AudioLava Premium

User Guide

Acon Digital Media GmbH

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1 Introduction

AudioLava Premium is the ideal solution for restoring and recording high quality audio from LP to CD. AudioLava Premium automatically removes noise such as tape hiss or clicks and crackle on LP records. The user friendly Cleaning Wizard helps you to find the best way to bring your analog and digital recordings back to life in an impressive quality and guides you through all the steps from recording and track splitting to restoration and CD burning.

With AudioLava Premium you can easily enhance your audio files to a quality you have never heard before. The digital emulation of phono preamplifiers allows you to record your music signal without buying an additional phono preamplifier.

1.1 Requirements

Before you install AudioLava Premium, 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
- 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
- Windows Media Player 10 or the LAME encoder is required for mp3 export with a bit rate higher than 56 kbit / s

2 Connecting Your Stereo Equipment

This chapter describes how to connect your tape deck (cassette tape player) or your record player to your computer.

2.1 Connecting a Tape Deck (Cassette Tape)

If you have an external tape deck or an 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:
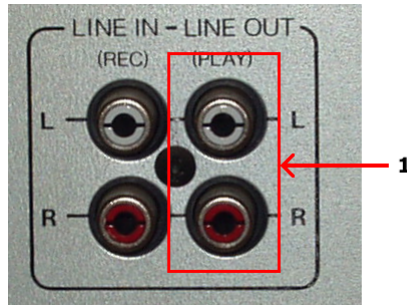
RCA connectors on a tape deck unit. Connect the sound card to the Line Out connector (1).

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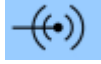. Connect the other end to the *Line In* connector of your computer. The *Line In* connector is either labelled as such or indicated with the following symbol: .

Connecting to a Sound Card or USB Audio Device with RCA connectors
External USB devices for audio input and ouput 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.

2.2   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*).
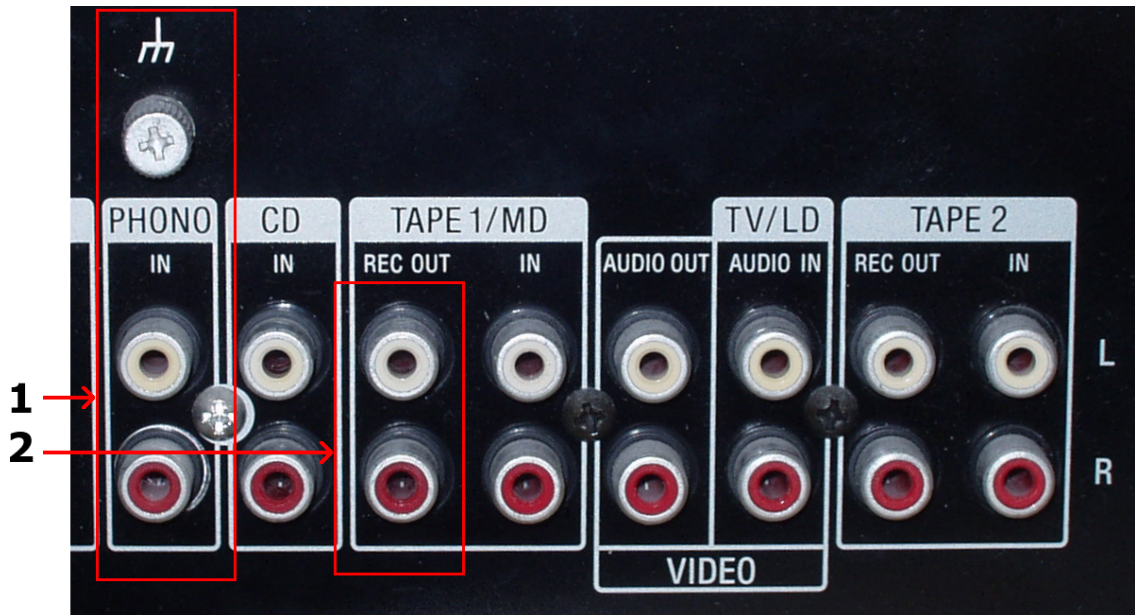
RCA connectors on an amplifier. Make sure the record player is properly connected to the phono input (1) and connect the sound card to the Rec Out connector (2).

2.2.1   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 ouput 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.

2.2.2   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 AudioLava Premium probably don't need any additional hardware. AudioLava Premium 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:
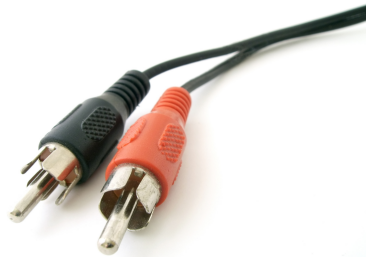
Cable with male RCA connectors.

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).
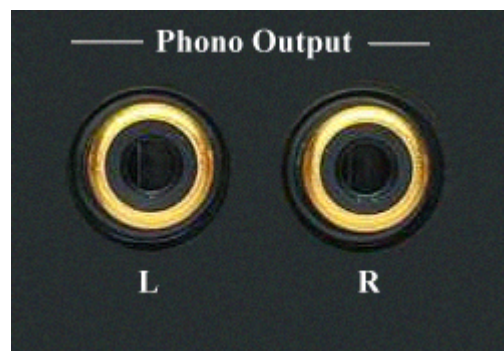
Connecting to a Sound Card with Mini Jack Connector

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:
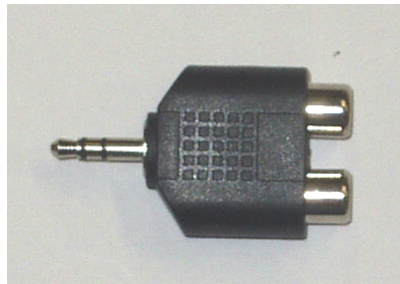
An RCA to mini jack adapter.

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 ouput 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.
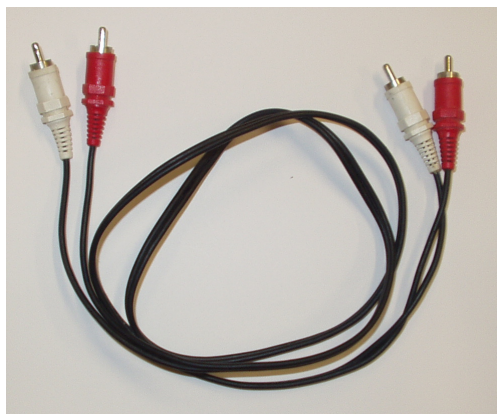
A connection cable with RCA connectors on both ends.

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.

3   Using the Cleaning Wizard

When you start AudioLava Premium, a Cleaning Wizard window is opened automatically. 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.

When you start AudioLava Premium or open the Cleaing Wizard, the following window appears:
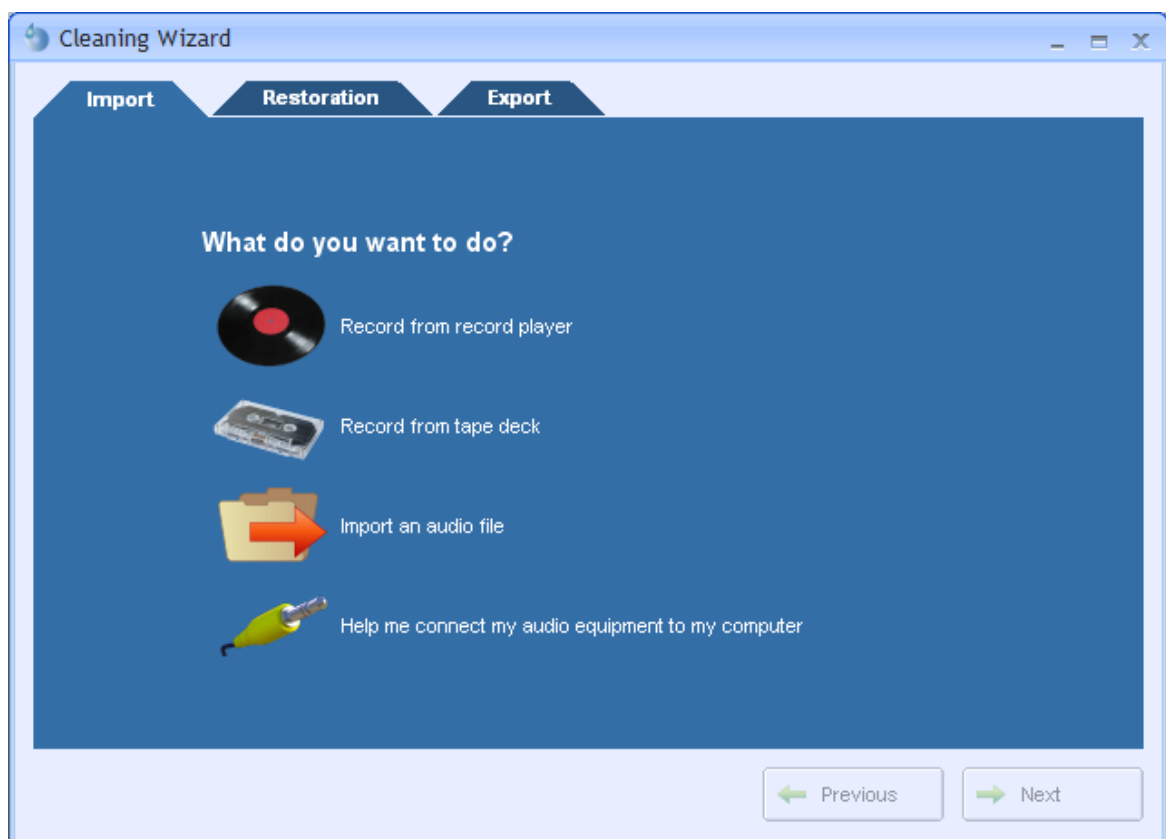
The Cleaning Wizard window

> **Note**   The Cleaning Wizard leads you step by step through the process of transfering 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.

3.1   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.

3.1.1   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 *quantisation*.

3.1.1.1   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 quantisation 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.

3.1.1.2   Quantisation

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 quantisation 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 quantasized 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.
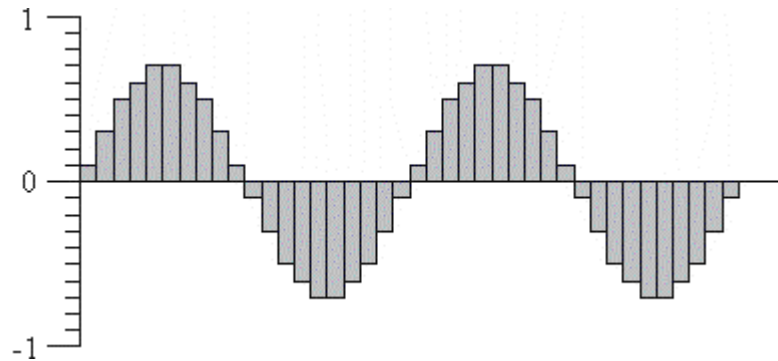
The digital representation of a sine wave.

3.1.1.3   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 AudioLava Premium, 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.
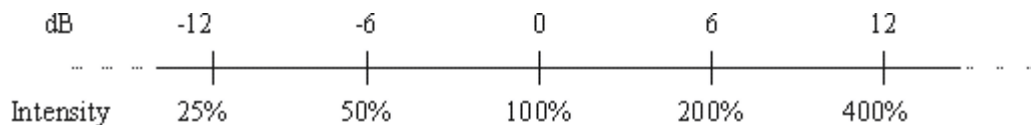
The decibel dB versus intensity change in percent

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

3.1.2   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 AudioLava Premium.

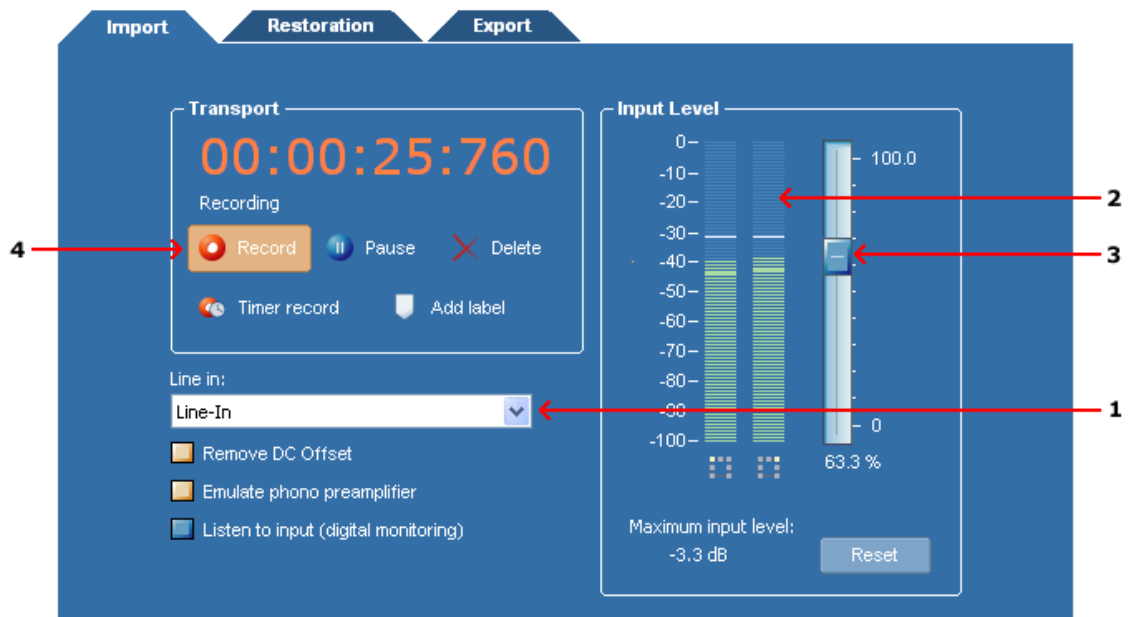The input level meter (2) shows the current input level. If you have connected your audio equipment and started playback, the meter should show a constantly changing input level. If the level is low and not changing, there is probably something wrong with the connection or the wrong input line is selected. 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.

3.1.3   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.

3.1.4   Advanced Recording Options

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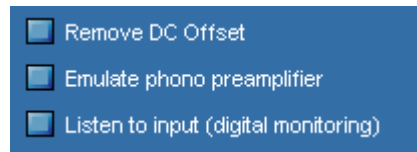
The advanced recording options in the recording dialog

- **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 inaudiable 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.

3.1.5   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.
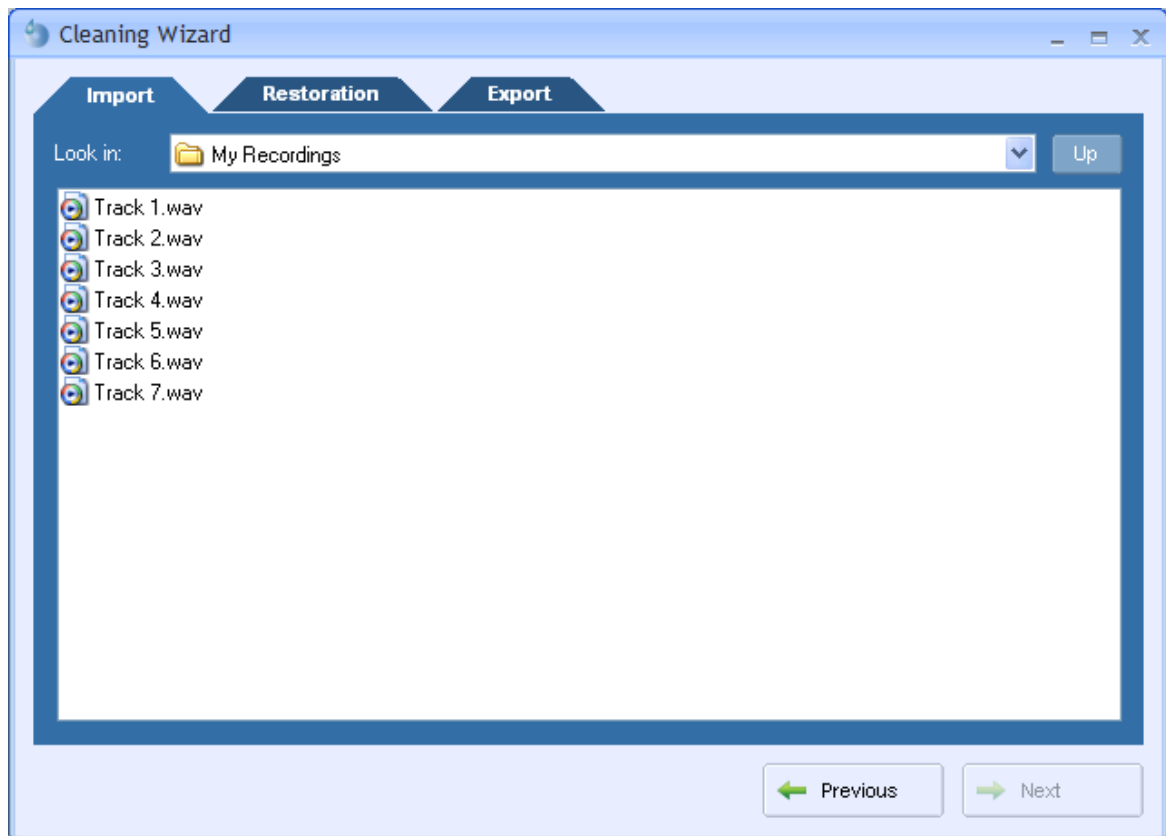
The File Import page in the Cleaning Wizard.

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.

3.2 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. By clicking the "Settings..." buttons to right of the adjustment sliders, you can adjust further and more advanced settings of the audio restoration tools.
6. 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.

3.2.1   Track Splitting

The Cleaning Wizard automatically searches for pauses 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.

3.2.2   Analysing Noise for Noise Reduction

If your recording contains pauses with pure noise without any other signal, the pause can be used to create a noise profile automatically. Please select a region containing noise only and click the button labelled "Noise Analysis" on the Restoration Page. The result of the analysis is automatically loaded in the profile graph of StudioDenoiser.

3.2.3   Restoration

There are four restoration tools integrated into the Cleaning Wizard:

- **Declicker (based on StudioDeclicker)**
  Removes loud clicks and pops.

- **Decrackler (based on StudioDeclicker)**
  Removes short but frequent clicks, referred to as crackle.

- **Declipper (based on StudioDeclipper)**
  Restores recordings that suffer from analog or digital clipping.

- **Denoiser (based on StudioDenoiser)**
  Removes static noise like tape hiss.
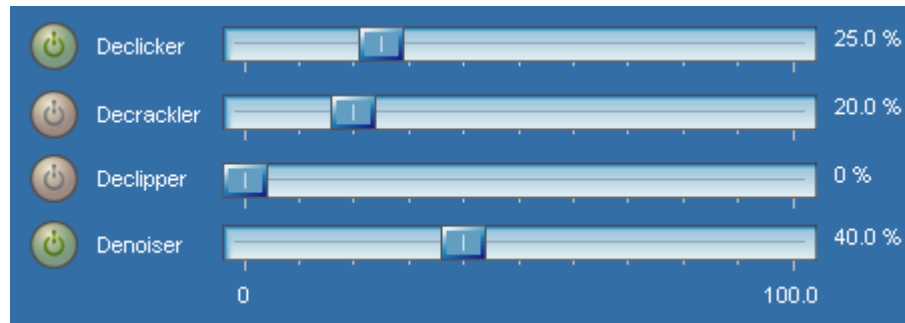
The restoration tools in the Cleaning Wizard

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. By clicking the "Settings..." buttons to right of the adjustment sliders, you can adjust further and more advanced settings of the audio restoration tools. The declicker and the decrackler settings are both identical to the settings of the StudioDeclicker tool. If you click the declipper settings, the StudioDeclipper settings opens and if you click the denoiser settings, the StudioDenoiser settings opens.

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).

3.2.4   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.
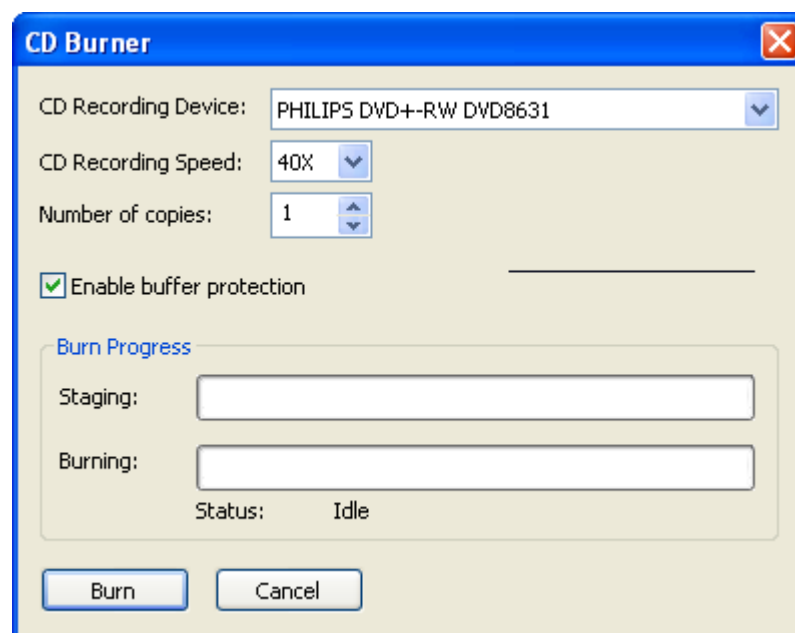
Removing an Effect from the List
1. Click the effect you want to remove.
2. Press the *Delete* button.

3.3   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.

3.3.1   Burn a CD

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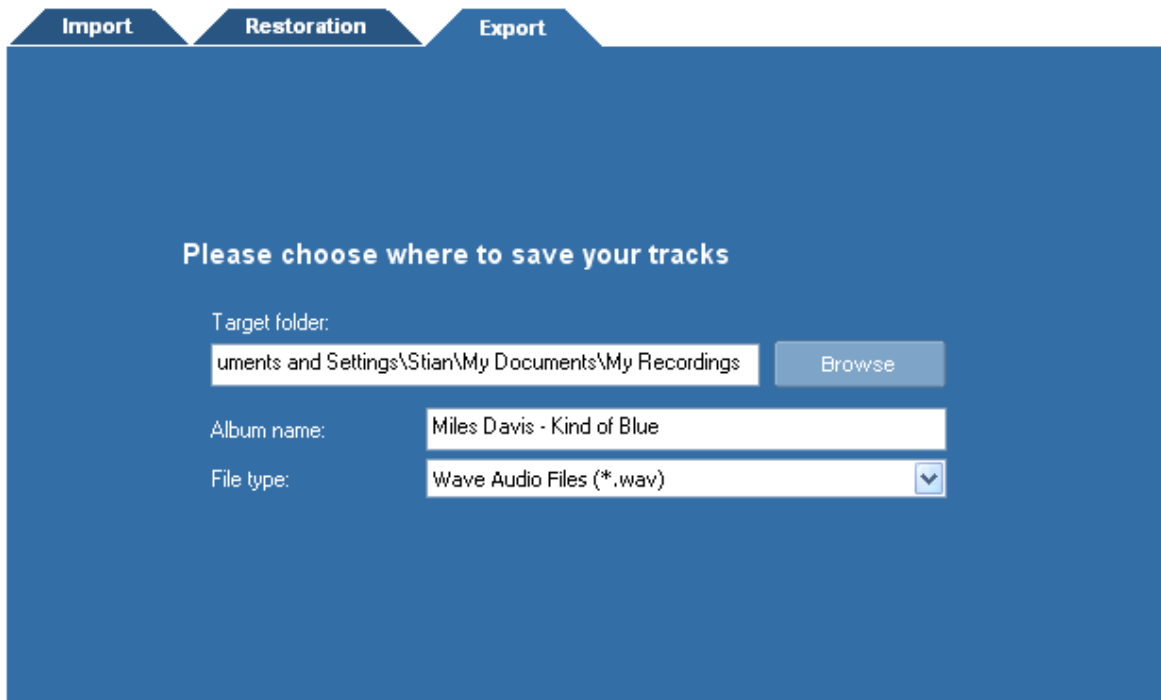
The CD burner dialog allows you to select a CD recording device, recording speed and the
number of copies.

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.

3.3.2   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.

4   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.

4.1   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.
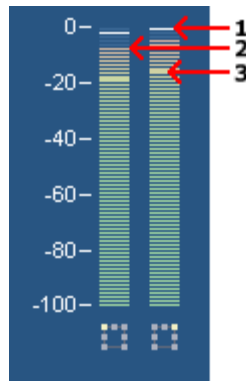
The level meter analyzer showing the peak hold value (1), peak value (2) and RMS value (3).

4.2   FFT Analyzer

The *FFT Analyzer* shows the frequency content of short analysis time frames. FFT stands for Fast Fourier Transform. FFT is an effecient way of calculating the frequency domain of a signal.
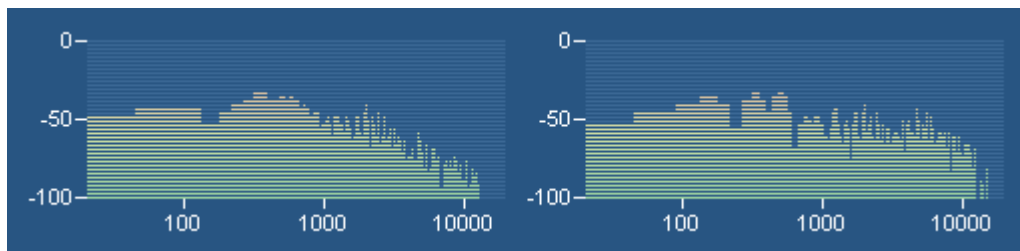
The FFT Analyzer shows the frequency content of the output audio signal.

4.3   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.
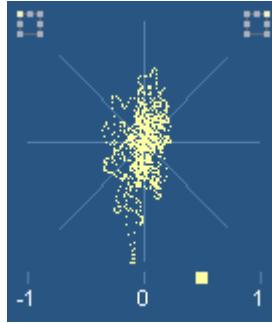
The phase correlation meter shows the releationship between the left and the right channel in a stereo recording.

4.4   Big Time Display

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

5   Audio Processing

In AudioLava Premium, 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.
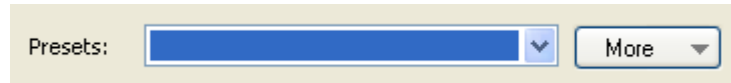
The Preset Manager found in most of the processing tools in AudioLava Premium

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.
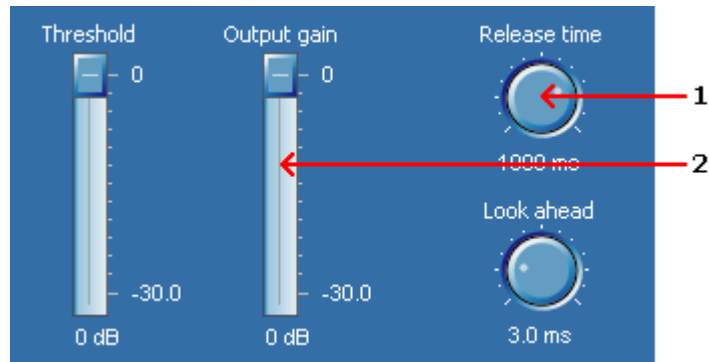
Examples of knobs (1) and level sliders (2).

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.
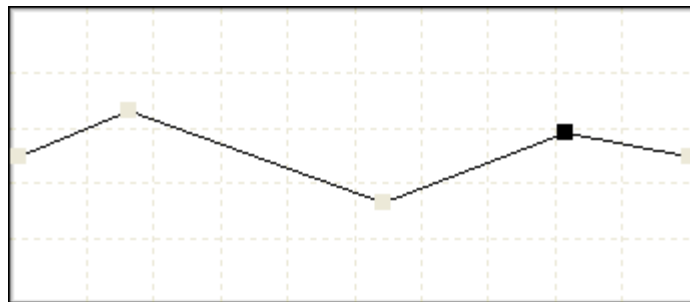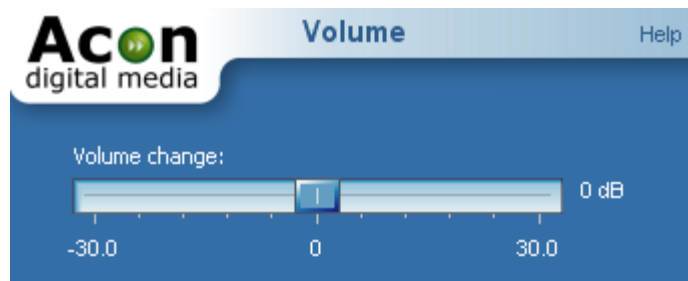

Examples of a curve edit control.

5.1   Audio Effects

5.1.1   Adjusting the Volume

The volume adjustment tool allows you to change the volume of the recording. 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   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.
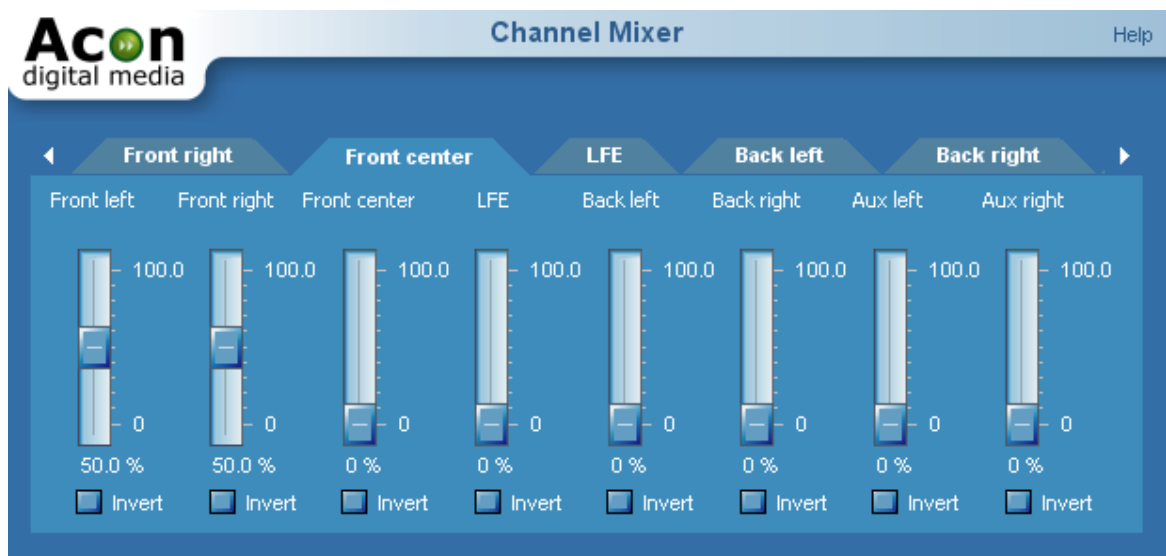
The Channel Mixer settings

5.1.3 Dynamic Processor

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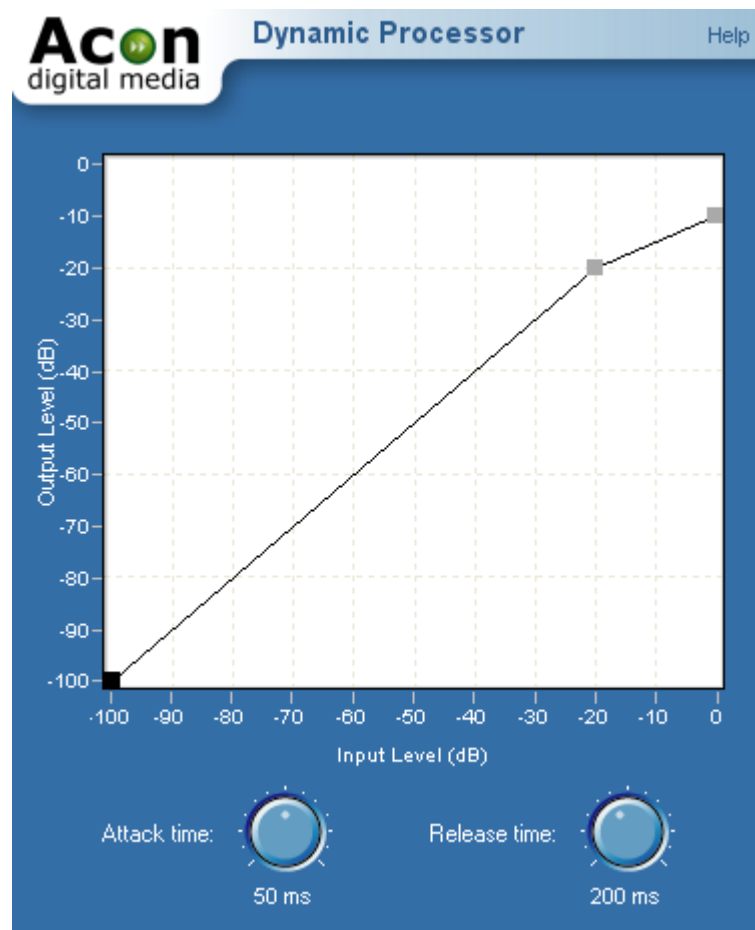sound engineer trying to maintain an as steady volume level as possible while doing a recording. When the input level increases he pulls down the volume slider, and he pulls it up when the input level decreases. A dynamic processor does the same thing automatically, only with a much faster reaction time.

The Dynamic Processor settings
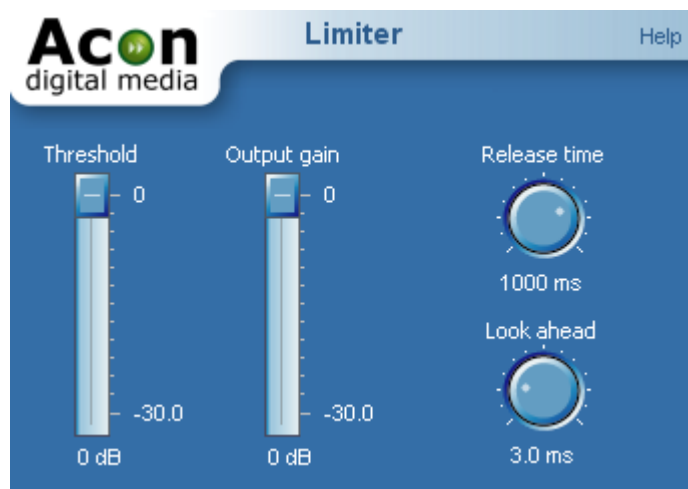
Settings

- 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.

5.1.4   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.

5.1.5   Echo

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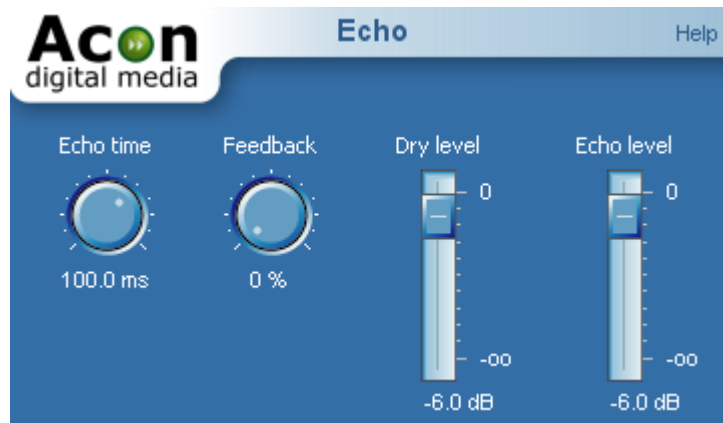
The Echo settings

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.

5.1.6   Reverb

The reverb simulates the reverberation effect of some natural surroundings.
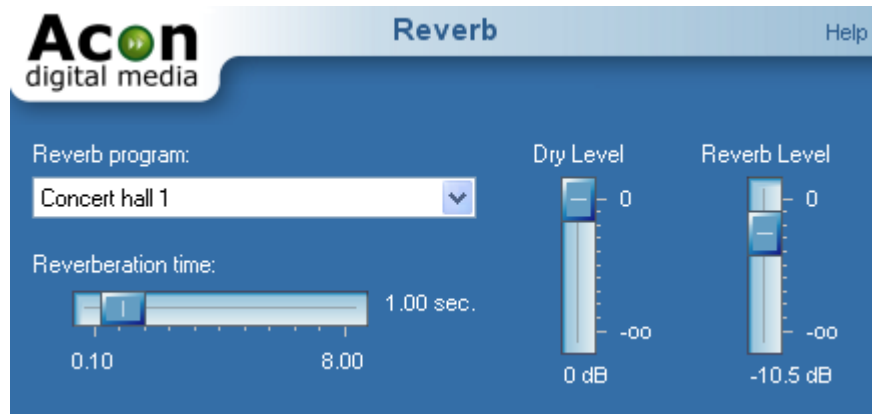
The Reverb settings

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, Stadion* 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.

5.1.7   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.
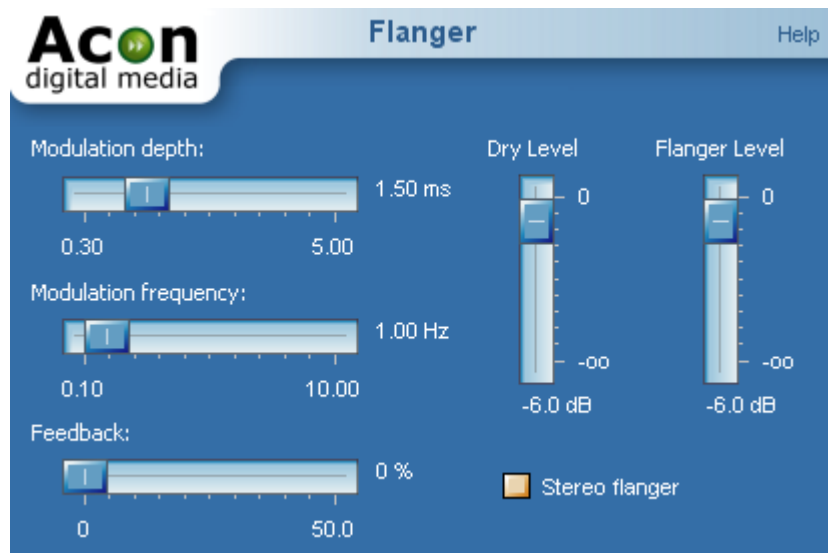
The Flanger settings

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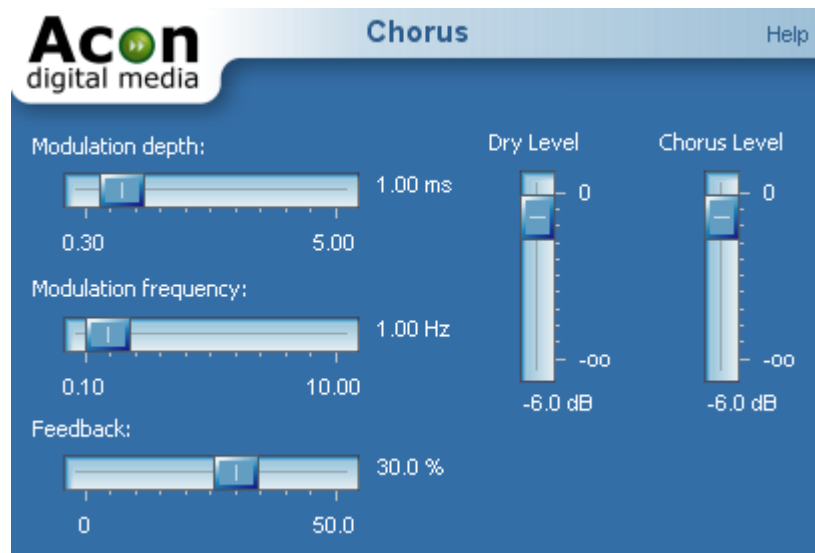
- **Dry Level**
  The amount of unprocessed signal in the output mix.

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

5.1.8   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.

5.1.9   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, AudioLava Premium 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.

5.1.10   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 AudioLava Premium time stretching algorithm is used to compensate for the change of length normally occurring during pitch shifting.
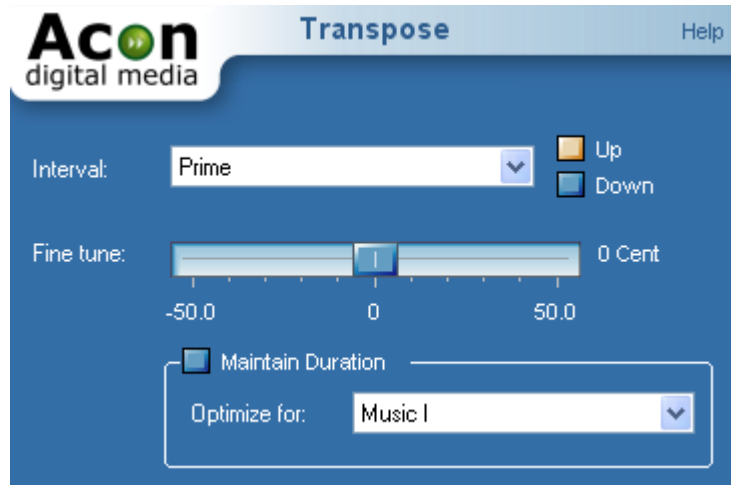
The Transpose settings

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.

5.1.11 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. AudioLava Premium 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

5.1.12   Reverse

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

5.2   Using Audio Plug-Ins

AudioLava Premium 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.

5.2.1   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*. AudioLava Premium then rescans the plug-ins on your computer and updates the menu.

5.2.2   Specifying VST Directories

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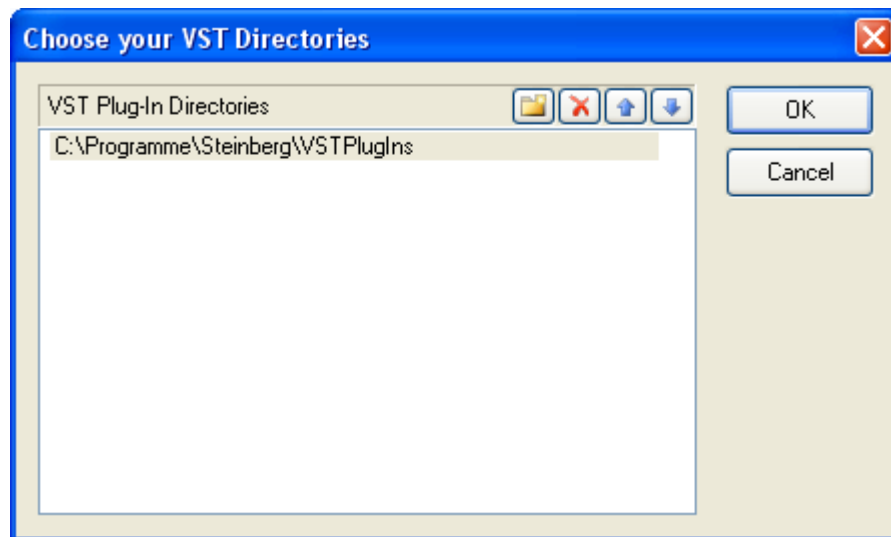
The VST Directories dialog lets you specify one or more VST directories to scan through.

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 [x].

5.3   Enhancement Tools

5.3.1   StudioDeclicker

StudioDeclicker is a tool specialized on removing impulsive noise such as clicks and crackle. These distortions are very frequently encountered on LP and 78 RPM records. StudioDeclicker contains two different algorithms to deal with clicks and crackle. The actual declicker algorithm eliminates large clicks and pops in the recording, while the decrackler algorithms eliminates the frequent, but short clicks that the human ear percieves as crackle. StudioDeclicker removes clicks by substituting the recorded signal in the short period of time during the click with a signal estimated using the undistorted audio surrounding each click.
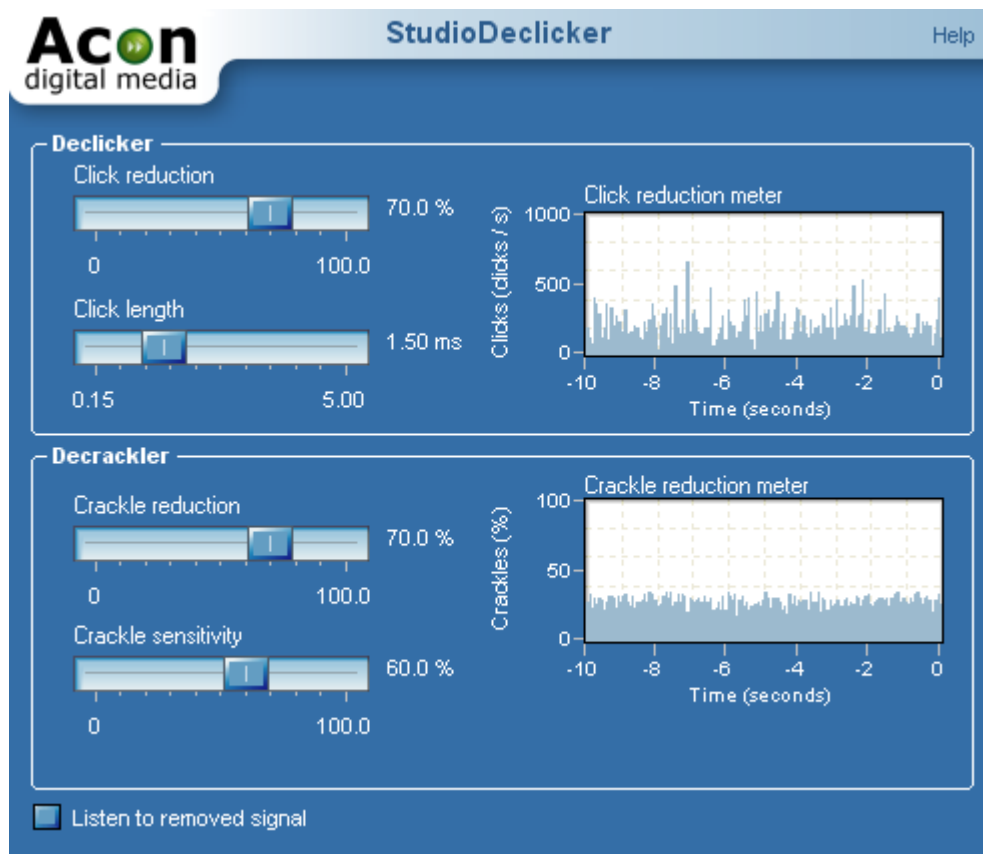
The StudioDeclicker user interface.

The StudioDeclicker user interface contains a click and a crackle reduction meter that give visual feedback of the restoration process. Both meters show a history of the reduction activity during the past ten seconds. The click reduction meter shows the number of clicks removed per second, whereas the crackle reduction meter shows the

percentage of input samples regarded as crackle distorted.

Settings

- **Click reduction**
  Sets the sensitivity of the declicker algorithm. Higher reduction levels result in more click reduction.

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

- **Crackle reduction**
  Sets the level of crackle noise to be removed by the decrackler algorithm. Higher crackle reduction levels result in more crackle reduction.

- **Crackle sensitivity**
  Sets the sensitivity of the decrackler algorithm. Higher crackle sensitivities result in more crackle reduction.

5.3.2   StudioDeclipper

StudioDeclipper 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. StudioDeclipper substitutes such distorted peaks by an estimation of the signal curve using almost the same mathematical methods as the StudioDeclicker when eliminating clicks.
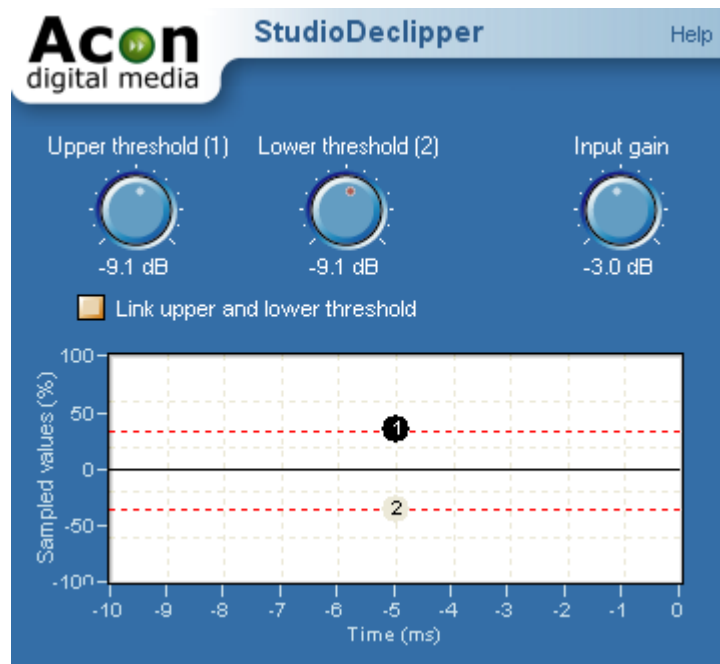
The StudioDeclipper user interface.

StudioDeclipper contains an oscilloscope view to visualize restoration. The oscilloscope shows the last ten milliseconds of the recovered audio signal. The most important parameters of the declipper are the upper and lower threshold levels. The declipper will substitute all recorded peaks above the upper and below the lower threshold value. The threshold values can be adjusted using their corresponding knob controls or directly from the oscilloscope view.

Settings

- **Upper threshold**
  All samples values above the upper threshold are substituted by a signal estimation.

- **Lower threshold**
  All samples values below the lower threshold are substituted by a signal estimation.

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

- **Link upper and lower threshold**
  Usually, the clipping introduced during recording will be symmetrical, which means that the upper and lower thresholds will have the same absolute value. By activating the upper and lower threshold link, the adjustment of the declipper is simplified in the case of symmetrical clipping.

5.3.3   StudioDenoiser

StudioDenoiser is a plug-in for broadband noise reduction. Because the algorithm takes the perceptual properties of the human hearing into account it achieves a high level of noise reduction with a minimum of audiable artifacts. The noise reduction algorithm is similar to the spectral subtraction technique. This means that the frequency distribution of the noise present (the *noise profile*) in the recording is needed.
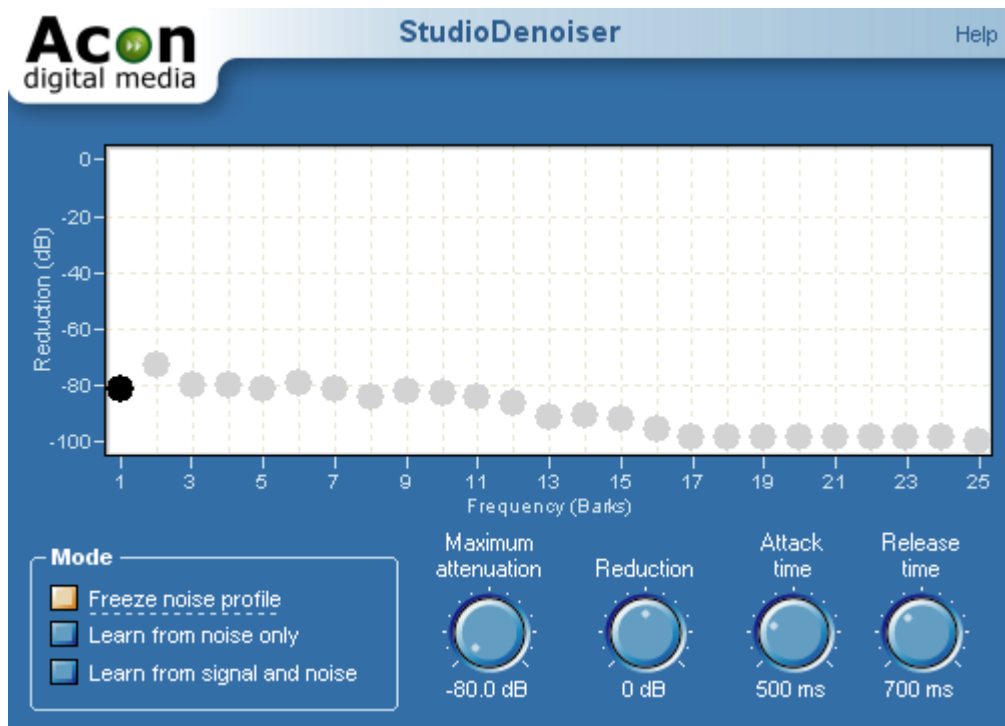
The StudioDeclicker user interface.

StudioDenoiser offers three ways of estimating the noise profile.

- **Estimation from Noise Signal**
  If you have parts of the recording containing only noise, you can automatically estimate the noise profile through analysis of a region containing noise only. Set the mode to "Learn from noise only" and play the part of the recording containing only noise. Select the "Freeze noise profile" when done.

- **Estimation from Noisy Audio Signal**
  The noise profile can also be estimated from the noisy audio signal. This method is not as accurate as the estimation from the pure noise signal, but if there are no parts available containing only noise, this is a good alternative. Furthermore, the results of

the estimation can be fine tuned by the user. Set the mode to "Learn from signal and noise" and start playing. After a couple of seconds, select the "Freeze noise profile".

- **Manual entry**
  Alternativly, the noise frequency distribution can be defined manually. It is recommended to perform an estimation from the noisy audio signal before manually editing the noise profile, because the estimation serves as a good starting point. Make sure the mode is set to "Freeze noise profile" before editing the noise profile. You can manually set the noise level of each frequency band by moving the circles with the mouse or by using the arrow keys.

Settings

- **Mode**
  Selects the working mode of the denoiser. "Freeze noise profile" should be selected during denoising. "Learn from noise only" and "Learn from signal and noise" should be selected only when estimating the noise profile.

- **Maximum attenuation**
  Maximum attenuation allows you to adjust a maximum attenuation factor for each frequency band. This parameter is also referred to as noise floor. By leaving a certain noise floor, you can mask artifacts from the noise reduction algorithm.

- **Reduction factor**
  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.

- **Attack time**
  The attack time is the response time of the noise suppresion when the signal level in a frequency band increases. Longer response times gives better noise reduction, but can in some cases lead to artifacts.

- **Release time**
  The release time is the response time of the noise suppresion when the signal level in a frequency band increases. Longer response times gives better noise reduction, but can in some cases lead to artifacts.

5.3.4 High Frequency Rebirth

Dull and lifeless recordings may be a result of lost high frequency content. By using the *High Frequency Rebirth* tool, 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.
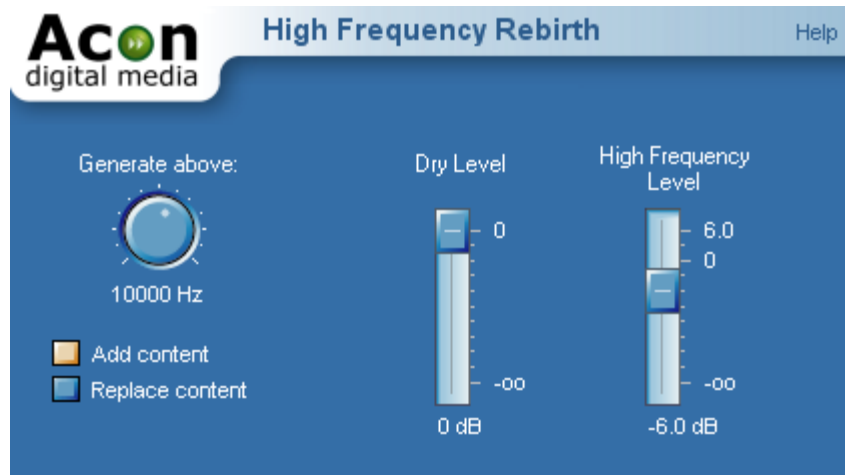
The High Frequency Rebirth settings

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.

5.3.5   Equalizer

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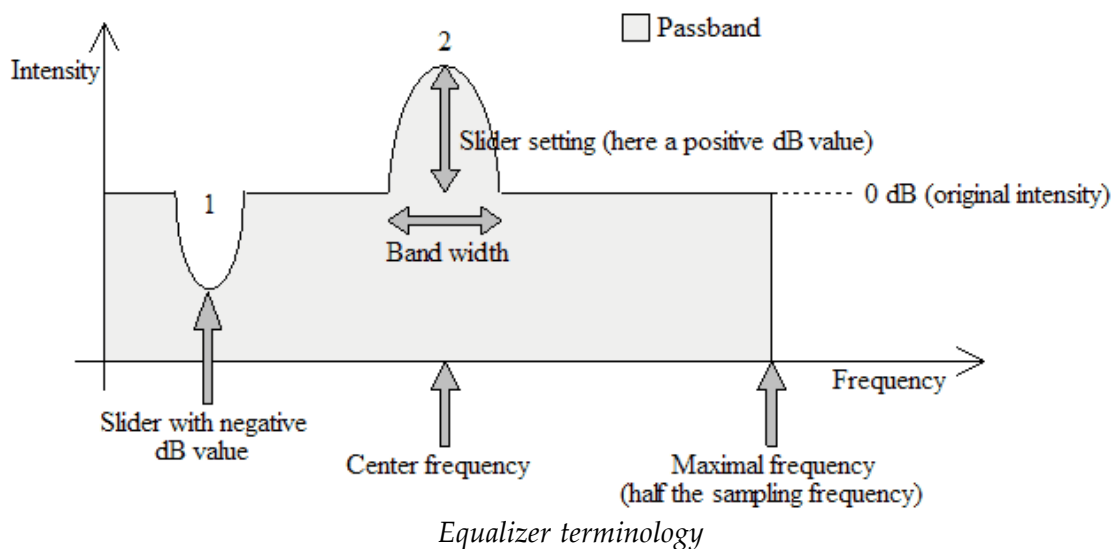
The lower part of the equalizer dialog box is used to display the filter response curve (gain versus frequency). The filter response curve is calculated mathematically and is updated as the parameters are altered.
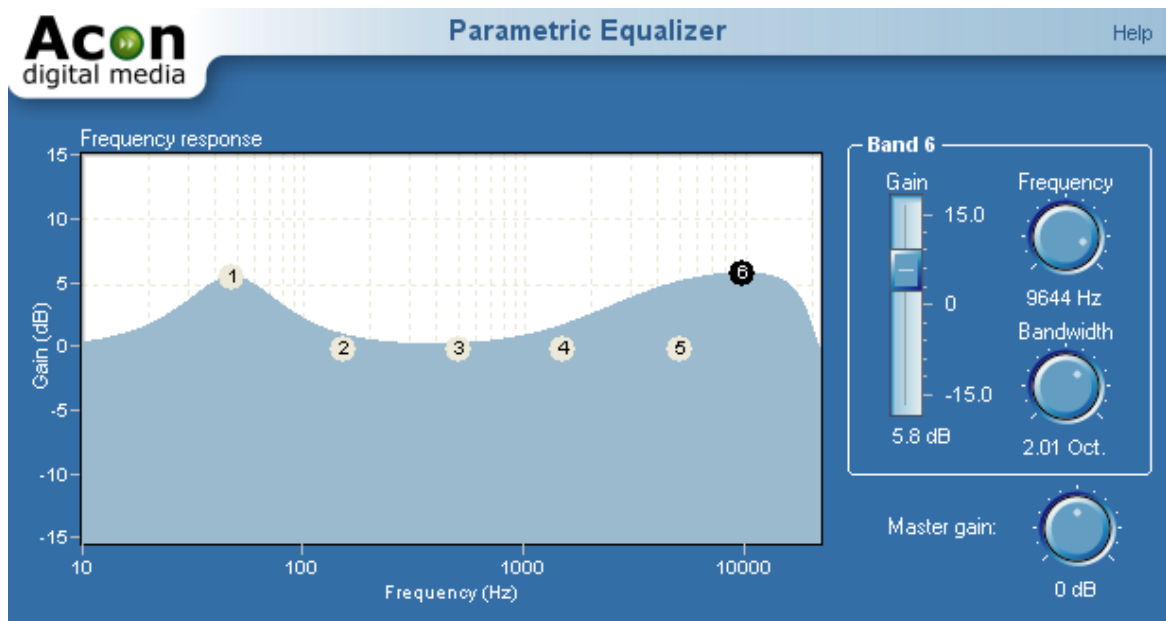
The Equalizer settings

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.
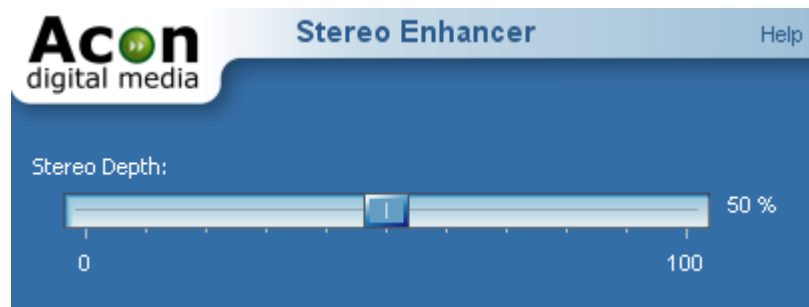
- **Frequency**

- **Gain**

- **Bandwidth**

### 5.3.6	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).

5.3.7   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, perfrom declicking and apply the deemphasis filter.
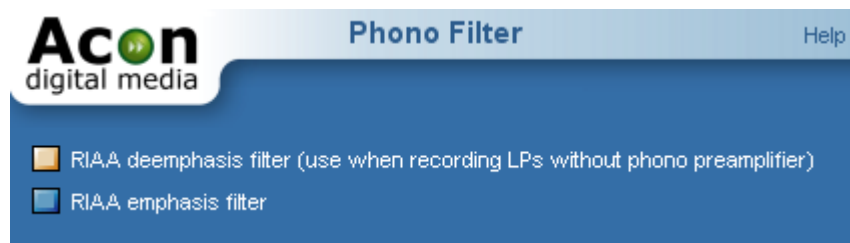
The Phono Filter settings

Settings

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

6   Preferences and Device Settings

6.1   Device Settings

AudioLava Premium support both the Windows MME and ASIO driver systems. You can select which driver to use or which sound card (if you have more than one) to use for recording and playback by selecting *Device Settings...* from the *Options* menu.

The drivers listed begins with either "[ASIO]" for ASIO drivers or "[MME]" for MME drivers. Choose input and output devices from the drop-down lists to change the current configuration. Click the button labelled *OK* when you are done.

ASIO Channel Map
If you use ASIO drivers, you can assign input or output channels to the internal channels in AudioLava Premium using the *ASIO Channel Map*. Open the *ASIO Channel Map* by clicking the "Settings..." button next to the device drop-down lists.

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