a connection cable with a mini jack connector on one end and RCA connectors on the other end is required to connect your tape deck or stereo system to your computer.
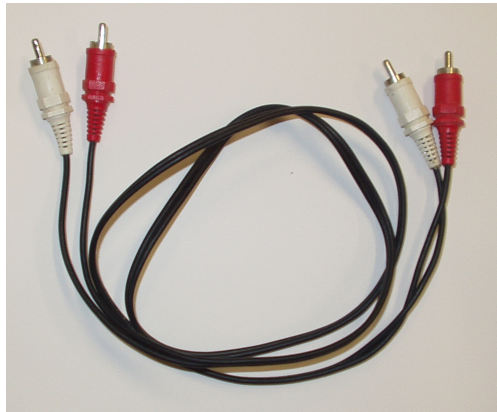

A connection cable with RCA connectors on one end and a mini jack connector on the other.

If you didn't receive such a cable with your sound card or computer, you can purchase one in normal audio or electronics stores.

To connect your tape deck or stereo system, connect the end with the RCA connectors to the *Line Out*, *Tape Out* or *Tape Rec* connectors on your tape deck or stereo system. Connect the other end to the *Line In* connector of your computer. The *Line In* connector is either labelled as such or indicated with the following symbol: .

Connecting to a Sound Card or USB Audio Device with RCA connectors

External USB devices for audio input and output are growing in popularity and many of these have normal *Line In* RCA connectors. In rare cases, internal sound cards might also have RCA connectors. In these cases, a connection cable with RCA connectors on both ends is required.

A connection cable with RCA connectors on both ends.

You can purchase RCA connection cables in normal audio or electronics stores if you don't have one already. In this case, all you have to do is connect the cable to the *Line Out*, *Tape Out* or *Tape Rec* connectors on your tape deck or stereo system and the *Line In* connectors of the sound card.

3.2   Connecting a Record Player (LP)

If you have a stereo system with an amplifier already set-up, the easiest way to connect a record player to your computer is to connect the computer to the amplifier's *Line Out* connectors (sometimes also labelled *Tape Out, Tape Rec* or *Rec Out*).
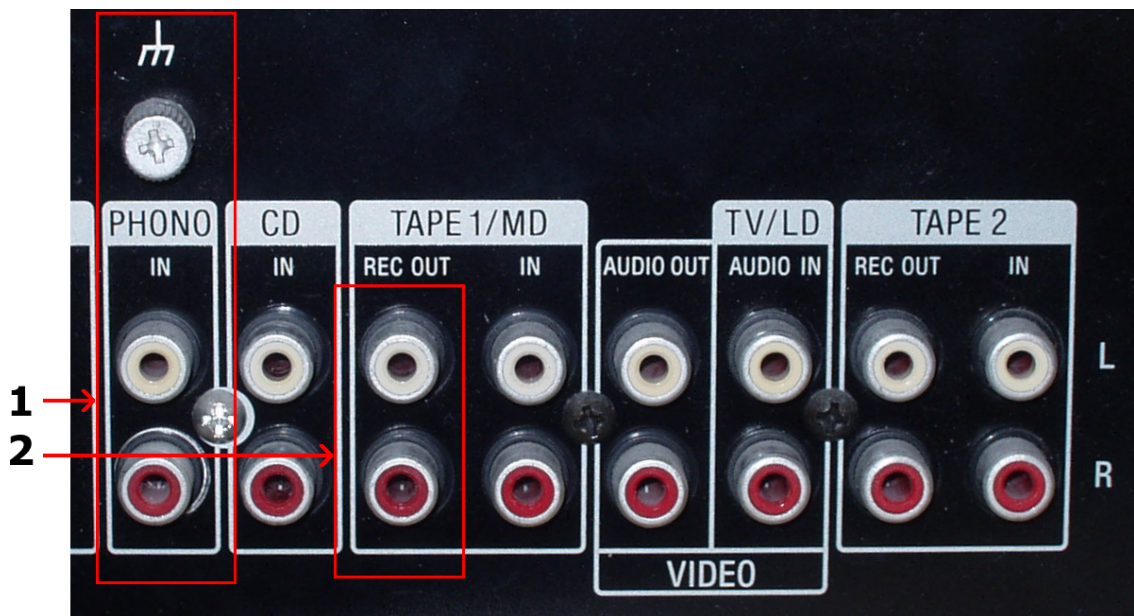
RCA connectors on an amplifier. Make sure the record player is properly connected to the phono input (1) and connect the sound card to the Rec Out connector (2).

3.2.1   Connecting a Record Player through an Amplifier

Before you proceed, please make sure your record player is properly connected to your amplifier and that you can listen to records on your stereo system.

The *Line Out* of the amplifier or stereo system will now need to be connected to the computer. Integrated sound cards are usually equipped with "mini jack" connectors (also called 3,5 mm jack).

The back pane of a computer with an integrated sound card and a line in mini jack connector (1).

Connecting to a Sound Card with Mini Jack Connector
If you computer has mini jack connectors, a connection cable with a mini jack connector on one end and RCA connectors on the other end is required to connect your amplifier or stereo system to your computer.

A connection cable with RCA connectors on one end and a mini jack connector on the other.

If you didn't receive such a cable with your sound card or computer, you can purchase one in normal audio or electronics stores.

To connect your amplifier or stereo system, connect the end with the RCA connectors to the *Line Out*, *Tape Out* or *Tape Rec* connectors on your amplifier. Connect the other end to the *Line In* connector of your computer. The *Line In* connector is either labelled as such or indicated with the following symbol: .

Connecting to a Sound Card or USB Audio Device with RCA connectors
External USB devices for audio input and output are growing in popularity and many of these have normal *Line In* RCA connectors. In rare cases, internal sound cards might also have RCA connectors. In these cases, a connection cable with RCA connectors on both ends is required.

A connection cable with RCA connectors on both ends.

You can purchase RCA connection cables in normal audio or electronics stores if you don't have one already. An In this case, all you have to do is connect the cable to the *Line Out*, *Tape Out* or *Tape Rec* connectors on your amplifier or stereo system and the *Line In* connectors of the USB audio device.

3.2.2   Connecting a Record Player Directly

It is usually not possible to connect a record player directly to the computer, because the audio on LP records is modified for technical reasons so that low frequencies (bass) are too soft and high frequencies (treble) too loud. Amplifiers with a phono input modify the signal using an exactly tuned equalizer so that the original signal is restored. It is also possible to purchase hardware units solely for this purpose. These are usually called phono preamplifiers.

The good news is that you with Acoustica 4 probably don't need any additional hardware. Acoustica 4 emulates the equalizer in phono preamplifiers and thus restores the original audio. You can connect the record player directly to the *Line In* on your sound card and enable the *Emulate phono preamplifier* option in the recording page (see

[Record Audio](#) for more information).

In most cases, the record player will have "RCA connectors" on the backside labelled *Phono*, as depicted below:
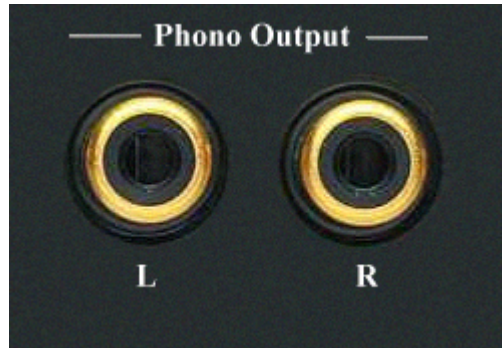
RCA connectors on a record player.

Some record players come with a fixed cable with male RCA connectors:

Cable with male RCA connectors.

The output from your record player will now need to be connected to the computer. Integrated sound cards are usually equipped with "mini jack" connectors (also called 3,5 mm jack).

The back pane of a computer with an integrated sound card and a line in mini jack connector (1).

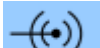### Connecting to a Sound Card with Mini Jack Connector
If you computer has mini jack connectors and your record player female RCA

connectors, a connection cable with a mini jack connector on one end and RCA connectors on the other end is required to connect your record player to your computer.


A connection cable with RCA connectors on one end and a mini jack connector on the other.

If you didn't receive such a cable with your sound card or computer, you can purchase one in normal audio or electronics stores.

To connect your record player, connect the end with the RCA connectors to the *Phono Out* connectors on your record player. Connect the other end to the *Line In* connector of your computer. The *Line In* connector is either labelled as such or indicated with the following symbol: .

If you record player has an integrated cable with male RCA connectors you will need an RCA to mini jack adapter (available in audio or electronic stores), as depicted below:
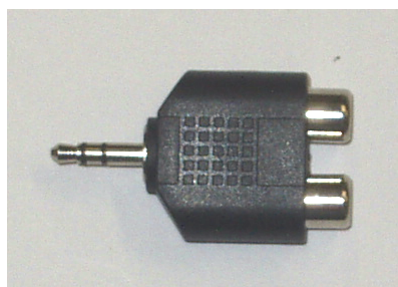

An RCA to mini jack adapter.

The mini jack end of the adapter can be connected to the *Line In* connector of the sound card as already described.

Connecting to a Sound Card or USB Audio Device with RCA connectors

External USB devices for audio input and output are growing in popularity and many of these have normal *Line In* RCA connectors. In rare cases, internal sound cards might also have RCA connectors. In these cases, a connection cable with RCA connectors on both ends is required if the record player doesn't have a cable with male RCA connectors attached.
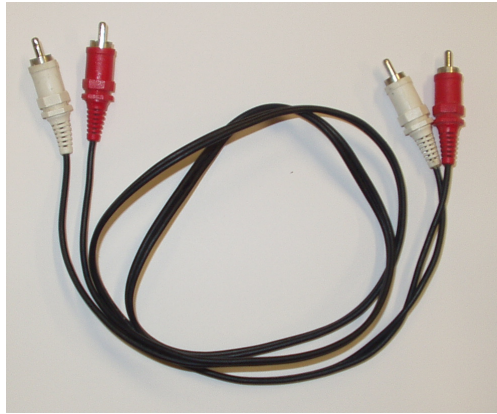
A connection cable with RCA connectors on both ends.

You can purchase RCA connection cables in normal audio or electronics stores if you don't have one already. An In this case, all you have to do is connect the cable to the *Phono* connectors on your record player and the *Line In* connectors of the USB audio device.

4   Recording

Recording audio in Acoustica is easy, but there are some terms you should know before starting you first recording project. The chapter <u>Concepts of Digital Audio</u> theoretically describes some important issues regarding digital audio. If you are familiar with digital audio, please proceed to <u>Recording through the Sound Card</u>.

4.1   Concepts of Digital Audio

Before audio can be edited on computers it must be digitized. The output from most audio equipment such as tape recorders, microphones or record players is analog. Analog means that the audio signal is represented by an alternating electrical voltage. The voltage is analog to the air pressure changes in the air during the performance, hence the term analog signals. The sound card in your computer is needed to convert the constantly changing electrical voltage to a stream of numbers at fixed rate intervals. This process is done in two steps called *sampling* and *quantization*.

4.1.1   Sampling

The conversion from a continuously changing measure to a series of measured values at discrete time instances is called sampling. The rate (or number of measurements per second) of which the sampling is done, is along with the quantization depth the most important quality factor of digital recording equipment. If it is set too low, disturbing artifacts occur. A CD quality recording is recorded with a sampling rate of 44 100 samples per second. We say that the *sampling frequency* is 44 100 Hertz (or short Hz).

In fact, all frequencies above half the sampling frequency, which is known as the Nyquist frequency, are substituted by frequencies below the Nyquist frequency. This effect is called aliasing. To avoid aliasing a sampling system contains of a low pass filter which ideally filters out all frequencies above the Nyquist frequency and leaves all frequencies below unaffected. In the case of the audio CD, the highest frequency that can theoretically be recorded is 22 050 Hz.

4.1.2   Quantization

After measuring an analog input signal at fixed time intervals we have a stream of samples. The samples exist in terms of a voltage measured at a certain point in time. The voltage can usually be one of an infinite number of possible voltages within the legal voltage range. Computers cannot accurately describe every single one of the infinite number of possibilities, so it is necessary to divide the voltage range of interest into fixed sized regions. All voltages within one region are given a certain number during the quantization process. If we have a large number of regions which implies a larger number of discrete voltage levels, we can describe a voltage more accurately than with fewer voltage levels. The audio CD is quantized with 65536 voltage levels, which is the maximum number of levels possible to archive with a binary number with 16 bits. Thus we say that the Audio CD has 16 bit resolution. Modern recording studios are frequently using 24 bit resolution or even higher during the mastering process.
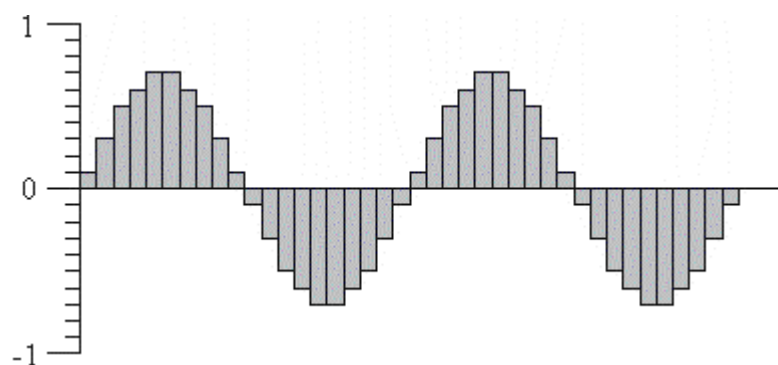

The digital representation of a sine wave.

4.1.3   The Decibel Unit (dB)

When the volume of the recorded sound is changed, the degree of change is usually expressed in terms of decibels or short dB. This is a common unit in connection with audio. In Acoustica, decibel is used to express the extent of change relative to the original level.

Special for the decibel unit is that it is based on a logarithmic scale. Zero dB represents no change, whereas an increase of six dB represents a doubling of the signal amplitude. Reducing by six dB results in half the signal amplitude.
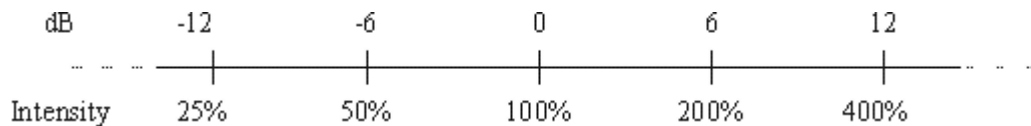

The decibel dB versus intensity change in percent

The decibel scale is chosen to suit the sensitivity curve of the human ear which have the same logarithmic property.

4.2   Recording through the Sound Card

Please follow the steps below to record audio from audio equipment such as record players, tape decks or microphones through your sound card:

1. Make sure the audio source is properly connected to the audio input of your computer. See Connecting Your Stereo Equipment for more information about how to connect your audio equipment.
2. Create a new an empty recording window by selecting *File | New...* or position the cursor where you want to insert recorded audio in an existing recording.
3. Select *Sound | Record...* or press Ctrl+R.
4. If you are recording to an empty editing window Acoustica needs to know what sample format you wish to use (see Concepts of Digital Audio for more information). The following dialog box appears:
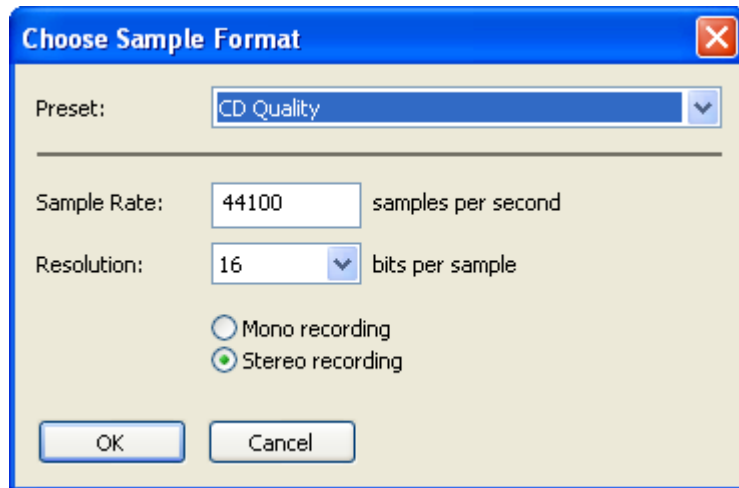
The sample format dialog box in Acoustica

Please choose the desired recording format and click the button labelled *Ok*.

5. The Recording dialog box now opens:
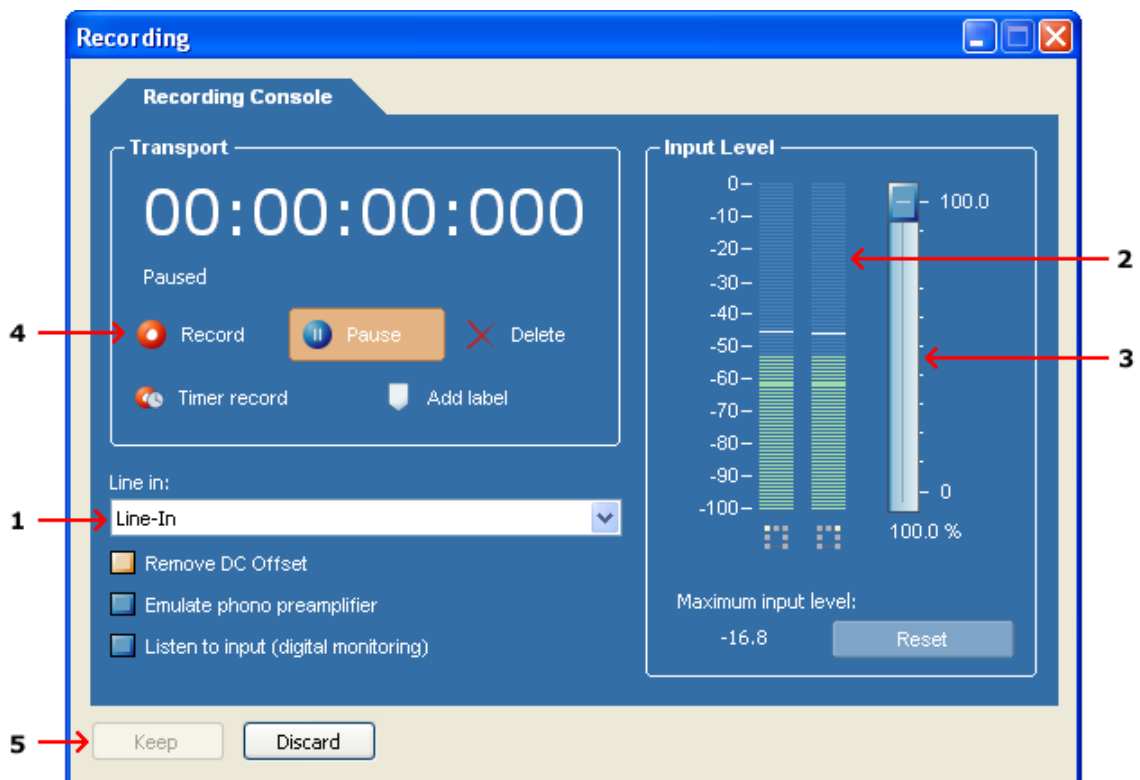
The Recording dialog box

Now make sure that the correct input line is selected in the line in combo box (1). You can monitor the input level using the level meters (2) and, if needed, adjust the

input level using the input level slider (3). The level meter should never be in the red area in order to avoid clipping distortions.

6. Click the button labelled *Record* (4) to start the recording.
7. When you are done recording, click the button labelled *Keep* (5) to accept the recording.

4.2.1 Timer Record

The *timer record* feature allows you to start and stop recording after a certain period of time or depending on the presence of an input signal. To start timer record, click the button labelled *Timer Record* in the recording dialog. The following dialog box appears:
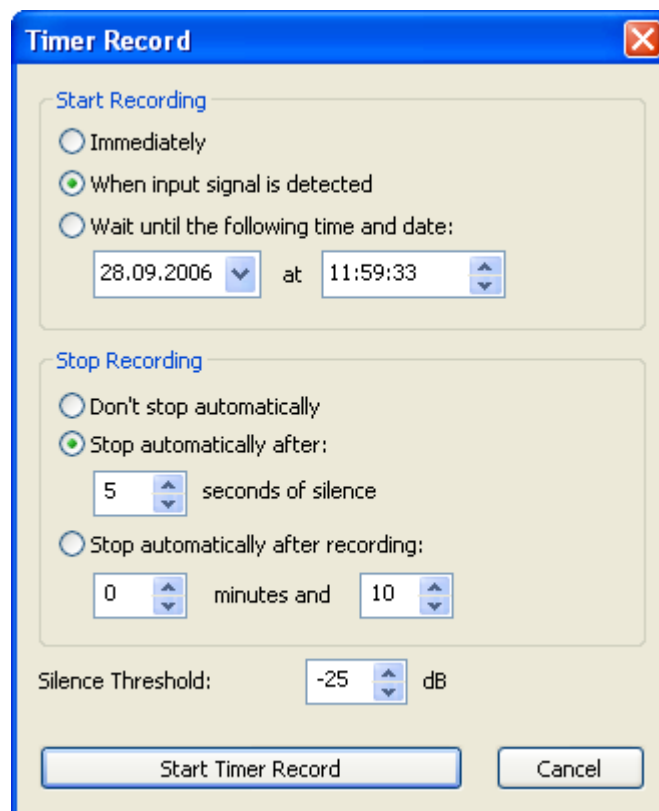

The Timer Record settings.

You can choose to start the recording immediately (after clicking the *Start Timer Record* button), at a certain time and date or when an input signal is present. The threshold value for the input signal detection can be defined using the *Silence Threshold* field at the bottom of the dialog.

The recording can also be stopped automatically, either after a certain period of silence or after a certain period of time.

4.2.2   Advanced Recording Options

The recording dialog in Acoustica also offers some advanced settings:
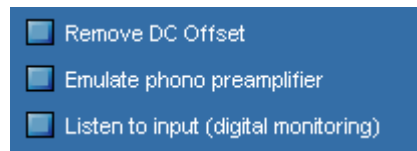
The advanced recording options in the recording dialog

- **Remove DC Offset**
  A DC offset (Direct Current offset) is present in the input signal when the audio signal isn't centered around the zero voltage line as it should be. The problem is quite common on low-end sound cards. DC offset are unwanted and harmful to the stereo equipment and it can lead to problems when processing the recording further. You can automatically remove the DC offset by enabling the option *Remove DC Offset* . Extremely low and inaudible frequency components (including the DC offset) are then filtered out of the signal.

- **Emulate phono preamplifier**
  This option should only be enabled when recording from a record player that is connected directly to the input of the sound card without a phono preamplifier.

- **Listen to input (digital monitoring)**
  You can listen to the signal being recorded (after DC offset and phono preamplifier processing, if enabled) by enabling this option. It is not recommended to leave this option on during the complete recording session, because it makes recording glitches more probable.

5   Audio Processing

In Acoustica, all the processing tools have some properties in common. The processing is performed on the selected region and the selected channels only. Furthermore, most tools offer a preset manager that allows you to save often used parameter settings for later use.

The Preset Manager found in most of the processing tools in Acoustica

To add a preset:

- Click the button labelled *More...*
- A drop down menu appears. Select *Add Preset...* from the menu.
- In the pop-up dialog, enter the name of the preset and click the button labelled *OK*.

Loading a preset is equally simple. Just select the preset from the drop-down list and the preset settings are loaded. User presets can be removed by clicking the button labelled *More...*, selecting *Remove Preset...* and selecting the preset you wish to remove.

You can bypass the effect processing by checking the check box labelled *Bypass transform* for a convenient A / B comparison.

The processing tools introduce some custom controls that you should get familiar with. The level slider is similar to the Windows track bar, but with some enhancements. The value range and the current value is always displayed. You can manually edit the value by pressing the space bar when the control is active or by clicking the value text below the level slider. An edit box pops up containing the current value. To change the value, enter a new value and press return. By pressing the Esc key the changes are discarded.
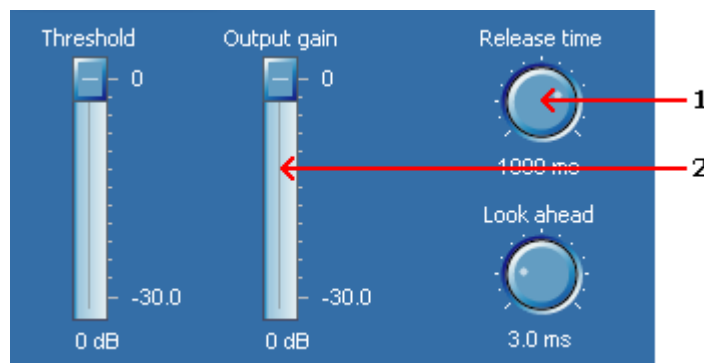
Examples of knobs (1) and level sliders (2).

The knob control is similar to the level slider, but it is round and is often used to save space in complex layouts. As with the level sliders, you can change the value by pressing the space bar or by clicking the value text below the knob.

A more complex custom control is the curve control which is used when a curve input is needed. The curve control allows the user to add, move or remove points. Straight lines between the points build the curve. You can add points by clicking the location where you want a new point to appear in the curve control. You can remove points by clicking the right mouse button over an existing point.
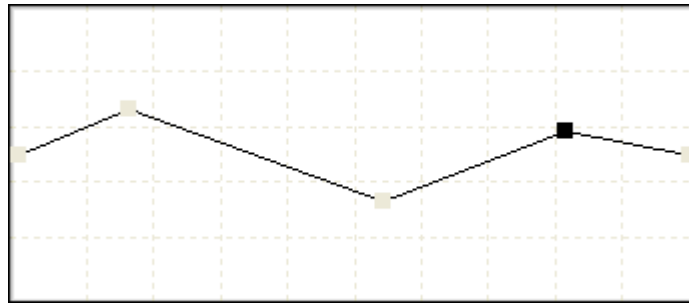
*Examples of a curve edit control.*

5.1   Manipulating Volume

The *Volume* menu contains several commands for manipulation the volume of a recording.

- Adjusting the Volume
- Normalize
- Applying a Volume Curve
- Fading In or Out
- The Channel Mixer

5.1.1   Adjusting the Volume

The most basic volume manipulation command is the *Adjust Volume...* command in the *Volume* menu. The only parameter is the volume change in decibel. For more information on the decibel unit, see The Decibel Unit (dB).
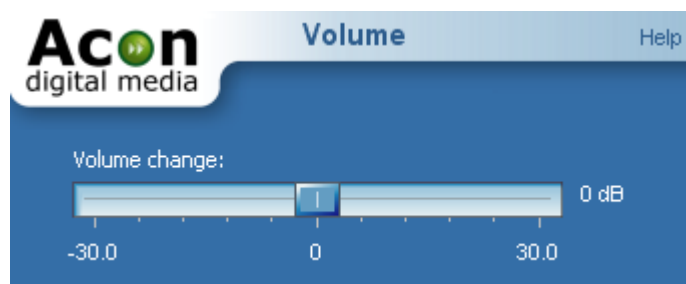
The Adjust Volume settings

5.1.2   Normalize

The *Normalize...* command in the *Volume* menu can be used to ensure a constants signal level in all your audio recordings. After selecting your desired maximum peak level output in decibel, Acoustica analyses the selected region for the loudest peak. The volume of the region is changed according to the selected maximum. If you set the
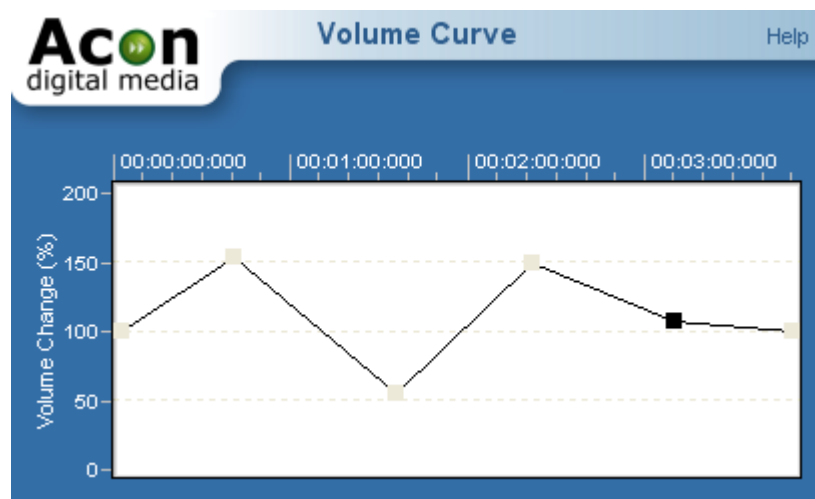
normalize level to 0 dB, the loudest part will be the maximum level reproducible without signal distortion. For more information on the decibel unit, see The Decibel Unit (dB).
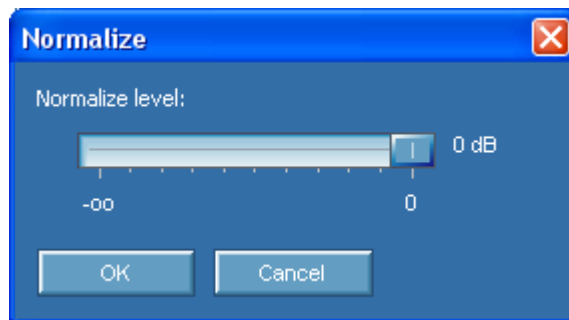
The Normalize dialog box

5.1.3   Applying a Volume Curve

You can apply a user drawn volume curve on the selected region by selecting *Draw Freehand Volume Curve...* from the *Volume* menu.
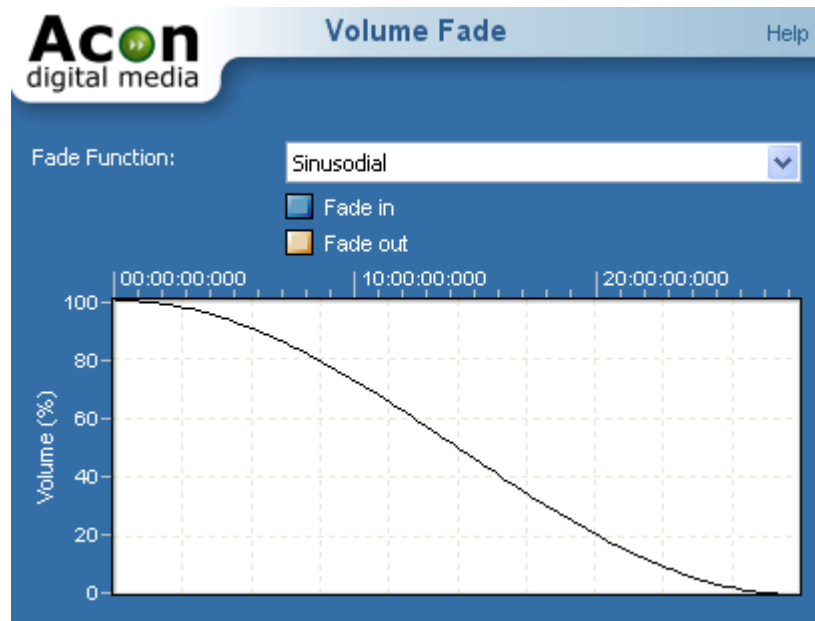

The Volume Curve settings

5.1.4   Fading In or Out

If you wish to create fade ins or fade outs, select *Fades...* from the *Volume* menu. You can select a fade curve from one of four mathematical functions from the *Fade function* drop-down list:

- Linear
- Logarithmic
- Exponential

- Sinusoidal

Select the direction of the fade using the *Fade in* or *Fade out* radio buttons. The resulting fade curve is visualized in the curve control.

The Volume Fade settings

5.1.5   The Channel Mixer

The channel mixer is a tool that works only on stereo recordings. The channel mixer mixes the content of the left and the right channels with a user defined weighting. The weightings are individually adjustable for each channel.
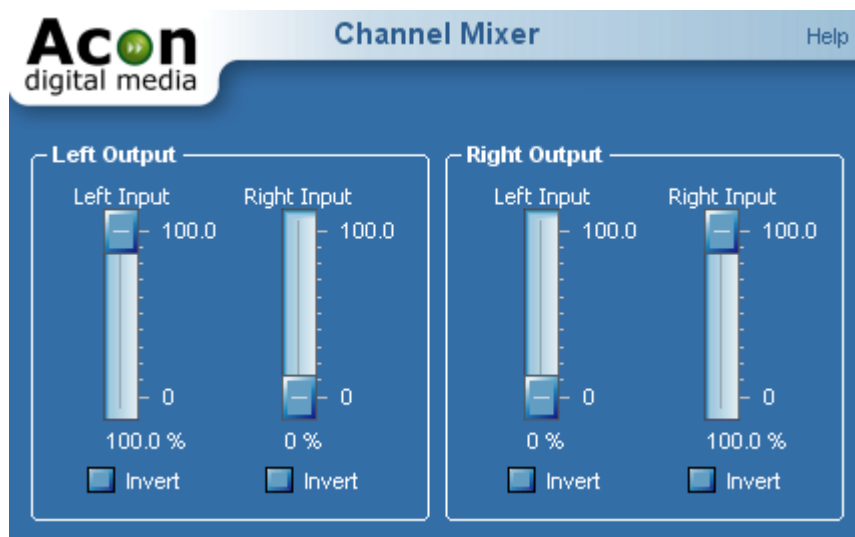

The Channel Mixer settings

5.2 Audio Effects

5.2.1 Dynamic Processor

A dynamic processor is used to alter the dynamic properties of the recording. To understand how a dynamic processor works, imagine a sound engineer trying to maintain an as steady volume level as possible while doing a recording. When the input level increases he pulls down the volume slider, and he pulls it up when the input level decreases. A dynamic processor does the same thing automatically, only with a much faster reaction time.
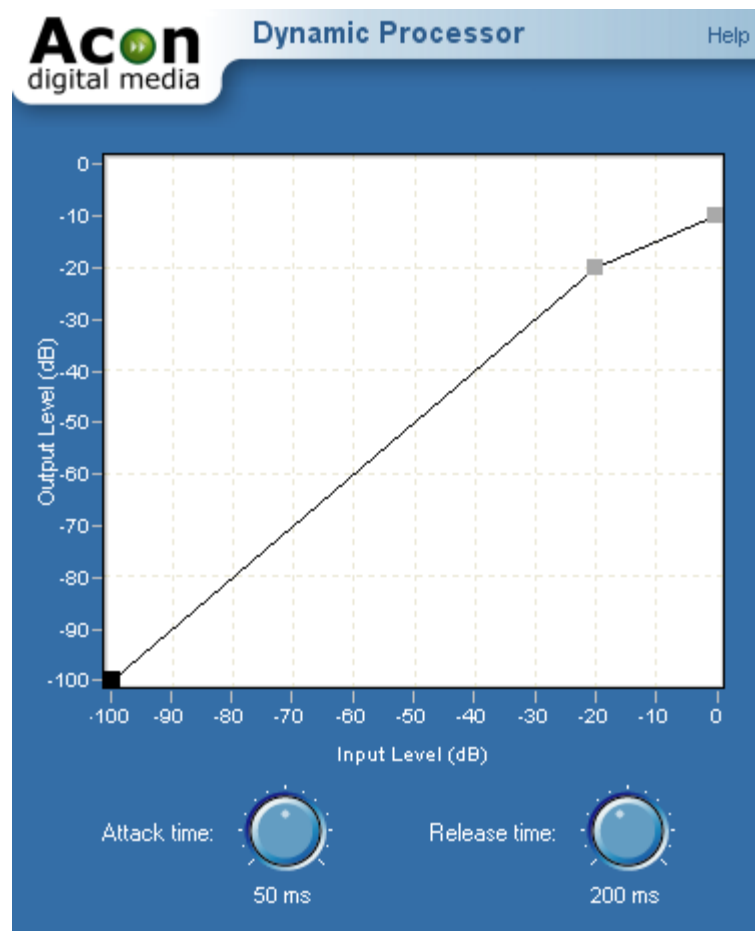
The Dynamic Processor settings

Settings

- An editable curve containing the output level as a function of the input level. By altering this curve you can change the dynamic characteristics of the recording to fit
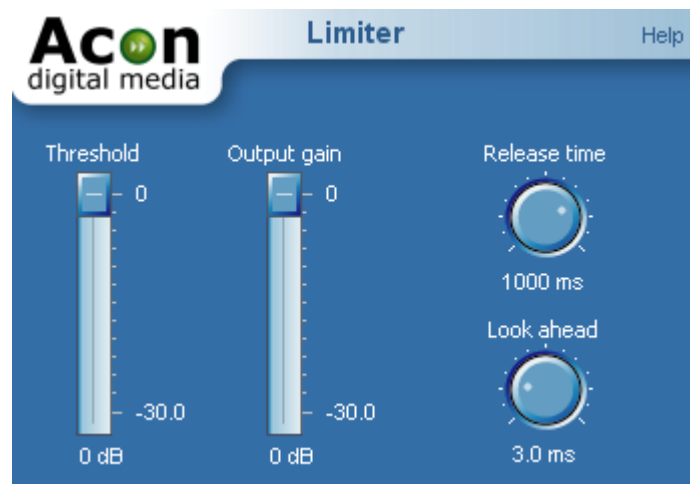
your needs.

- **Attack time**
  The reaction time when the input level of the source material increases.

- **Release time**
  The reaction time when the input level of the source material decreases.

5.2.2   Limiter

Limiters belong to the dynamic processing tools and ensure that the signal level doesn't exceed a user selectable boundary while minimizing any possible distortions. To achieve this, limiters introduce a certain latency, called look-ahead. The look-ahead ensures that the limiter can respond in time when the signal level suddenly rises. Many CD producers also use limiters to push the perceived volume to a maximum without introducing audible distortions.

The Limiter settings

Settings

- **Threshold**
  All signals above the threshold level will be attenuated so that the threshold level is not exceeded.

- **Output Gain**
  The output gain of the limiter which corresponds to the highest possible output signal level.

- **Release time knob**

You can use the release time knob to adjust the response time when the input level of the source material decreases. Longer release times will result in a smoother sound.

- **Look ahead knob**
  You can adjust the number of milliseconds the limiter uses to determine its internal gain settings with the look-ahead knob. The limiter will have a latency equal to the look ahead time.

5.2.3 Echo

The echo effect adds delays or echoes to the selected region.
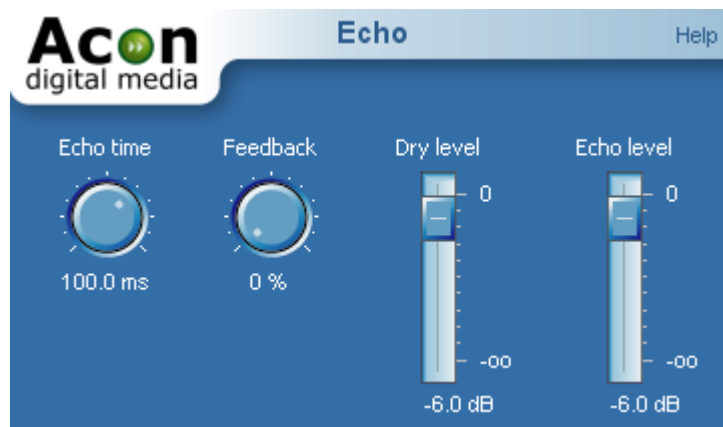
The Echo settings

Settings

- **Echo time**

- **Feedback**
  The feedback percentage specifies the volume attenuation since the last delay interval.
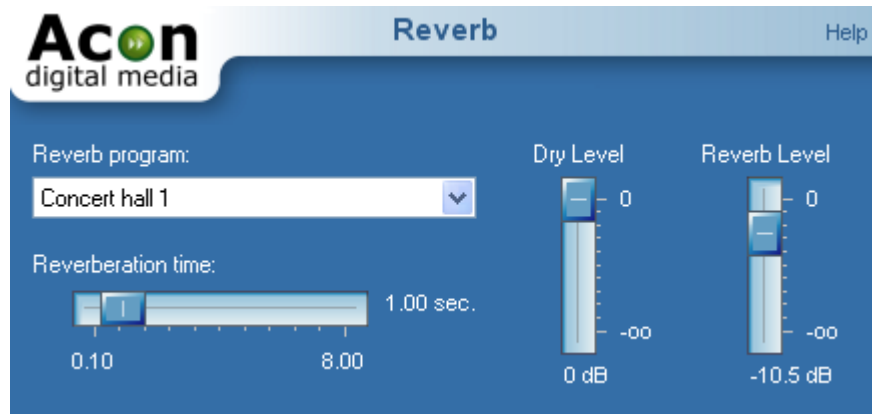
- **Dry Level**
  The amount of unprocessed signal in the output mix.

- **Delay Level**
  The amount of processed signal in the output mix.

5.2.4 Reverb

The reverb simulates the reverberation effect of some natural surroundings.

The Reverb settings

Settings

- **Reverb program**
  Select the reverberation program from the drop-down list that fit you needs. You can choose between *Small room, Medium room, Large room, Concert Hall 1, Concert Hall 2, Cathedral, Stadium* or *Plate* reverb. Only the Plate reverb needs further explanation. The plate reverb program simulates the sound of analog plate reverbs used frequently before digital reverbs were available.

- **Reverberation time**
  The time before the reverb tail drops below 1 / 1000 of its original amplitude.
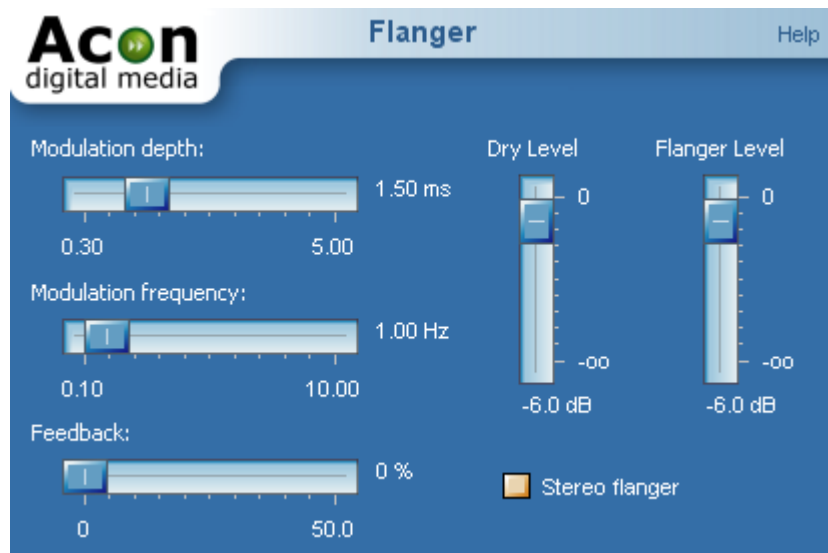
- **Reverb level**
  The amount of processed signal in the output mix.

- **Dry out level**
  The amount of unprocessed signal in the output mix.

5.2.5 Flanger

Flanger is an artificial effect. Flanger will occur when two sources playing the exactly same recording with a short time delay are mixed together. The result is that some frequencies are canceled, while others are amplified. Changing the time delay between the two sources will result in other frequencies being canceled or amplified.

The Flanger settings

Settings

- **Modulation depth**
  Amplitude of the modulating function.

- **Modulation frequency**
  Frequency of the modulation function. A higher modulation frequency will result in faster changes in the tone quality.

- **Feedback**
  The feedback percentage specifies attenuation in the internal delayed feedback loop.

- **Stereo Flanger**
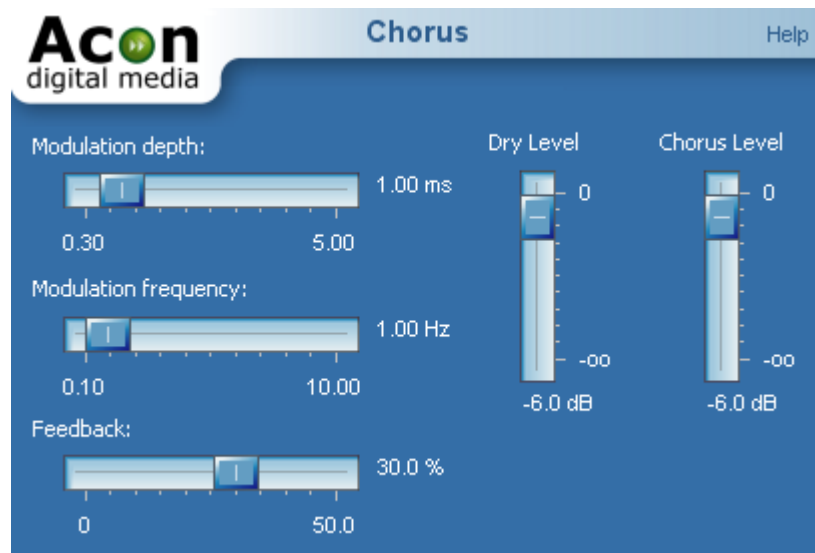 Toggles stereo or mono processing mode.

- **Dry Level**
  The amount of unprocessed signal in the output mix.

- **Flanger Level**
  The amount of processed signal in the output mix.

5.2.6   Chorus

Chorus adds warmth to recordings by dividing the audio source into several voices with a random vibrato. These artificially created voices are then mixed together.

The Chorus settings

Settings

- **Modulation depth**
  Amplitude of the modulating functions.

- **Modulation frequency**
  Frequency of the modulation functions. A higher modulation frequency will result in faster changes in the tone quality.

- **Feedback**
  The feedback percentage specifies attenuation in the internal feedback loop.

- **Stereo Chorus**
  Toggles stereo or mono processing mode.

- **Dry Level**
  The amount of unprocessed signal in the output mix.

- **Chorus Level**
  The amount of processed signal in the output mix.

5.2.7 Harmonizer

The harmonizer mixes several pitch shifted voices to create interesting harmonies. You can mix up to four pitch shifted voices. The often experienced "chipmunk" effect which occurs when transposing the human voice or musical instruments can be reduced using

the *Maintain timbre* option. When *Maintain timbre* is checked, Acoustica creates a smooth spectral envelope estimation of the signal and whitens the signal before pitch shifting. The original smooth spectral envelope is applied after transposing and the original timbre is preserved.
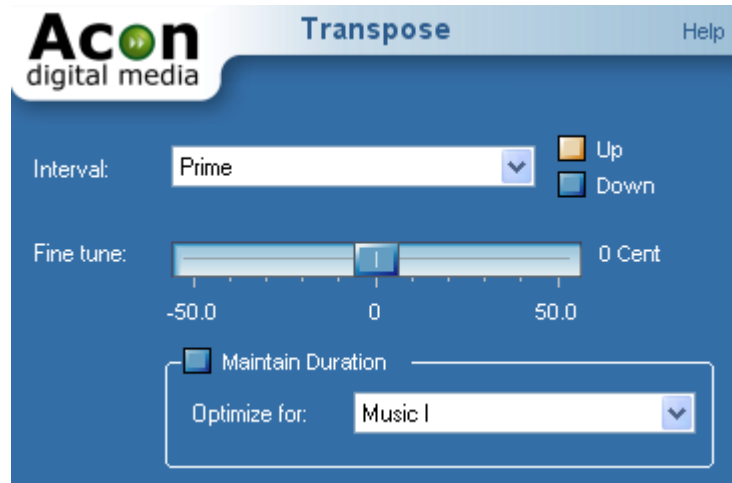

The Harmonizer settings.

Settings

- **Activate**
  Check this to activate the current voice.

- **Interval**
  The musical interval to transpose. Use the radio buttons to set the transpose direction to *up* or *down*.

- **Fine tune**
  With this slider you can fine tune the pitch shift factor in cents which are 1/100 of a semitone.

- **Volume**
  Volume of the current voice in dB.

- **Pan**
  Left / right panning of the current voice in percent.

5.2.8   Transpose

The transpose command allows you to transpose the content of the selected region with or without changing its length. When transposing without change of length, the Acoustica time stretching algorithm is used to compensate for the change of length normally occurring during pitch shifting.

The Transpose settings

Settings

- **Interval**
  The musical interval to transpose. Use the radio buttons to set the transpose direction to up or down.

- **Fine tune**
  With this slider you can fine tune the pitch shift factor in cents which are 1/100 of a semitone.

- **Maintain Duration**
  Check this check box if you want the length of the section to be the same after pitch shifting.

- **Optimize for**
  The inner settings of the time stretching algorithm are affected by choosing an optimization from one of the following: Speech, Music I or Music II, monophonic signals or percussive instruments only.

5.2.9   Time Stretching

The time stretch algorithm allows you to change the length of the recording without changing the perceived pitch. It is not physically possible to do this without some sort of distortion and the quality of these time correction algorithms lies in their ability to reduce the audible distortions. Acoustica uses a method that works well on monophonic signals like speech and single musical instruments as well as on polyphonic music. However, some artifacts occur on high expansion or compression rates like in all other time stretching software.

The Time Stretch settings

Settings

- **Percentage**
  The time stretching factor in percent.

- **Optimize for**
  The inner settings of the time stretching algorithm are affected by choosing an optimization from one of the following: *Speech, Music I or Music II, monophonic signals* or *percussive instruments only*.

- **Duration change**
  Here you can enter the new length after time stretching.

- **Tempo change**
  Here you can enter the original tempo in BPM and the target tempo after time stretching

5.2.10 Reverse

The reverse effect processes the selected region in such a manner that it will be played backwards.

5.3 Using Audio Plug-Ins

Acoustica 4 supports DirectX as well as VST plug-ins. Both formats have become widely used standards for audio processing plug-ins on the Windows platform. The menu *Plug-Ins* contains a list of the plug-ins currently installed on your computer. If you have installed a plug-in that does not appear in the list, please refresh the plug in list (see Refreshing the Plug-in List). Using DirectX or VST plug-ins is as easy as using the internal audio processing tools. You will see that previewing works exactly the same way as with the internal audio processing tools.

5.3.1 Refreshing the Plug-in List

If you have installed new DirectX or VST plug-ins on your computer that do not appear in the *Plug-Ins* menu you should refresh the menu by selecting *Plug-Ins | Refresh Plug-In List*. Acoustica then rescans the plug-ins on your computer and updates the menu.

5.3.2 Specifying VST Directories

VST plug-ins are not registered on your computer like DirectX plug-in, hence Acoustica needs to know where to find them. You can specify one or more directories to scan through.
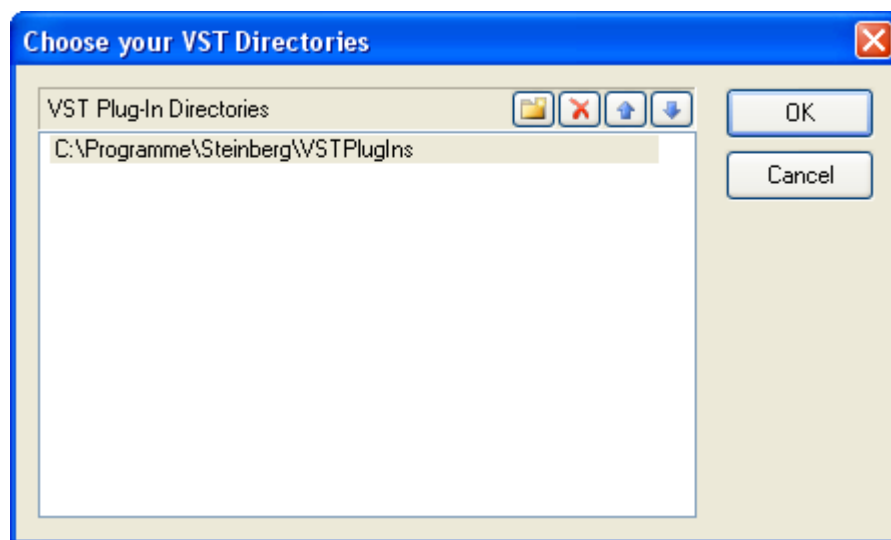
The VST Directories dialog lets you specify one or more VST directories to scan through.

To add a new entry, click the add directory button [ ]. You can edit a directory entry

by double clicking with the left mouse button. Delete an entry by clicking the delete button [ ].

5.4   Noise Reduction

Acoustica provides tools for removal of both stationary noise such as hiss and impulsive noise like clicks and pops. The frequency spectrum of stationary signals remain pretty constant over time. In other words, tape hiss, humming or other constant disturbances are removable using the Acoustica noise reduction system. The noise reduction is based on the spectral subtraction technique. This means that a spectrum of the noise present in the recording is needed. Acoustica can automatically estimate the noise profile or the spectrum of the noise can be obtained through analysis of a region containing noise only. Alternativly, the user can define the noise profile manually.

- Automatic Noise Reduction
- Removing Noise with a Measured Profile
- Removing Noise with a User Drawn Profile

Impulsive noise on the other hand consists of short unpredictable noise burst. It has become usual to define clicks as single noise bursts, whereas crackle is defined as a series of very short clicks with low amplitude. Acoustica is able to remove impulse noise due to the short duration of the noise bursts. Acoustica substitutes the recorded signal in the short period of time during the click with a signal estimated using the undistorted audio surrounding each clip.

You can remove clicks either automatically or manually:

- Automated Click Removal
- Manual Click Removal

5.4.1   Automatic Noise Reduction

The easiest way of removing stationary noise is to use the automatic noise reduction. Select *Enhancement | Automatic Noise Reduction...* Acoustica then performs a statistical analysis of recording in order to estimate the noise profile. This process might take a little while, depending on the length of your recording. After the analysis phase, the noise reduction window appear:
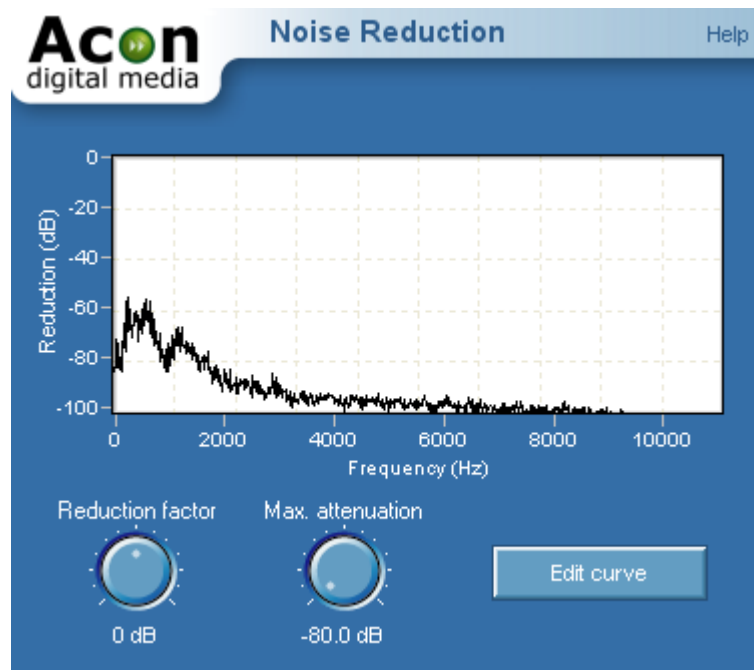
The Noise Reduction settings. The noise profile graph contains the result of the noise analysis.

There are two parameters you can adjust:

- Reduction factor scales the noise profile obtained in the analysis phase and allows you to remove more (positiv values) or less (negative values) noise than the analysis algorithm detected.
- Max. attenuation allows you to adjust a maximum attenuation factor for each frequency component. This parameter is also referred to as noise floor. By leaving a certain noise floor, you can mask annoying artifacts from the noise reduction algorithm.

Initially, you cannot edit the noise profile graph after the noise analysis. If you want to edit the result:

- Click the button labelled *Edit Curve*
- The number of points in the curve is reduced and you can add, move or remove the curve points.

5.4.2   Removing Noise with a Measured Profile

If your recording contains pauses with pure noise without any other signal, the pause can be used to create a noise profile automatically. Select *Enhancement | Noise Analysis...* after selecting the region containing noise only. After the analysis phase, select Enhancement | Noise Reduction... The result of the analysis is automatically loaded in the profile graph.
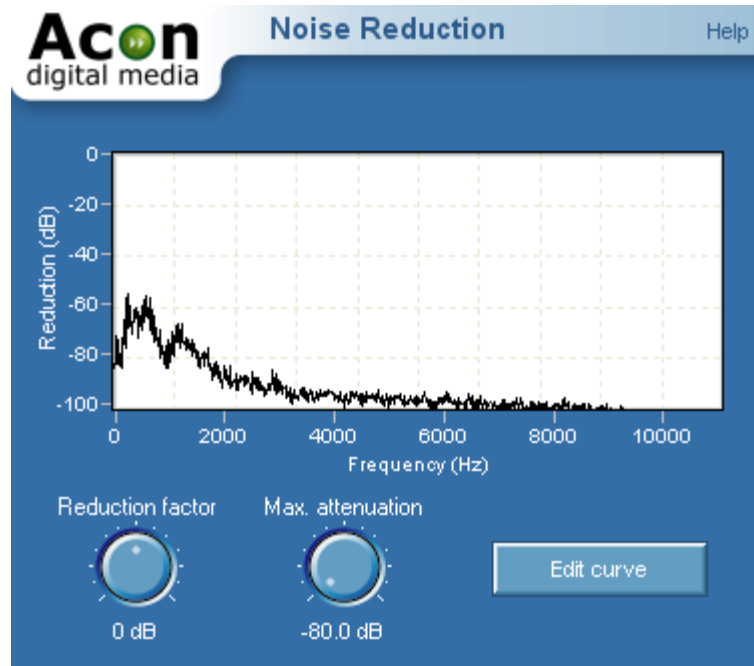
The Noise Reduction settings. The noise profile graph contains the result of the noise analysis.

Apart from the analysis method, the noise reduction algorithm is exactly the same as with the <u>automatic noise reduction</u>. Hence, the parameter settings have the same meaning.

5.4.3   Removing Noise with a User Drawn Profile

If no pauses are available for noise analysis you can draw a noise profile by hand. Select *Enhancement | Noise Reduction*. A curve editor shows up where you can manually enter a noise profile.

> **Note**   If you performed noise analysis in a previous editing step, you will need to reset the curve by selecting Reset from the preset list.

For tape hiss this method works quite well. The tape noise is more or less evenly distributed in the spectrum, thus a flat line should to the job. Finding the right threshold can be tricky, but after a bit of try and fail, good sounding settings should be possible to find.
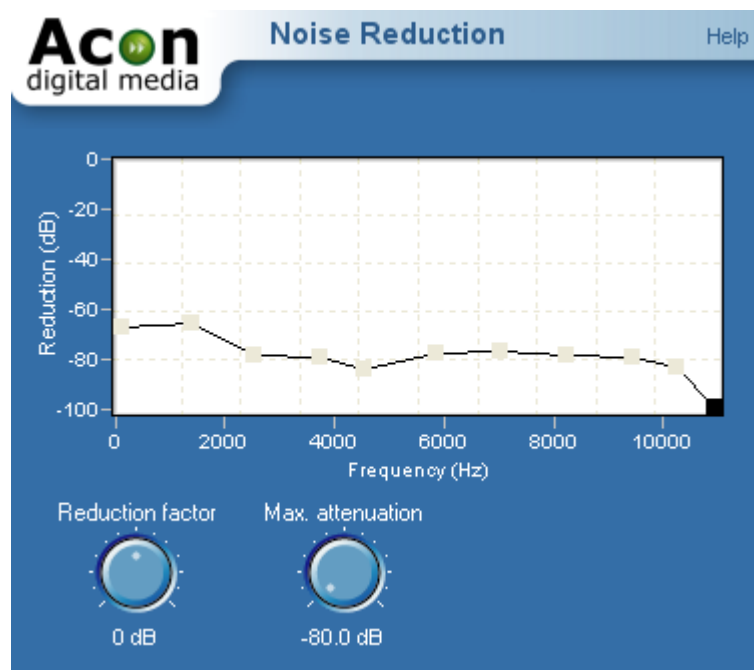
The Noise Reduction settings with a manually drawn noise profile.

5.4.4   Automated Click Removal

The declicker is a specialized tool for removing impulsive noise such as clicks and crackle. It is based on two algorithms, one that targets rare but loud clicks, and one that targets very short and frequent clicks referred to as crackle.
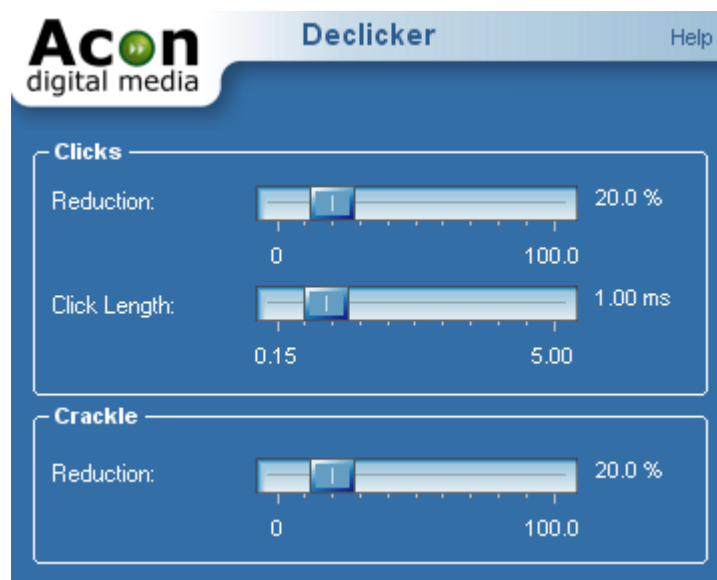
The Declicker settings

Settings

- **Click Reduction**
  Sets the sensitivity of the click filter. Higher reduction levels result in more click reduction.

- **Click length**
  The maximal length of the clicks that are to be removed.

- **Crackle Reduction**
  Sets the sensitivity of the crackle filter. Higher reduction levels result in more crackle reduction.

Optionally, clicks may be removed manually by selecting one single click and using the *Interpolate* command in the *Enhancement* menu.

5.4.5 Manual Click Removal

You can remove clicks manually be selecting the smallest region which fully includes the complete click and then select *Interpolate* from the *Enhancement* menu or press Ctrl+I. Acoustica substitutes the region with an estimate of the signal without the click.

> **Note** Interpolation is only possible on short regions. If the selected region is too long, the Interpolate command is grayed out.

5.5 Enhancement Tools

5.5.1 Declipper

The declipper restores audio recordings distorted by clipping. Clipping occurs during recording when the recording level is too high and the highest peaks cannot be correctly recorded. The declipper substitutes such distorted peaks by an estimation of the signal curve using almost the same mathematical methods as the declicker when eliminating clicks.
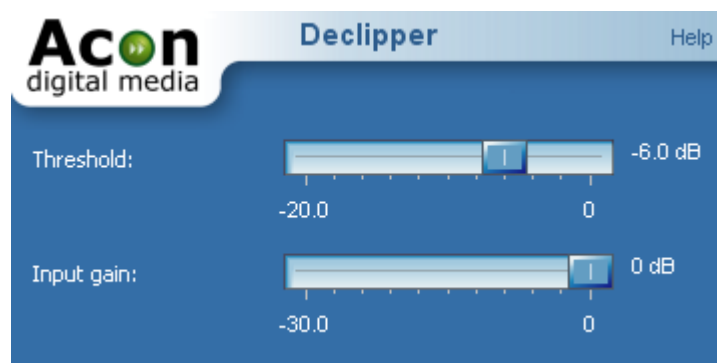
The Declipper settings

The most important parameter of the declipper is the threshold level. The declipper will substitute all recorded peaks louder than the threshold value. Furthermore, the input gain of the declipper can be adjusted. That is helpful if the signal level is so high that the restored clips get louder the maximum signal level range.

Settings

- **Threshold**
 All sample values louder than the threshold are substituted by a signal estimation.

- **Input gain**
  The input gain is useful for adjusting the signal level before declipping.

5.5.2   Equalizer

Acoustica features a powerful six band parametric equalizer. Unlike a graphic equalizer, a parametric equalizer allows the user to select the center frequency of the bands. The equalizer in Acoustica also has variable bandwidths on each band.

The center frequency is specified in Hertz. Sound characters of the different frequencies are best discovered through experimentation. The bandwidth may be specified in octave bands. One octave band equals the frequency range of one octave on the piano keyboard. The bandwidth is the same parameter as the Q-factor found in some analog equipment. A higher Q-factor represents a narrower bandwidth.

Equalizer terminology
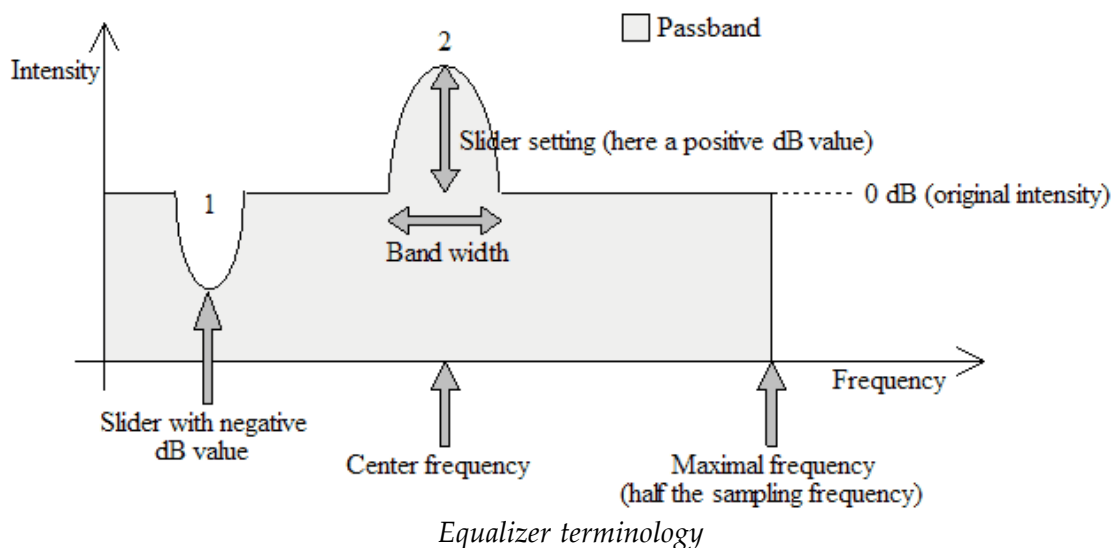
The lower part of the equalizer dialog box is used to display the filter response curve (gain versus frequency). The filter response curve is calculated mathematically and is updated as the parameters are altered.
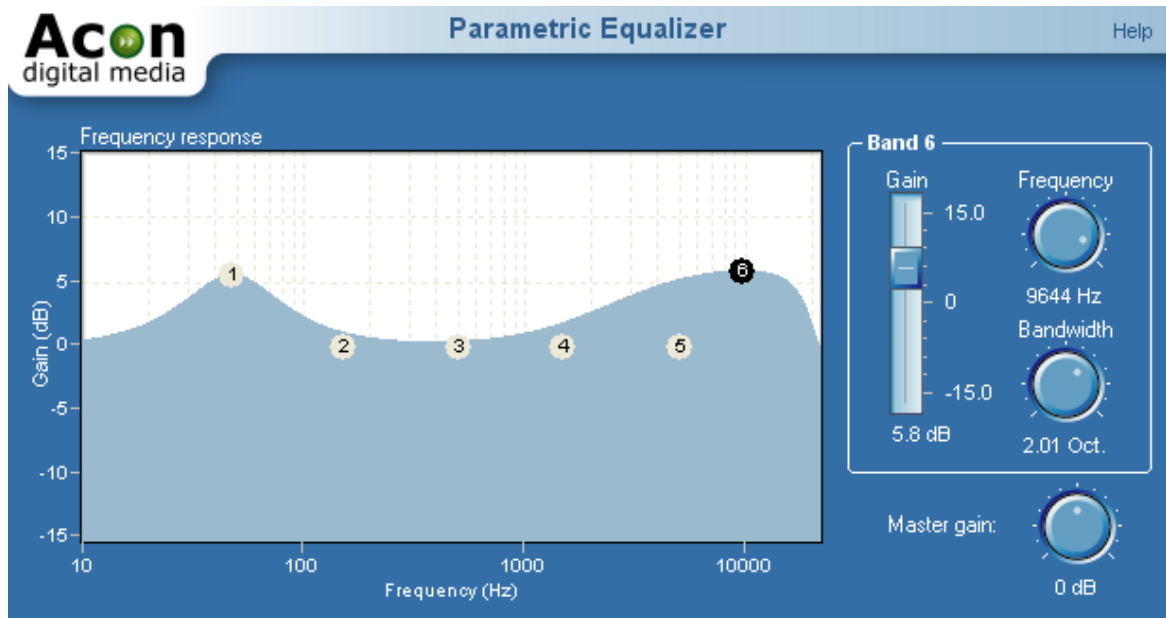
The Equalizer settings

Settings

- **Frequency response**
  The frequency response of the current equalizer settings is visualized. Each band is visualized with a small circle with the number of the band. The currently active band is highlighted. You can activate band by clicking the circles representing each band. If you keep the mouse button down while moving the mouse cursor, you can change the gain and the center frequency of each band directly in the frequency response.

- **Frequency**

- **Gain**

- **Bandwidth**

5.5.3   High Frequency Rebirth

Dull and lifeless recordings may be a result of lost high frequency content. By using the
High Frequency Rebirth tool (select *Enhancement | High Frequency Rebirth...*), you can add
artificially created harmonics to brighten up the recording. Note that these are merely
synthetic harmonics based on the low frequent content of the recording and this feature
can only be used give the listener the impression of stronger harmonics.
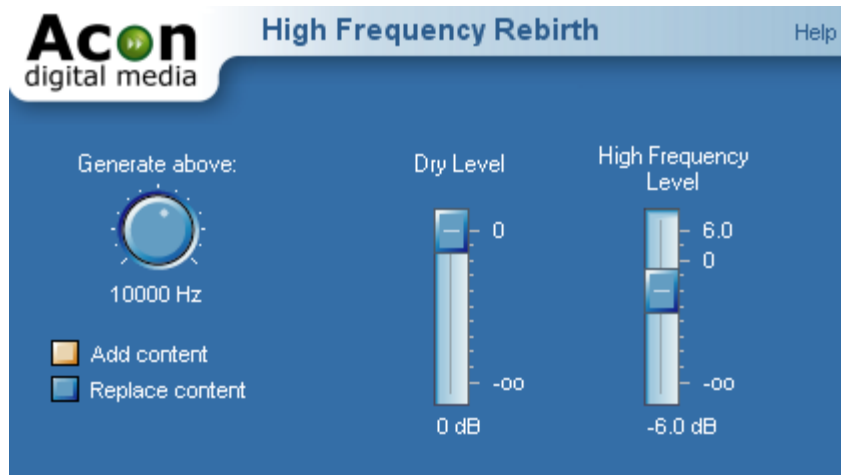
The High Frequency Rebirth settings

Settings

- **Generate above**
  Specifies the lower limit of the frequency band where harmonics should be
  synthesized.

- **Add content / Replace content**
  If add content is selected, the original content in the synthesis frequency band is
  mixed with the synthesized harmonics. If you choose replace, Acoustica removes the
  original content before adding the synthesized harmonics.

- **Dry level**
  The amount of unprocessed signal in the output mix.

- **High frequency level**
  The amount of processed signal in the output mix.

5.5.4   Stereo Enhancer

The Stereo Enhancer enhances the stereo image by filtering the left and the right differently. The filters are designed to maintain mono compatibility.
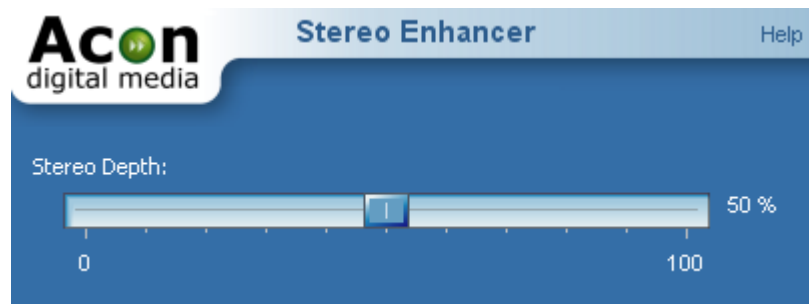
The Stereo Enhancer settings

Settings

- **Stereo Depth**
  Sets the amount of stereo enhancement from 0 (no enhancement) to 100% (maximum enhancement).

5.5.5   Remove DC Offset

Wrongly calibrated recording equipment may result in a signal that is not centered around zero as it should be. The *Remove DC Offset* tool (select *Enhancement | Remove DC Offset*) calculates the DC offset of the selected region and subtracts the offset if necessary.

5.5.6   Phono Filter

The phono filter emulates the effect of a phono preamplifier (deemphasis filter) or the opposite process applied when creating a master record (emphasis filter). In some cases, you can achieve better results from the declicker if you record an LP without the emulation of a phono preamplifier, apply the declicker and then apply the phono filter. You can post process any recording originating from an LP this way by first applying the emphasis filter, perform declicking and apply the deemphasis filter.
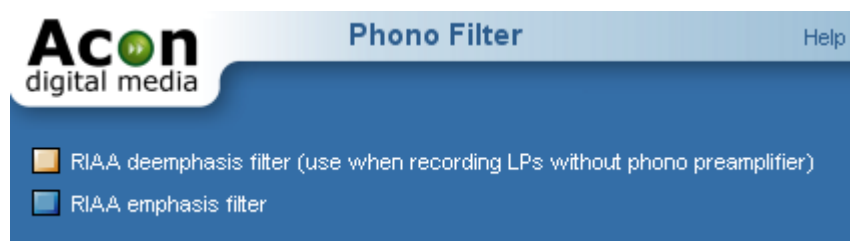
The Phono Filter settings

Settings

- **Emphasis or deemphasis mode**
  Choose deemphasis mode if you have a recorded an LP record without phono
  preamplifier.

5.6   Converting the Sample Format

You can change the sample format of an audio recording by selecting *Sound | Convert
Sample Format...* If you are not familiar with the terms sample rate or resolution, please
read <u>Concepts of Digital Audio</u> before proceeding.

To convert the sample format of a recording, select *Sound | Convert Sample Format...* A
dialog box appears where you can define the sample rate, resolution and the number of
channels in the new sample format. Click the button labelled *Ok* when you are done.
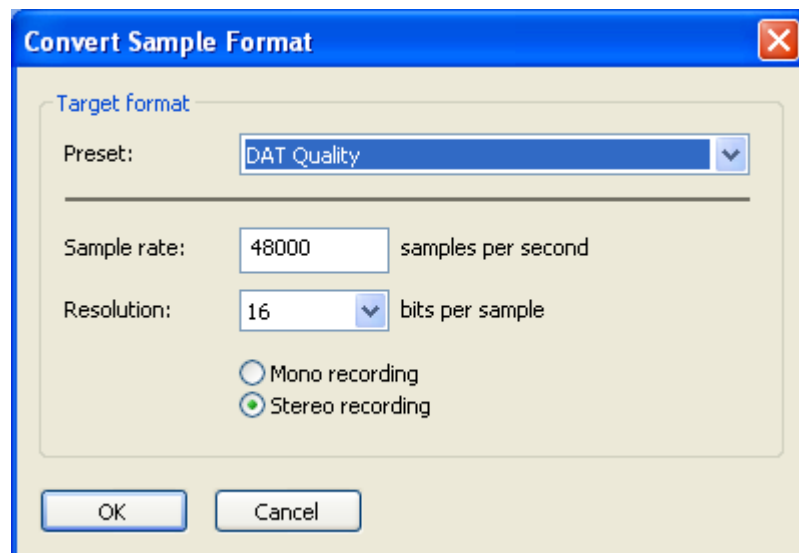
The Convert Sample Format dialog box

5.7   Effect Chains

The effect chain editor allows you to link effects, processing tools, VST and DirectX
plug-ins. The chains can be saved including the settings of the effects for later use.
Furthermore, each element can easily be bypassed and the order of the elements
changed using drag and drop.
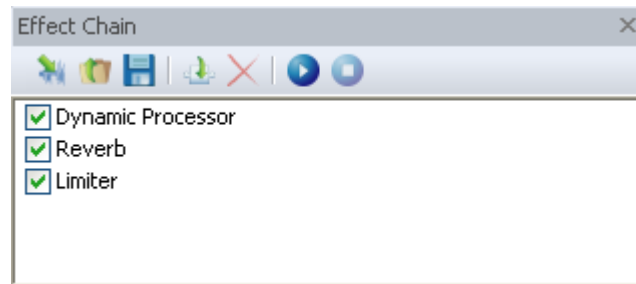
The effect chain editor in Acoustica.

Adding Effects to the Chain

To add a new effect to the chain, click the add effect button [ ]. A browser window appears where you can select the effect, plug-in or processing tool you want to add to the chain.

Removing Effects from the Chain

To remove an effect, select the effect you wish to remove and click the remove button [ ].

Editing the Effect Settings of an Element in the Chain

To open the effect settings window of an element in the chain, double click its entry in the list.

Playing the Audio Processed by the Effect Chain

You can play audio processed by the effect chain if you have an open editing window. Select the region you want to play in the editing window and click the play button [ ]. The selection will be played looped. You can stop the playback by clicking the stop button [ ].

Bypassing an Element

You can bypass an element by clicking the checkmark left to the effect entry in the list.

Saving and Loading Effect Chains

You can store a complete effect chain including all parameter settings for later use. To store the effect chain, click the save button [ ]. A standard file save dialog box appears where you can enter the file name. To open an effect, click the load chain button [ ] and select the file in the file browser.

6   Working with Audio CDs

You can create Audio CDs (see [Creating Audio CDs](#)) that you can play in normal CD players directly within Acoustica if you have a supported CD burner. Furthermore, Acoustica allows you to import audio tracks from existing CDs digitally and without quality loss for further editing or archiving on the computer (see [Importing Audio Tracks from CDs](#)).

6.1   Creating Audio CDs

Acoustica allows you to create audio CDs containing your edited recordings. The first step towards your own CD is to create a CD-Project by selecting *File | New | CD Project…* or by clicking the new button and select *CD Project…* from the drop-down list. The CD Project contains a list of the audio tracks to be written on the CD and an additional toolbar for commands related to the CD Project.

An empty CD Project window

The CD Project window contains a list of the audio tracks to be written to the audio CD.

You can add either

- [existing audio files](#) - or -
- [the content of an editing window](#)

as a separate tracks in the track list of the CD project.

6.1.1   Adding an Existing Audio File

You can add an existing audio file to the CD project by clicking the import audio file button (![icon]) or using drag and drop from the Windows Explorer. If you click the import button, a standard file open dialog box appears (see [Loading Audio Files](#)) where you can selected the audio file or files you wish to add to the CD project.

6.1.2   Adding the Content of an Editing Window

The content (or parts of the content) of an editing window can be added as a track in the CD project without first creating an audio file:

1. Select the region in the editing window you wish to add as an audio track on the CD.
2. Click the left mouse button anywhere on the highlighted area and keep the mouse button down.
3. Move the mouse cursor to the track list in the CD project window.
4. Release the mouse button.
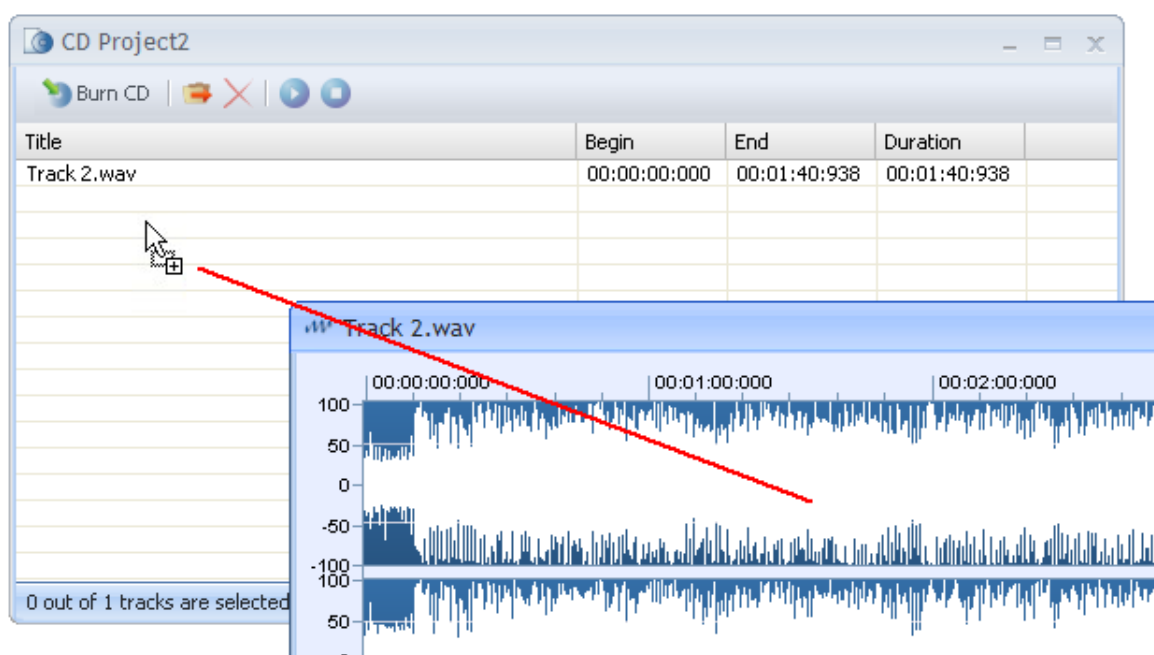
Adding the selected region of an editing window to the CD project using drag and drop

6.1.3   Burning the CD

When you are finished assembling your CD project, click the button labelled *Burn CD* from the toolbar of the CD Project window. The *CD Burner* dialog appears:
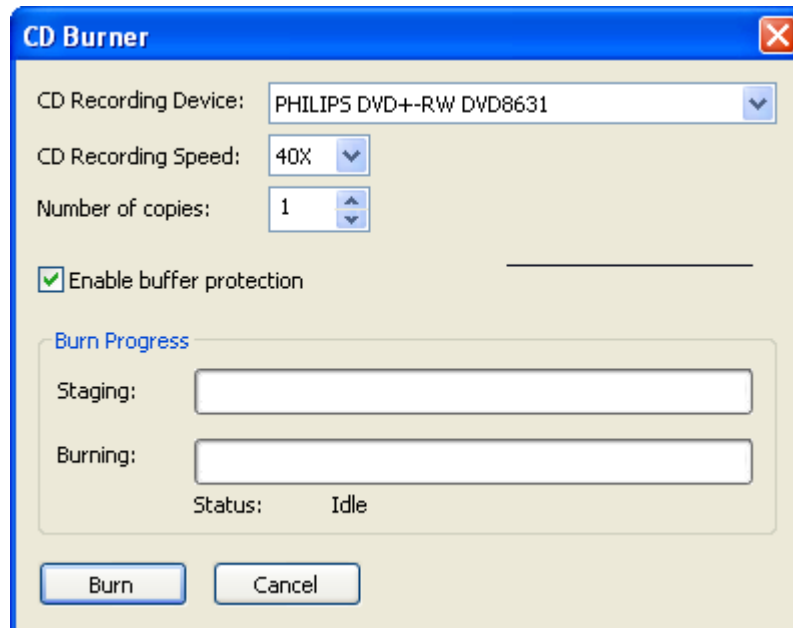
The CD burner dialog allows you to select a CD recording device, recording speed and the number of copies.

Insert a blank CD into the CD recording device. If you have several CD recording devices installed on your computer, make sure you choose the correct one from the *CD Recording Device* list. You can choose among different recording speeds and set the number of copies to burn. It is recommended to keep the *Enable buffer protection* option enabled, if supported (if not, the check box appears inactive). Click the button labelled *Burn* to start burning.

6.2 Importing Audio Tracks from CDs

It is possible to digitally import audio data from audio CDs with most CD-ROM readers. To import one or more audio tracks:

1. Select Import Tracks from Audio CD... from the File menu. The CD Track Extraction dialog box appears.
2. Choose the device that contains your source audio CD from the CD-ROM Device drop-down list.
3. Select the track or the tracks you wish to import from the track list.
4. Click the button labelled Extract.
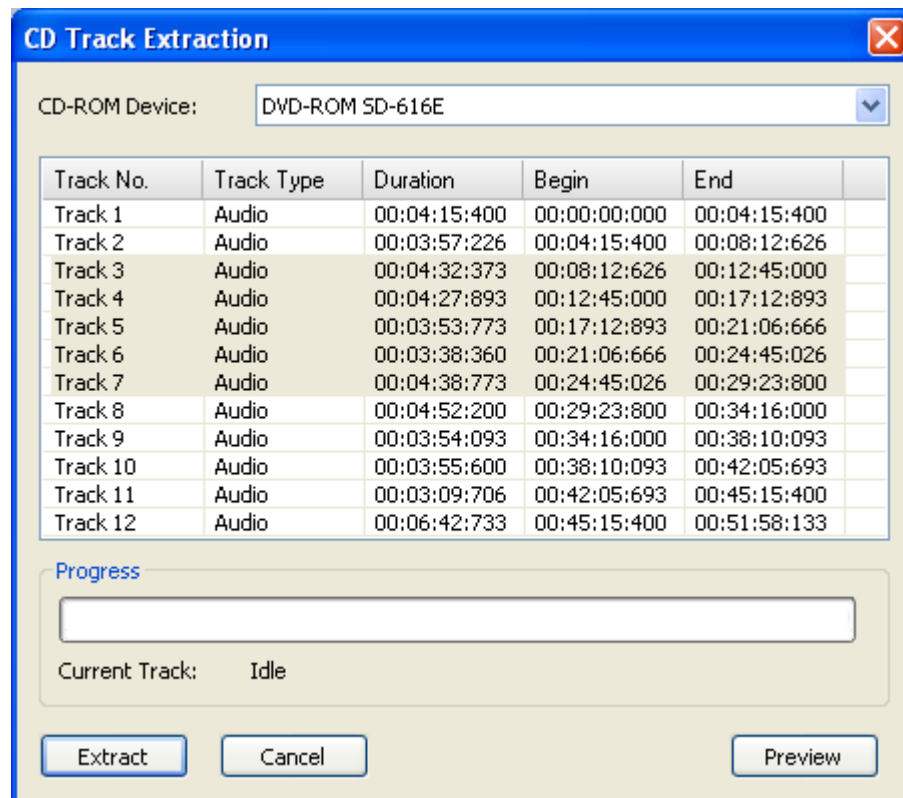
The CD Track Extraction dialog box

The imported audio tracks appear as separate editing windows in the Acoustica workspace.

> **Note** You can listen to the content of a track by clicking the button labelled Preview

7 Using the Cleaning Wizard

If you are new to digital audio recording, the Cleaning Wizard simplifies LP or cassette to CD transfers by guiding you through all the steps from recordings, track splitting, restoration and CD burning. To open the Cleaning Wizard, select *File | Cleaning Wizard...* and the following window appears:
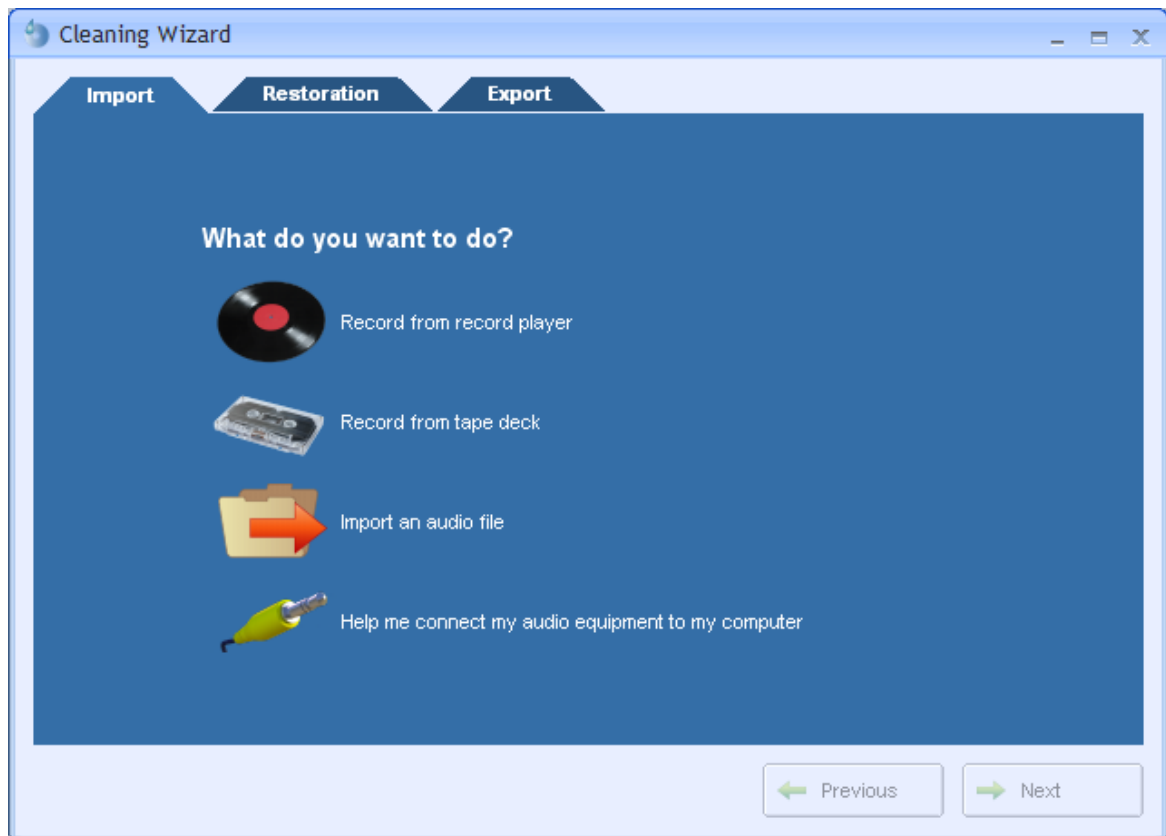
The Cleaning Wizard window

> **Note**   The Cleaning Wizard leads you step by step through the process of
> transferring your analog audio to CD. However, if you can choose to go back
> to an earlier step or skip one or more steps at any time by clicking at the
> Import, Restoration or Export tabs at the top of the wizard.

7.1   The Import Page

The import page is the first step in the Cleaning Wizard. From here you can choose to
record an LP, record a cassette, import an audio file or get help about connecting your
audio equipment. Click on one of the options to proceed.

7.1.1   Record Audio

If you choose to record from either LP or cassette the Cleaning Wizard will proceed to
the recording page:
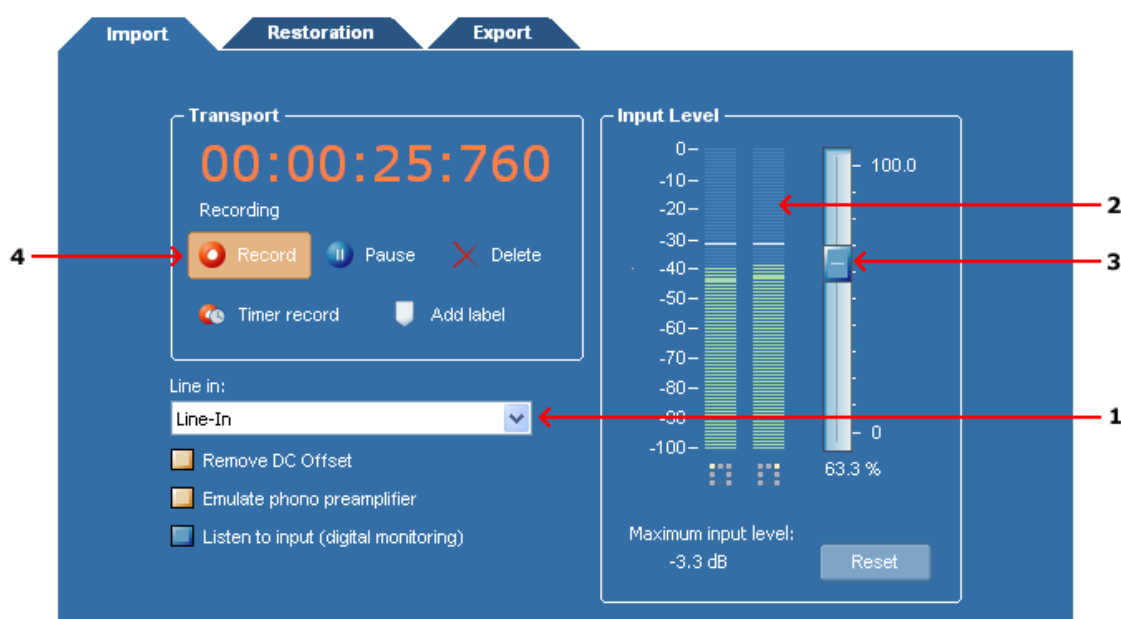
The recording page in Acoustica 4.

The input level meter (2) shows the current input level. If you have connected your audio equipment and started playback, the meter should show a constantly changing input level. If the level is low and not changing, there is probably something wrong with the connection or the wrong input line is selected. You can usually choose between several different input lines, like *microphone* or *line in*. The Cleaning Wizard tries to set *Line In* as input line automatically, however, you should check the input line setting (1) before proceeding. If you are uncertain, please try through the options and see if there is any response on the input level meter (2).

Recording Step by Step
1. Make sure your audio equipment is properly connected to your computer
2. Check that the input line (1) is correctly set. In most cases, the line input should be set to *Line In*
3. Check that the input level is in the correct range. You can use easily check the input level using input level meter (2). You can adjust the input level using the input level slider (3). The meter should never go up to 0 dB, otherwise digital clipping will be introduced. Check with the loudest part of the record or cassette tape you are recording that the input level meter doesn't go higher than about -6 dB.
4. Click the button labelled Record (4) to start the recording.
5. Press play on your tape deck or record player.
6. Press the Next button in the lower right part of the Cleaning Wizard when you have recorded the whole record or cassette tape.

> **Note** The recording page has several advanced features not mentioned here. Please see the chapter <u>Recording through the Sound Card</u> for more details about the advanced features such as timer record, digital monitoring and DC offset correction.

7.1.2 Import Files

If you choose to import an existing audio file, a file browser window appears where you can select the audio file you want to open.
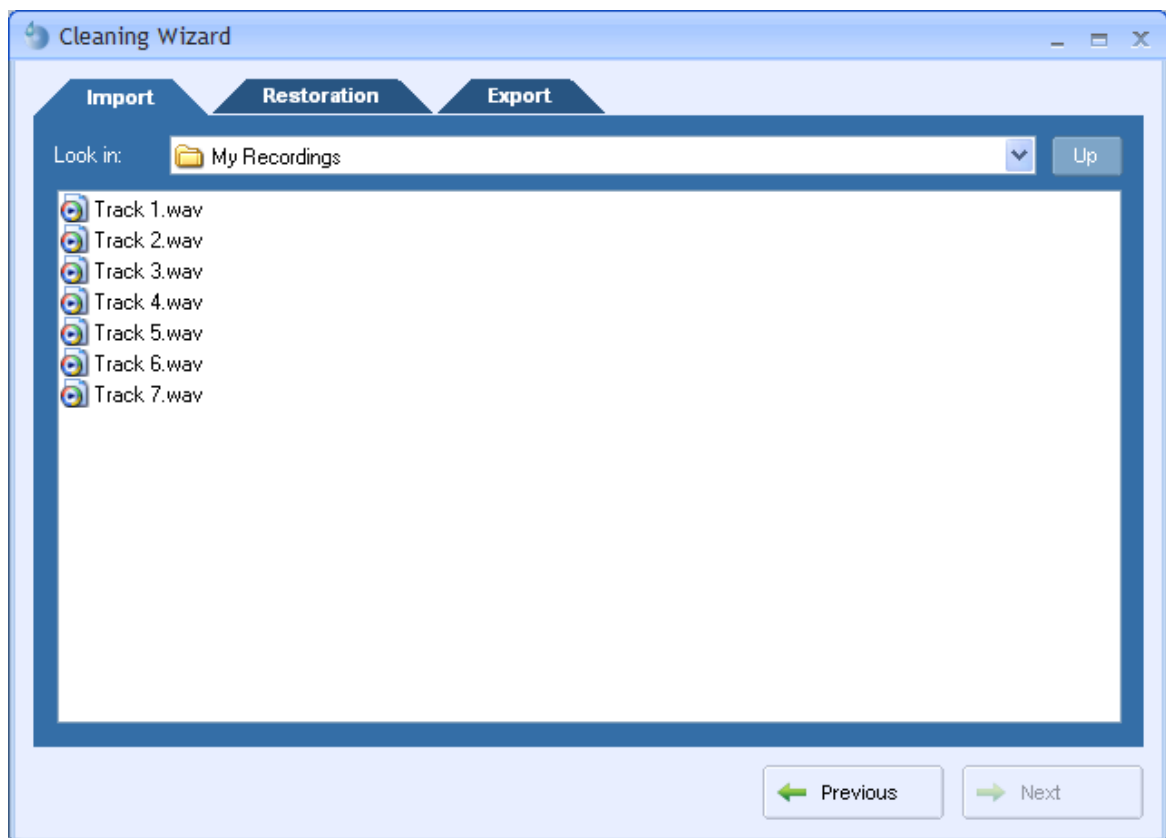
The File Import page in the Cleaning Wizard.

To import an audio file, please do the following:

1. Choose the folder in which your file is located from the Look in drop-down list.
2. Click the audio file you wish to open and click the button labelled Next.

7.2   The Restoration Page

The *Restoration Page* allows you to adjust the settings of the audio restoration tools and split the recording into several tracks.
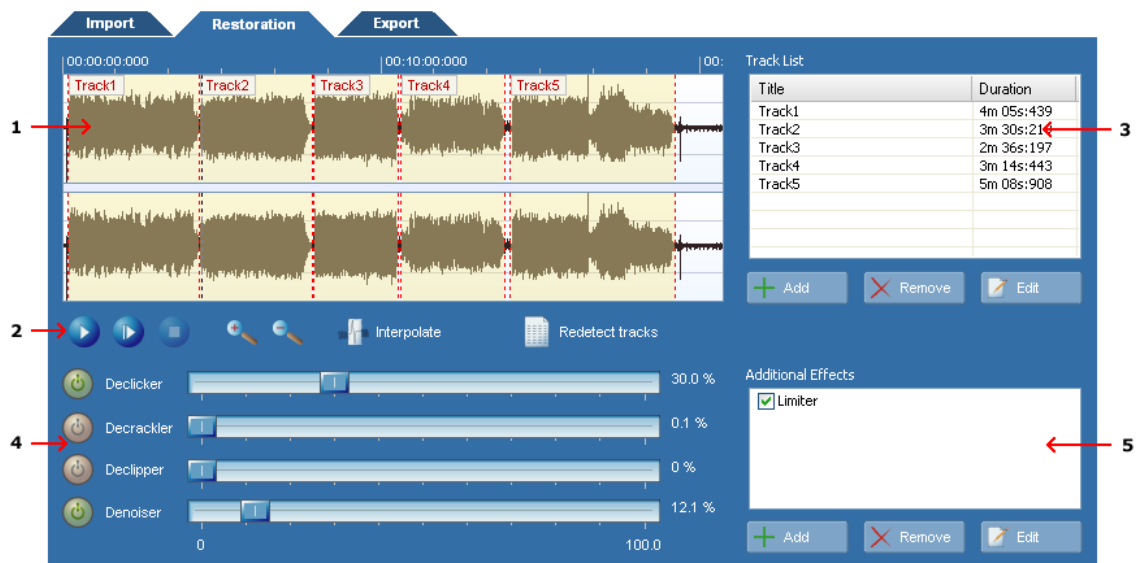
The Restoration Page contains a waveform view of the recording and list of the tracks, as well as audio restoration and processing options.

1. The waveform view shows you a graphical representation of the recording. Tracks are indicated with a different color.
2. The transport bar allows you to control playback.
3. The track list shows you the tracks defined. The Cleaning Wizard automatically suggests tracks, however, you can easily add, move or remove track markers.
4. The audio restoration tools, declicker, decrackler, declipper and denoiser. You can adjust the amount of restoration using the sliders and activate or deactivate a tool using the on / off buttons.
5. You can add further effects and processing tools, like equalizing or reverb. You can choose among all the internal tools and effects as well as VST and DirectX plug-ins.

7.2.1   Track Splitting

The Cleaning Wizard automatically searches for pauses and suggests tracks when recording or importing audio files. However, if the recording is very noise or tracks are blended seamlessly into each other, the tracks suggested by the Cleaning Wizard might not be identical to the original tracks on the source record or cassette.

Changing the Region of an Existing Track
1. Move the mouse cursor to the beginning or the ending of a track. The mouse cursor turns into a left-right arrow.

2. Keep the left mouse button pressed while moving the mouse cursor to the new position.
3. Release the mouse button.

Adding a track
1. Move the mouse cursor the beginning of the track you want to add in the waveform view.
2. Keep the left mouse button pressed while moving the mouse cursor to the end of the track.
3. Release the mouse button.
4. Click the button labelled *Add Track.*

Removing a Track
1. Click the track you want to remove in the track list.
2. Press the *Delete* key on your keyboard.

Renaming a Track
1. Click the track you want to rename in the track list.
2. Wait more than two seconds without moving the mouse cursor or press the F2 key on your keyboard.
3. The entry in the track list turns into an edit box. Enter the new name of the track.

7.2.2   Restoration

There are four restoration tools integrated into the Cleaning Wizard:

- **Declicker**
  Removes loud clicks and pops.

- **Decrackler**
  Removes short but frequent clicks, referred to as crackle.

- **Declipper**
  Restores recordings that suffer from analog or digital clipping.

- **Denoiser**
  Removes static noise like tape hiss.
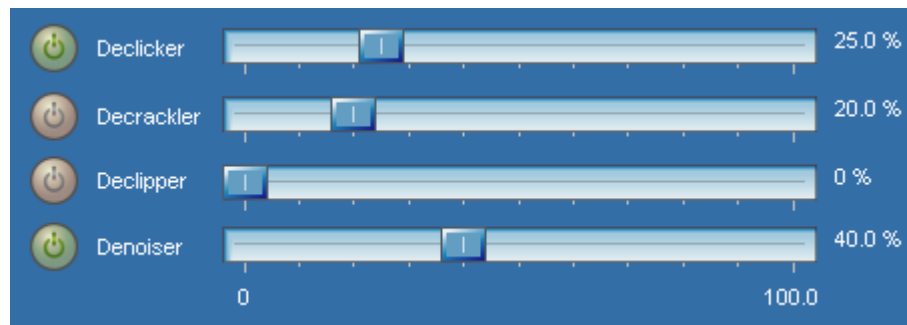
The restoration tools in the Cleaning Wizard

You can adjust the effect of each tool by moving the sliders in the range from 0% (no effect) to 100% (full effect). Furthermore, each tool can be activated or deactivated by clicking the on / off button left to the slider.

Playing Restored Tracks

The restoration tools in the Cleaning Wizard are processed in real time during playback so that you can listen to the effect of different restoration settings during playback. You can control the playback from the transport buttons in the Restoration Page:
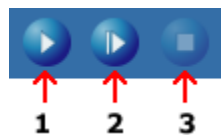
The transport button bar in the Restoration Page gives you full control over the audio playback.

Start the playback by clicking the play button (1). The playback will start from the current cursor position in the waveform. You can also use the play all button (2) to play the complete recording. Stop the playback by clicking the stop button (3).

7.2.3   Further Editing and Processing

The Restoration Page in the Cleaning Wizard also lets you add further effects (or processing tools) to the recording. You can choose among all the internal effects as well as VST or DirectX effects. When you play the recording in the Restoration Page, the effects are processed in real time so that you can check the results immediately.

Adding Further Effects
1. Click the button labelled *Add Effect* below the additional effects list.
2. Select the effect you want to add to the effects list.

Editing the Effect Parameters
1. Double click the effect in the additional effects list.
2. The effect settings page is shown.

3. Make your changes and close the window when done.

Removing an Effect from the List
1. Click the effect you want to remove.
2. Press the *Delete* button.

7.3   The Export Page

You can export your cleaned up tracks to audio files or burn them directly on a CD. Please choose one of the two.

7.3.1   Burn a CD

If you choose to burn a CD, the *CD Burner* dialog appears:
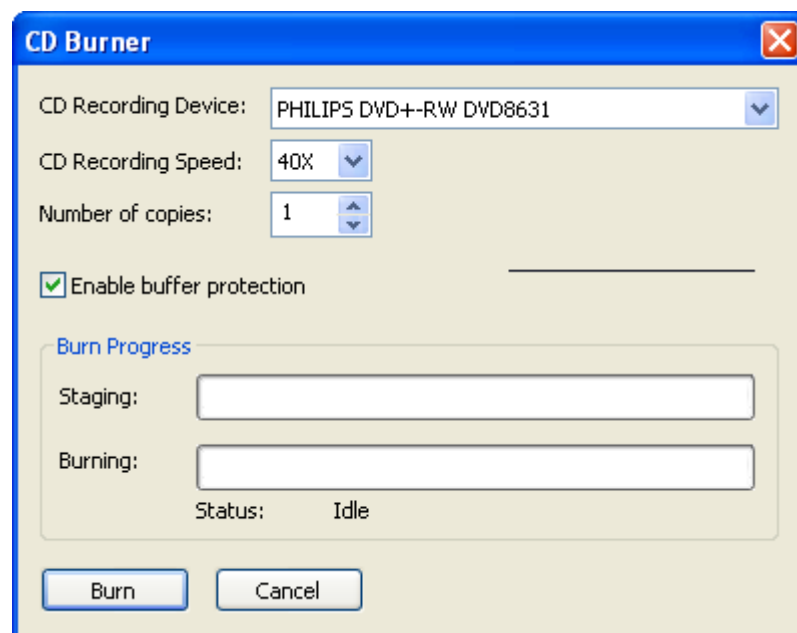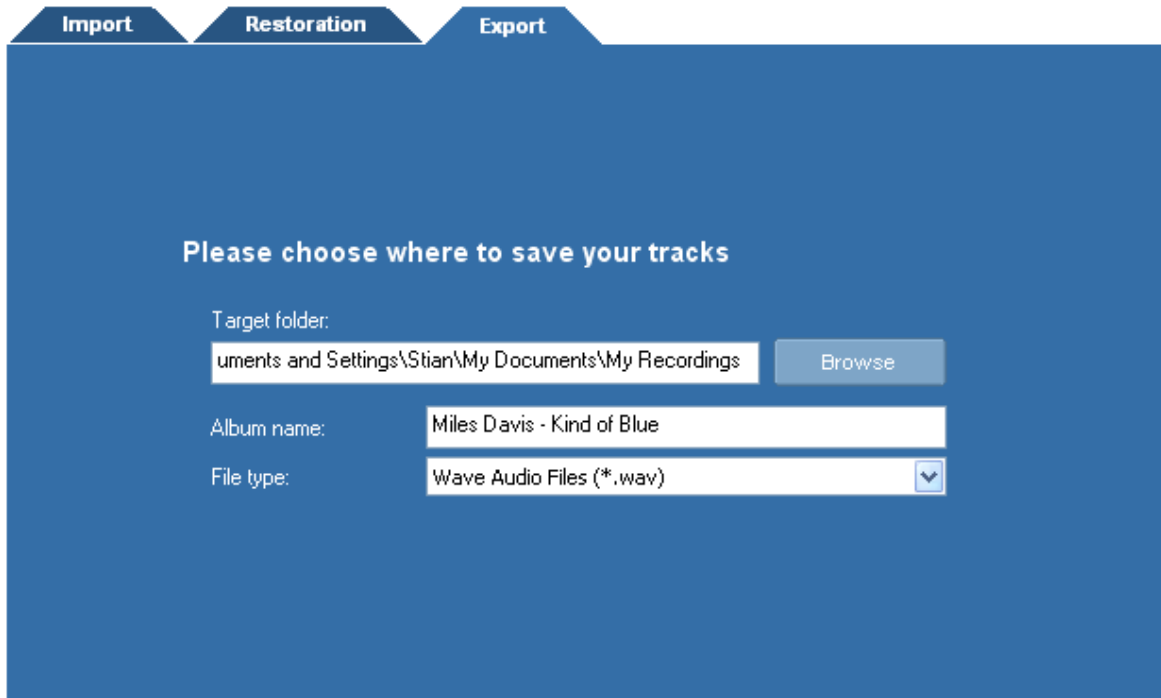
The CD burner dialog allows you to select a CD recording device, recording speed and the number of copies.

Insert a blank CD into the CD recording device. If you have several CD recording devices installed on your computer, make sure you choose the correct one from the *CD Recording Device* list. You can choose among different recording speeds and set the number of copies to burn. It is recommended to keep the *Enable buffer protection* option enabled, if supported (if not, the check box appears inactive). Click the button labelled *Burn* to start burning.

7.3.2   Export to Audio Files

If you choose to export your tracks to audio files, the *File Export Page* appears:

The File Export Page in the Cleaning Wizard

You can choose a destination directory for your tracks (1), an album name (2) and the file format of the exported tracks (3). During the export, an directory will be created with the album name and the tracks are written to audio files with the name of the tracks within the album directory.

8   Audio Analysis

Acoustica features a range of tools that allows you to study your recorded material in the time domain (see Time Domain), frequency domain (see Frequency Domain) or a combination of these (see Combining Time and Frequency).

8.1   Time Domain

The normal wave plot shown when making a recording in Acoustica is a time domain representation of the signal. When recording, Acoustica has taken samples of the signal at certain intervals, quantized them, and stored them as series of digitized values. The wave plot is the result of drawing these samples on the screen with the time evolving along the horizontal axis.
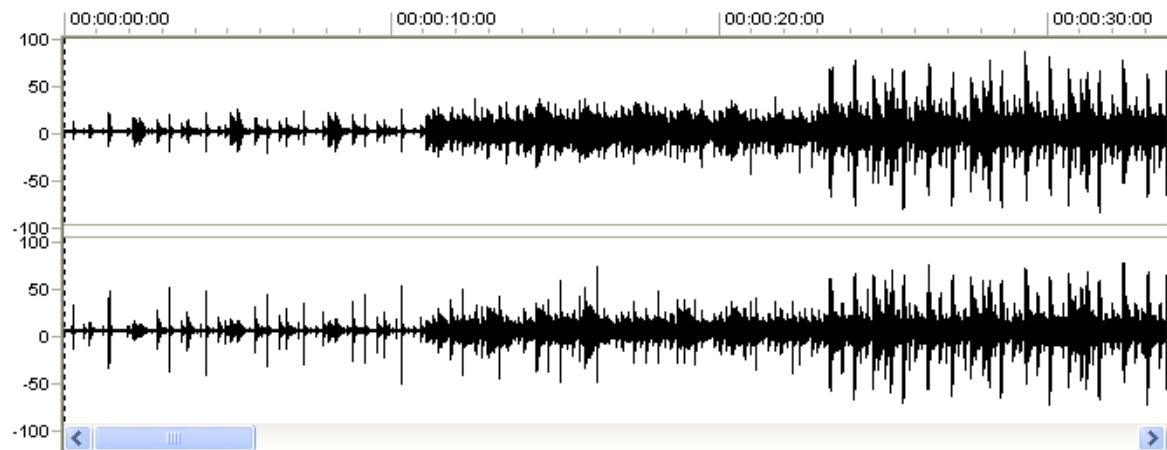
A plot of a signal represented in the time domain

8.2   Frequency Domain

All natural sounds can be described as an infinite sum of sine functions. The frequency of a sine function is related to what we sense as pitch. Our ears are not able to hear frequencies above 20kHz (a sine function with 20 thousand completed wavelengths within one second), so the mentioned infinite sum turns into a finite sum which is possible to handle on a computer. The signal in the frequency domain is represented through the weight of each sine function needed to recreate the signal, rather than the sampled values from the time series. These weights are visualized in Acoustica by selecting *Analysis | Spectrum Analyzer*.
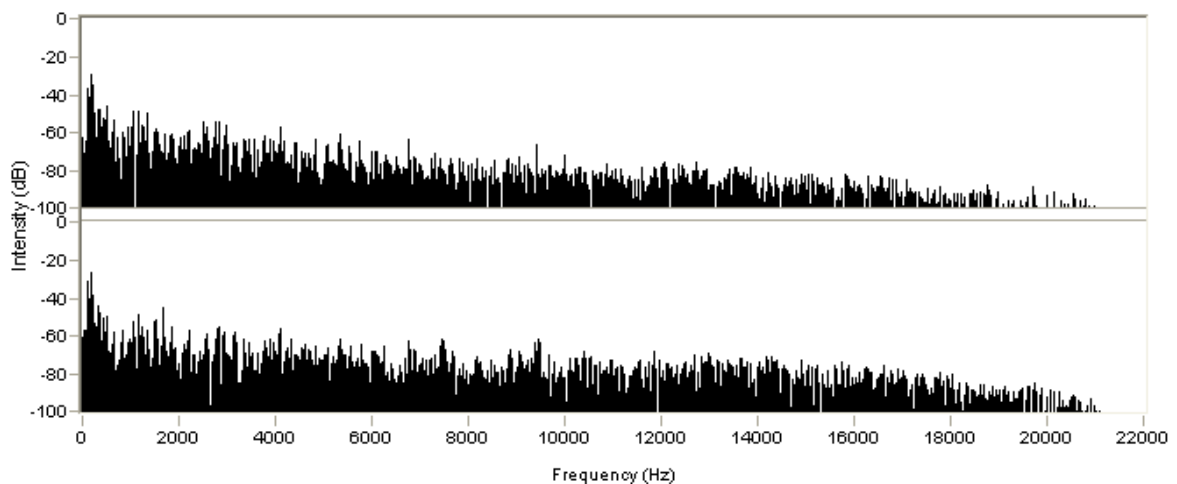
A plot of a signal represented in the frequency domain.

8.3   Combining Time and Frequency

We have a tool for examining the frequency content (the spectrum) of our recording and we have the normal wave plots for examining how our recording evolves over time. Is there a possibility to combine these features, in order to study how the frequency

content evolves over a period of time? Actually, Acoustica features two ways of displaying so called time-frequency plots. The spectrogram and the wavelet transform (based on the Morlet class of wavelets for the advanced reader). They differ mainly in the frequency scale. The spectrogram has a linear frequency scale whereas the wavelet transform has a logarithmic frequency scale.
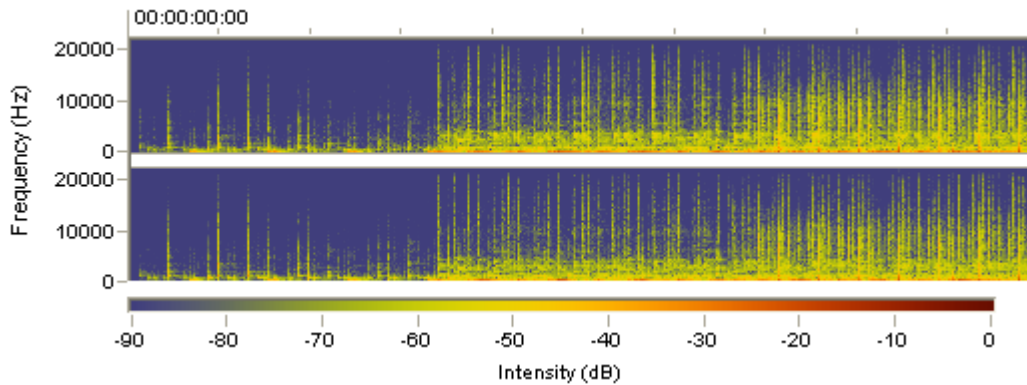
A spectrogram generated by Acoustica. The vertical axis represents the frequency, the horizontal represents time.

You can adjust the resolution and the color representation in both the spectrogram and the wavelet transform in the Preferences dialog box (see Changing the Preferences).

9 Preferences and Device Settings

9.1 Device Settings

Window's multimedia system allows several different audio cards being installed at the same time. You can select which sound card to use by selecting *Device Settings...* from the *Options* menu. Choose input and output devices from the drop-down lists in the device settings dialog to change the current configuration and click the button labelled *Ok*.

9.2 Changing the Preferences

You can set your personal preferences with the command *Preferences...* in the *Options* menu.
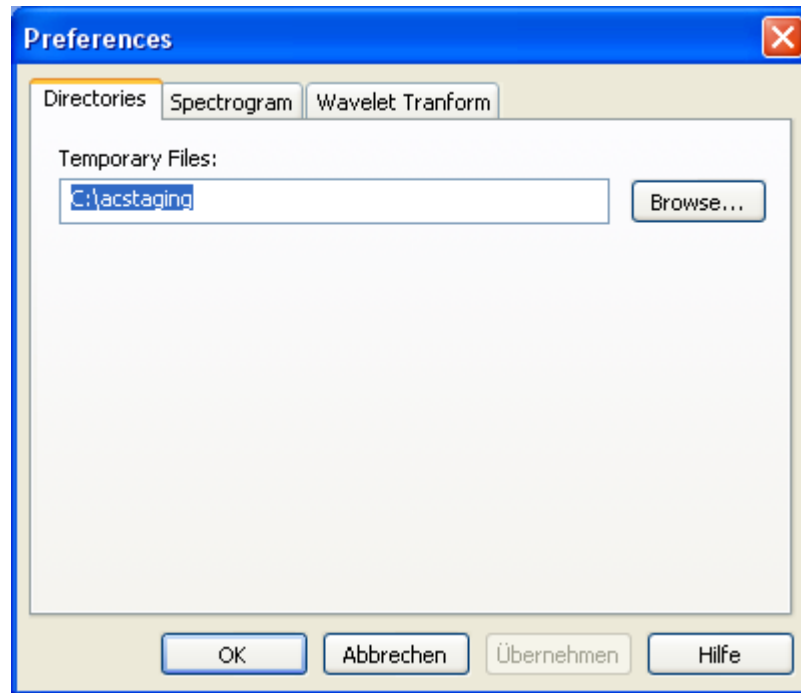
The Preferences dialog box

The preferences are organized in different pages. Only one page is shown at a time and you can show another page by clicking the tabs in the upper part of the dialog box. The following pages are available:

- Directories
- Spectrogram
- Wavelet Transform

9.2.1   The Directories Page

The *Directories* page contains only one setting, the directory for all temporary files created by Acoustica. You should set this directory path to a directory on a fast hard disk with sufficient free space. Acoustica works extensively with temporary files and the speed of the programs depends to a large degree on the speed of the hard disk where the temporary files are situated.
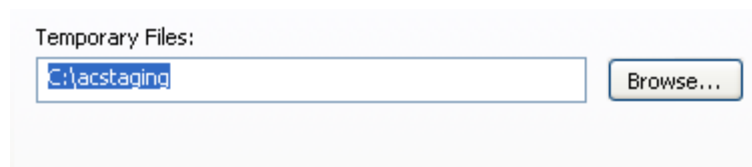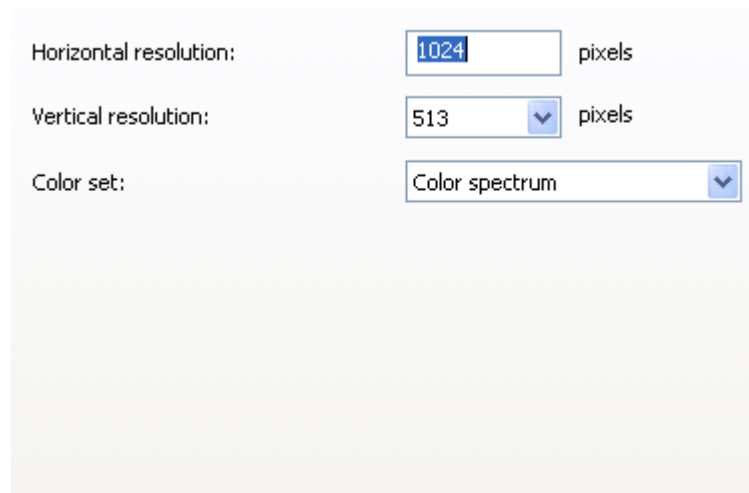
The directories preferences page

9.2.2   The Spectrogram Page

You can change the resolution and the color settings of the spectrogram in the *Spectrogram* preferences page. You can enter an arbitrary horizontal resolution, but the vertical resolution is limited to 65, 129, 257, 513 or 1025 pixels (due to the internal realization using the fast Fourier transform for the advanced reader).

You can choose the color settings from the *Color set* drop-down list. You can choose between:

- Color spectrum
- White on black
- Black on white
- Sepia

The spectrogram preferences page

9.2.3   The Wavelet Transform Page

You can change the resolution, the color settings and other properties of the wavelet transform in the *Wavelet Transform* preferences page. Both the horizontal and the vertical resolution in pixels can be set freely.

You can choose the color settings from the *Color set* drop-down list. You can choose between:

- Color spectrum
- White on black
- Black on white
- Sepia

In addition, you can set the frequency range to be analyzed in Hertz and the so-called coherence. A higher coherence value leads to an analysis with better a frequency resolution on behalf of the time resolution.

Horizontal resolution:	640	pixels
Vertical resolution:	200	pixels
Upper frequency limit:	20000	Hertz
Lower frequency limit:	40	Hertz
Coherence:	10	%
Color set:	Color spectrum	

The wavelet transform preferences page

Index