AudioLava 2

User Guide

Acon AS

AudioLava 2 User Guide

Table of Contents

1 Introduction

AudioLava is the ideal solution for restoring and recording audio from LP to CD. AudioLava automatically removes noise such as tape hiss or clicks and crackle on LP records. The user friendly wizard style application 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.

1.1 What's new in AudioLava 2

AudioLava 2 has been re-implemented from scratch for the highest quality demands and cross-platform support and is now also available on Mac and as 32 and 64 bit applications on Windows. A large range of new features and usability improvements are implemented.

- Available for Mac, Windows 32 bit and Windows 64 bit
- Redesigned and modern looking user interface
- Better quality restoration tools based on the acclaimed Acon Digital Restoration Suite

1.2 Requirements

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

PC Version (Windows)

- Windows 7 / 8.x / 10
- Intel Core i3 or AMD multi-core processor (Intel Core i5 or faster recommended)
- 1366 x 768 display resolution (1920 x 1080 or higher recommended)
- 1 GB RAM (4 GB or more recommended)
- 1 GB free HD space

Macintosh Version (OS X)

- OS X 10.8 or later
- 1 GB RAM (4 GB or more recommended)
- 1 GB free HD space

2 Working with 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. An audio interface connected to 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.

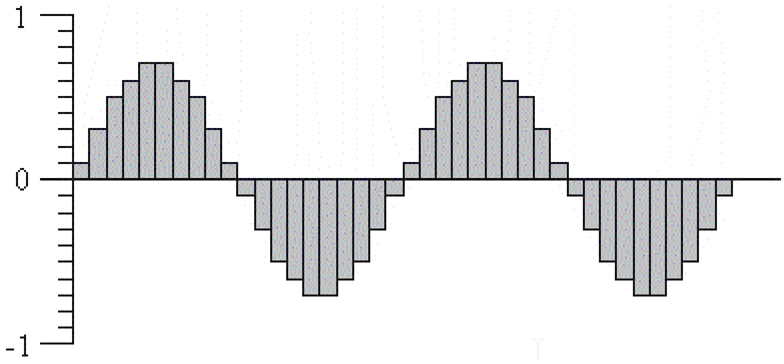
2.1 Sampling

The conversion from a continuously changing measure to a series of measured values at discrete time instances is called *sampling*. The rate at which the sampling is done, is along with the quantization depth the most important quality factor of digital recording equipment. You will not be able to record the highest audible frequencies if it is set too low. A CD quality recording is measured at a rate of 44 100 samples per second. We say that the sampling rate (or 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 unless the audio input is filtered. 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 CD audio, the highest frequency that can theoretically be recorded is 22 050 Hz.

2.2 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 achieve 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.



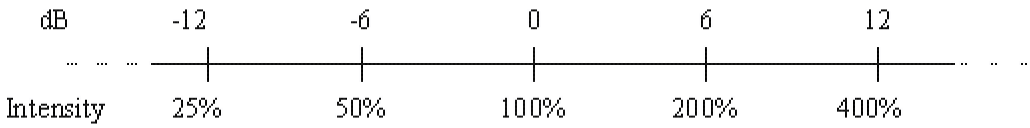
The digital representation of a sine wave.

AudioLava works with 32 bit floating point resolution internally. That allows enough precision even through several processing steps and the audio signal won't be clipped or otherwise distorted before playback or saving the audio file with a lower resolution.

2.3 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, 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.

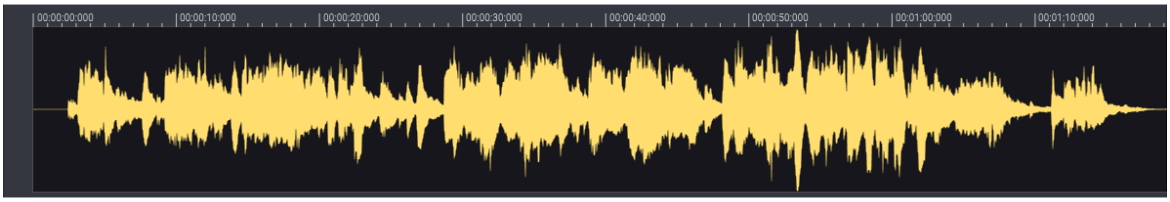


The decibel dB versus intensity change in percent

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

2.4 Visualization of Audio

The normal wave plot shown when making a recording in AudioLava is a time domain representation of the signal. When recording, AudioLava 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.



The waveform visualization of an audio signal in AudioLava.

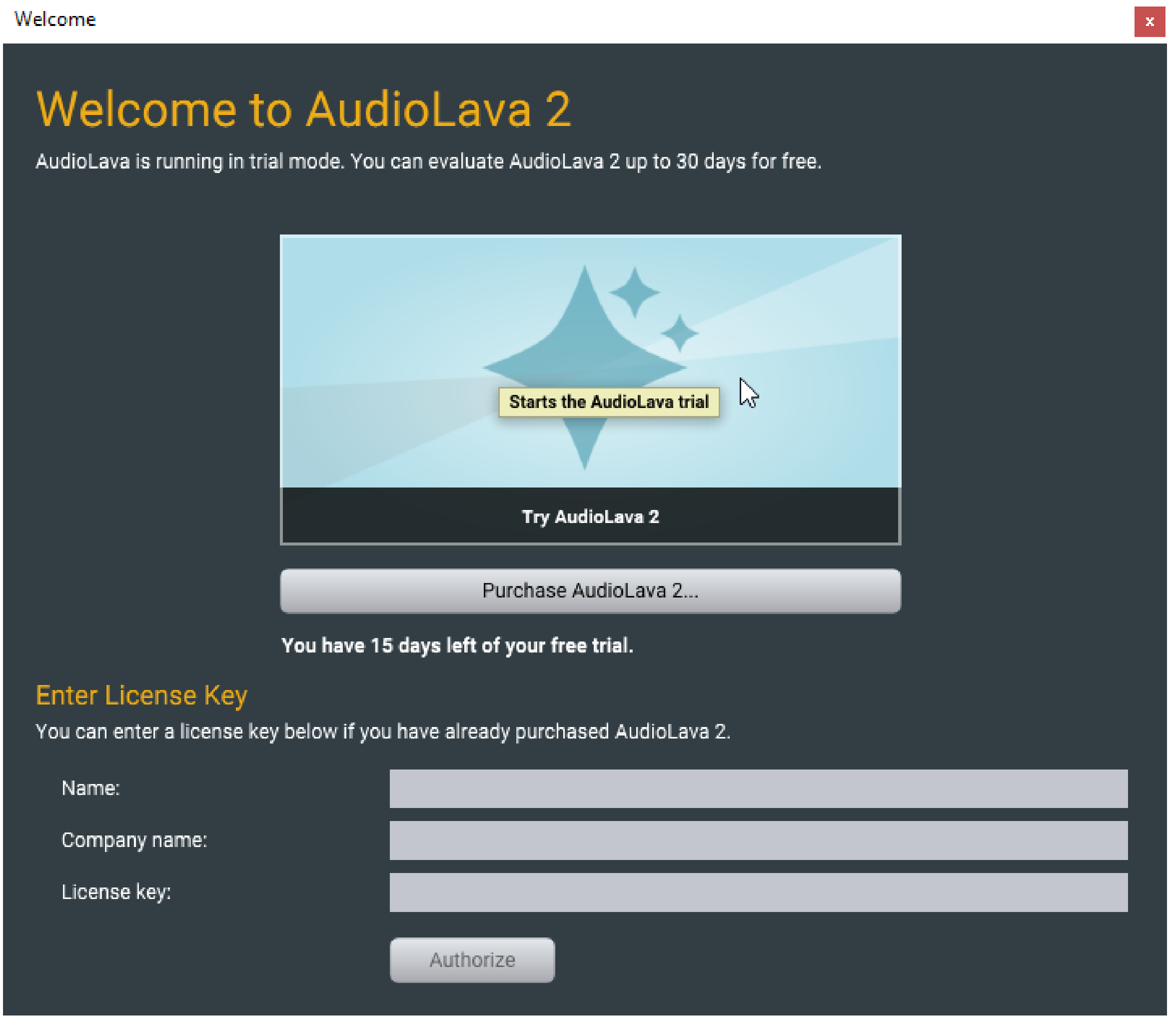
The waveform visualization is very convenient in audio editing because it provides a good overview of the recording while allowing you to select time regions.

3 Using AudioLava

AudioLava simplifies the transfer of analog recordings to computer or CD for users new to digital audio recording. It guides you step by step through all the steps from recording, track splitting, restoration and CD burning.

Autorization

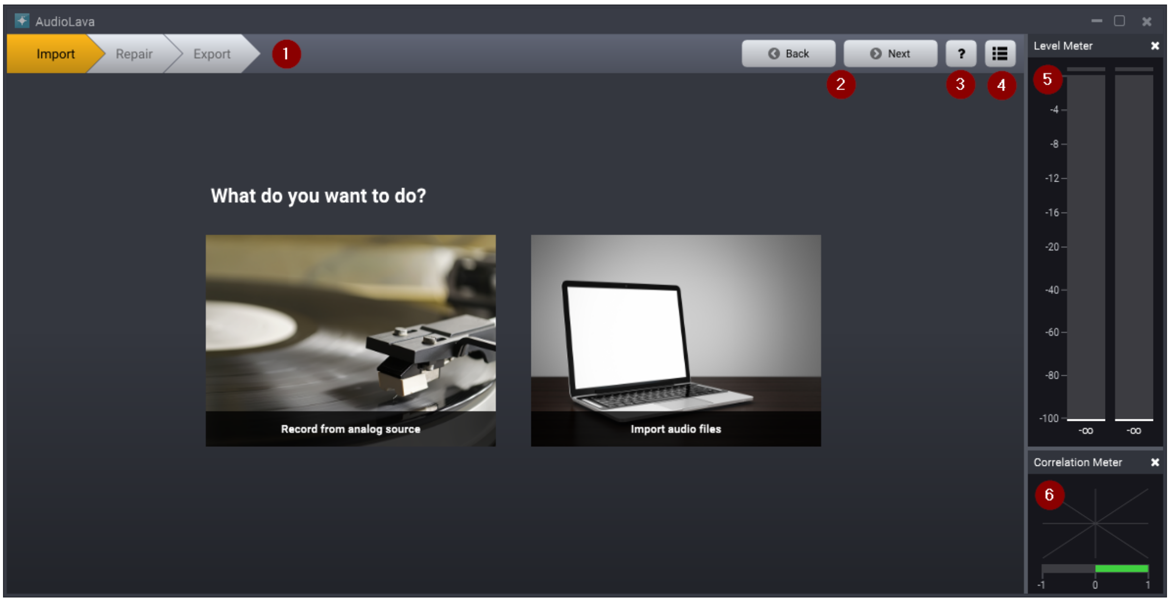
AudioLava starts in trial mode the first time you open it. You can try *AudioLava* for free in up to 30 days before purchasing a license. To use the trial, please click the large cyan trial button. You can also choose purchase a license online or enter a license key if you have already purchased a license. The *Authorize* button gets activated as soon as you have entered a valid license key. Please click this button when activated to authorize your license.



The AudioLava trial message window. You can use AudioLava in up to 30 days for free.

The User Interace

Now the *AudioLava* main interface is displayed. Wizard pages guide you through the different steps of digitizing, restoration and archiving.

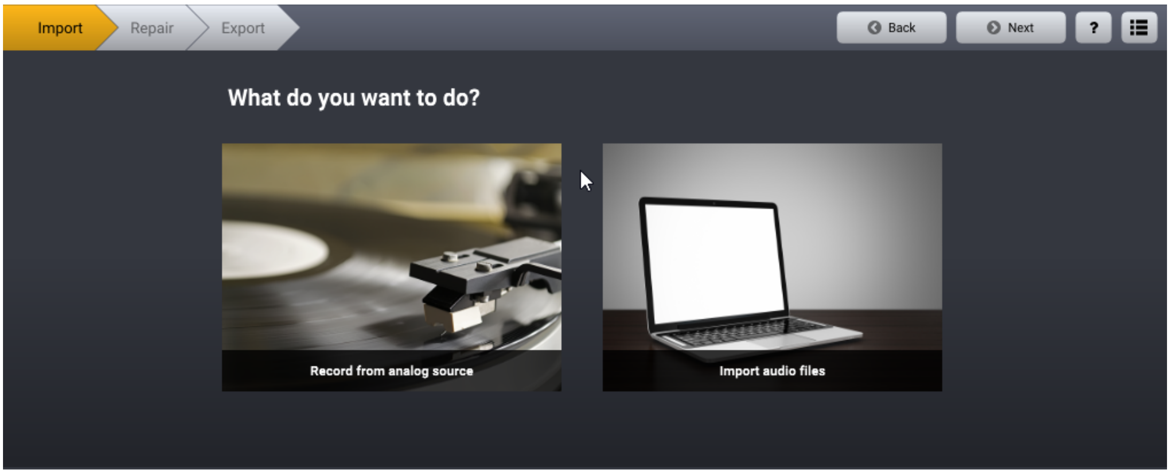


The AudioLava main window with the wizard style user interface and audio analyzers docked at the right hand side of the window.

1. The header tabs with the currently active wizard page indicated in orange. You can choose to go back to an earlier step or skip one or more steps by clicking at the *Import*, *Repair* or *Export* tabs. Some of these may be inactive if no content is recorded or loaded.
2. Back and next buttons allows you do move between the wizard pages, one at a time.
3. The help button for easy access to this help.
4. The application menu button. Click this for a drop-down menu. You can change Audio Device Settings 48, open the Audio Plug-in Manager 49 and more.
5. The level meter 51
6. The correlation meter 56

3.1 Import

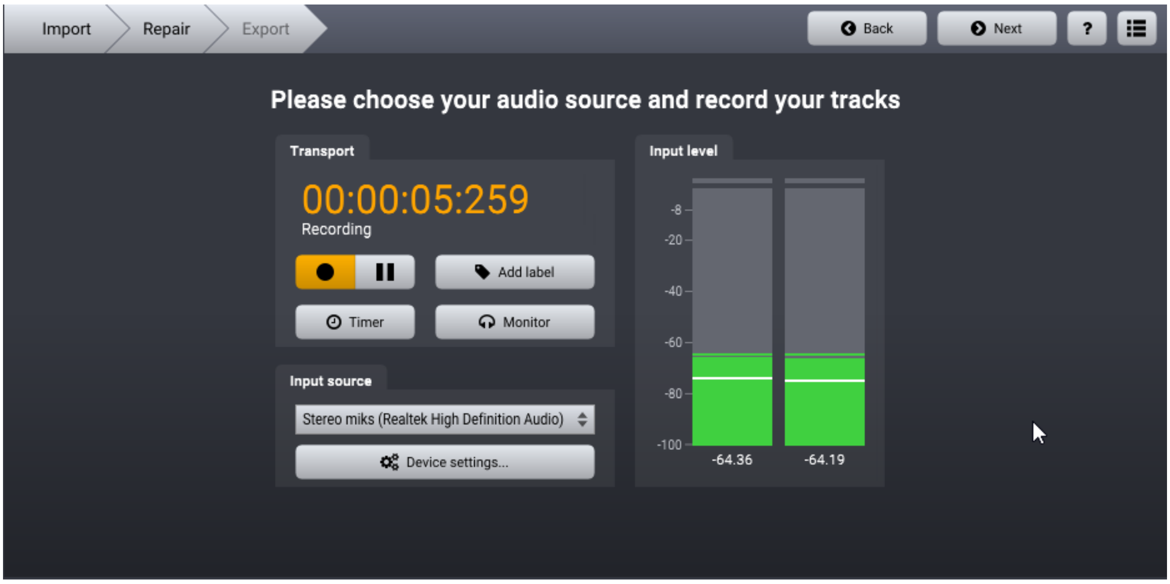
The first screen you see after opening the *AudioLava* lets you choose whether to record from an analog source or open an existing file:



The start page in AudioLava.

3.1.1 Record Audio

If you choose to record from an analog source the *AudioLava* will proceed to the recording page :



The recording page in the AudioLava.

The *Input level* meter 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 sources, like microphone or line in.

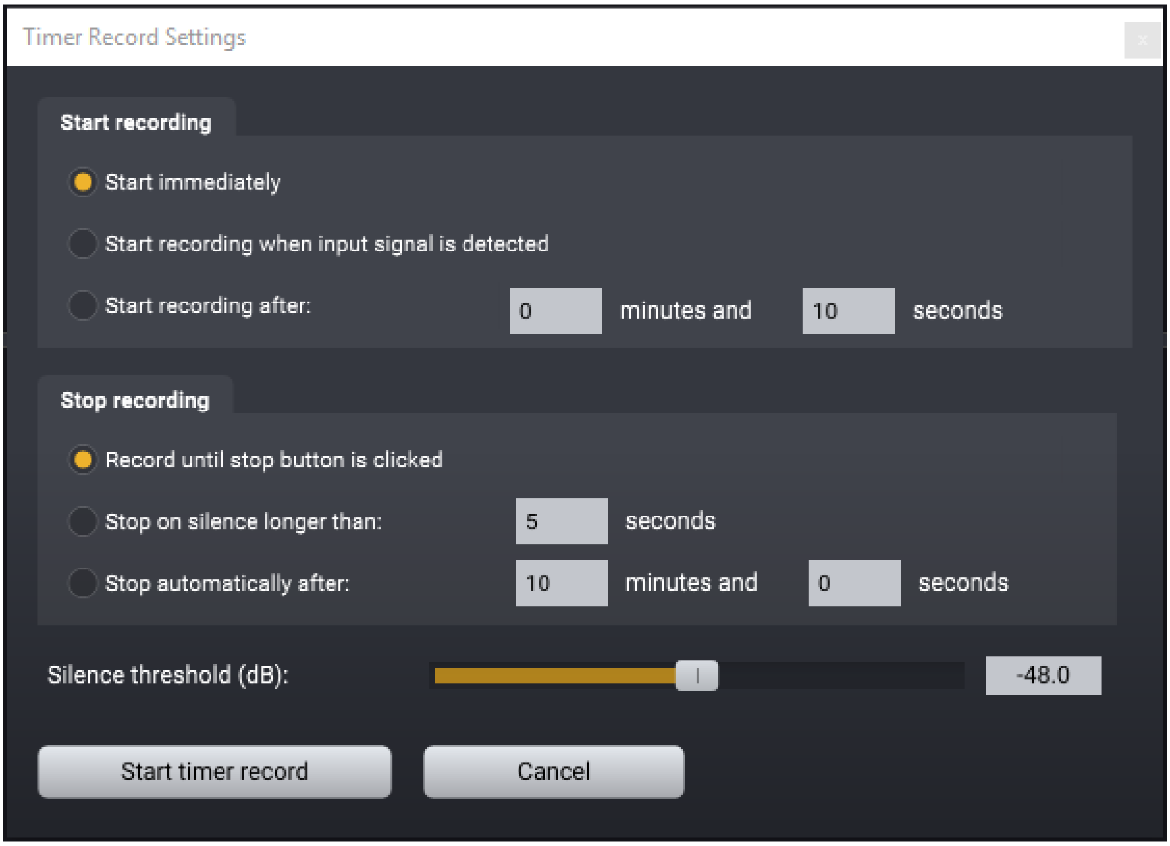
Recording Step by Step

1. Make sure your audio equipment is properly connected to your computer

2. Check that the *Input source* is correctly set. You should see the name of your external audio interface or the internal sound card on your computer if you use that.
3. Check that the input level is in the correct range. You can use easily check the input level using *Input level* meter. 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 -12 dB to allow some headroom. Most audio interfaces will have an input level knob where you can adjust the level, otherwise adjust the output level of your source.
4. Click the record button [•] to start the recording.
5. Press play on your tape deck or record player.
6. Once you have recorded the whole record or cassette tape press the *Next* button in the upper right part of *AudioLava's* main window.

3.1.1.1 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* in the recording dialog. The following dialog box appears:



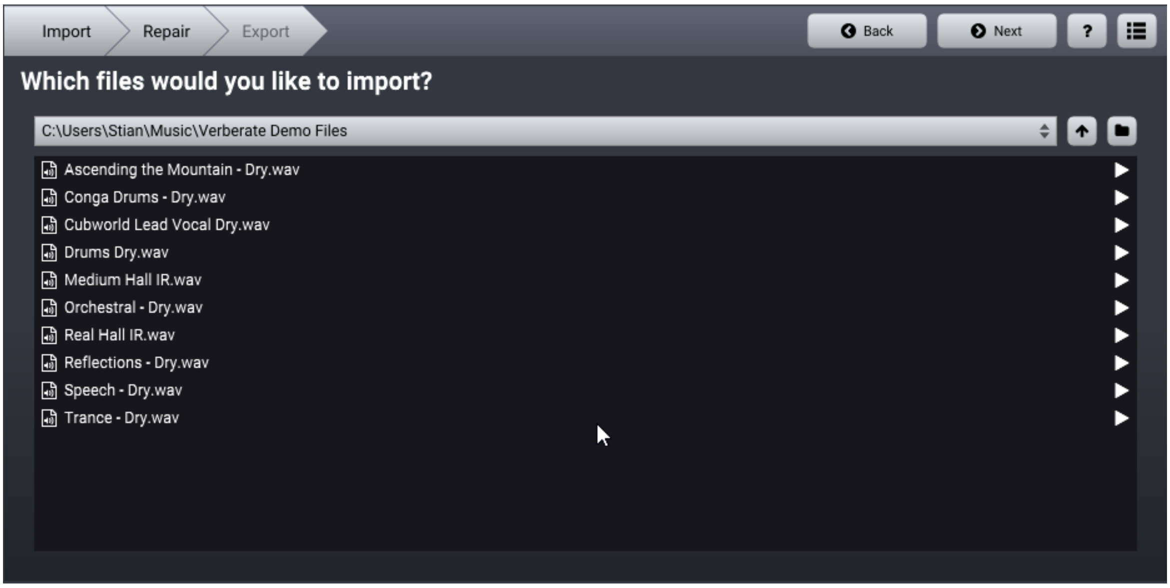
The Timer Record settings.

You can choose to start the recording immediately (after clicking the *Start timer record* button), after a specified period of time or when an input signal is present. The threshold value for the input signal detection can be defined using the *Silence threshold* slider 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.

3.1.2 Import Files

If you choose to import an existing audio file, a file browser window appears where you can select the audio file(s) you want to open.



The File Import page.

To import one or more audio files, please do the following:

1. Choose the folder in which your file is located from the drop-down list above the file list. You can go up one level or browse the folders on your computer using the buttons to the right.
2. Click the audio file you wish to open. If you want to open multiple files, press Ctrl while selecting multiple files. Click the *Next* button in the upper right corner once you are done.

Tip
You can click the play icons to the right of each file entry to listen to the file before importing.

3.2 Repair

The Repair 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.

The Repair Screen Elements

1. The waveform view shows a graphical representation of the recording. Tracks splits are indicated with a green line and the name of the track.
2. The toolbar with transport and command buttons.
3. The track list shows the tracks you defined. The *AudioLava* automatically suggests tracks, however, you can easily add, move or remove track markers.
4. The audio restoration tools, DeClick, DeCrackle, DeClip, DeNoise and DeHum. By toggeling the buttons on the left, you can activate or deactivate a tool using the toggle buttons to the left. The amount of processing is adjustable with the sliders.
5. You can add further effects and processing tools, like equalization or reverb in the *Processing Chain*. See The Processing Chain 14 for more information.

3.2.1 Track Splitting

AudioLava automatically searches for pauses and suggests track split positions when recording or importing audio files. However, if the recording is very noisy or tracks are blended seamlessly into each other, the tracks suggested by *AudioLava* might not be identical to the original tracks on the source record or cassette.

Moving the Position of an Existing Track Split

1. Move the mouse cursor to an existing track split indicator in the waveform. The mouse cursor turns into a left-right arrow.
2. Keep the mouse button pressed while moving the mouse cursor to the new position.
3. Release the mouse button.

Adding a Track Split

1. Move the mouse cursor the beginning of the track you want to add in the waveform view.
2. Click the *plus* button below the track list.

Removing a Track Split

1. Click the track you want to remove in the track list.
2. Press the Delete key on your keyboard or the *minus* button below the track list.

Renaming a Track

1. Double click the track you want to rename in the track list. The entry turns into a text editor field.
2. Enter the new name of the track and press enter.

3.2.2 Restoration

AudioLava offers five restoration tools:

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

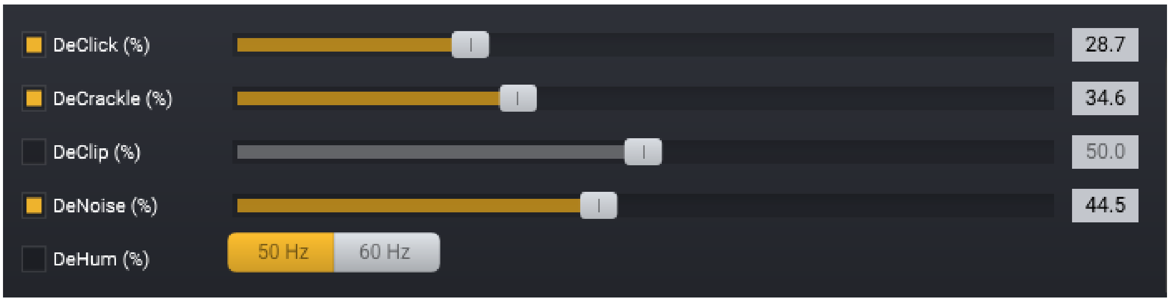
- **DeCrackle**
  Removes short but frequent clicks, referred to as crackle.

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

- **DeNoise**

Removes static noise like tape hiss. If the automatic noise detection does not remove satisfactory, you can select a portion of audio, that only contains noise. Then click the Analyze button. Once you have done this, play the whole recording and adjust the DeNoise slider until you find a good balance between noise reduction and alteration of the desired audio.

- **DeHum**
  Removes 50 or 60 Hz hum originating from electric power lines



The restoration tools in AudioLava.

You can adjust the effect of each tool by moving the sliders in the range from 0% (no effect) to 100% (full effect). The tools can be activated or deactivated by clicking toggle buttons to the left of the sliders.

Playing Restored Tracks

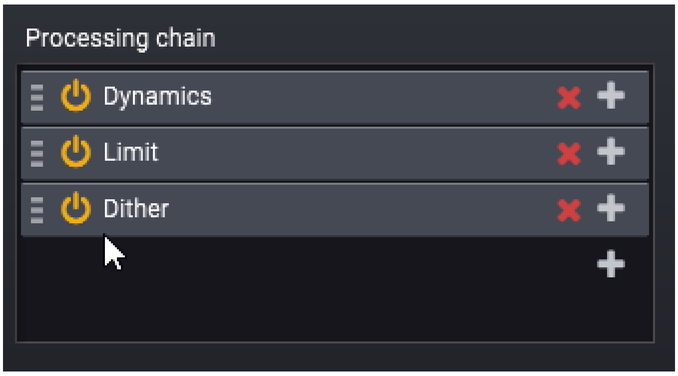
The restoration tools in *AudioLava* are processed in real time during playback so that you can listen to the effect of different restoration settings immediately. You can control the playback from the transport buttons in the Restoration Page.

3.2.3 Further Editing and Processing

The *Repair* page in the *AudioLava* lets you apply further processors to the recording using the *Processing Chain*. You can choose among all the internal processors as well external plug-ins. When you play the recording in the *Repair* page, the effects are processed in real time so that you can hear the results immediately.

3.2.3.1 The Processing Chain

The *Processing Chain* allows you to create a chain of processors or plug-ins. The processing chains can be saved including the processor settings for later use. Furthermore, each element can easily be bypassed and the order of the elements changed using drag and drop.



The Processing Chain editor in AudioLava.

Adding Processors to the Chain
To add a new processor to the chain, click the plus icon in the list. A pop-up menu appears where you can choose among internal processors or plug-ins.

Removing Effects from the Chain
To remove a processor, click the red **X** icon in the processor item in the list.

Editing the Effect Settings of an Element in the Chain
Double click the name of an entry in the list to open the processor window of an element in the chain.

Bypassing an Element
You can bypass a single element in the chain by clicking the *power on* icon in the processor list entry.

Saving and Loading Processing Chains
You can store a complete processing chain including all parameter settings for later use. To store the processing chain, click the right mouse button over the processing chain editor and choose *Save Processing Chain File*. A standard file save dialog box appears where you can enter the file name. To load a processing chain, click the right mouse button over the processing chain editor and choose *Load Processing Chain File.*

3.2.3.2 Audio Processing

Most processing tools in AudioLava have some properties in common such as preset management, A/B comparisons and bypassing. You can find a common set of buttons and controls in the header section of the included processors and these are described below.

Preset Manager

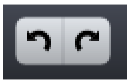
AudioLava is shipped with a set of factory presets that serve as a starting point for further adjustments. You can browse through preset categories and presets as well as create and manage your own presets using the preset management section:

The preset management section available in all the integrated processors.

You can browse through the presets using the arrow buttons. Alternatively, you can click the current preset name and a drop-down menu appears. You can also save your own presets by choosing "Save user preset file..." from the menu. A file chooser dialog box appears where you can enter the name of the preset you wish to save. You can create sub folders and place your preset files inside, and these will appear as categories in the user presets.

Undo and Redo

You can undo (or redo) changes to the parameter settings by clicking the circular arrow back or forward buttons:



Undo and redo buttons

A / B Comparisons

It is frequently useful to be able to quickly compare different parameter settings. You can do this using the A / B comparison buttons:

The A / B comparison buttons allows you to quickly compare different settings

You can keep two independent sets of parameter settings, the A and B settings, and switch between them using the corresponding buttons. The arrow button copies the settings from A to B or the other way around depending on which parameter set that is currently active.

The Processor Menu

The last button in the header section displays the processor menu:

You can click the processor menu button for the processor specific menu

The processor menu allows you to access the processor help topic directly along with other processor specific features.

Using Sliders to Adjust Parameters

Horizontal, vertical and rotary sliders (or knobs) are frequently used in the built-in processors in AudioLava. Clicking with the mouse on a knob and moving up or down will allow you to modify the setting. You can also use the scroll wheel of your mouse. For more precise control, you can hold down the Ctrl key on your keyboard at the same time as you use your mouse to modify the setting.

3.2.3.3 Tools

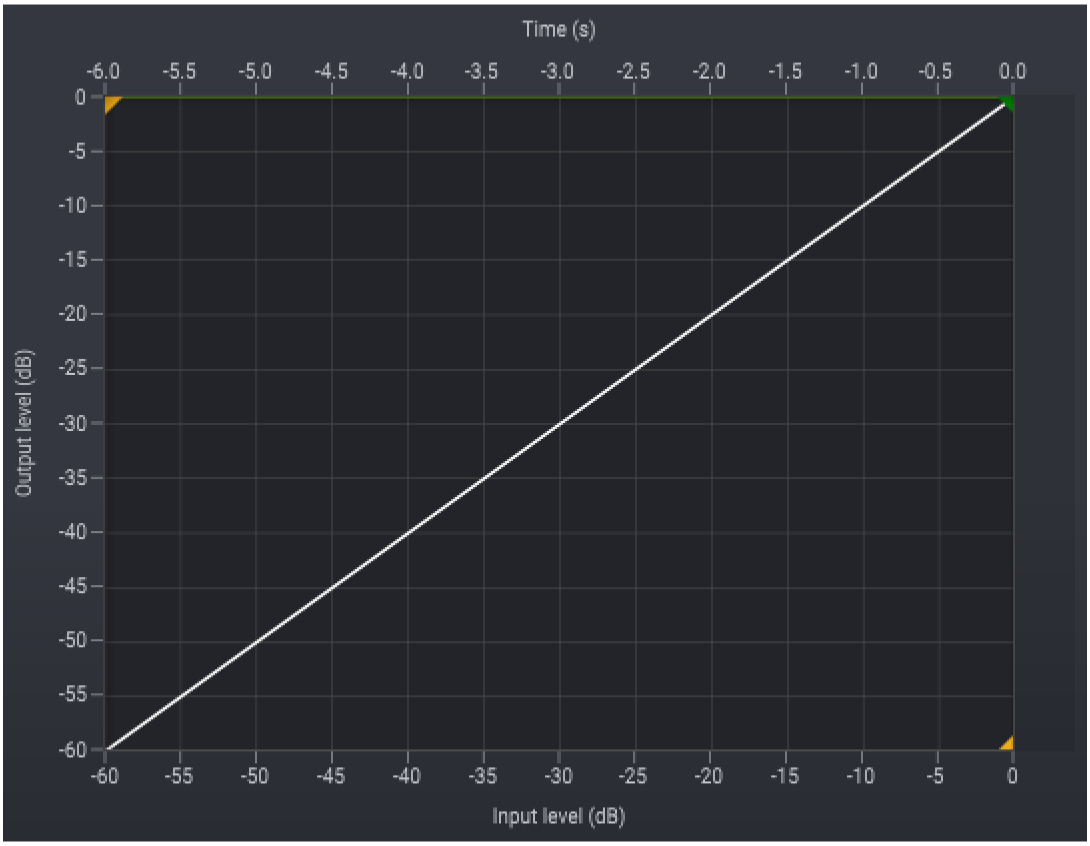
The *Tools* menu in in AudioLava contains the most common audio processing tools such as dynamic processing, equalization, sample format conversions and more.

3.2.3.3.1 Dynamics

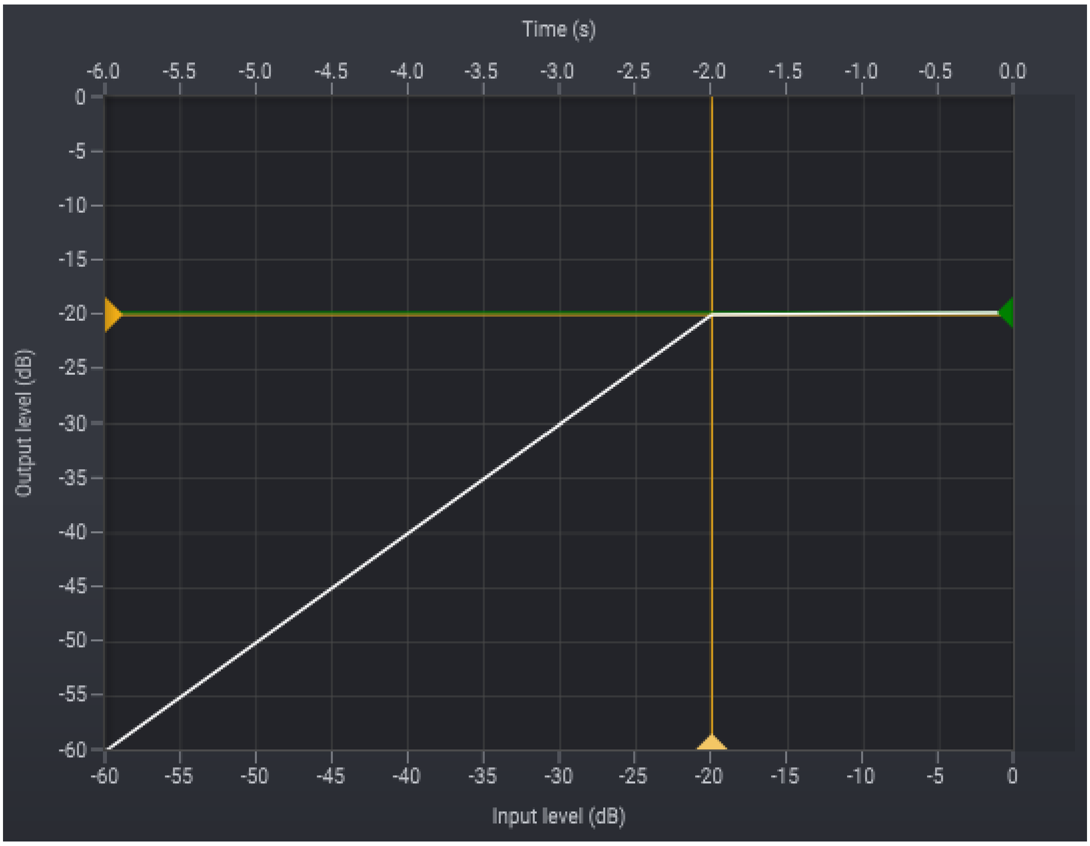
About Dynamic Processing

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 as steady a volume level as possible while doing a recording. When the input level increases he pulls down the volume fader, and he pushes it up when the input level decreases. A dynamic processor does the same thing automatically according to its settings, only with a much faster reaction time.

Modern dynamic processors allow you to set a ratio between the input levels and the output levels. This ratio is visualized as a curve where the horizontal axis represents the input level and the vertical axis represents the output level. A straight line as shown below represents a 1:1 ratio.

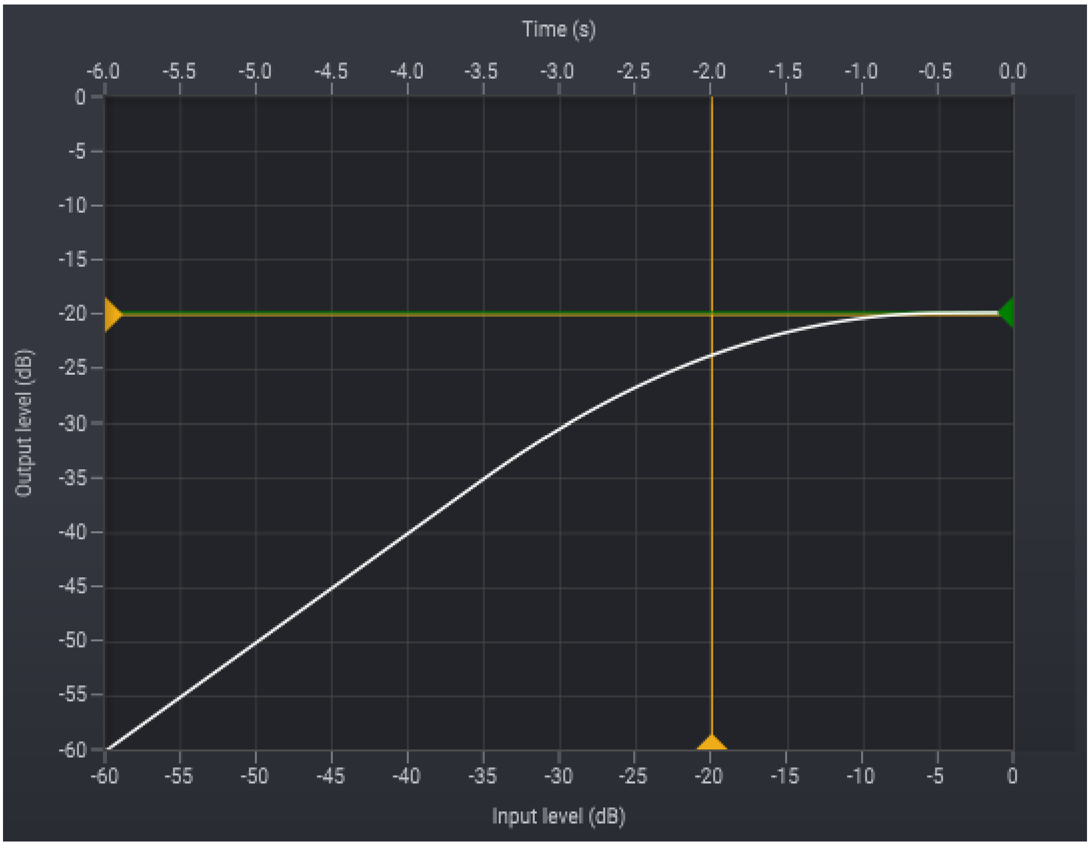


With such a setting, no change is made to the level as it is processed. Changes are made to the dynamics by altering the ratio and the threshold. In the example below, all signal levels above -20 dB are attenuated with a 100:1 ratio, so that the output level only raises 1dB when the input raises 100dB above the threshold. This setting would be comparable to a piece of hardware known as a limiter. You can see from the graph below that once the input level reaches –20 dB, the output level is barely going any higher than -20 dB, even when the input level increases.



If the dynamic processor changes the level too fast, low frequency signal components might become distorted. How quickly the dynamic processor adapts to changes in the input level is called response time. The response time is divided into the amount of time when the input level rises (the attack time) and when it falls (the release time).

When applying compressor ratios that lead to extreme changes in the dynamics, audible artifacts might become noticeable (often referred to as "pumping and breathing"). Smoother compression curves will generally reduce the artifacts of a dynamic processor. Soft kneeing automatically softens the curve to reduce such artifacts. A high level of soft kneeing was used in the image below.



User Interface

Parameter Settings

- **Threshold level (-60 dB to 0 dB)**
  When the input level exceeds the set threshold, the compressor starts to apply compression. For example, if the threshold is set at -12 dB, it will only respond to incoming audio that exceeds -12 dB. However, as soon as you activate the Soft knee, it's possible that compression takes place even before the input level exceeds the set threshold, due to the nature of the soft knee.

- **Ratio (0.01:1 to 100:1)**
  By setting up a ratio, you decide how much gain reduction should take place for

every dB the input level exceeds above the set threshold. For example, if the threshold is set on 2.00:1, there will only be a 1 dB output increase for every 2 dB the input level exceeds above the set threshold. You could also say that the compressor will do 1 dB of gain reduction for every 2 dB the input level exceeds the set threshold.

It's also possible to set a ratio that is below 1.00:1. This basically turns the compressor into a gate or expander. Audio levels below the threshold will now be reduced.

- **Soft knee (0 dB to 24 dB)**
  A soft knee allows you to create a smoother compression curve, which can help to prevent or reduce audible artifacts. With the soft knee set to 0 dB, there is no soft knee applied to the compression curve. When the soft knee is set to 8 dB for example, the compression curve will be smoothly interpolated for input levels in the range -8 dB to +8 dB relative to the threshold level.

- **Attack time (0.01 ms to 200 ms)**
  You can use the attack time knob to adjust how quickly the compressor will react to incoming audio that is above the threshold. The faster the attack time, the faster the compressor will react. If compression is applied on a snare for example which has a very loud attack, setting up a very fast attack time might help to reduce the loud attack of the snare, as the compressor can almost instantly apply gain reduction. However, in many scenarios a slightly slower attack time might provide you with a more natural sounding compression behavior, for example during mastering.

- **Release time (ms)**
  You can use the release time knob to adjust how quickly the compressor will stop processing audio that is no longer above the threshold. A fast release time can greatly enhance the sustain/decay of certain softer elements of the audio, but it can also introduce a "pumping" effect. Slow release times (>200ms) often result in a more smoother/natural compression behavior.

- **Hold time (0 ms to 500 ms)**
  As soon as audio is below the set threshold, the release time kicks in. However, the hold time allows the compressor to postpone the activation of the release time for up to 500ms, which might provide you with a smoother compression behavior in certain audio material. Although you are free to experiment with the hold function, we would like to suggest to use a hold time that is significantly shorter than the release time.

- **Side chain low cut (0 Hz to 2000 Hz)**
  Low frequency, like in a kick drum for example, has a lot of energy and this can make a compressor work very hard and can cause a "pumping" effect. By increasing

the side chain low cut frequency, you can let the compressor react less to frequencies that are below the set frequency, which helps to prevent or reduce the pumping effect caused by excessive low frequencies.

- **Make-up (0 dB to 32 dB)**
  You can use the make-up knob to compensate for the level reduction caused by the compressor. It's possible to set the make-up in auto mode, by pressing the A button in front of the knob. When the A button is yellow, the auto make-up is activated and the make-up knob is disabled.

- **Latency (0 ms to 15 ms)**
  It is possible to reduce harmonic distortion of low frequency content even with short attack and release times if the dynamic processor is allowed to examine the signal ahead of time. The downside is a slightly increased latency. You can adjust the maximum allowed latency in milliseconds.

- **Channel Linking (0% to 100%)**
  With the channel linking set to 100%, the amount of gain reduction will be the same for the left and right channels, even if there is a difference in level between these channels. The more you dial the knob away from 100%, the more the channels will be independently treated, with complete independence between the channels once you have a 0% channel linking. Channel linking values below 100% can result in shifts in the stereo image. During mastering, the best results will be most likely achieved with the channel linking set between 70% and 100%.

- **Oversampling (off, 2x, 4x)**
  The internal sample rate of the dynamic processor can be multiplied by 2 or 4, depending on the sample rate of the project. For example, if you have a project with a sample rate of 48 kHz, setting the oversampling to 4x will result in an internal sample rate of 192 kHz.

3.2.3.3.2 Limit

The Purpose of a Limiter

Limit is a two-stage dynamic processor, designed to transparently increase the perceived loudness of audio, while at the same time making sure no audio clipping occurs. First, there is a compressor stage (Pre-Compressor) that helps to keep the number of exceeding peaks within an acceptable range, which helps to prevent or reduce audible distortion. The second stage is the actual peak limiting stage (Peak Suppression) which makes sure that no audio will pass through above the threshold, also known as *brickwall* limiting. In this stage, peaks are reduced with a very quickly responding algorithm in the most transparent way as possible.

User Interface

Parameter Settings

- **Input (0 dB to +32 dB)**
  The internal threshold in Limit is always 0 dB. In order to increase the perceived loudness, you need to increase the input level of the audio. As soon as the input audio will exceed the internal threshold, gain reduction will be applied in order to keep the audio from clipping. Excessive amounts of gain reduction can lead to audible distortion and/or a pumping effect. For mastering duties, we recommend to keep the amount of gain reduction to a minimum.

- **Output (0 dB to -32 dB)**
  The output of Limit is set to 0 dB by default, but can be used to compensate for the increased level when the input is boosted.

- **Attack time (0.01 ms to 500 ms)**
  You can use the attack time knob to adjust how quickly the Pre-Compressor stage will react to incoming audio that is above the internal threshold. The faster the attack time, the faster the compressor will react. A fast attack time will be more transparent,

at the cost of a slightly lower increase in perceived loudness. A slow attack time will increase the perceived loudness, at the risk of introducing audible distortion or a pumping effect. For mastering duties, we recommend to start with an attack time of 40 ms and adjust it to taste.

- **Release time (1 ms to 5000 ms)**
  You can use the release time knob to adjust how quickly the compressor will stop processing audio that is no longer above the internal threshold. A fast release time will greatly increase the perceived loudness at the risk of audible distortion. A slow release time will be more transparent, at the cost of a lower increase of perceived loudness. For mastering duties, we recommend to start with a release time of 150 ms and adjust it to taste.

- **Pre-Compressor Channel Linking (0% to 100%)**
  With the channel linking set to 100%, the amount of gain reduction will be the same for the left and right channels, even if there is a difference in level between these channels. The more you dial the knob away from 100%, the more the channels will be independently treated, with complete independence between the channels once you have a 0% channel linking. Channel linking values below 100% can result in shifts in the stereo image. During mastering, the best results will be most likely achieved with the channel linking set between 70% and 100%.

- **Look ahead time (0 ms to 15 ms)**
  Limit uses a sophisticated look ahead algorithm to ensure the highest possible transparency. The higher this value, the more transparent the processing will be. Please keep in mind that look ahead time will also introduce latency, which will be automatically compensated for by AudioLava.

- **Peak Suppression Channel Linking (0% to 100%)**
  With the channel linking set to 100%, the amount of limiting will be the same for the left and right channels, even if there is a difference in level between these channels. The more you dial the knob away from 100%, the more the channels will be independently treated, with complete independence between the channels once you have a 0% channel linking. Although channel linking values below 100% can result in shifts in the stereo image, it is not as evident as during the pre-compressor stage. During mastering, the best results will be most likely achieved with the channel linking set between 50% and 100%.

- **Oversampling (off, 2x, 4x)**
  The internal sample rate of Limit can be multiplied by 2 or 4, depending on the sample rate of the project. For example, if you have a project with a sample rate of 48 kHz, setting the oversampling to 4x will result in an internal sample rate of 192 kHz.

3.2.3.3.3 Dither

Introduction to Dither

Whenever you reduce the resolution (bit depth) of an audio signal you will introduce truncation errors, which can be a very unpleasant artifact if audible. The quantization noise is correlated to the audio signal which we perceive as more disturbing that uncorrelated noise. Dither introduces low level random noise to de-correlate the noise. Additionally, it is possible to alter the frequency distribution of the noise signal. The human ear has a frequency dependent sensitivity and it makes sense to "move" the noise to frequency regions where the ear is less sensitive. This process is called noise shaping and the *Dither* tool in AudioLava offers detailed control over the noise shaping process.

User Interface

Parameter Settings

- **Target resolution**
  Here you have to set the target resolution. If you want to change the resolution from 24 bit to 16 bit for example, you need to select 16.

- **Dither level (%)**
  Dither amplitude in percent of 1 LSB (least significant bit). This should be at 100% for complete de-correlation of quantization noise.

- **Filter length (ms)**
  This allows you to set the length of the noise shaping filter in milliseconds. Longer filters allow more accurate shaping, but comes at CPU cost.

- **Max. noise shaping (dB)**
  This sets he maximum gain allowed in the noise shaping filter. The higher the setting, the less audible the added noise will be, by moving the noise up in the frequency spectrum. However, too much energy concentration in the high frequency bands should be avoided as this can cause issues with certain digital to analog convertors.

3.2.3.3.4 Phono Filter

About 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). It can also apply PCM type emphasis and deemphasis as found on DAT and early CD recordings. The analog amplitude frequency response is calculated accurately using analytic calculations and applied using a minimum phase filter to match the analog counterpart as closely as possible.

User Interface

Parameter Settings

- **Filter mode**
  Choose deemphasis mode if you have a recorded an LP record without a phono preamplifier. Choose Emphasise if you want to prepare an audio file for an LP master.

- **Output level (dB)**
  Use the output level slider to compensate for the increase or decrease in audio level.

Advanced Settings

You can adjust the time constants as specified in analog emphasis and deemphasis circuitry. These constants (T1, T2 and T3) are often specified in literature about restoration of vintage recordings (before the RIAA standard). You can enter these time constants yourself under *Time constants*.

3.2.3.3.5 Equalize Light

Introduction

Equalize Light is a versatile, user friendly and great sounding equalizer with several unique features. Unlike other equalizers, you can freely adjust not only center frequency, gain and bandwidth, but also the filter slope for each band. The filter slope can be set anywhere from 3 to ultra sharp 120 dB per octave. Equalize Light is a zero latency plug-in with a minimum phase response.

Great care has been taken to provide a user interface that is straight forward to use. Band parameters can be adjusted using handles directly in the graphical representation of the frequency response, including bandwidths and filter slopes. A flexible real-time analyzer lets you monitor every aspect of the processing. You can easily switch between full, mid, side, left or right channel processing for each band and Equalize automatically routes the audio signal internally to ensure the best results and lowest possible latency.

User Interface

The graphical user interface of Equalize Light is designed to hide the complexity of the plug-in and provide an efficient workflow as well as intuitive control over the plug-in parameters.

Adding and Removing Bands

You can add up to 12 individual bands in *Equalize Light*. Each equalizer band represents a filter type, which can be either low cut, low shelf, bell, notch, high shelf or high cut. To add an additional band, click the + button below the curve display or double click in the curve display. A bullet shaped handle appears in the curve that you can move around using the mouse. The currently selected band also shows additional handles that you can use to manipulate the bandwidth (if applicable) and filter slope. You can remove the band by clicking the x button or by double clicking the handle.

Soloing and Bypassing Bands

You can solo or temporarily bypass an equalizer band in order to monitor its effect on the audio signal. You can enable the solo mode by pressing the Ctrl key while moving an equalizer band handle. The Shift key enables the bypass band mode. Alternatively, you can use the buttons to enable or disable the solo and bypass modes:

: Solo mode
: Bypass mode

Band Parameter Settings

The parameters that are related to a specific band are placed within the band group and the header indicates which band is currently active. The parameters apply to the currently active band only. To change the currently active band you can either click its bullet handle in the frequency response visualization or use the arrow buttons to browse through the active bands.

- **Frequency (Hz)**
  The center (for peak and notch filters) or threshold frequency of the currently active equalizer band in Hertz.

- **Band gain (dB)**
  The gain of the currently active equalizer band in decibels. This parameter is not available for high or low cuts, or the notch filter.

- **Bandwidth (oct.)**
  The bandwidth of the currently active equalizer band in octaves. This parameter is only available for peak and notch filters.

- **Slope (dB/oct.)**
  The filter slope of the currently active equalizer band in decibels per octave. This controls the steepness of the filter.

- **Gain to Bandwidth Link**
  The perceived bandwidth of an equalizer band is dependent on the gain setting. You can link the bandwidth to the gain setting, so that the bandwidth is automatically adjusted when you change the gain to preserve the perceived bandwidth.

  : Click this button to activate or deactivate the gain to bandwidth link

- **Filter Type Buttons**
  You can choose between six different filter types:

  : Low cut, removes frequency content below the band frequency
  : Low shelf, boosts or attenuates frequency content below the band frequency
  : Peak filter, boosts or attenuates frequency content around the band frequency
  : Notch filter, removes frequency content around the band frequency
  : High shelf, boosts or attenuates frequency content above the band frequency
  : High cut, removes frequency content above the band frequency

- **Channel Mode Buttons**

You can choose any of the following channel modes for each equalizer band individually:
L : Apply to left channel only
M : Apply to mid channel only. The mid channel is the sum of the input channels multiplied by a scaling factor.
: Apply to both channels
S : Apply to side channel only. The side channel is the difference of the input channels multiplied by a scaling factor.
R : Apply to right channel only

Global Parameters

- **Master gain (dB)**
  The master gain controls the overall gain of the equalizer in decibels. Equalize can automatically adjust the overall gain to compensate for any gain changes in the equalizing process. This is based on an average distribution typically found in music and cannot be completely accurate. To activate the automatic gain compensation, click the button labelled A to the left of the master gain knob.

Analyzers and the Frequency Response Visualization

Equalize visualizes the frequency response of the current equalizer settings and contains two separate spectrum analyzers that you can set up to monitor the effects of the processing.

- **Analyzer 1 & 2**
  You can choose to analyze different signal sources using the drop-down lists under the Analyzer 1 and Analyzer 2 headers. Either input or output signals can be analyzed and you can also choose if the left, right, mid or side signal should be analyzed.

- **Drop (dB/s)**
  The drop parameter sets how quickly the analyzer adapts to lower signal levels and is specified in decibels per second.

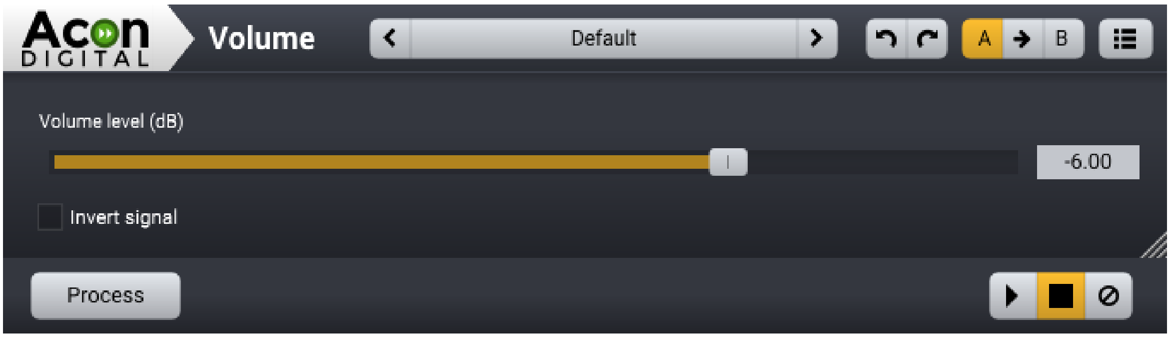
- **Range Buttons**
  You can change the range of the frequency response visualization or the spectrum analyzers by clicking the buttons at the lower end of the level axes. A drop down list appears with the alternatives.

3.2.3.4 Volume

The *Volume* menu in in AudioLava contains a collection of audio processing tools related to volume manipulation.

3.2.3.4.1 Volume

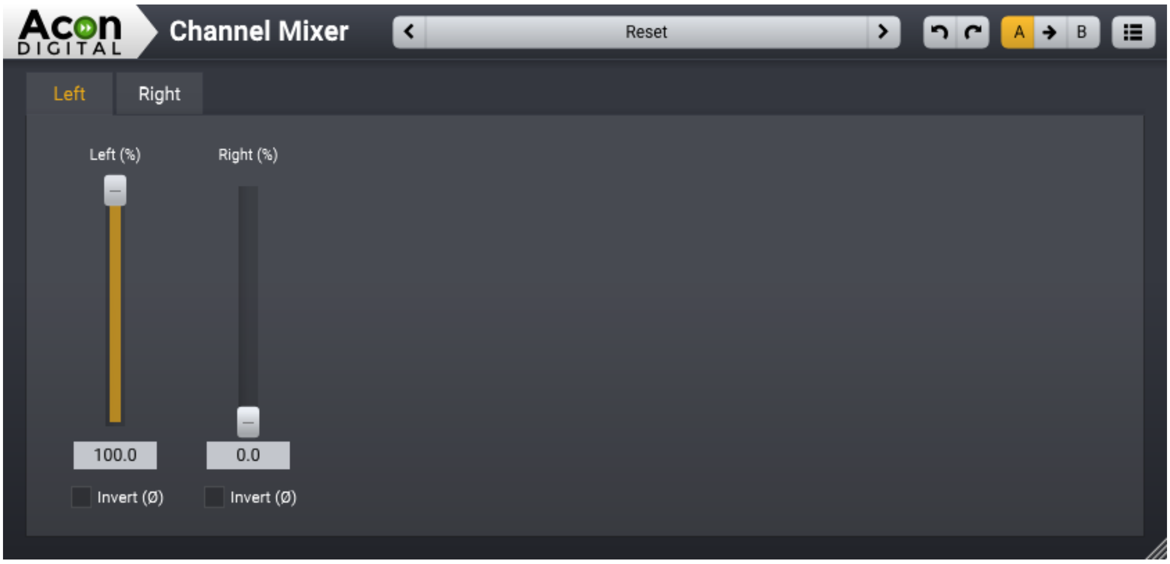
The most basic volume manipulation tool in AudioLava is *Volume*. The only parameters are the volume change in Decibel and the *Invert signal* toggle:



The Volume level slider has a range from -96 dB to +32 dB

3.2.3.4.2 Channel Mixer

The channel mixer is a tool that works only on stereo and multichannel recordings. Each output channel is represented by a tab in a tab control and you mix several input channels to each output channel. The input channel levels are adjusted using the sliders. The input signal from each source channel can also be inverted using the *Invert* check boxes.



The Channel Mixer settings

3.2.3.5 Effects

The *Effects* menu in in AudioLava contains a collection of common effect processors such as reverb, echo, chorus and more.

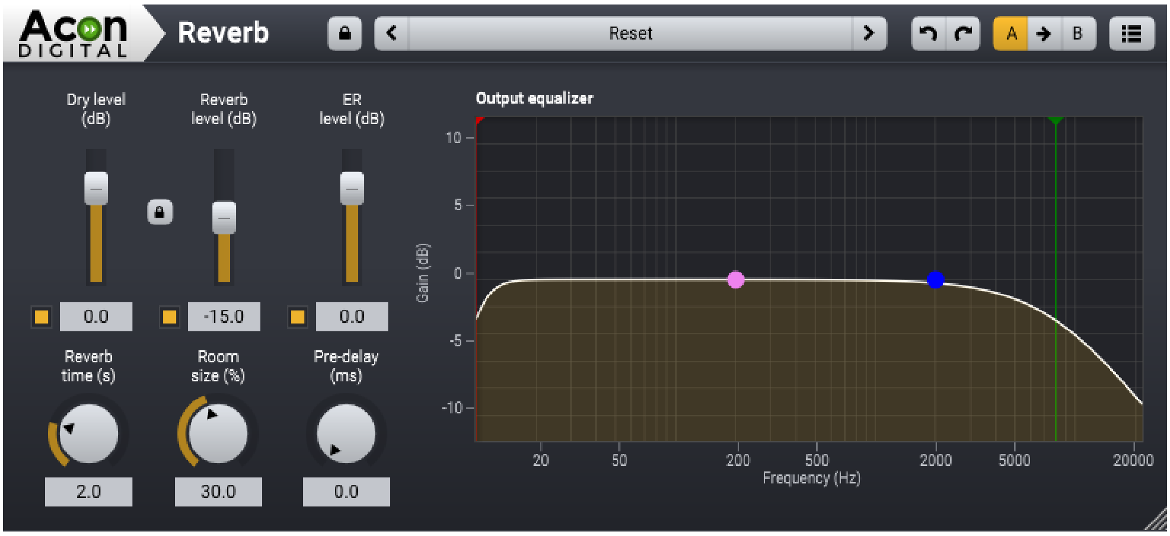
3.2.3.5.1 Reverb

About Reverb

Reverberation occurs when sound is produced in an enclosed acoustical environment. Even outdoors, there is likely to be some level of reverberation, however subtle. The sound propagates through the air before it arrives at the listener, but the sound is also reflected when it hits walls or other objects. Due to the propagation time, these reflections arrive at the listener later than the sound from the direct path. After a certain build-up time, there are usually so many reflections that no distinct echoes are distinguishable, but rather a smoothly decaying sound.

The first few reflections, usually called early reflections, are important cues for our perception of an acoustical environment. For that reason, most digital reverberation units differentiate between early reflections and the dense late reverberation.

User Interface



Parameter Settings

- **Dry level (dB)**
  The amount of dry (unprocessed) signal to send to the output specified in decibel. You can use the toggle button to exclude the dry signal completely. You can lock the relation between the dry and reverb level by clicking the lock button between the dry and reverb level sliders.

- **Reverb level (dB)**
  The amount of reverberation signal to send to the output specified in decibel. You can use the toggle button to exclude the late reverberation completely.

- **ER level (dB)**
  The amount of early reflections to send to the output specified in decibel relative to the reverb level. The early reflections are important for our perception of distance to the sound source. Increasing the early reflection level gives the impression of getting closer to the source. You can use the toggle button to exclude the early reflections completely.

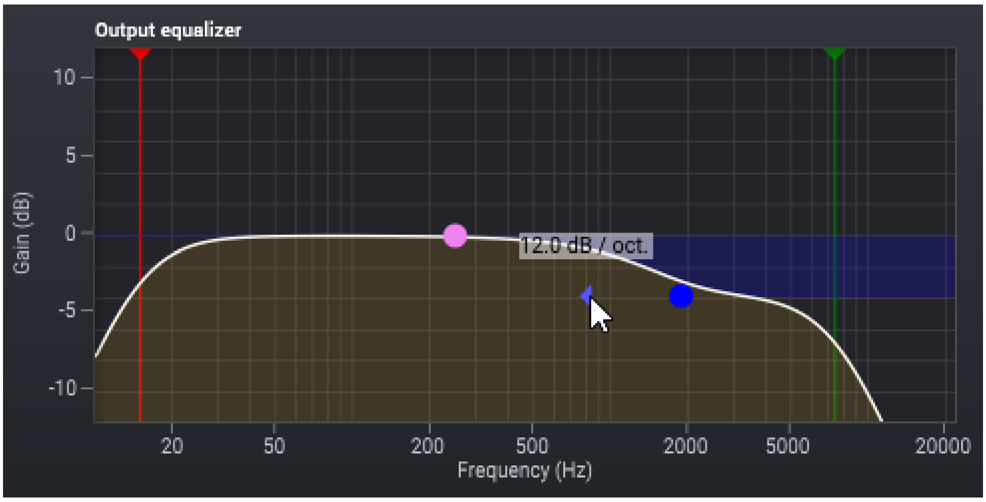
- **Early reflections mode**
  You can choose the early reflection program from a set of predefined programs, such as rooms, chambers, halls and plates.

- **Reverb time (s)**
  The reverb time specifies the duration of the reverberation and is specified by the number of seconds before the reverb tail level drops below -60 dB or 1/1000 of its initial amplitude.

- **Room size (%)**
  The room size determines the size of the simulated acoustical environment. In general, the room size should match the reverberation time. Using a very long reverberation time combined with small room sizes may result in resonances causing a slightly "metallic" quality. In some cases, as with a couple of the plate presets, the resonances are desired to achieve a specific sonic quality.

- **Pre-delay (ms)**
  The pre-delay slider allows you to adjust the time is milliseconds before the reverberation signal arrives.

- **Output Equalizer**
  You can use the output equalizer to apply filtering to the reverberation signal. Both the early reflections and the dense reverberation are filtered using the output equalizer. The equalizer consists of high and low pass filters as well as high and low shelving filters. As in the decay editor, you can use handles to change the filter settings and the frequency in Hertz as well as the gain in dB are displayed as cursor information while making adjustments. The filters in the output equalizer have variable filter slopes. You can modify the filter slope by clicking the filter section you want to edit. A small arrow appears in the same color as the filter section. You can move this arrow using the mouse to modify the slope as depicted below:



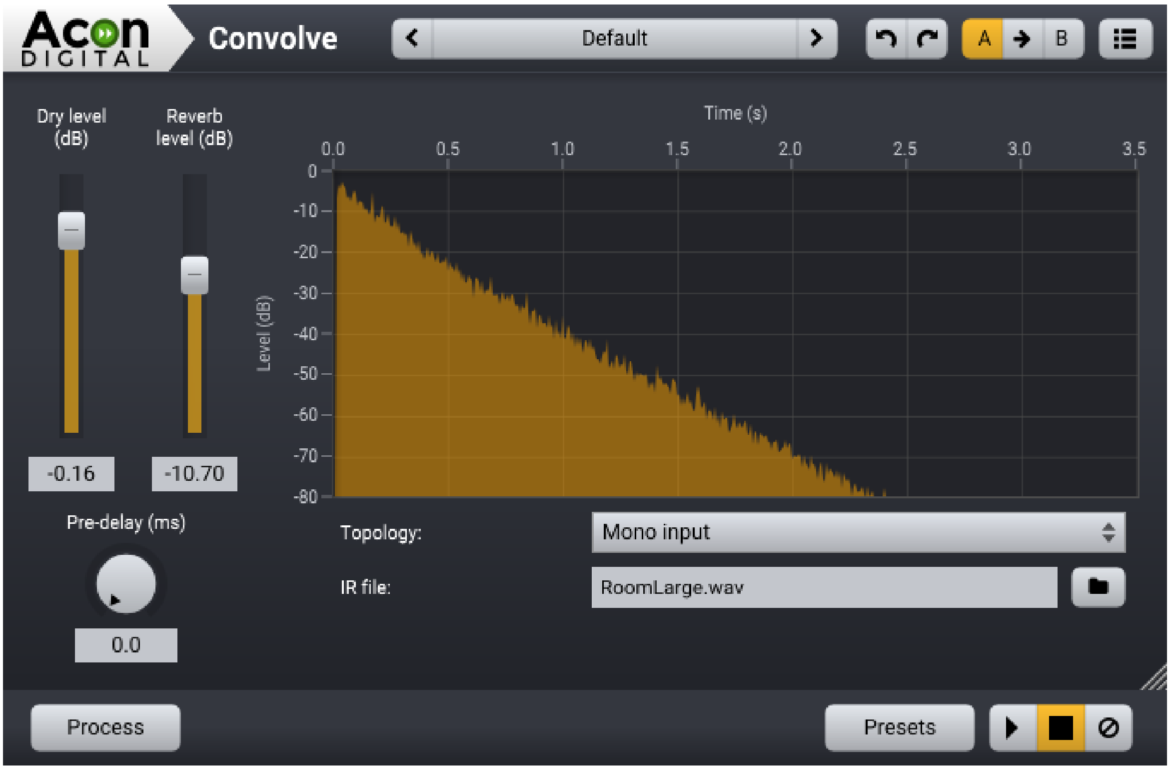
3.2.3.5.2 Convolve

Introduction to Convolve

Convolve is a convolution reverb which can be used to apply recorded impulse responses from real acoustic spaces. This makes it fundamentally different from algorithmic reverbs, like *Verberate*. Convolution reverb is perfect for adding lifelike ambiance to a recording, like the ambiance from a room, a hall or a parking garage for example. However, compared to an algorithmic reverb which often offers control over the decay time and room size for example, *Convolve* doesn't offer you this control, it totally relies on the impulse response you load.

There are a lot of amazing free impulse responses available on the net. The following two websites are recommended:
http://www.openairlib.net/
http://www.echothief.com/

User Interface



Parameter Settings

- **Dry level (-48 dB to 12 dB)**
  With this slider, you can control the amount of dry (unprocessed) signal to send to the output, specified in decibel.

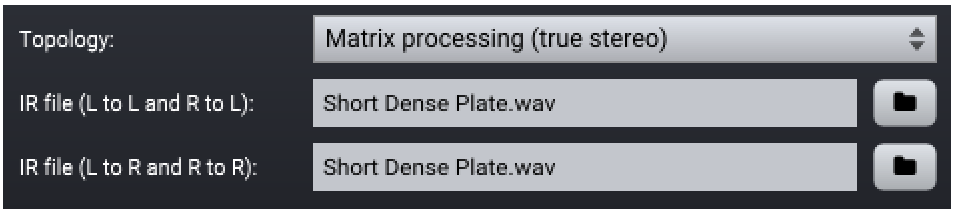
- **Reverb level (-48 dB to 12 dB)**
  With this slider, you can control the amount of processed signal to send to the output, specified in decibel.

- **Pre-delay (0 ms to 1000 ms)**
  The pre-delay knob allows you to adjust the time is milliseconds before the reverberation signal arrives.

- **Topology**
  This drop down menu offers three different topologies. *Mono input* will sum the input and processes the sum using the left and right channel of the IR file to create a stereo output. *Parallel channels* will processes the left input channel with the left channel of the IR file and the right channel with the right channel of the IR file independently. *Matrix processing* requires one IR file for each input channel where each IR file defines how to process the input channel to the set of output channels. This is the only option

that is "true stereo", which means that you will be able to hear the panning of the input signal in the reverb signal in a natural way.

When you select Matrix processing (true stereo) under Topology, you will need to load two IR files, one for the left channel and one for the right channel. This will look like this:



- **IR file:**
  Here you can load an impulse response file. This file can be of any audio file format supported by AudioLava.

3.2.3.5.3 Echo

About Echo

The echo effect is a multi-tap delay effect. Multi-tap means that you can add several delays (up to eight) with arbitrary delay times and gains. Two different timing modes are offered, the BPM (Beats per minute) mode or the milliseconds mode. In the BPM mode, the time delay of each tap is specified in beats.

User Interface

Parameter Settings

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

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

- **Echogram**
  This depicts the gain versus time for the configured taps. You can add new delay taps by clicking the left mouse button where you want the new delay tap to appear. To move a delay tap, click an existing point and keep the mouse button down while moving the mouse pointer to the new location. You can remove an existing delay tap by double clicking the tap you want to remove.

- **Range**
  This button allows you to modify the amount of time shown in the echogram. With the time mode in seconds, this can be set between 0.1 second and 5 seconds. With the time mode in beats, this can be set between 1/16 beat and 8 beats.

- **Delay (s) or Delay (beats)**
  This lets you specify the amount of time audio is delayed (in seconds or beats) before it is sent to the output. In the echogram, it affects the horizontal position of the tap.

- **Gain (dB)**
  The individual tap is present by this amount in the output. In the echogram, it affects the vertical position of the tap.

- **Feedback (dB)**
  The feedback percentage specifies the amount of attenuation since the last delay interval.

- **High cut frequency (Hz)**
  This allows you to change the cut-off frequency of the low pass filter in the feedback loop. The low pass filter can be enabled or disabled by clicking the check box underneath the knob.

- **Ping pong echo**
  This check box provides for a bouncing stereo delay.

- **Time Mode**
  Select either beats per minute for the BPM mode or seconds.

- **Beats per minute**
  Here you can specify the tempo by entering the number of beats per minute.

- **Snap mode**
  To make it easier to adjust the delay of each tap without loosing the alignment to the tempo, you can optionally turn on the snap mode. This option will only be enabled when the time mode is in beats.

3.2.3.5.4 Multiply

Introduction to Multiply

AudioLava comes with Acon Digital Multiply, which is a versatile chorus effect with a unique twist. Each simulated voice is processed with a phase randomizing filter so that unpleasant comb filter effects are omitted. The effect can be used to simulate the effect of several performers playing the same tones simultaneously, to widen the spatial image or to create special effects for sound design. Multiply can simulate up to 6 additional voices and both the pitch and the loudness of the voices can be modulated. There is also an integrated equalizer consisting of a low cut, low shelf, high shelf and high cut filters that can be applied to the effect signal. An integrated pre-delay section makes it possible to create modulated and diffuse echo effects.

The graphical user interface of Multiply is designed to give quick and intuitive access to all the parameter settings defining the sonic quality of the effect.

User Interface

Parameter Settings

- **Dry level (dB)**
  The amount of dry (unprocessed) signal to send to the output specified in decibel. You can use the toggle button to exclude the dry signal completely. You can lock the relation between the dry and effect levels by clicking the lock button between the dry and effect level sliders.

- **Effect level (dB)**
  The amount of processed signal to send to the output specified in decibel. You can use the toggle button to exclude the effect signal completely.

- **Frequency modulation rate (Hz)**

The frequency modulation rate controls how rapid the tone fluctuations in the simulated voices should be. It is specified in Hertz.

- **Frequency modulation depth (%)**
  The frequency modulation depth controls the amount of tone fluctuations in the simulated voices. It is specified in percent, ranging from no modulation (0%) to full modulation (100%).

- **Amplitude modulation rate (Hz)**
  The amplitude modulation rate controls how rapid the loudness variations in the simulated voices should be. It is specified in Hertz.

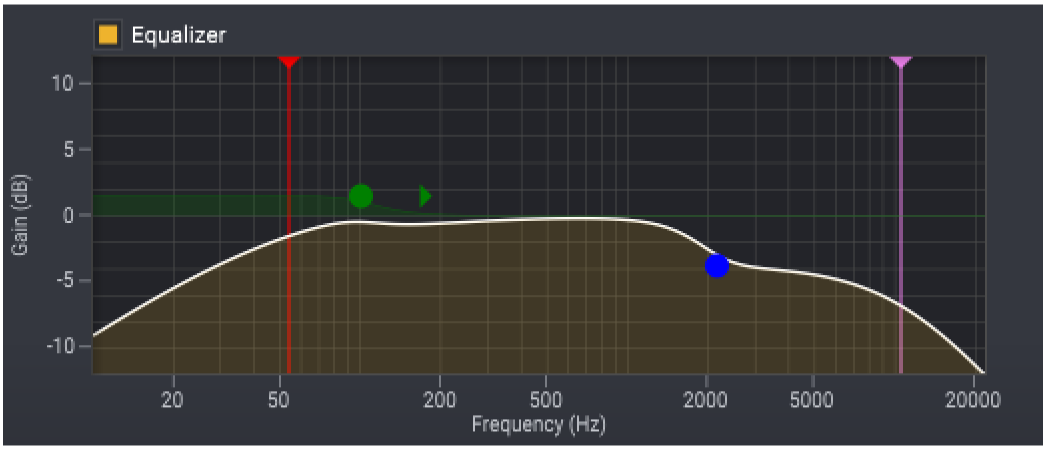
- **Amplitude modulation depth (%)**
  The amplitude modulation depth controls the amount of loudness variations in the simulated voices. It is specified in percent, ranging from no modulation (0%) to full modulation (100%).

- **Voice count (#)**
  The amount of simulated voices.

- **Stereo spread (%)**
  You can use the stereo spread parameter to control the stereo width of the effect signal. If the stereo spread is set to 0%, the effect signal will be mono when the input signal is mono and at 100% the full stereo width is achieved.

- **Pre-delay (ms)**
  The pre-delay slider allows you to adjust the time is milliseconds before the effect signal arrives.

- **Equalizer**
  You can use the output equalizer to apply filtering to the effect signal. The equalizer consists of high and low pass filters as well as high and low shelving filters. You can use handles to change the filter settings and the frequency in Hertz as well as the gain in dB are displayed as cursor information while making adjustments. The filters in the equalizer have variable filter slopes. You can modify the filter slope by clicking the filter section you want to edit. A small arrow appears in the same color as the filter section. You can move this arrow using the mouse to modify the slope as depicted below:



3.2.3.6 Using Audio Plug-Ins

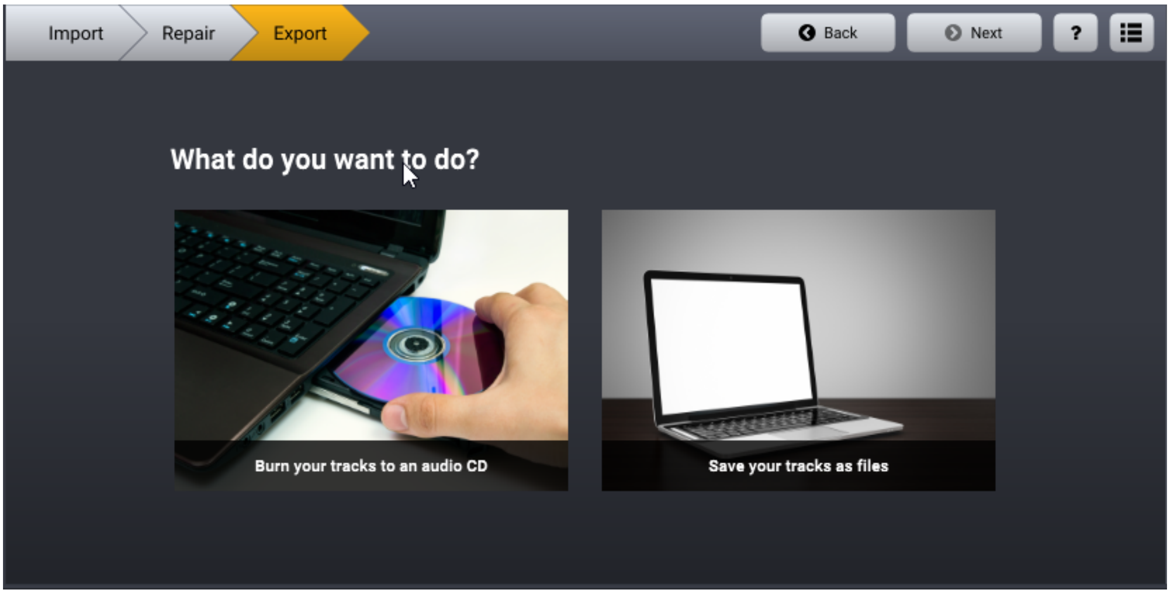
AudioLava supports the plug-in formats VST, VST3 and Audio Units. The latter is only available on Mac. You can scan for and manage plug-ins using the Plug-in Manager 49.

3.2.3.6.1 Accessing the Plug-Ins

Before you can use plug-ins, you have to scan available plug-ins. How this is done is described in The Plug-in Manager 49. After you have scanned for plug-ins you can access the plug-ins through the add processor drop-down menu of the chain editor (see The Processing Chain 14).

3.3 Export

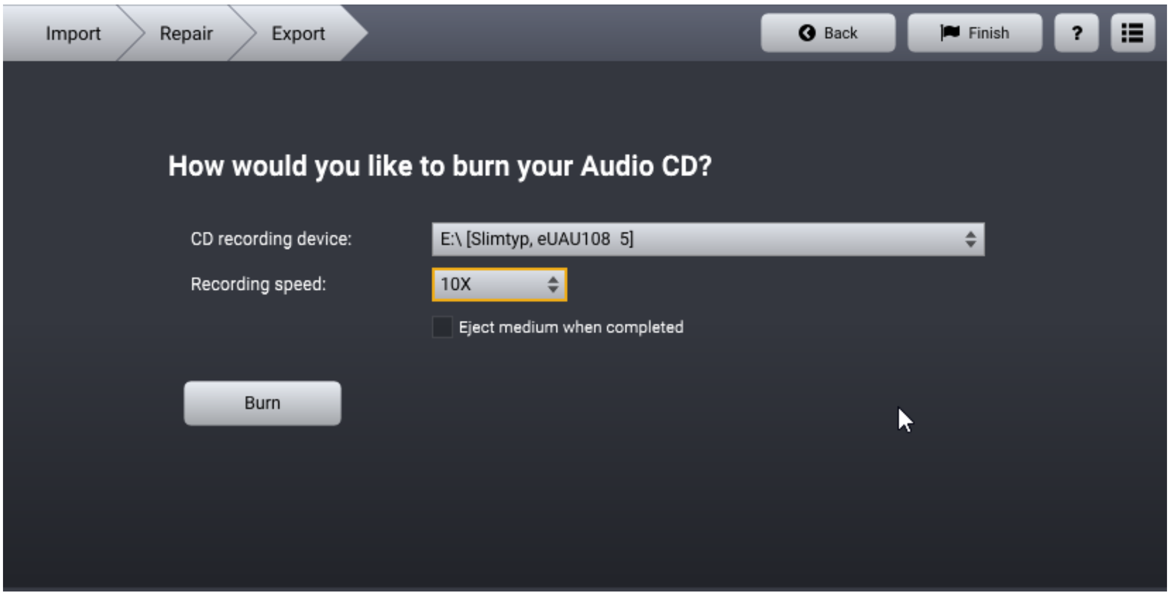
After you have revised the track list, set up the restoration tools and possibly added additional processing, you can proceed to the *Export* page using the *Next* button in the top right corner of *AudioLava's* main window. You can export your restored tracks to audio files or burn them directly to a CD. Please choose one of the two in the *Export* page:



The Export page where you can burn your tracks to an audio CD or save them as files on your computer.

3.3.1 Burn a CD

If you choose to burn a CD in the *Export* page, *AudioLava* proceeds to the burning page:

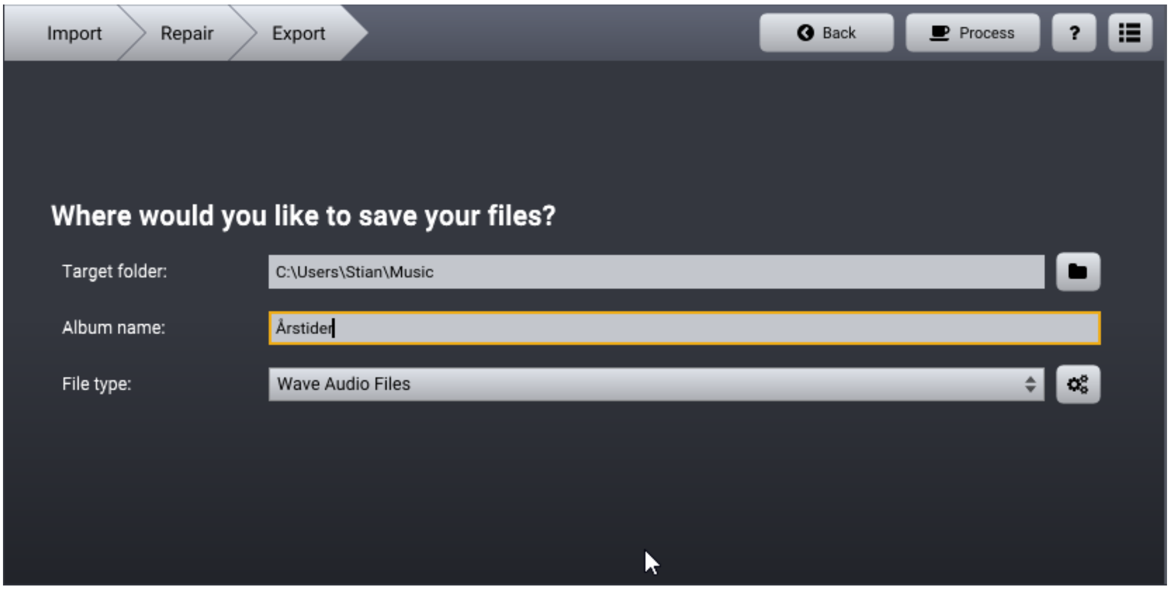


The CD burner dialog allows you to select a CD recording device and recording speed.

Insert a blank CD-R or CD-RW into the CD recorder. If you have several CD recording devices installed on your computer, make sure you choose the correct one from the *CD recording device* drop-down list. You can choose among different recording speeds. Click the *Burn* button to start burning.

3.3.2 Export to Audio Files

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

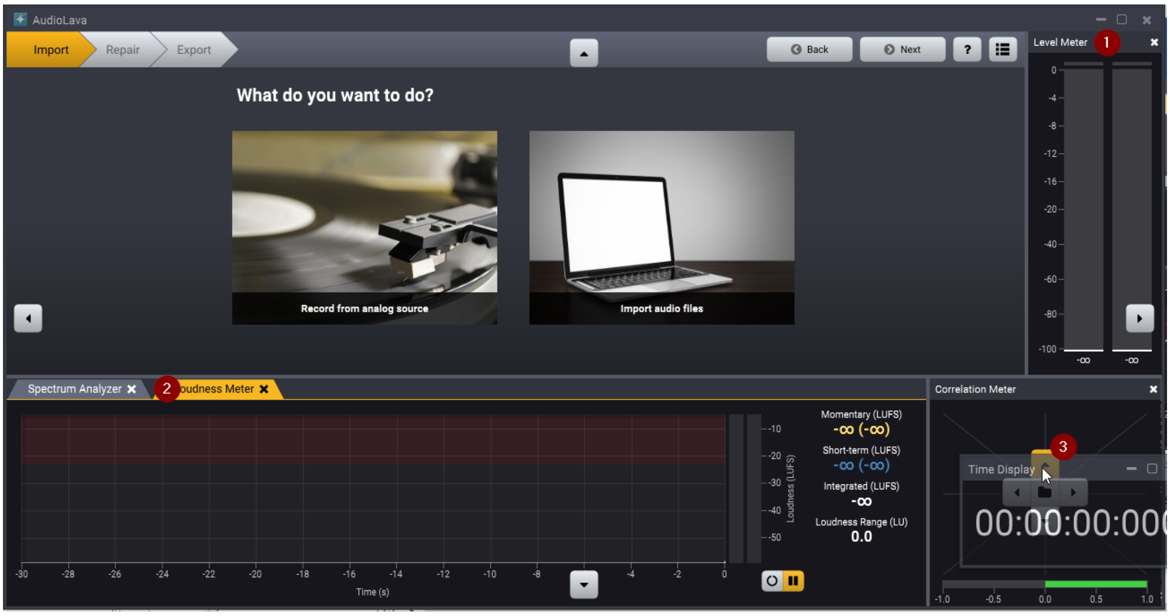


The File Export Page.

You can choose a *Target folder* for your tracks, an *Album name* and the *File type* of the exported tracks. During the export, a 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. Click the *Process* button in the top right corner of *AudioLava's* main window to start exporting your files.

4 Customizing the Workspace

AudioLava has a flexible docking system which allows you to dock pane windows such real-time analyzers or anchor editors in order to customize the workspace. Pane windows can be docked at one of the sides of an existing pane window or the main window.



Docking options in AudioLava. You can move the mouse cursor over one of the dock buttons to dock a pane window.

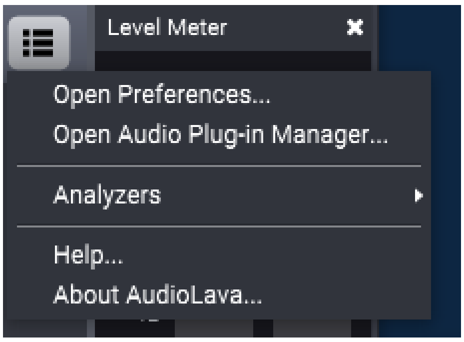
To dock a pane window, click a header of a pane window (1) or the tab header (2) if it is grouped and keep the mouse button pressed. As soon as you move the mouse, dock buttons are shown (3) to indicate possible docking position. Move the mouse cursor over any of the available dock buttons and release the mouse button. You can release the mouse button anywhere else on the screen to remove the window from the dock and create floating window.

Restoring the Default Layout

You can always restore the default pane layout by choosing *Restore Default Layout* from the *View* menu.

5 Application Menu

You can access the *AudioLava* application menu using the menu button on the right side of the wizard header:



The AudioLava application menu can be accessed through the menu button as shown here.

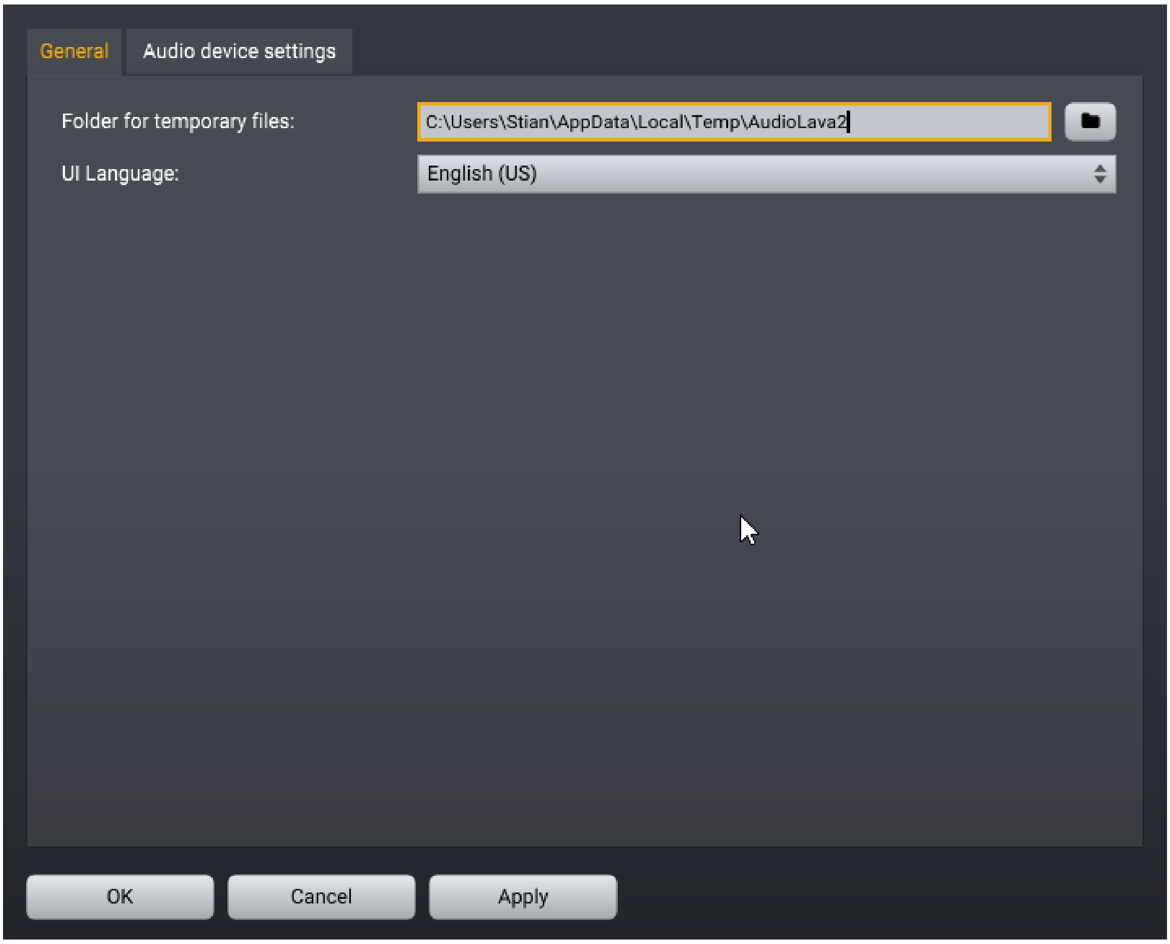
5.1 Preferences

Choose *Open Preferences...* from the application menu to open the preferences editor. There are two categories:

- General 47
- Audio device settings 48

5.1.1 General Settings

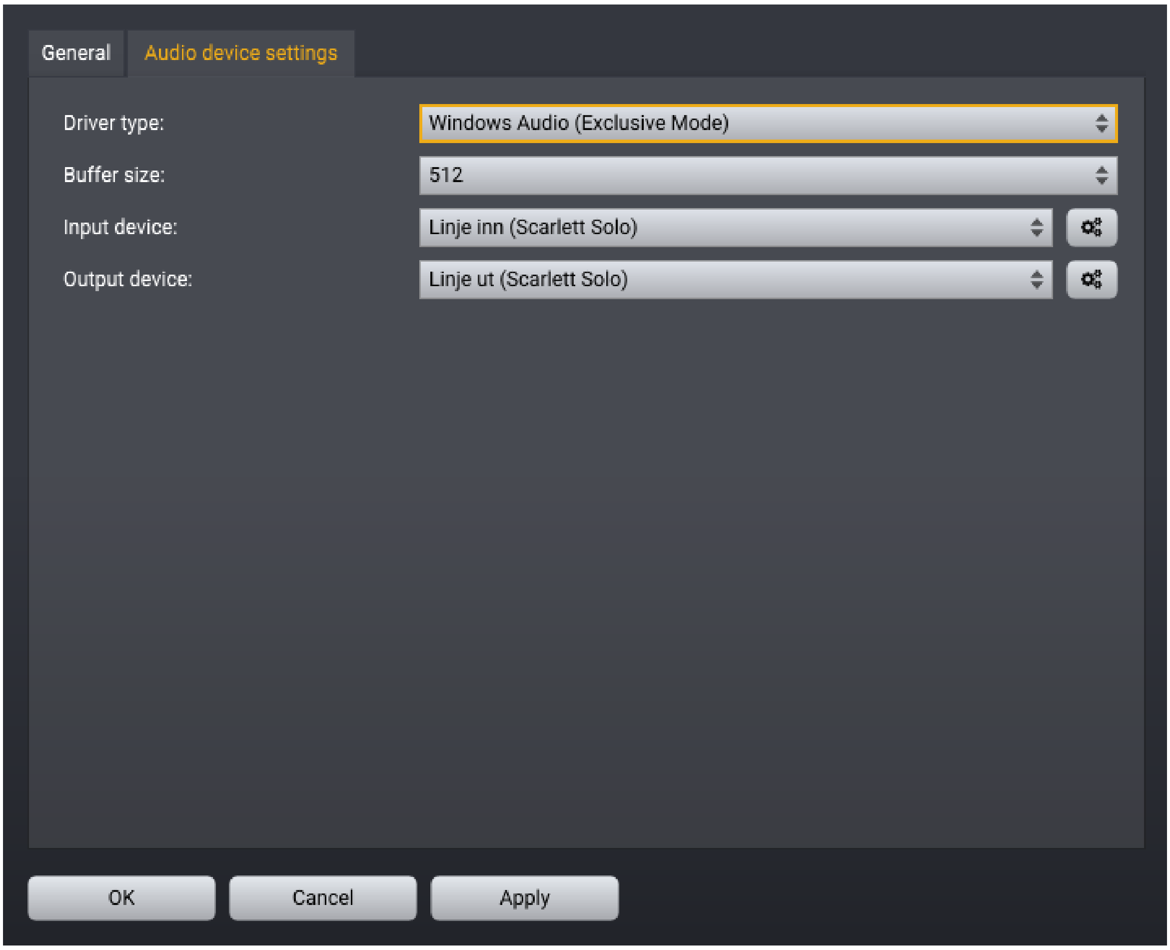
The general settings tab lets you define a folder for temporary files and choose the language for the user interface. Temporary files should be stored on a quick hard or SSD drive with plenty of free space.



The General settings where you can choose the directory for temporary files and the user interface language.

5.1.2 Audio Device Settings

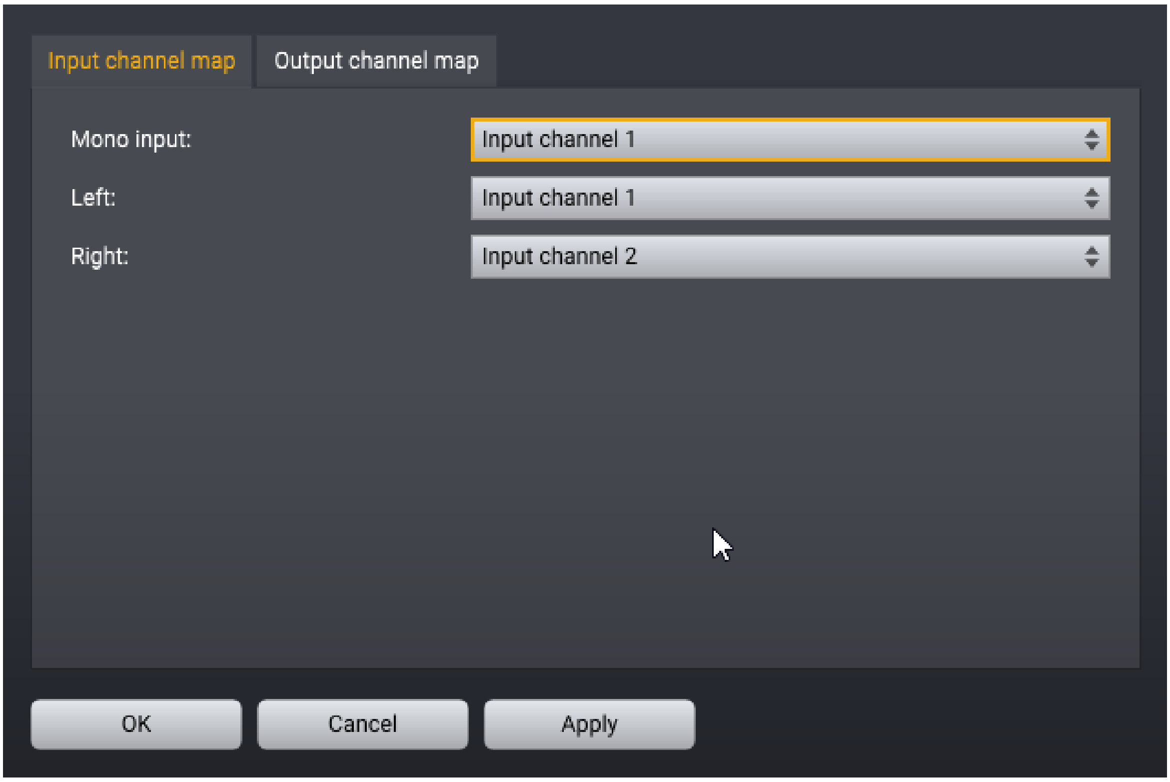
You can set up your audio device and assign input and output channels by choosing *Open Preferences* from the application menu and clicking the *Audio device settings* tab.



The audio device settings let you choose a driver type, buffer size and the input and output device to use.

We recommend ASIO as driver type on Windows if available and CoreAudio on Mac. The recommended buffer size is 512 samples.

If you have an audio device with several input and output channels or you are working with surround audio, you can set up the channel routing by clicking the button with the cogs icon to the right of the input / output device settings and the following window appears:

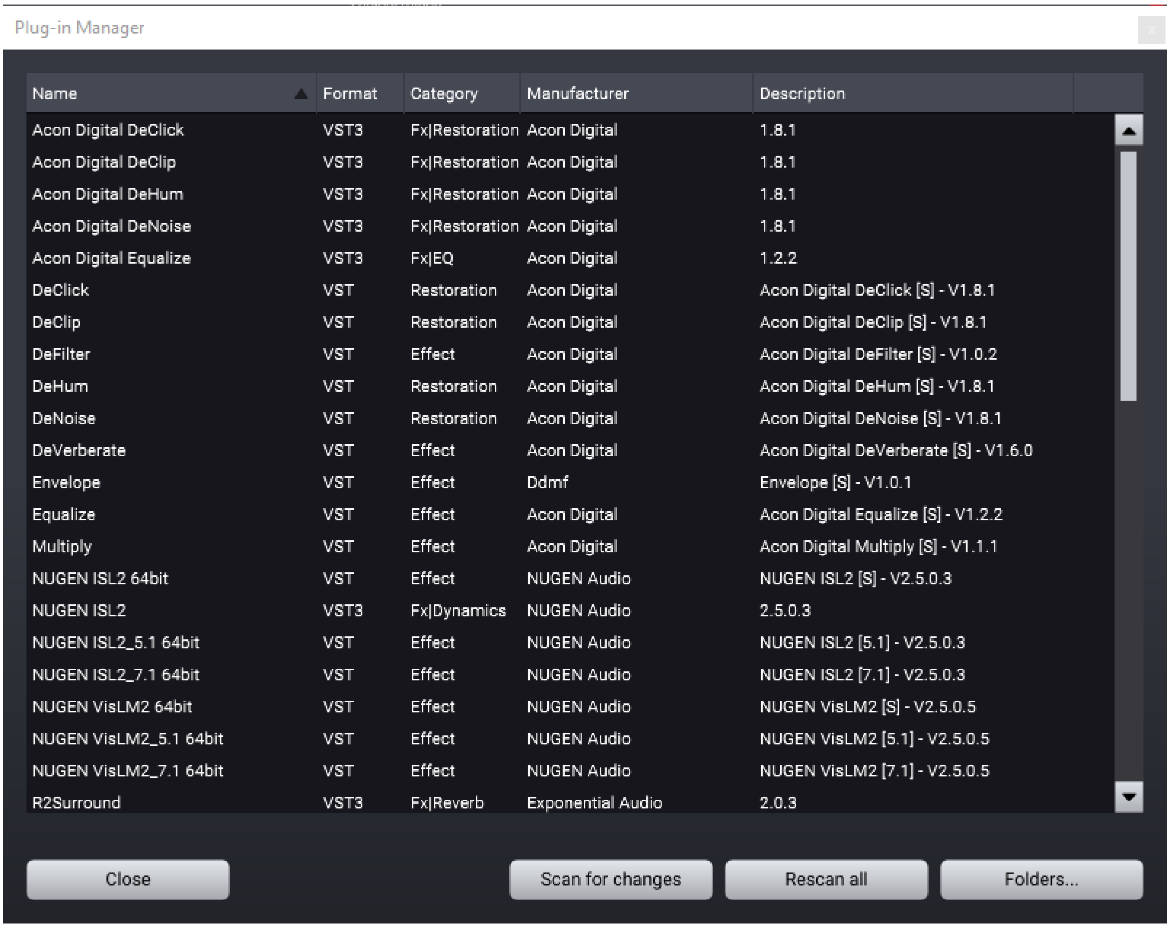


You can set up the input and output channel routing maps in AudioLava.

The channel map editor lets you assign channels on your audio interface to the speaker positions defined in AudioLava.

5.2 The Plug-in Manager

The *Plug-in Manager* lets you scan for new plug-ins and manage the existing ones. Choose *Open Audio Plug-in Manager...* from the application menu to access the plug-in manager.

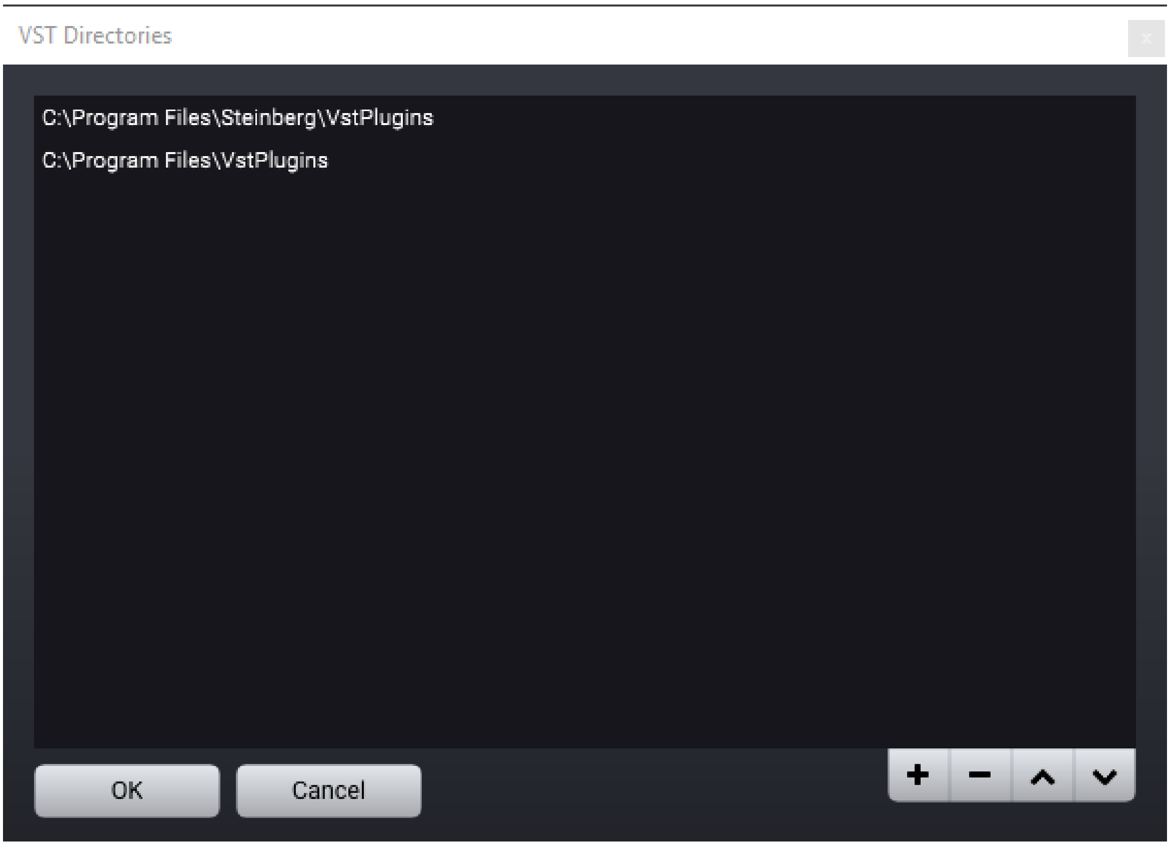


The plug-in manager in AudioLava with a list of detected plug-ins.

You can scan for new plug-ins using the *Scan for changes* button. Only newly added plug-ins will be validated and no longer existing plug-ins will be removed. If you click the *Rescan all* button, AudioLava will validate all plug-ins including those who have already been validated successfully.

Note: You can remove one or more plug-ins from the list by clicking it's entry in the list and press the delete key.

The VST3 and AU plug-in standards define the directory where plug-ins are situated. That's not the case for VST prior to version 3, though, and you can add or remove VST search folders by clicking the *Folders...* button. The following window appears:



List of VST search folders.

You can click the button with the plus icon to add a folder and the button with the minus icon to remove an existing entry. Furthermore, you can rearrange the search order using the up and down arrow buttons. Click *OK* when you are done.

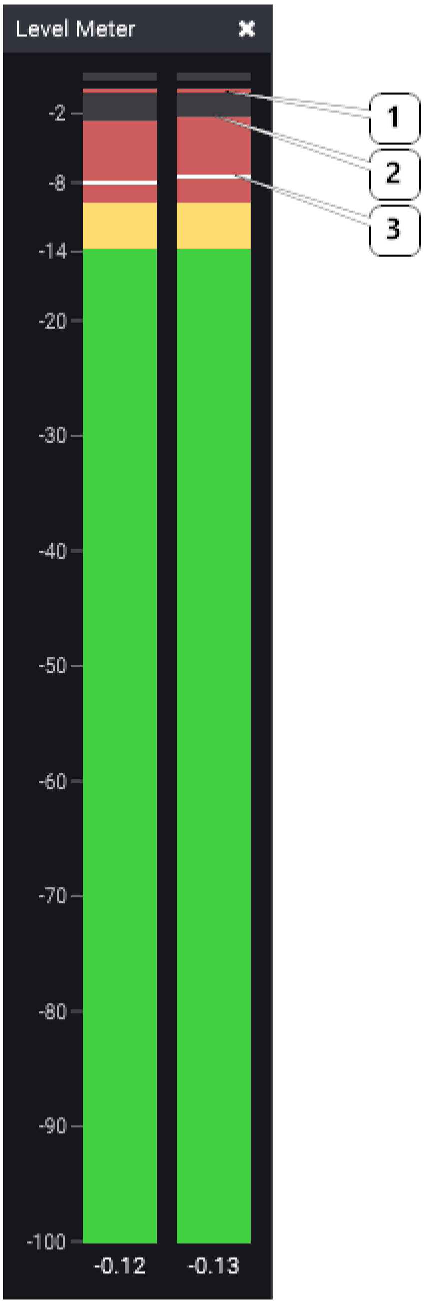
5.3 Realtime Analyzers

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

5.3.1 Level Meter

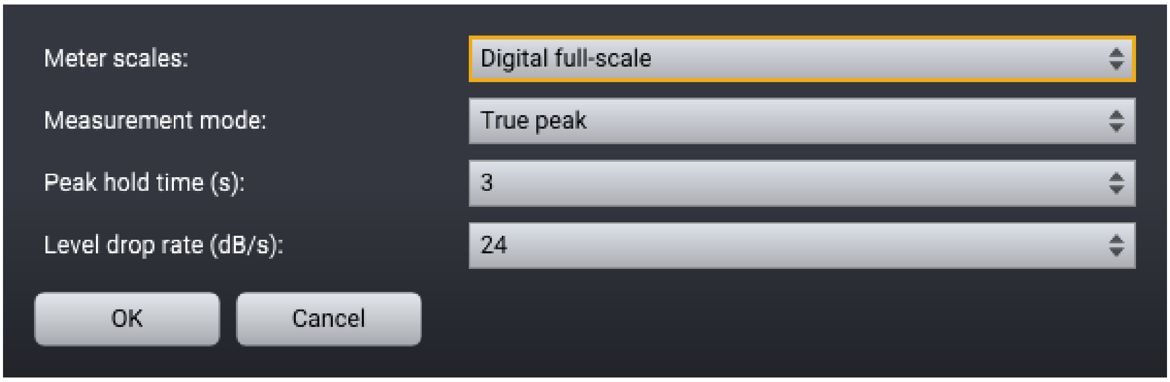
About Level Meter

The level meter lets you analyze the output level in terms of true peak, sample peak, peak hold and RMS values. The sample peak value is the maximum sample within a short analysis interval and is the value defining the height of the level meter bars. The true peak value takes into account how digital to analog convertors (DACs) would reconstruct an audio waveform, where it is possible for digital clipping to occur, even when the sample peak is never beyond 0dB. 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 RMS level is calibrated according to the AES17-1998 standard which is 3 dB higher than the mathematical RMS level.



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

You can configure the level meter to use different scales or change ballistics by clicking the left mouse button somewhere in the level meter. The following dialog box appears:



The settings dialog for the level meters.

AudioLava supports the K-System metering standard proposed by the audio engineer Bob Katz. The K-System is an attempt to standardize leveling practices throughout the audio industry. Three standards are available, K-20, K-14, and K-12 which are intended for different listening environments. You can choose to use one of the K-System meters or use the digital full scale meter as in earlier versions of AudioLava.

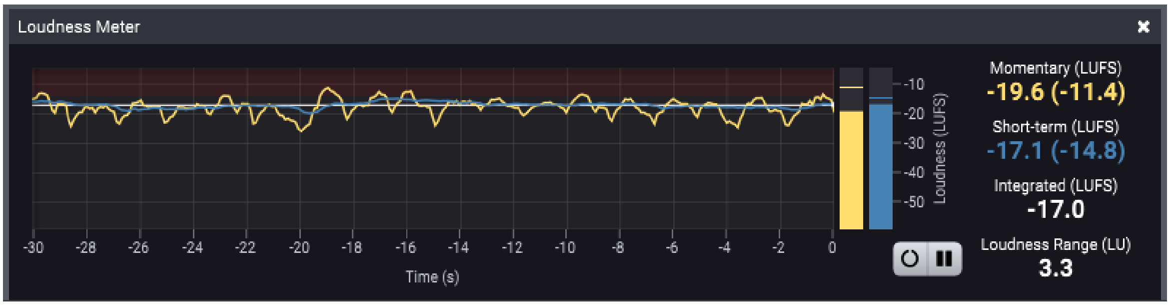
5.3.2 Loudness Meter

About Loudness Metering

Peak and RMS level metering don't necessarily match what we perceive as loudness very well. Short term RMS levels are related to loudness, but the sensitivity of the human auditory system is frequency dependent and that needs to be taken into account when measuring perceived loudness. The EBU R128 (based on the ITU-R BS.1770) recommendation defines a more suitable way of measuring loudness and deals with important issues such as how to react to loudness changes over time and how to measure loudness in multichannel audio. These two recommendations are becoming increasingly popular and are important to ensure a consistent listening experience when switching between audio tracks, television channels, radio programs and similar. Most music streaming providers and broadcasting organizations now specify loudness requirements that content providers need to adhere to.

The Loudness Meter in AudioLava

AudioLava has loudness metering built in that follows the EBU R128 and ITU-R BS.1770 recommendations.



The loudness metering in AudioLava showing the loudness curves from the last 30 seconds of output audio along with the current momentary loudness, short-term loudness, integrated loudness and the loudness range as defined by the EBU R128 recommendation.

There are four different loudness measures defined by the EBU R128 recommendation and all of these are displayed in the loudness meter in *AudioLava*:

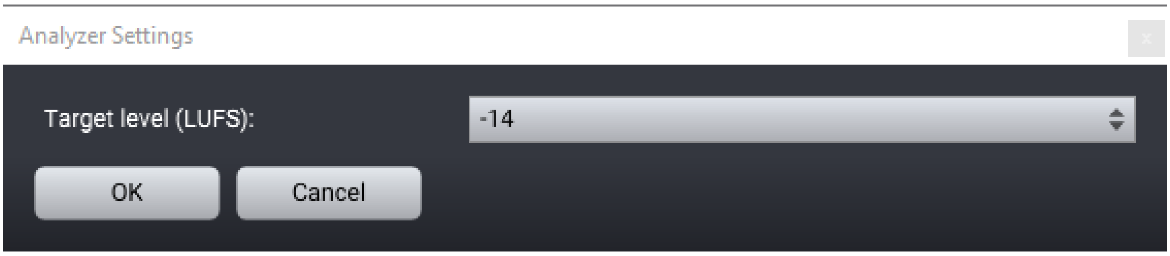
- **Momentary (LUFS)**
  The momentary loudness is calculated based on audio from the past 400 milliseconds and is visualized with a yellow curve in the loudness history graph.

- **Short-term (LUFS)**
  The short-term loudness is calculated based on audio from the past 3 seconds and is visualized with a blue curve in the loudness history graph.

- **Integrated (LUFS)**
  The integrated loudness is a measurement of the loudness of a complete audio track or program. Absolute and relative gates are used to avoid silent or very soft periods from affecting the loudness measurement. The integrated loudness is visualized with a white or red horizontal line in the loudness history graph. The indicator is white when the integrated loudness is below the target level and red if it is above. You can set the target level in the analyzer settings (see description below).

- **Loudness Range (LU)**
  The loudness range is a measure for the loudness variance over time. Rapidly changing levels will result in a higher loudness range measurement.

AudioLava also indicates the illegal loudness range with a red background color in the loudness history. You can change the target level by clicking anywhere on the background of the loudness meter and the following window appears:

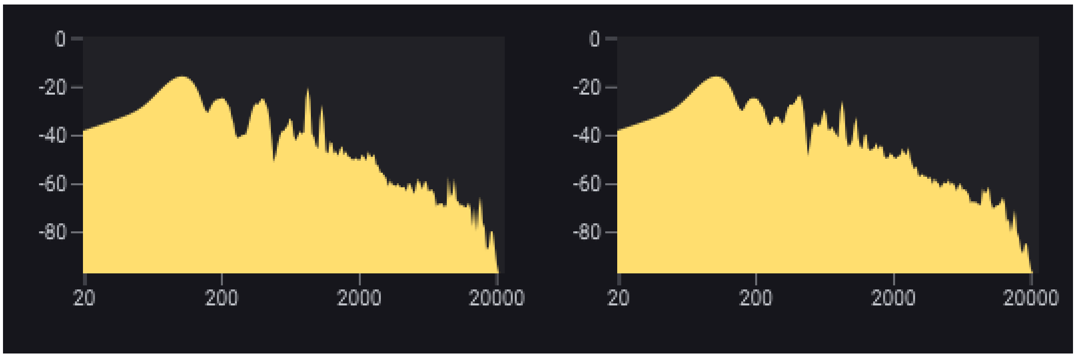


The Loundess Meter Settings lets you select the target level in LUFS

5.3.3 Spectrum Analyzer

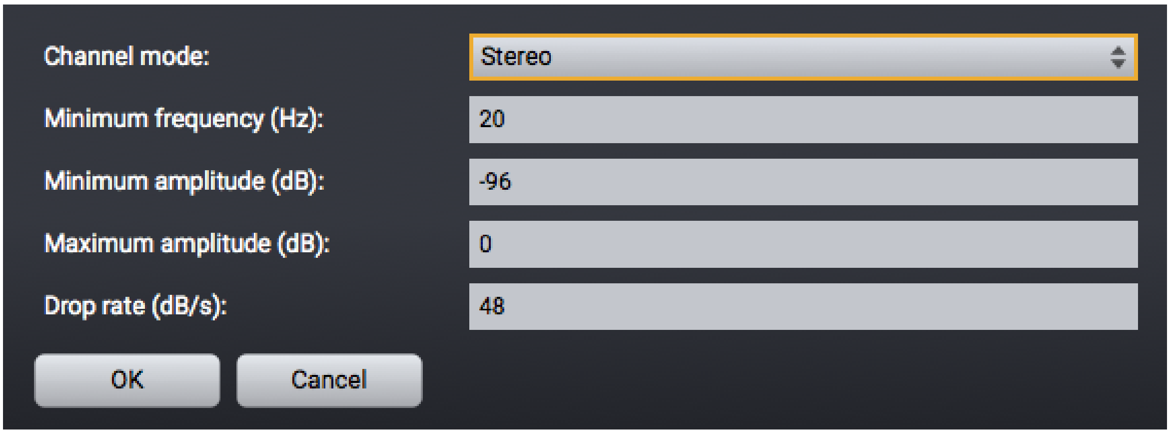
About Spectrum Analyzer

The Spectrum Analyzer shows the frequency content of short analysis time frames, using FFT, which stands for Fast Fourier Transform. FFT is an efficient way of calculating the frequency domain of a signal.



The Spectrum Analyzer shows the frequency content of the output audio signal.

You can configure the Spectrum Analyzer by clicking the left mouse button somewhere in the Spectrum Analyzer. The following dialog box appears:



The spectrum analyzer settings dialog.

Channel mode: defines, weather a single summed display or two displays for both

stereo-channels are shown.
Minimum frequency: defines the lowest frequency covered by the FFT analysis
Minimum and Maximum amplitude: set the dynamic range of the displayed area.
Drop rate defines the speed for the curve to fall back to lower levels.

5.3.4 Phase Correlation Meter

About 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 for 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 and the correlation value indicated in the bar graph at the bottom should range on the positive side.

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

5.3.5 Time Display

The big time display shows the current playback position in a dockable window pane

Index

- S -

- T -

- U -

- V -

- W -