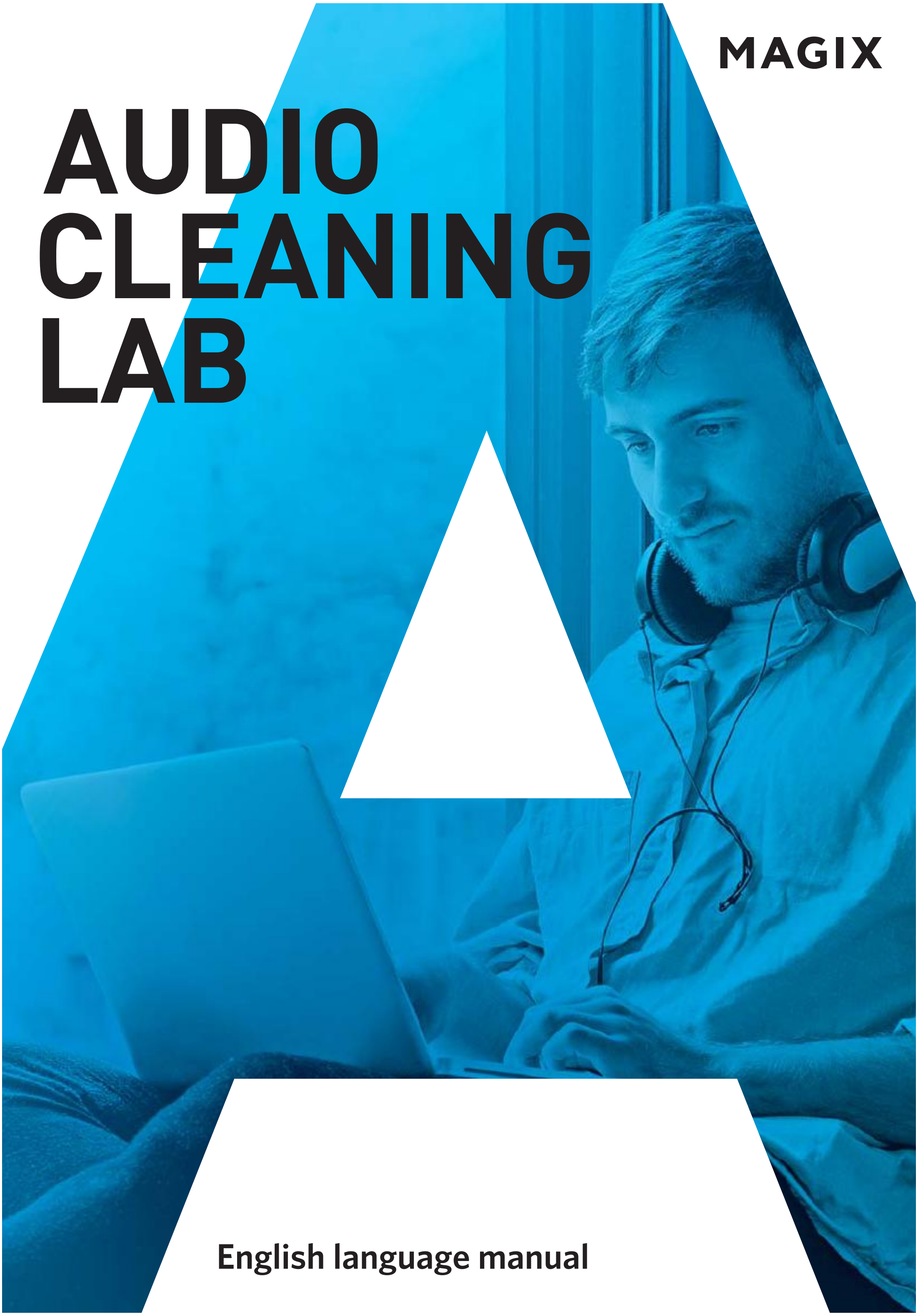
MAGIX

# AUDIO CLEANING LAB



English language manual

# Copyright

# Preface

Congratulations! You now own a high-performance digital audio lab, which is perfect for cleaning your records, tapes, CD tracks, MP3 collections, audio or video files, enhancing the sound, combining your media in any order, and burning or exporting everything in optimum quality directly onto CD or DVD.

The volume and sound on each track can be perfectly synced and equalized. Various audio formats can be combined, simultaneously edited, and burned. The method is especially easy and clear, since automatic settings, assistants and step-by-step instructions make sure that the process goes smoothly and easily. Neither previous experience using the software nor long processing time are necessary.

The print manual provides an introduction and tutorial that explains all the most important features with step-by-step instructions.

In addition, an electronic manual is supplied in PDF, which systematically explains all of the program's components one after the other. You can also use the program's help file by pressing "F1". If you prefer to discover the many possibilities of the program by yourself, then use the PDF manual and help file simply as a reference. An alphabetical index is included for this purpose.

Have fun with MAGIX Audio Cleaning Lab.

The MAGIX team.

# Table of Contents

# Support

Dear MAGIX customer,

Our aim is to provide fast, convenient, solution-focused support at all times. To this end, we offer a wide range of services:

- **Unlimited web support:**
  As a registered MAGIX customer, you have unlimited access to web support offered via the convenient MAGIX service portal on http://support.magix.net, including an intelligent help assistant, high-quality FAQs, patches and user reports that are constantly updated.
  The only requirement for use is product registration at www.magix.com
- **The online community, on-the-spot support and a platform for exchange:**
  MAGIX customers have free and unlimited access to the online community at www.magix.info, which includes approx. 150,000 members and offers the opportunity to ask members questions concerning MAGIX products as well as use the search function to search for specific topics or answers. In addition to questions & answers, the knowledge pool includes a glossary, video tutorials and a discussion forum. The multiple experts, found round-the-clock at www.magix.info guarantee quick answers, which sometimes come within minutes of a question being posted.
- **Email support for MAGIX products:**
  For each new MAGIX product you will automatically receive 12 months free customer support by email.
- **Premium Email Support:**
  For priority support, or if you want the support team to help with non-MAGIX related hardware problems you can purchase a premium email support ticket. Proceed as follows:
  1. Go to the MAGIX customer support page at http://support.magix.net.
  2. Sign in using your login details.
  3. Click on "Purchase access code" in the navigation bar.
     Each ticket corresponds to a specific problem, it will remain valid until the problem has been solved. A ticket is not limited to a single email.

**Please note:** To be able to use the Premium email support and free product email support via the Internet, you have to register your MAGIX product using the serial number provided. The serial number can be found on the sleeve of the installation disc or on an insert card included in the package.

- **Additional telephone service:**
  Besides the large number of free customer service offers, we also offer a fee-based telephone customer service.

Here you can find a summary of our technical support telephone numbers:

http://support.magix.net/

**Mail (Europe):** MAGIX Development Support, P.O. Box 20 09 14, 01194 Dresden, Germany

**Mail (North America):** MAGIX Customer Service, 1105 Terminal Way #302, Reno, NV 89502, USA

**Please make sure you have the following information at hand:**

- Program version
- Configuration details (operating system, processor, memory, hard drive, etc.), sound card configuration (type, driver)
- Information regarding other audio software installed

### MAGIX Sales Department

You can reach the MAGIX Sales Department workdays for help with the following questions and problems:

- Orders
- Product consulting (pre-purchase)
- Upgrade requests
- Returns

## Europe

Monday - Friday, 09:00-16:00 GMT

**U.K.:** 0203 3189218

**Denmark:** 45 699 18763

**Sweden:** 46 852 500713

**Finland:** 35 89 42419023

**Norway:** 47 210 35843

## North America

9 am to 4 pm EST Mon-Fri

**Phone:** 1-305-722-5810

# Introduction

## What is MAGIX Audio Cleaning Lab?

MAGIX Audio & Music Lab Premium is a gentle but powerful cleaner for all kinds of acoustic material including records, tapes, CD tracks, speech recordings and MP3s. Digital cleaning removes everything from light crackling to severe interference on scratched records, old cassettes and MP3s that have been compressed too many times. A fine polish of the sound brings new life to every song and the disc-burning function lets you save your sensitive sound material on audio or data CD to protect it from further damage.

Individual tracks, even individual passages within a track can be given their own effect settings. Additionally, the entire sound can be cleaned up, refreshed, and especially important for compilations, the volume can be balanced.

MAGIX Audio & Music Lab Premium is fast, easy-to-use and very gentle: Almost all effects are calculated in realtime during playback without damaging the recorded material in the least. The original recordings and songs remain untouched on the hard disk. This means you can experiment as much as you like without having to worry about causing any lasting damage to your audio material.

## MAGIX Audio & Music Lab Premium and MAGIX Audio Cleaning Lab

Audio & Music Lab comes in two versions, MAGIX Audio Cleaning Lab and MAGIX Audio & Music Lab Premium. This program help file and manual describes the functions in MAGIX Audio & Music Lab Premium. Program functions that are not available in the "smaller" version MAGIX Audio Cleaning Lab are indicated throughout document.

The following functions are only available in MAGIX Audio & Music Lab Premium and are not in MAGIX Audio Cleaning Lab:

- **24-bit Recording:** The default setting for recording is CD quality (44.1 kHz sample rate, 16-bit resolution). MAGIX Audio & Music Lab Premium can record at higher resolutions and bit rates if the sound card supports it.
- **More tracks:** You can switch to two or four audio tracks which enables you to mix music with speech and other sounds. This makes it possible to include spoken commentary parallel to the other audio in a video recording.
- **More analyzer types:** MAGIX Audio & Music Lab Premium offers nine different analyzers - MAGIX Audio Cleaning Lab only offers two.
- **VST Effects:** MAGIX Audio & Music Lab Premium supports the integration of third-party VST effects, some effect plug-ins (AM-Track SE, Energizer and the Essential FX) are included.

- **Video editing:** Load and edit the video sound, the Video Sound Optimizer effects and the video preview monitor are only available in MAGIX Audio & Music Lab Premium.
- Recording **web radio stations** (view page 52)
- **A scrubbing slider** (view page 36)
- **Batch processing** (view page 131)
- **Spectral view and spectral editing** (view page 32)
- Voice-over function (view page 130) for automatic volume curves on background music
- **Improved Auto Mastering:** MAGIX Audio & Music Lab Premium uses object effects for the Auto Mastering process which leads to better results, especially in projects that contain audio from various sources.

# What is new in MAGIX Audio Cleaning Lab?

### User interface improvements

We implemented many user suggestions to improve the user interface:

- A better readability of the effects presets.
- Dedicated Buttons for edit functions (Copy, Paste...)
- The order of the effects was optimized, the most important effects are placed topmost in the fx sections, so you can acess them withou scrolling also on smaller screens. You can even change this order to suit you needs.
- You can now switch between the effects via button from within an effect dialog.
- Options regarding the program display are moved into a new "View" menu, there are new keyboard shortcuts to maximize single screen areas.

### More and better effects

- The detailed effect views are now also available in MAGIX Audio Cleaning Lab.
- Auto Cleaning was integrated in the main programm interface and has better setting options.
- The Timestretching now uses the zPlane Elastique Pitch algorithms for better quality.

Only in MAGIX Audio & Music Lab Premium

- There is a secondary reverb effect available, the essentialFX Reverb
- In the mouse mode "Edit spectrum directly" (spectral cleaning) there is a secondary play button for a comparison between filtered and unfiltered audio.

### Better edit functions

- A new mouse mode Range Mode. For faster cutting there is now a Range Mode in MAGIX Audio & Music Lab Premium where you can select a time range in the audio material for editing.
- You can have up to four tracks in MAGIX Audio & Music Lab Premium.
- You can change the curve form also for single fades (not just crossfades).

### More Improvements

- A better video import (only MAGIX Audio & Music Lab Premium): Audio & Music Lab is capable of importing a lager range of videos directly, without converting it first. The calculation of the video preview strip is done in a separate process so you can play the video immediately after loading.
- Search function: With the search function you can search for a key word in every menu item, help topic or effects preset.
- No fee based activation for using the AAC codec (except Windows Vista).

# Features

## Import

You can either import existing audio files in many conventional formats into MAGIX Audio Cleaning Lab or simply use MAGIX Audio Cleaning Lab to record your own music. Whether from cassette, tape, LP, or Internet streaming, with only a few clicks you can digitize your music and edit it further with MAGIX Audio Cleaning Lab.

## Cleaning

The key feature of MAGIX Audio Cleaning Lab is its ability to remove unpleasant noise in music and enhance the overall sound. There are numerous professional tools available for this such as the "DeClicker", "DeCrackler", "DeClipper", "DeNoiser" (including DeRumbler preset), and "DeHisser". You can also add several sound effects to your music.

## Mastering

So that your recordings sound optimal, a selection of mastering tools are available to you once you have cleaned up the audio material. Try them out yourself to see which settings are the best, or let MAGIX Audio Cleaning Lab do the work for you by searching for the best settings automatically. For more detailed information about the tools, please read MAGIX Audio Cleaning Lab's help file.

## Export

Of course, you can also export your recordings. There is a wide range of formats available which enable you to enjoy your recordings anywhere you like.

**Audio Import:** WAV, MP3, WMA, AIFF, OGG Vorbis, M3U, CUE, CD-A, FLAC, AAC[3]

**Audio Export:** WAV, MP3[1], WMA, AIFF, OGG Vorbis, CD-A, FLAC, AAC[3], Data-CD/DVD

**Video import/export[2]:** MPEG-2[4], MPEG-4 (incl. AVCHD)[4], MOV, WMV, MXV

[1] Requires Microsoft Media Player
[2] Not in MAGIX Audio Cleaning Lab
[3] Fee-based activation under Windows Vista
[4] Requires free activation under some circumstances

# Quick start

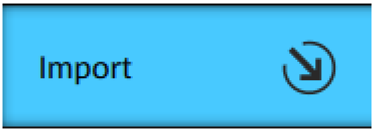
In this chapter, we will guide you through all of the important functions in MAGIX Audio Cleaning Lab step-by-step. You don't need any special experience; just some time for recordings and hard drive space.

With MAGIX Audio Cleaning Lab you can load audio material from a number of different sources into projects in order to clean it up, edit it, or export it.

Analog material like records or tapes have to be digitized first so that it can be processed by the computer. The recording function in MAGIX Audio Cleaning Lab can be used for this purpose.

## Load and playback audio files

After starting the program, MAGIX Audio Cleaning Lab displays an empty project window. You are in the "Import" section, which is recognizable by the lit button below the project window.

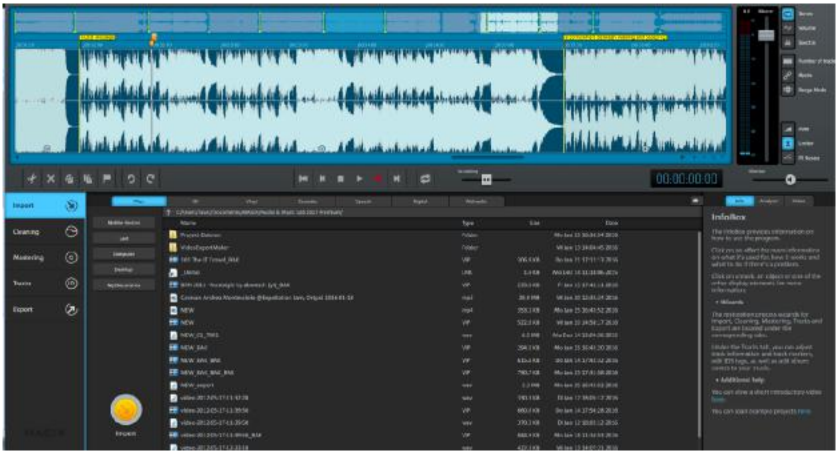



In the import area you can choose from seven (six in MAGIX Audio Cleaning Lab) different options, depending on what type of audio material is to be edited:


- You can load your audio files, e. g. in MP3 format, by clicking the "Audio files" button. This will open the file browser.
- You can import audio CDs via the "CD" button.
- You can record records or audio cassettes using the "Record" or "Cassette" buttons.
- Speech recordings are created using a connected microphone in the "Speech" area.
- "Digital" can be used to record the current playback of the computer, e.g. the Internet browser.
- "Web radio" allows you to record web radio stations.

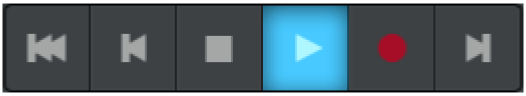
The file browser works like the Windows file explorer. Search for the files in the file system of the computer and select them with the mouse. Several individual files can be selected with Ctrl + mouse click. For a group of files, click on the first file first, then press shift and click on the last file. Click on "Import".

After import, a waveform display of the audio track appears in the main window. Each file is symbolized by an individual checkbox, a so-called "object". More information on the objects can be found in the chapter Cutting and arranging objects (view page 54).



The waveform display gives you an idea of the audio track's progression. Even some noises can be recognized. Crackling, for example, creates a clearly identifiable spike in the waveform.

Below to the right, you'll find the transport controls for playback, stop and moving the playback marker. You'll be familiar with these functions from other playback devices.

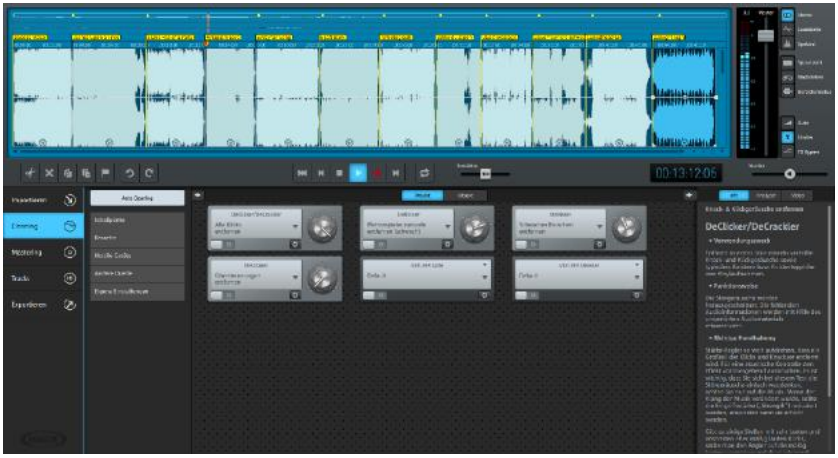


Use the mouse for quicker navigation: If you click on a position in the timeline above the waveform display, the position line will be moved exactly to this position. You can start and stop the track by pressing the spacebar of your keyboard.

# Cleaning

There are many different types of noise. Let's start with the easiest ones: Constant humming or irritating hissing. These kinds of noise are usually caused by tape recorders, record players or microphones and are audible throughout the complete track. Find out how to get rid of these irritating noises:

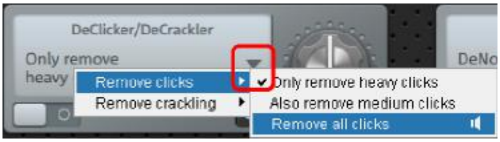
- Switch to the "Cleaning" section.




In the middle of the function area you can now find various function modules for noise removal that can be used individually or combined. If you click on one of these modules, you will find explanations regarding purpose and workflow of the individual module on the info monitor located in the bottom right-hand corner. To the left side of them, you can find presets for the entire Cleaning area, sorted into headings depending on the sound source of the source material (record, cassette,...).

The modules are operated in the same way:

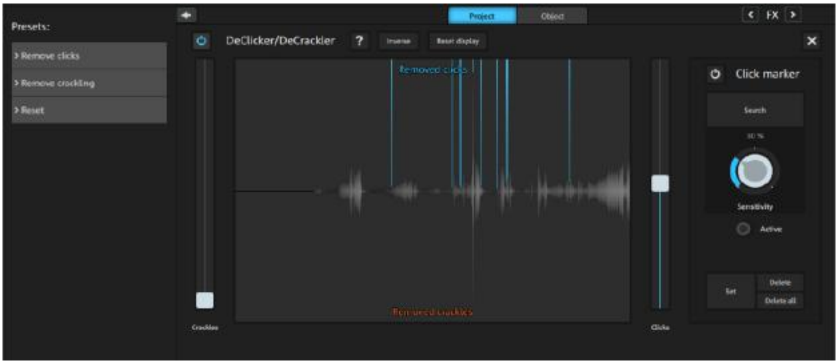

You can use the knob to control the amount of each cleaning effect.




If you are not satisfied with the presets in the start dialog, you can choose further presets using the arrow menu.


To get an idea of how the selected effect affects your audio material, you can turn it on and off by clicking on the symbol in the left-hand corner.


The cog wheel button at bottom right is used to open a user interface where you can set other parameters of the effect.




Depending on the effect unit, you can specify very effective audio editing settings here. To do so, it's necessary to know a bit about audio editing though. Usually, however, you will not need these special functions, but they can be useful in complex cases. You can find more information in the "Cleaning effects" (view page 62) chapter.

# Mastering

- Click on the "Mastering" button.

"Mastering" is the second effects area and is meant for optimizing audio material. For example, you can increase the volume of the music (Multimax), copy the audio characteristics of a particular musical style (Sound Cloner (view page 80)) or sonically "freshen up" old MP3s with a low bit rate (Brilliance Enhancer (view page 75)).

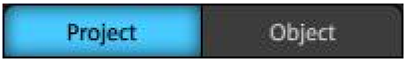
Here you will also find a column with preset categories and the various effect modules on the right.

The functionality of the optimization effects is the same in principle: Adjust the strength of the effect via the slider, switch the effect on and off to compare the advanced settings of an effect by clicking on the symbol in the bottom right-hand corner.

# Object effects

If you combine the audio material from various sources, it's normally not sensible to apply cleaning and mastering effects to all objects in the same way. Tape recordings typically have different noise than LP recordings. Therefore, there are two different ways you can use the effects: Project effects and Object effects.

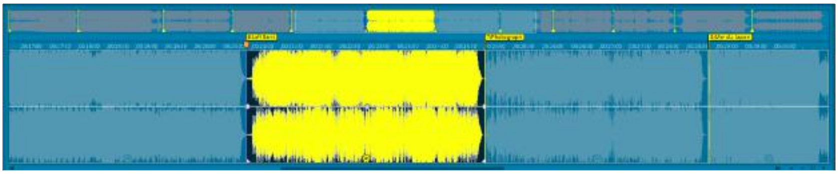
In the cleaning and mastering area you can see how the two types of effects are applied. "Project" is the default setting.




The settings for project effects are applied to the entire sound, i.e. for all objects in the master track.

Each object can also have its own individual effect settings. To add an effect to an object, click on "Object".

This will highlight the object that you want to edit. There are also many audio effects available. These effects can be set separately for every object in the master track .



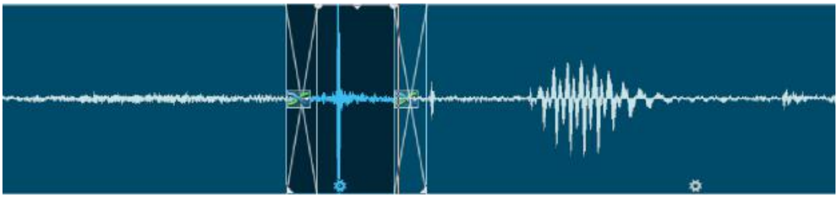
# Remove unwanted sections

Now let's take a look at some special kinds of audio noise that occur when something bangs against the microphone.

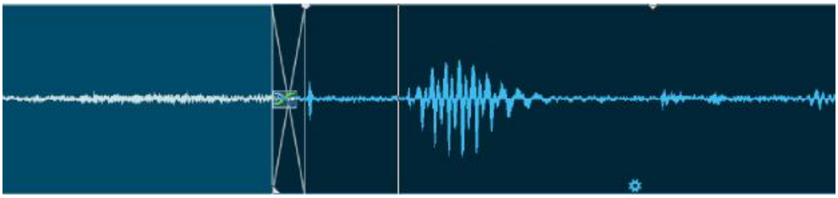
These types of clicks, pops, rustling or even longer unwanted sounds can be cut out of the track by using the scissors tool in the track window.

> **Note:** This technique is only useful if the noise is the only sound and nothing else should be heard. If a click or pop occurs in the middle of the music, it's better to use theRetouching clicks and pops (view page 21) method described in the section below.

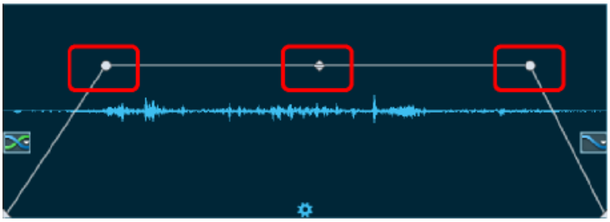
1. Play the audio track and search for the section that you want to cut out.
2. Set the position line in front of this section.
3. Now click on the scissors icon. This will cut the object at this position.
4. Now place the playback marker behind the section that you want to remove and cut again.
5. This creates three separate objects on the track.



- Select the object in the middle and delete it by pressing the "Delete" key.



The back of the object automatically moves back and slightly overlaps with the front part of the object. You can use the handles to make fine adjustments to the transition.



- There are handles at the top left and right hand corners, which allow you to create fade ins and outs.
- The handle at the top center is for adjusting the volume. If you pull it downwards, the volume of the object is reduced.
- The handles at the bottom left and right corners adjust the object borders. You can fine tune your cuts by extending or contracting the object.

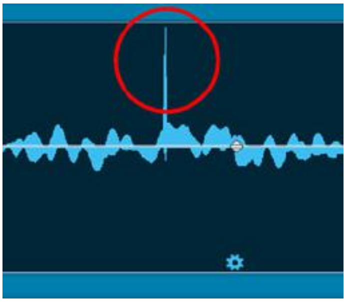
# Retouch short noises such as clicks or pops

Let's take a look at a quick and elegant way of retouching short noises such as pops and clicks using a pen tool.

**Note:** You can't undo editing that you did with the waveform drawing tool. After selecting the pen tool, you can decide if you want play it safe and edit in a copy of the file or proceed directly in the original material.

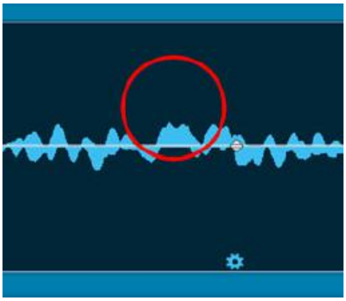
- Search for a section in the material where you hear crackling and set the playback marker to that position.
- Zoom into that section so that you can see the details of the waveform display. You can either use the zoom button in the bottom right corner of the track window.

- Take a close look. Can you see the click or pop? Look for a "mountain" that is flat on top or exceeds its surroundings.



- Select the Wave Draw mode ("Options" > "Wave Draw mode").

The mouse pointer turns into a pen which you can use to draw directly into the waveform. If you click, the zoom level increases automatically and you can draw. It's about trying to turn the flat form a bit more into a peak and moving it slightly towards the bottom. The critical section should then look like this at the end:

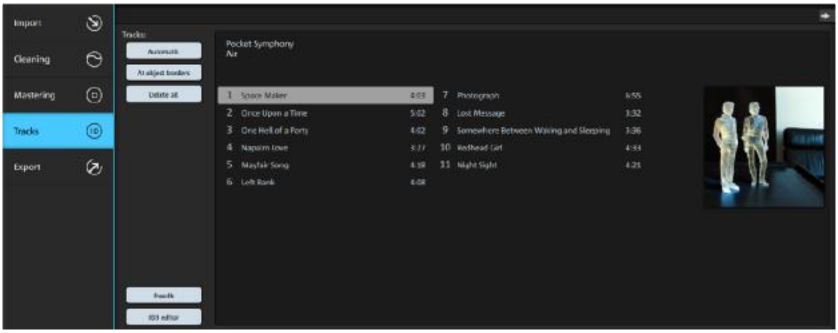


If you play back the material now, the cracks have disappeared.

# Set track marker

Your recording is now ready and should be burned to CD. For burning to CD and also exporting to individual MP3 files, you now need track markers in the project that indicate where a new song starts in a recording.

To do so, switch to the "Tracks" view.




Here you have the opportunity of checking and changing track markers and names.

For records or cassettes, there is an automatic system that already sets the track markers during the recording using the silence between the tracks. This means that all track markers are probably already set after the recording, but you may have to move one or other of them slightly. And, of course, the tracks still need to be correctly named.

Click on freedb. Audio & Music Lab tries to determine the names of the tracks from an online database using the special combination of track lengths. If that doesn't work, whether you like it or not you will have to enter them by hand. Double-click on a name (album, artist, name of the individual tracks) in the middle of the window in order to change it.

Under **ID3 Editor** you can access other editing options, e.g. for changing the order of the tracks. Click on **CD Cover** to automatically load a cover from the Internet.

# Export

After you have "cleaned up" the material, you can export it either song by song as MP3 or WAV files or all songs on the track as an audio CD.

To do this, switch to the "Export" interface.

There are various export options available. "Files" saves the music on the hard drive as an MP3 or Wave file. Use "Audio CD" or "Data disc" to burn your project to CD. The "Share" option can be used to upload files to social networks.

To save your project (view page 54) go to "File" > "Save project". This does not save the actual music, but instead saves all of the edits made in Audio & Music Lab in case you want to change something later or burn another CD.

# Overview of the program interface


*Program interface of MAGIX Audio Cleaning Lab*

1. **Menu bar:** Here you'll find all the features available in MAGIX Audio Cleaning Lab
2. **Overview track:** The entire project is displayed here. The area that you are currently editing is highlighted.
3. **Track:** Here you can perform detailed editing of the audio material (view page 54).
4. **Editing buttons:** Here you'll find buttons for important editing functions like split objects (view page 57) or delete and set track marker. The "Undo" and "Redo" buttons are also found here
5. **Work areas:** Here you switch between the Import (loading and recording of audio material), cleaning, mastering (sound cleaning and improvement), tracks (names and sequence of the songs) and exporting (saving ready-to-use audio files, burning or sharing on the Internet) work areas.
6. **Transport console:** Controls track playback.
7. **Presets:** Presets for the cleaning and mastering effects according to the audio material. You can hide the presets using the up arrow.
8. **Project/Object:** Here you can determine whether an effect will be applied to the whole project or only to a selected object
9. **Effects section:** Here you can access the individual cleaning (view page 62) or mastering effects (view page 74).

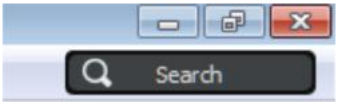
- **Search function:** Click on „Search" and enter a key word, the search lists all menu items, help topics and effect presets that contain the key word.
- **Master volume:** Here you can set the master volume of the track. The limiter prevents clipping and the auto function enables the volume to be automatically optimized.
- **Info area:** Displays the InfoBox with help texts on the effects that are offered, a visualization of the audio signal or the video preview (not in MAGIX Audio Cleaning Lab.

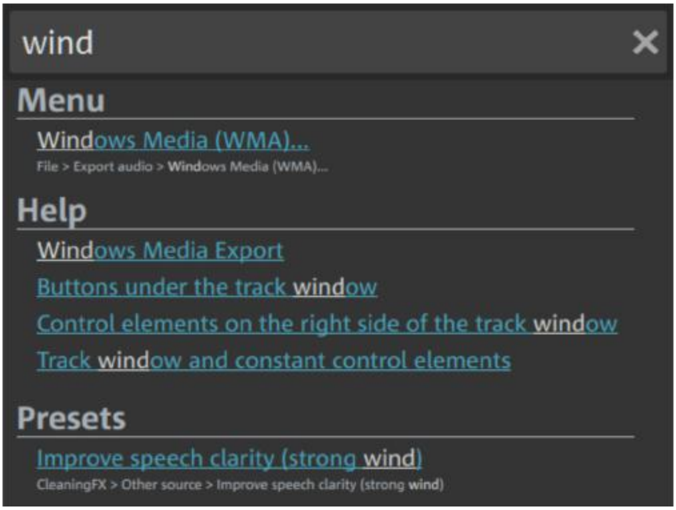
# Track window and constant control elements

This chapter describes the display and control elements which are available to you independently from the selected section in MAGIX Audio Cleaning Lab

## Search

At the top right of the program window is a search function.




Click on "Search" and enter a key word, the search list all menu items, help topics and effect presets that contain the key word.

Keyboard shortcut: Crl + F

# The master track

## Audio material display

All of a project's audio material is displayed in the master track of the track window as a waveform. The waveform corresponds with the acoustic properties of the material. This means that there isn't anything to listen to at places where there isn't anything visually; higher waves mean high volumes. The tracks waveform display forms the most important basis for locating specific passages.

The display is compressed, meaning that the waveform is displayed as a ratio of loud passages to quiet passages. This ensures correct display for quieter sections at the beginning or end of a song.

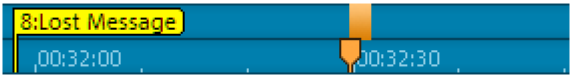
## Position line

During playback a thin line will move horizontally from the left to the right over the master track. This is the position line, which indicate, which part of the wave form is currently reproduced.

The last starting point of the playback is indicated by a small triangle in the master track. The position line will jump back to that position, once playback has finished.

## Timeline/Marker

The timeline is located above the track. It shows the time position in the project. The measurement unit for the time position can be changed in "Options" > "Measurement Unit". You can choose from: samples, milliseconds, hours/minutes/seconds, and CD frames.



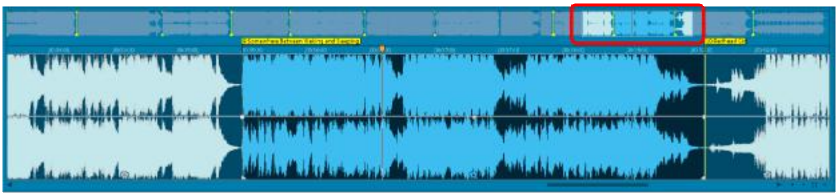

The markers are also displayed in the timeline. There are two different types of marker.

**Track markers** show where new CD tracks begin and where new files start when exporting MP3s. More information about this is available under Set track marker (view page 60).

Simple **markers** can be used to tag specific positions in a project. To set a simple marker, place the position line at the according time position and chose "Set marker" from the Edit menu.

To move markers, simply click on them and move them with the mouse. Clicking on a marker moves the position line to that position in the project. If the position line is on a marker, it can be removed using the Del key.

## Overview track



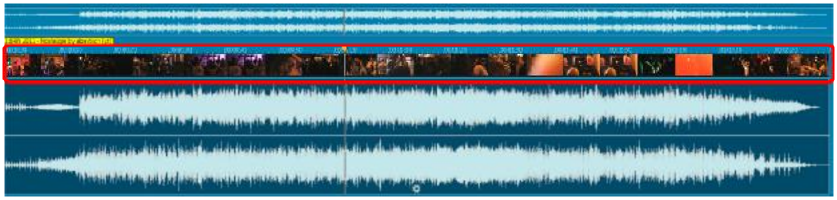
*The overview track lets you select the project section which is displayed in the track display (displayed in blue).*

The track window now includes an additional overview track with a reduced display of the complete project. Now you can work at a specific position or on a certain song while maintaining an overview of the complete project and quickly navigate to areas that need more work. In the overview track you can also select the song or position in the material to be displayed in the track window.

Move the section to the part of the project simply by clicking in the overview track; the zoom level remains the same. Define a new range in the overview track by clicking and dragging. Vertical yellow lines indicate the markers in the overview track.

Use the "Overview mode" entry in the "Options" menu to show/hide the overview track.

## Video preview strips



When a video file is loaded the video footage will be displayed as a filmstrip. This makes it easier and quicker to find specific scenes in the video. The further you zoom into the video the more images will be displayed, all the way until you reach individual frames.

Note: Video sound editing is not available in MAGIX Audio Cleaning Lab.

## Navigation

The task of the transport controls is to help you navigate through the audio material in the master track. Here you can find functions that you will already now from your old tape recorder. You can get more information in the chapter Transport controls (view page 34).

The simplest solution, however, is to navigate directly in the track: a mouse click on the time ruler will set the starting point for the playback (even if playback is already running!) The space bar starts and stops playback.

## Zoom

The waveform display allows you to recognize certain parts of the material from the shape. For many tasks it is quite useful to enlarge the waveform display. For detailed editing, e.g. editing with the scissor tool, there are several zoom options:

**Quick zoom:** For quick zooming it is sufficient to click with the mouse in the timeline, keep the mouse button pressed down and move it up or down. This way, you can quickly zoom the cursor in and out at any position without releasing the mouse.

The +/- zoom button at the bottom right corner of the track window zoom in/out the display.

This enlarges the central area of the track window. Clicking on the adjacent triangle opens the zoom menu. Here you can open different zoom and navigation commands (view page 29).

By selecting an area in the overview track (view page 28), you can also determine the visible section of the project. The overview track is located above the track window and displays the entire project irrespective of the selected zoom level.

You can click on this symbol or the "a" key to quickly restore the fullscreen view of the project.

Moving the scroll bar forward and backward allows you to quickly navigate through the project. Dragging the scroll bar ends adjusts the size, which also zooms in and out of the project.

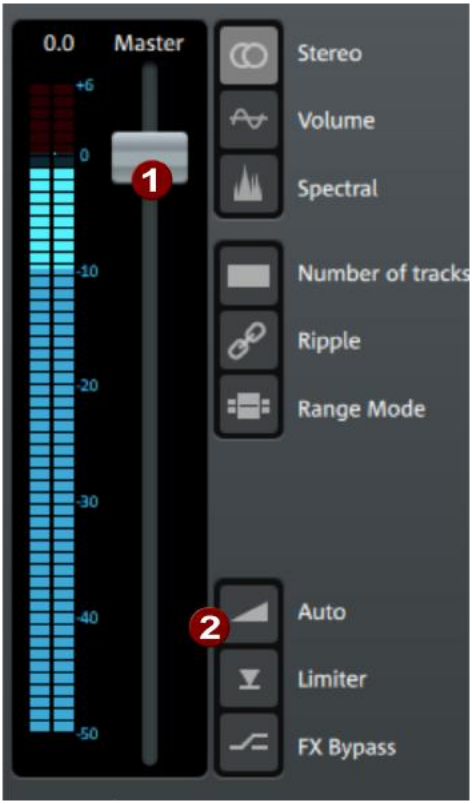
## Zoom settings

| Command | Keyboard shortcut | Description |
|---|---|---|
| **Play/Position marker** | | **Commands for quickly editing the playback position** |
| Marker left | Alt + right | The play marker can be moved quickly between the markers. |
| Marker right | Alt + left | |
| Object border left | Shift + Alt + right | The play marker can be quickly moved from object edge (object start and end) to object edge. |
| Object edge right | Shift + Alt + left | |
| **Zoom ranges** | | |
| Show all | A | The entire project is visible. |
| Zoom 1s | 1 | The visible section of the project is quickly set to the selected value. |
| Zoom 10 s | 0 | |
| Zoom 60 s | 6 | |
| Zoom 4 min | 4 | |
| Zoom 10 min | Shift + 0 | |
| **Vertical zoom** | | Vertically zooms in and out of the wave shape. This is useful for locating the crossover point (for precise sample editing). |
| Zoom into waveform | Ctrl + Cursor down | |
| Zoom out of waveform | Ctrl + Cursor up | |

# Control elements on the right side of the track window

On the right side of the waveform display you'll find various buttons that influence the display and playback of the project.

## Master fader




1. This slider adjusts the output level of the track. For maximum sound quality, the project should be optimally controlled.

2. Use the "Auto" button to find this optimum level automatically. To do this, set the position line just in front of the loudest position in the project and start playback. (The loudest position can be found by locating the highest peak of the waveform display).

   After you have played the loudest part of the range, click on the "Auto" button to the right of the volume control. MAGIX Audio Cleaning Lab automatically adjusts the level so that the loudest part of the range just played back is exactly 0 dB - this will be the maximum volume.

**Note:** The master fader adjusts the volume of the project which means that it will be exported at this volume. The Monitor fader (view page 36) can be used to set the playback volume for editing.

## Peak meter display

The display beside the master fader is a peak meter and shows the peak level of the audio in the track during playback. For stereo tracks the the left bar shows the level of the left channel and the right bar shows the level of the right channel. Both bars react together for mono tracks.

## Stereo


If the stereo button is activated, the waveform display of the audio material for both stereo channels will be displayed separately.

This view is useful to visually monitor processing of the material in the stereo panorama, or to locate precise crossover points during editing operations.

## Volume curve


Use the **"Volume"** button to activate a volume curve (view page 59).

## Spectral


In addition to the waveform display, this button allows a spectral display (view page 152) of the audio material to be shown as well.

At the same time the mouse mode „Edit spectrum directly" (view page 147) is set. By using this mode you can mark noises directly in the spectral display and filter it out. See for more information on mouse modes in the Options menu chapter (view page 146).

Hint: You can activate the Spectral display separately by using the according menu option in the View menu.

Note: The spectral functions are not available in MAGIX Audio Cleaning Lab.

## Number of tracks


Use the "Number of tracks" button to create a second or three additional stereo tracks, e.g. if you want to mix several audio sources (narration, music and sound effects) or to "park" a song on the additional tracks for the meantime.

Additional track provide a better overview when there are multiple objects in play. All objects can be moved randomly between the tracks. If moved between tracks while holding the "Shift" key, the horizontal position will be retained, i.e. only the track is changed. Materials which are placed over one another will be played back simultaneously and burned on CD as well. If this is desired, then the volume level at this position should be monitored, since objects which are playing back simultaneously add to the overall volume.

Note: The second track is not available MAGIX Audio Cleaning Lab.

## Ripple

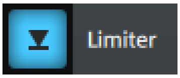

This button controls the behavior of subsequent objects when objects are deleted and moved (view page 57).

## Range Mode


This button switches to the Range Mode (view page 58) for audio editing.

> Note: The Range Mode is not available in MAGIX Audio Cleaning Lab.

## Limiter



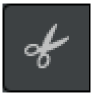

The limiter can be switched on to reduce clipping. This device works sound-neutrally and provides a final guard against extreme levels.

## Bypass


You can use this button to turn all effects on or off to compare the edited material with the original.

# Buttons under the track window

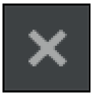
## Split



A selected object is split into two objects at the position line. This also works during playback.

Keyboard shortcut: T

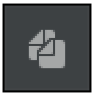
## Delete



The currently selected object will then be deleted from current project. The subsequent objects are moved forward so that there is no gap in the track.

Keyboard shortcut: Del

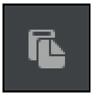
## Copy



The selected object is copied from the project into the clipboard. It can then be re-inserted elsewhere.

Keyboard shortcut: C

## Paste



The content of the clipboard is inserted into the project at the position line.

Keyboard shortcut: Ctrl + V

## Markers

Opens a menu with different functions to set track markers (view page 141) automatically.

## Undo

In the project you can undo the last changes you made. This way, it's no problem if you want to try out critical operations. If you don't like the result, you can always revert to the previous state using "Undo".

Keyboard shortcut: Ctrl + Z

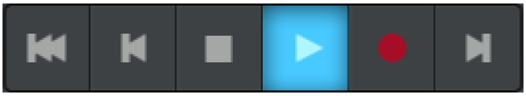
## Redo

The “Redo” function undoes the previous “Undo” function.

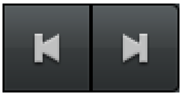
Keyboard shortcut: Shift + Y

## Transport console

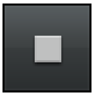
The transport console determines the playback position in the project. The current playback position is indicated by the position line, a thin vertical line in the track window. Use the transport console to start and stop playback, move forward and backward within the project and set the playback position back to the very beginning.



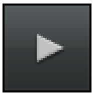
**Return to start:** Resets the position line to the start of the project (also works during playback).



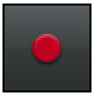
**Next Track/Previous Track:** Sets the playback position at the next or previous track marker.



**Stop:** Stops playback, the position line jumps back to the starting position



**Play:** Starts playback; clicking again stops playback at the current playback position.



**Record:** Begins recording. Right-clicking opens the settings dialog for recording where you can define the recording format, level and other.

**Loop:** Display and control for playback in a loop (see Loop Playback (view page 35)

**Controlling with the computer keyboard and mouse:**

- The space bar starts or stops playback.
- Clicking on the timeline above the track moves the position line to the corresponding position (also works during playback).
- Other special keyboard commands can be used to jump between markers and object edges quickly (see Zoom settings (view page 29)).

## Loop playback

During loop playback, a range of the project is played back in a repeating loop. This function is useful if you want to listen to parts of the audio repeatedly, e.g. to optimize transitions or effect settings in critical sections.

The loop range can be dragged out in the timeline using the mouse. The loop button lights up to indicate that a loop is activated. Now when you start playback using the spacebar, the loop range will be played back.


Tip: The borders of the loop range can be adjusted during playback.

If no range has been defined and you click on the loop button, a default range of two seconds will be set.

## Scrubbing slider

Note: This function is not available in <Program name_classic>.

Scrubbing refers to "searching" through the audio material at different speeds to locate a certain position "acoustically".

In this case, MAGIX Audio Cleaning Lab behaves like the editing function of a tape player. The motor is switched off, but the tape remains at the sound head. Turning the tape reels by hand can slowly move the tape past the sound head to make it easier to find specific positions on the tape.

By varying the playback speed, it's possible to quickly approach a position, but also to arrive at the exact position at a reduced speed.


You can use the Scrubbing slider to control the playback with variable playback speed. The further you drag the slider to the right or to the left, the quicker the project is played forward or rewound.

## Time display

The time display shows the playback position in the project. The measurement unit is the same as in the timeline and can be adjusted in the Options menu.

## Monitor fader


The monitor fader controls the monitoring volume during playback. In contrast to the master fader (view page 31) it does not influence the volume of exported files or songs burned to CDs. Instead, it acts as a practical extension of the volume control on the sound card or on the computer speakers.

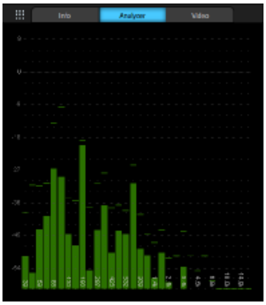
Attention: The monitor fader works in addition to the volume control on the sound card mixer. This means that if the volume is set to 0 on the sound card, the monitor fader will have no effect.

# Info area

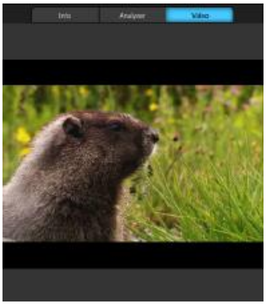
The info area at bottom right can be switched between three different views:



**InfoBox.** The InfoBox provides information on how to use the selected effect.


**Analyzer.** The analyzer displays the audio material graphically.


**Video monitor** (not in MAGIX Audio Cleaning Lab). The video monitor shows previews of videos (view page 124).

This can be completely hidden using the arrow to the right of the monitor.

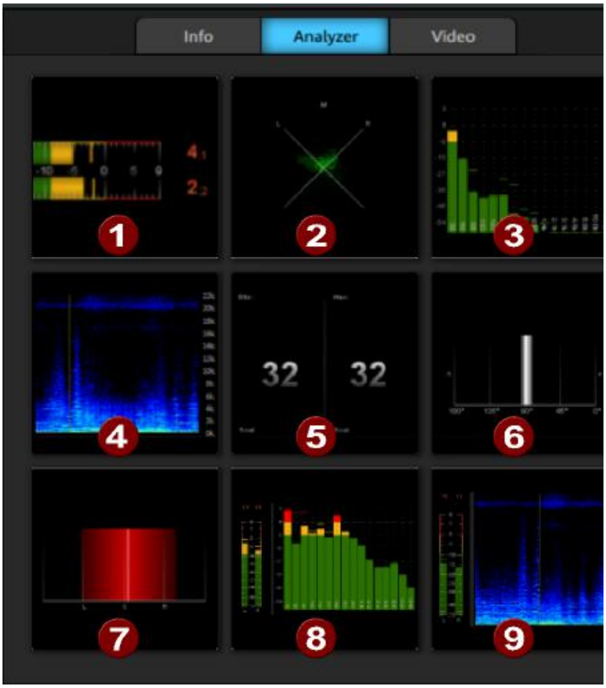
## Info Box

The Info Box offers fast and immediate help when working with MAGIX Audio Cleaning Lab. You can open it with the arrow button on the right side of the Cleaning and Mastering section. Click on an effect to get information about its range of application, functionality, correct handling and possible sources of error.

Clicking on an object displays the applied Object effects (view page 72), if available. The Info Box also offers useful tips for other program areas.

## Analyzer

The analyzer provides a graphical display of the played back audio material. Here you can choose from the following display options:




1. Peak meter
2. Vectorscope
3. Spectroscope
4. Spectrogram
5. Bit meter
6. Correlation gauge
7. Direction meter
8. Peak meter + spectroscope combined
9. Peak meter + spectrogram combined

Note: Views 5-9 are not available in MAGIX Audio Cleaning Lab.

Click on one of the graphics to select the appropriate display mode.

 Use this button to select the display mode.

You can also change the display mode directly by right-clicking in the monitor window; Here there are also different presets under "Presets".

**Peak meter:** Display of the peak level and the RMS value (average loudness, for more about this, see also Loudness normalization).

**Vectorscope (phase oscilloscope):** The vectorscope gives you information on the distribution of the stereo image in your recording. The signal of the left channel is depicted above the left axis, that of the right channel above the right axis. A mono signal is depicted as a vertical dash in this display; the more the signals of the two channels in a stereo signal differ, the wider the displayed "cloud". Please note that a broadening of the display implies more deletions, and in turn that the signal is not as mono compatible.

**Spectroscope:** In the spectroscope the signal is divided into individual frequency ranges (frequency bands). The level amplitude of each frequency band displays volume of the relevant frequency range. This is a way of telling whether certain frequency bands are being strained.

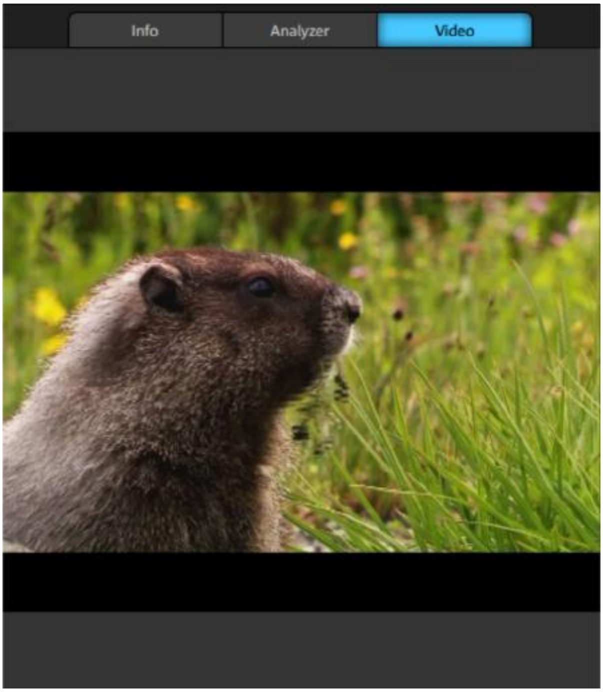
**Spectrogram:** In the spectrogram, the signal is specified as frequencies against time. The volume of frequencies is visualized by its brightness.

**Bit meter:** The bit meter shows you the bit rate at which the played signal is calculated and the maximum editing rate which is possible.

**Correlation meter:** The correlation meter can be used to read the phase shifts between the two stereo channels and thus the degree of erasure. If the signal display is in the left, red area between 90° and 180°, the mono signal is no longer played back perfectly.

**Direction meter:** The direction meter indicates the location direction of the signal. The width corresponds to the degree of correlation.

## Video Monitor



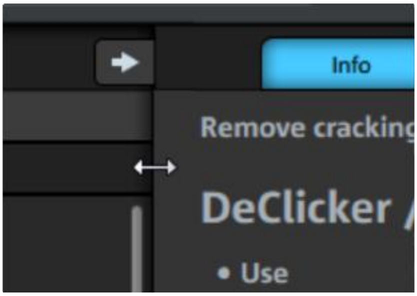

The video monitor displays a preview of the loaded video. The playback position in the project corresponds to the position in the video. The video monitor, therefore, serves as a kind of orientation help so you can see where in the video you are currently working.

**Detach video window:** Double click on the "Video" tab to make the video window floating. This means, you can then move it around and resize it.

Note: The function „Load video sound" is not available in MAGIX Audio Cleaning Lab.

# Changing the size of sections in the program interface

The program interface of MAGIX Audio Cleaning Lab automatically tries to optimize the use of the available space on screen, depending on the screen resolution.




By dragging the horizontal border between the lower and the upper area (audio track and effects) or dragging the horizontal border between the lower area and the info area you can adapt this screen spacing to your needs. The space distribution will be saved when you close the program.

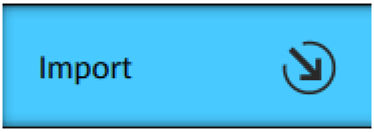
With the keyboard shortcuts **F5** (upper area), **F6** (lower area) and **F7** (info area) you can maximize the according screen area.

Upper area maximized

Lower area maximized

Info area maximized

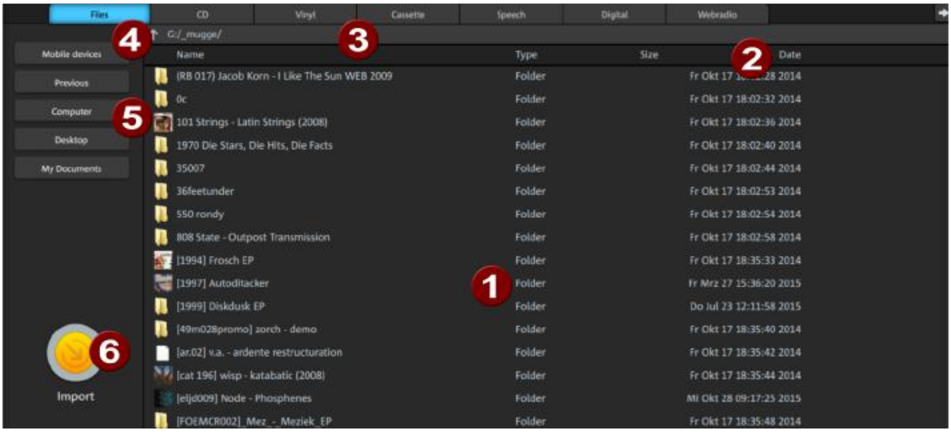
# Import




In this section audio material is imported into MAGIX Audio Cleaning Lab. You can load audio files ("Files"), import tracks from audio CD ("CD") or record audio material ("Record", "Cassette", "Language", "Digital", "Web radio").

## Files

In the "Files" view you can import audio files from the hard drive or connected data storage devices into MAGIX Audio Cleaning Lab. This feature works in much the same way as Windows Explorer.

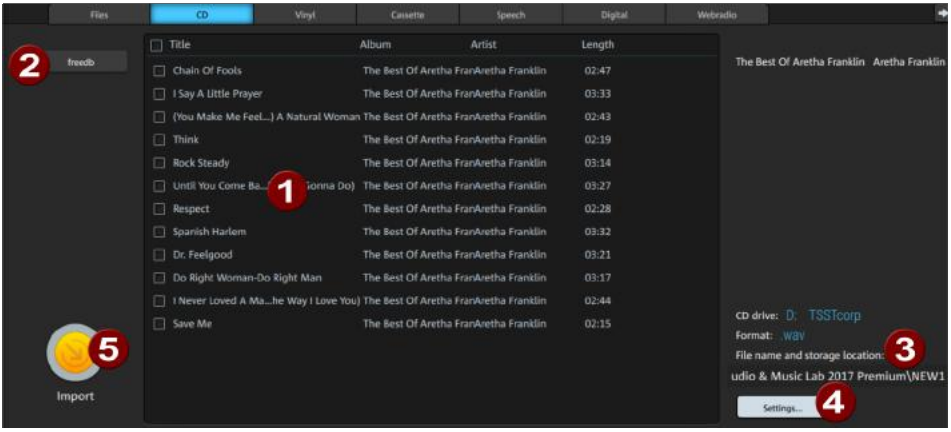



1. **File list:** Here all of the files in a folder that are supported by MAGIX Audio Cleaning Lab are displayed in a list. They can be selected by clicking on them. Just like with Windows Explorer you can extend your selection with "Ctrl" + click and select a series of files with "Shift + click".
2. One click on the file properties (name, type, size, date) in the table header sorts the files according the corresponding property.
3. Go one layer up
4. The file path of the displayed folder
5. Shortcuts for predefined folders
6. **Import**: This button can be used to import the selected files.

The selected file is added to the project at the position 2 seconds after the last object. This value can be changed in the CD/DVD menu with the option Set automatic pauses (view page 142). If you have tracks which blend over one another and are distributed on several files, you should change this value to "0".

# CD

With **CD** you can import music from a CD into the program. You can import entire Audio CDs or individual CD tracks into the project. Unlike data CDs, audio CDs require special treatment while importing ("grabbing" or "ripping"). The data is imported digitally which eliminates any loss in sound quality.




1. **Track list**: Tracks can be selected for import by checking the box in front of them. The box at the very top selects all the tracks.
2. **freedb**: Request for track information from the freedb Online CD Database (view page 143)
3. Here the parameters for the import are displayed, e.g. the CD drive name, the file format, the storage location and the file name. In the default setting the CD content is imported as a wave file into the project folder (see Path settings (view page 151))
4. **Settings:** To select a different CD drive, file format or storage location you can open the advanced settings dialog (view page 43) here. Changes made here will be applied to all subsequent imports.
5. **Import:** The audio material is copied from the drive onto the hard disk. A progress bar is displayed.

When this process is finished, the tracks are added to the project as individual objects with corresponding track markers.

## CD Import Settings

In this dialog you can adjust the settings for CD import, e.g. to change the save location or the file format.

In the upper part of the dialog you'll find a file explorer which you can use to set a new name or target folder for the audio file. Under „**Last storage location**" you can choose from a list of save locations that are already being used.

**Reading speed:** The default setting is „Maximum", i.e. as fast as the drive allows. However, if reading errors occur, you can reduce the speed here.

**CD drive options:** Here you can change the settings and select the drive for importing the CD if you have installed several CD drives.

**File options:** Here you can determine whether all of the CD tracks will be imported as a single file or each track as a separate file. In the case of separate files you can select from various file name schemes in the list field.

**Import:** This closes the settings dialog and starts the import process. The changes to the settings will be applied to any subsequent imports.

# LP/Cassette/Voice recording

In these views music or speech from an analog source can be recorded.

> Note: All three views are identical to use, only the settings of the automatic track marker detection on the medium that is used are different; for cassettes, there is the additional High Speed Dubbing (view page 50) option.

To record from analog sources, proceed as follows:

1. Connect the recording source now if you haven't already done so.
2. Switch to the corresponding import view (in this example "LP") and play back the audio from the source to set the level. It is best to use a loud section for this.
3. Adjusting the recording level is essential when recording digitally via sound cards in order to achieve optimum sound quality. The peak control can be checked using the "Recording level" LED chain (**1**). They should not move beyond the "safe" area. If the level moves into the "too loud" or "too quiet" areas, correct it with the yellow level fader (**2**). But be careful. If you lower the input level, you also lower the precision of the resolution with which the analog signal is digitized. For this reason, the fader should also be set as loud as possible.
4. Now check the threshold values for automatic CD track recognition. To do this, play back one song until the end and into the pause. The level must fall under the green zone (**3**) so that the next song will be recognized. Correct the threshold value if necessary.
5. The recording function in the import section is reduced to only the most important features. For recordings in other file formats and various other options, open the Recording settings (view page 46) dialog under "Settings..." (**4**).
6. You can start the actual recording with the large red record button (**5**). The duration of the recording and the remaining storage capacity is displayed on the right (**6**). You can monitor the recording level on the LED display during recording. If the display reaches the upper LEDs, clipping occurred at some point. In this case, you should definitely check the recording for clipping and, if necessary, repeat the recording using a lower recording level.
7. If you do not hear your source although it is correctly connected and also the peak meters deflect, you can activate the "Monitor" (**7**) button. Internal monitoring is realized in this way. (For more information on the subject of Monitoring while recording (view page 49), see below).
8. At the end of the recording you will be asked if you want to use the recording. The newly-recorded material will be placed at the current position of the playback marker in the project.

> Note: If there is no level displayed, chances are the wrong input has been selected. Open the record settings dialog using the "Settings..." button and click on "Audio line-in" to correct the error with Automatic input and level (view page 49).

## Basic knowledge about recording with the PC

The record function converts analog audio signals – records, tapes, sounds, speech – into digital data, which can be saved on the PC and edited with MAGIX Audio Cleaning Lab.

The device which is used to digitalize the audio signals is already built into most sound cards and aptly called an analog-digital converter, often abbreviated with A-to-D, ATD or A/D. In order to record sounds, the A/D converter takes samples of the sound to be digitalized at fixed intervals by measuring the voltage level of the signal. The frequency of the sampling is called the sample rate and naturally lies within the kHz frequency range; several thousand times per second. The higher the sample rate, the more samples are recorded by the A/D converter, thus making the sound conversion closer to the original.

The precision with which the A/D converter measures the voltage level of the analog signal is determined by the sample resolution. The same principle applies here: The finer the resolution, the better and more natural the digital conversion.

Audio recordings in CD quality are recorded with a sample rate of 44.1 kHz and a resolution of 16 bits.

## About the term "sound card"

When the first version of MAGIX Audio Cleaning Lab, the MAGIX cleaning lab was released in 2001, having audio functionality integrated on the computer mainboard was rather new and not the default option, and the sound quality of these on-board-sound devices was really bad. For a decent recording an additional expansion card, from specialized vendors, was obligatory: the so-called "sound card".

Today every PC is equipped with audio inputs and outputs by default, nevertheless we continue to talk about sound cards for simplicity. And even though the sound quality of these built-in devices was improved over the years, to obtain a really good recording quality we would still recommend the use of a dedicated "sound card" (which is nowadays mostly an external USB device) from a music shop.

## Connecting the recording sources

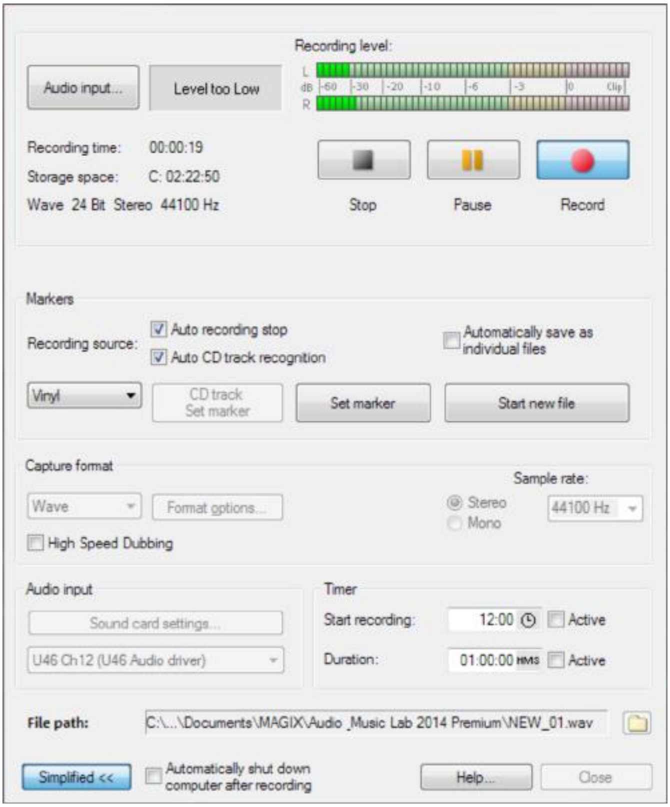
First, the recording source must be connected to the computer. There are several possibilities depending on the equipment you are using:

- If you would like to import music from a stereo system, use your sound card's line-in input. If the stereo system amplifier comes with separate line-out or AUX out connectors, they should be used. In this case, they have to be connected to the computer's line-in. Hi-Fi amplifiers usually have RCA (composite) connectors and the computer has mini stereo connectors. You need to then get a suitable cable with two small RCA (composite) connectors and a mini stereo jack.
- If the amplifier comes without separate outputs (other than the speaker connectors), you can also use the headphone jack for recording. Normally, you will require a cable with stereo or mini stereo jacks. This type of connection has the advantage of a separate volume control to set the headphone input signal level. However, headphone jacks are often not very high-quality. For this reason, you should use line-out outputs whenever possible.
- To record cassettes from a tape deck, connect the tape deck’s line-outs directly to the computer's line-in.
- When recording LPs, the record player's outputs should not be connected directly to the computer because the phono signal has to be pre-amplified. In

this case, connecting through the headphone jack or an external preamp is recommended. Some record players also have normal line-out connections.

- If you would like to make recordings with a microphone, connect it to the computer using the microphone input (usually a red jack).

## Recording settings




**Audio input:** Opens the input and level automation (view page 49) for automatic selection of the correct input signal and level.

**Recording level:** Displays the recording level; corresponds to the peak meter in the main window (see above (view page 43)) .

**Record:** This button starts the actual recording. During recording, the recording time and remaining space on your hard disk are indicated.

**Record pause:** Pauses the recording. Click the "Pause" button again to resume.

**Stop**: This button ends recording.

## Markers

**Automatic recording stop:** If this button is activated, recording will cease automatically after approx. 16 seconds of silence. This lets you make long recordings without having to worry about stopping it once the recording is finished, e.g. recording an entire LP or cassette.

**Automatic CD track recognition:** If this button is activated, then track markers are automatically placed at the end of the pauses after a track. In order for pause recognition to function seamlessly, you will have to set the proper source in the selection box (LP, cassette, CD, or Internet). You can specify the detection parameters even further using theAutomatic track marker recognition options (view page 150)in the Options menu.

**Save automatically in individual files:** If this feature is active, then every individual track that is recognized will be saved as a unique file.

**Set CD track marker:** Even during recording, you can set track markers by clicking the corresponding button in the recording dialog.

**Start new file:** If you want to record very long sessions or multiple sources one after the other, the recording file can become extremely large. This button will create a new sequentially numbered file.

## Recording format:

In the front list field (preset is "Wave") you can change the file format giving you the option of recording directly in compressed formats such as MP3 or OGG VORBIS. You can set details such as the bit rate and compression method for each format in the "Format options".

**Resolution/Sample rate:** Here you can set the sample rate and bit resolution for the recorded audio file as well as define whether it will be recorded in mono or stereo. 24-bit recording (view page 50) requires a high-quality audio card with 20 or 24-bit conversion, plus a 24-bit capable MME driver. Audio cards with SPDIF digital interfaces can also record audio material in 24-bit quality.

> Note: MAGIX Audio Cleaning Lab always records in 16-bit resolution with a maximum sample rate of 44.1 kHz.

**High-Speed Dubbing:** Activates the Recording at double speed (view page 50) function.

## Audio input

**Audio input:** The button "Sound card settings" opens a dialog with special settings (view page 48) for whatever sound card is present. The name of the selected sound card is also displayed. If you are using several sound cards (or such with several inputs), then you can select one from the menu.

**Timer:** Enter a starting time for a recording and the length of the recording. The recording won't begin immediately after pressing the "Record" button, but rather at the specified time. This way, time-delayed recordings (for example, at night or when you're out) are now possible. Of course, the system clock has to be set correctly. If "Record length" is also activated, then the recording will end automatically after the indicated period.

**File name/file path**: The name of the audio file to be created and the preset folder are displayed in the recording window. Both can be changed by clicking the folder button.

**Shut down computer automatically after recording:** If you are working with timer recordings, you might as well have the computer shut down automatically after the recording is complete.

**Advanced/Simple...:** The settings dialog can be "folded up" so that only the peak meter and control elements are visible.

**Help**: Opens the program's help file for the recording dialog.

**Close:** Closes the recording dialog.

Keyboard shortcut: R

## Record properties

This dialog provides you with information regarding the currently selected sound card. Supported audio formats of the sound card and the sound card driver's information is also displayed.

**Driver system:** Here you can switch between driver types (MME and WDM).

> Note: Adjust this setting only if you have problems with audio playback or recording.

**Special:** Some sound cards or audio devices (for example, USB turntables) do not offer mixer support. With the "Monitor input signal" option you can listen to the sound during recording (monitoring).

"Filter DC offset" allows you to remove the DC offset section (view page 73) of the input signal, even during recording.

# Monitor while recording

Monitoring while recording (referred to as 'Monitoring') can take place in two different ways, either directly inside the sound card or provided by MAGIX Audio Cleaning Lab ("Monitor" button).

Most sound cards, including those integrated in the computer ("Onboard sound"), provide support for the Windows sound card mixer. Within the mixer there is also a signal path that passes an input signal directly to the sound card output.

To activate monitoring using the sound card mixer:

1. Open the "Control Panel" > "Sound" and switch to the "Record" tab, or right-click on the loudspeaker symbol in the right lower corner of the screen and select "Recording device" from the context menu
2. Double-click the required recording device. If your source is connected and playing, the device can be recognized from the appropriate peak on the peak meter.
3. Switch to the "Listen" tab and activate the "Use this device as playback source" checkbox.
4. If you now click "Accept", you should hear their input signal.

Some sound cards or audio equipment, such as USB record players, lack this possibility for monitoring.


For this reason, the "Monitor" button appears in the Import section of all areas.

If it is activated, MAGIX Audio Cleaning Lab passes the input signal to the output.

**Attention:** If the monitoring sounds hollow and strange or you hear echos, probably both ways of monitoring are active and are interfering with each other. In this case, either deactivate the "Monitor" button or "Listen" in the sound card driver.

# Input and level automation

Every sound card has a least two inputs (microphone and line), as well as various "internal" inputs for the CD drive or the signal from another program, for example, Internet radio. With input and level automation you can automatically select the correct input for your recording without having to search and adjust the input level in order to avoid distortions.

To do so, click on "Audio input" in the record dialog. If you had already connected your source and begun playback, the correct input will be determined immediately. Otherwise do this now and click on "Search channel again".

If this didn't work, use the "Windows mixer" button to open the Windows Mixer and select the channel manually.

Adjusting the recording level is essential when recording digitally via sound cards in order to achieve optimum sound quality. If the adjustment is set too high, distortion occurs and the incoming signal must be reduced. If you reduce input sensitivity, the resolution at which the analog signal is digitized is also reduced. The level controllers of your sound card should generally be set as high as possible in order to achieve optimum results. Yardstick for an optimal level is the loudest part of the material. The loudest part should be adjusted to the maximum. You can now adjust the recording level with the help of the LED display in the record dialog.

You can adjust the level of the source manually using the "Volume" controller. If you activated "Automatic level adjustment", the level controller will automatically be set to the correct value.

## High Speed Dubbing

Some double cassette decks have a "High Speed" copy function. On one deck a cassette will be played at double speed and on the other recorded at double speed. Doubling the speed cancels itself out so you will end up with a completely normal cassette recording. Thereby making it possible to copy cassettes in half the time.

With the "High Speed Dubbing" option you can also use this function with MAGIX Audio Cleaning Lab. If it's activated and you record played audio material at double speed, after recording the speed of the recorded material will be automatically halved in order to end up with the original speed.

Note: Since, with a cassette deck, it cannot be assumed that exactly double the speed is used for high speed dubbing, it's best to compare the tempo of the recording with the original and to correct it with Resampling/Timestretch mouse mode (view page 147) if necessary.

## 24-bit Audio Support

Note: 24-bit recording is not available in MAGIX Audio Cleaning Lab.

Not only can audio data be recorded in MAGIX Audio & Music Lab Premium in 16-bit CD quality, but also in higher-quality 24-bit resolution. You can select this 24-bit resolution audio format in the recording dialog. 24-bit recording requires a high-quality audio card with 20 or 24-bit conversion, plus a 24-bit capable MME driver. Audio cards with SPDIF digital interfaces can also record audio material in 24-bit quality.

The high-resolution audio data are saved and edited by MAGIX Audio & Music Lab Premium in 32-bit floating point data format. This ensures full 24-bit quality regardless of the level. The dynamics increase to more than 140 dB, the signal-to-noise ratio of the recordings drops to 110 dB and more depending on the sound card. Thanks to Floating Point Processing there is no need to worry about clipping during internal processing. A Floating Point Signal only begins to clip at approximately 1500 dB above 0 whereas a 16-bit signal clips as soon as 0 dB is exceeded.

Even if the audio material is to be burned on a 16-bit audio CD in effect, it pays to use a 24-bit audio resolution for the recording because all effect calculations also then run with a higher quality and do not generate any rounding errors in the audible 16-bit range.

Compared with 16-bit recordings, 24-bit recordings take up double the memory on the hard drive (by saving them as 32-bit float data), but considering the capacity of modern hard drives this is a good compromise for the increase in quality.

High-resolution audio data can be imported and exported as 24-bit WAV files so that the data exchange with other high-quality audio systems – e.g. MAGIX Samplitude – is not a problem.

# Digital

The "Digital" view is used for recording digital sources. This can be the digital input of the sound card, for example, but also the output of other programs on the computer. Operation is analog to the other views. The option for automatically setting the track markers is adjusted accordingly.

## Record output


The "Record output" option allows you to use the current output of the computer as a digital recording source.

> Note: Many sound cards also offer the entire sound output for selection as a recording device, often under the title of "Stereo mix" or "What You Hear". This also works, but the result of such a recording is not really a digital recording. Technically, the analog output signal is simply fed back here to the analog input, i.e. on the sound card, that is to say a double conversion takes place - from digital to analog and back.

"Record output" is used to carry out a genuine digital recording, i.e. exactly those digital data are saved that the playback software (e.g. the player in the browser) provides to the sound card driver.

This function uses the WASAPI driver model in "Shared" mode for the sound card, which means one set of audio equipment can be used by several programs at the same time. Here, however, it is a condition that all programs must work with the same sample rate (view page 44). Some sound cards (especially many onboard sound cards) are preset with a sample rate of 48 kHz for system playback. In this case a warning message appears and Audio & Music Lab automatically adapts the sample rate of the recorded material. In this case, for optimum sound quality, however, we recommend changing the sample rate used in Audio & Music Lab to 48 kHz prior to recording in the Options menu > Playback parameters (view page 149) dialog.

## Typical use

**Recording a copy-protected audio CD**: According to the copyright act, it is forbidden to copy a CD with or without copy protection. However, an owner of a CD may create a backup copy for himself - even if it's a copy-protected CD. The problem with copy-protected CDs is that they cannot be imported using conventional PC drives. To create a backup of such a copy-protected CD, you can play it in the CD drive as an audio CD and use the "Record output" option.

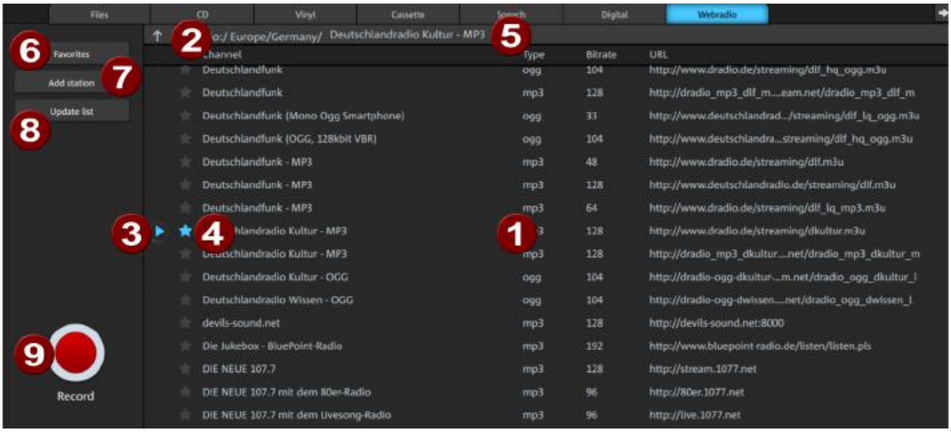
**Record Internet Stream:** Many Internet contents are only available as "streaming media" and can, in fact, be played but not downloaded for listening to them later offline. With the "Recording output" option you can record such streams, thus safeguarding them for posterity.

**Note:** It is possible that this function will be prevented by the appropriate player software for copyright reasons. In this case, you can only use the above-mentioned analog recording of the output signal via the "stereo mix" recording source.

# Web radio

Note: This function is not available in <Program name_classic>.

In this view you can play back web radio while also recording it at the same time.




1. **List of stations:** All of the available stations are listed here. At the top level there is a series of folders where the stations are classified according to musical style as well as geographically. Double-click a folder in order to change to it.
2. **Current folder:** The current folder is displayed at top left. You can move up a level using the arrow.
3. **Playback:** A click on the Play symbol starts (after a short loading time) the playback of the station.
4. **Add to favorites:** Select your favorite station with an asterisk.
5. **Current track:** The track just played is displayed here. (Not all web radio stations make this information available.)
6. **Favorites:** Use this button to switch to the Favorites folder. All stations with an asterisk are listed here.
7. **Add station:** If you know the URL of a web radio, you can use this button to add other stations. Even entered stations automatically end up in the Favorites folder. To remove them again, simply click on the asterisk.
8. **Refresh list:** A click on this button loads an updated list of stations from the MAGIX server.
9. **Record:** The current web radio playback is recorded.

# Cutting and arranging objects

This chapter is about working with objects and track markers. Descriptions of the individual control elements can be found further on in the „Track window“ (view page 26) chapter.

## What are objects?

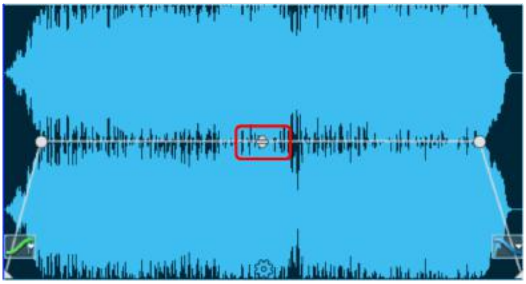
Objects provide a waveform display of audio material. The starting point of an object is assigned to an audio file. The length of the object determines the length of the part of the audio file. This means that an object does not actually contain the audio material and is only a set of instructions for audio playback. When objects are edited (transitions, volume adjustments, cuts (view page 57)) more instructions are defined that are applied in realtime during playback. The actual audio file is not changed ("non-destructive editing") but the adjustments to the settings are saved permanently.

As objects contain simple playback instructions and only reference audio material, they can be easily moved to any position within the track window, or deleted without changing the audio file.

## Project

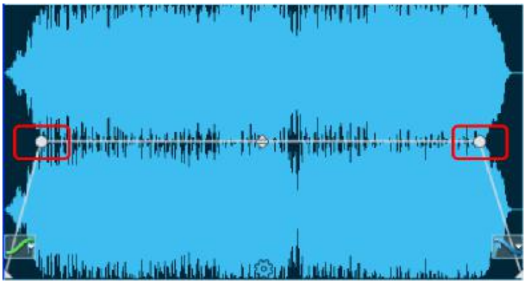
The project (*.vip file) contains all the data from MAGIX Audio Cleaning Lab. It does not contain audio data but only the names of the imported and recorded audio data and the saved locations on the hard drive, the data for the contained objects and most of the effects processing. To keep things simple the object display on the tracks is subsequently referred to as a project.

## Adjust object volume



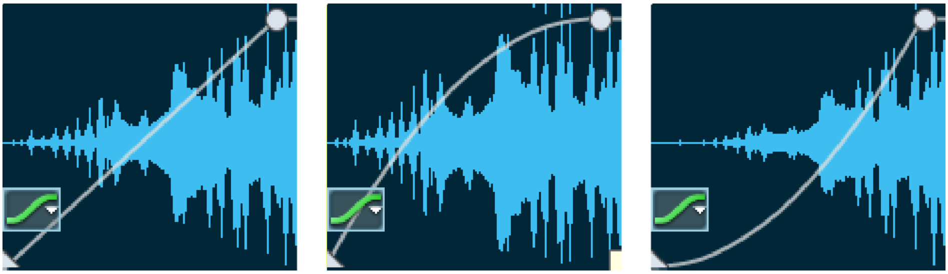
The handle at the top center can be used to adjust the volume of the objects. This handle is particularly important for synchronizing the volume of songs originating from different sources. The volume of audio CDs may also differ.

# Fading objects in and out



At the top corners of every object there are two fade handles that can be adjusted to fade an object in or out. It is particularly useful to use the fade handles to avoid hard transitions or crackling when you have cut passages out of a recording.

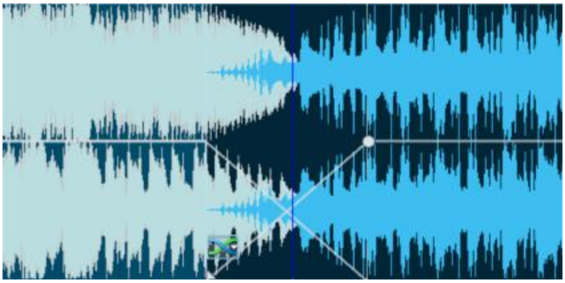
 Click on this symbol to change the fade curve.



*Linear Fade* *Exponential Fade* *Logarithmic Fade*

# Fading objects

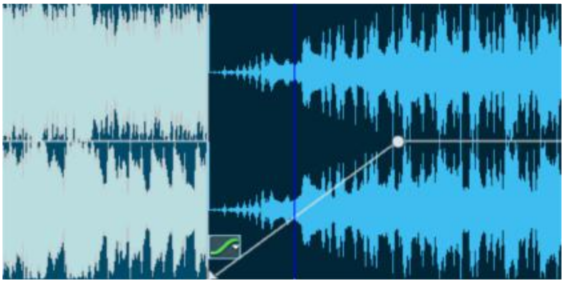
With every cut the two objects that are created are slightly crossfaded in order to avoid crackling. This is called Auto crossfade. A crossfade is also added if two objects in a track are moved into each other or overlap each other.



The top handle of the right object allows you to regulate the fading in and out of both objects.

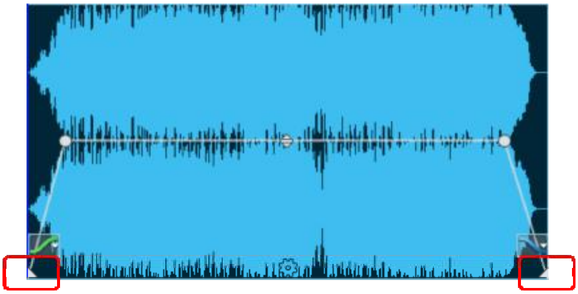
The bottom handle controls the length of the two objects. If you move them, one of them is extended whereas the other is shortened. The length of both objects together remains the same.

A crossfade symbol is added to each crossfade. By clicking on the symbol, you can change the curve of the transition like with the simple object's fade in and out.



"No crossfade" deactivates the crossfade. The front object is cut by the back object: if the playback reaches the back object, the front object is muted immediately (not fades out). However, the fade in of the back object is still active.

# Reducing and increasing the length of objects



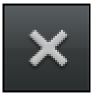
Objects can be shortened by dragging the lower right corner to the left. During this process the mouse pointer turns into a double arrow. This can be used to shorten songs or recordings without having to cut them. To make the object longer again, simply drag the handle to the right. (When you can't move it any further, the full original length has been reached).

An object can be shortened from the start in the same way, e.g. if too much silence has been recording before the actual audio begins. To do this, simply use the handle in the lower left corner. If too much audio material was removed, you can restore it by moving the object border back in the other direction.

# Splitting objects

To remove unneeded audio material (e.g. from recordings) you can split an object into several smaller objects and delete the ones you don't want.

This can be done with the function „Split objects". Set the position line by clicking on the timeline at the position in the waveform where the object should be split. Now click on this button or press the **T** key to split the object.



The unwanted objects can then be deleted using the delete button. The subsequent objects will then move to fill the gaps. (This behavior can also be deactivated; for more about this see below).

If you split an object, a short transition (view page 55) ("Crossfade") is created automatically at the position of the cut to avoid crackling being caused by the cut.

To find the best positions for object cuts we recommend working with a zoomed waveform display. There are various options for adjusting the display size in the Zoom (view page 29) section.

Tip: Use the commands "Remove object beginning" or "Remove object end" (keys **D** and **U**) in the "Edit" menu to remove unnecessary beginnings and ends of recordings.

## Object effects

Splitting objects also has another application: All effects are also available as object effects (view page 72) and can be additionally added. This means you can remove noise from the entire project or a humming sound that only occurs at one point in a specific object.

# Deleting and moving objects

To delete an object select the object and click on the delete button (or press the Del key). No gap will be created in the project, i. e. the subsequent objects are moved. If you move an object, all subsequent objects will be moved along in unison automatically so that no gaps appear.


The "Ripple" button at the top right can be used to change this behavior.

The button has 3 modes that can be set by clicking them one by one:

**Ripple objects** When objects are deleted or moved, subsequent objects are moved in unison.

**Move objects freely** When objects are deleted or moved, subsequent objects are not moved in unison.

**Lock objects** The objects (view page 54) are generally locked and cannot be moved.

This option is important for video editing to prevent the audio track of the video from being moved out of sync with the images. (The audio comes too early or too late). This mode is automatically activated when videos are loaded.

This option is not available in MAGIX Audio Cleaning Lab.

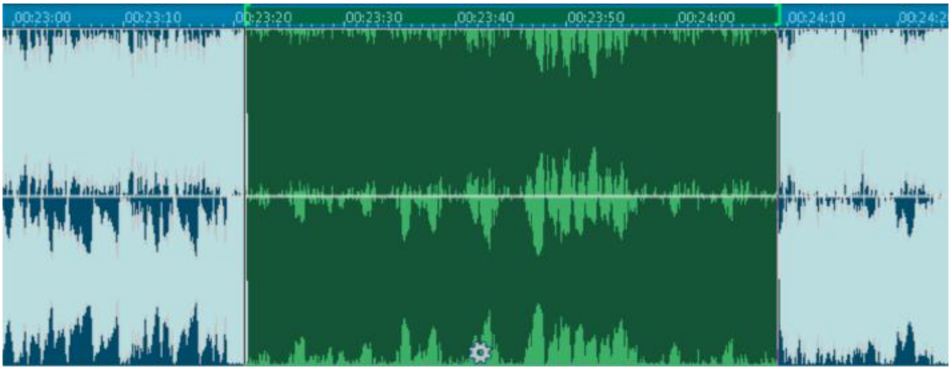
# Range Mode

The Range Mode is a different method to cut and edit audio material on the track.


To activate the range mode, click this button to the right of the track view.

The mouse pointer changes to the range selection symbol when you move the mouse over the audio material.

When you drag the mouse over an object you won't move the object but select a range over the audio..




*You can change the range selection by dragging the green brackets on the timeline over of the wave form.*

The edit menu commands "Split", "Cut", "Copy", "Paste" and "Delete" and the buttons below the track view no longer apply to a selected object but to a selected range of audio.

This makes working on audio faster if there are many cuts necessary. The work flow as shown in Remove unwanted sections (view page 20) in the tutorial chapter, for instance, is now shortened to just selecting the unwanted noise and clicking the delete button.

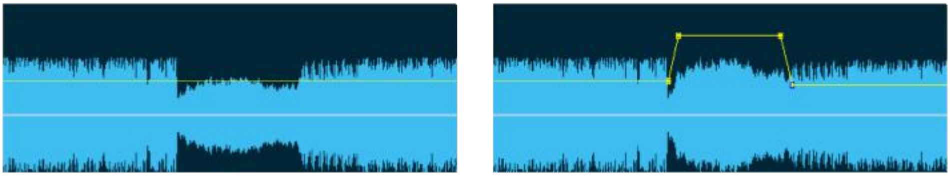
Note: The Range Mode is not available in MAGIX Audio Cleaning Lab.

# Draw volume curves


Use the **Volume** button to activate a volume curve.

You can use it to add volume curves to your audio material, for instance, for compensating volume fluctuations or increasing the volume of quiet passages.

The waveform display adjusts to the curve, giving you visual aid to help work out how much you need to raise specific sections.



*On the left side you can see that a section in the audio material is quieter than the rest, on the right this is corrected using a volume curve.*

There are two ways to edit these volume curves: Create a "handle" by clicking on the curve. This can be moved using the mouse to create volume progressions. This method should be used if gradual volume changes over longer passages are needed.

Alternatively you can use the Volume Draw mouse mode (view page 146). This allows you to "draw" any volume curve using the mouse. This way, you can create irregular volume progressions quickly. This mouse mode can be found in the "Options" menu.

**Note:** Sometimes increasing the volume isn't enough even though the curve is at the top end of the object. You then need to zoom out of the track vertically (e.g. Shift + Mouse wheel), which will create more space above allowing you to drag the handle to the necessary height.

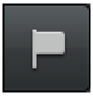
**Tip:** The voice over effect (view page 130) creates volume curves for automatic fading in/out based on the audio material in the second track.

# Set track marker

Recording the entire side of an LP most often involves several songs one after the other. If the record is going to be burned to CD, track markers must be used to define the beginning of each song. Otherwise the whole side of the LP will appear on the CD as a single track. Even when the project is being exported as an MP3, MAGIX Audio Cleaning Lab still has to know where each MP3 starts.

MAGIX Audio Cleaning Lab is preset to divide audio material into CD tracks automatically directly after recording (view page 46). Track markers can also be set "by hand" in the recording dialog while the recording is in progress.

Another option is to set the track markers directly in the track view:



To do this simply set the position line at the beginning of a new song and click on the button. (or you can press the **m** key on your keyboard or select "Set track marker" in the CD menu).

The new track marker will appear over the position line in the timeline.

## Automatic track recognition - How it works

The process goes like this: MAGIX Audio Cleaning Lab searches for points where a new track starts, i.e. the end of a pause. Pauses are normally 0.5 - 3 seconds long.

The second step checks to make sure the interval between the pauses is long enough. For example, it is very improbable that a recording of the Top Ten hits will contain pauses a minute long. If this sort of thing is detected, the marker for the second pause is removed.

The third step examines the start and end of the audio material more precisely. Records always feature a loud bump when the needle is placed on the record and another one when it is removed from the record at the end of the recording. MAGIX Audio Cleaning Lab attempts to recognize these noises and to exclude them, i.e. object edges are automatically moved inwards to match the start and end of the actual music.

There are sensible presets for thresholds and times in the track marker automation which depend on the selected recording source (records, cassettes, CDs/DVDs, Internet). If these don't work properly, you can change them in the Track marker recognition options (view page 150) dialog ("Options" menu).

Sometimes it is helpful to place the first one or two markers manually and to separate the objects using the "T" key, especially if the volume levels are very different. MAGIX Audio Cleaning Lab will examine the objects individually.

MAGIX Audio Cleaning Lab cannot recognize the correct track markers in every case (e. g. during live recordings or with classical music). If you've tried using a number of different settings and still aren't satisfied with the results, you can set the track markers manually (see above).

### Check and move track markers

Before burning the audio material to CD, it is best to check and make sure that the track markers are in the right places. To do so, the "Tracks (view page 109)" section offers an easy-to-read list with all track markers and names. Use the ID3 Editor (view page 110) for detailed editing.

To adjust the track markers directly in the track window, move the position line using the skip button on the transport console (or Alt + Arrow keys) from marker to marker and play back the audio material from that point. If the marker is unnecessary or incorrectly positioned, you can click on it and delete it or move it using the mouse button.

## Change song order

Sometimes it is necessary to change the song order. The easiest way to do this is when you have already assigned track markers. Then you can resort the tracks in the section Tracks (view page 109) in the ID3 Editor (view page 110) (Marker/Positions tab). In this case objects may be split automatically.

Note: When using multiple tracks (not in MAGIX Audio Cleaning Lab) all the objects on all tracks that are between two track markers are moved together.

Use also the feature of placing the track markers automatically ("Automatic" button in the "Tracks" section or keyboard shortcut Ctrl + M).

Hint: For a quick sorting directly in the track, you can drag a track marker ahead of or behind another track marker. This will regroup the corresponding objects in the track.

In MAGIX Audio & Music Lab Premium there is also a different method: to split the material into separate objects and arrange them as you like, you could also use the additional track (view page 152)s as a clipboard for the objects.

Note: MAGIX Audio Cleaning Lab does not have a second track. Here you can cut an object (Ctrl + X) and paste it at the desired position (Ctrl + V).

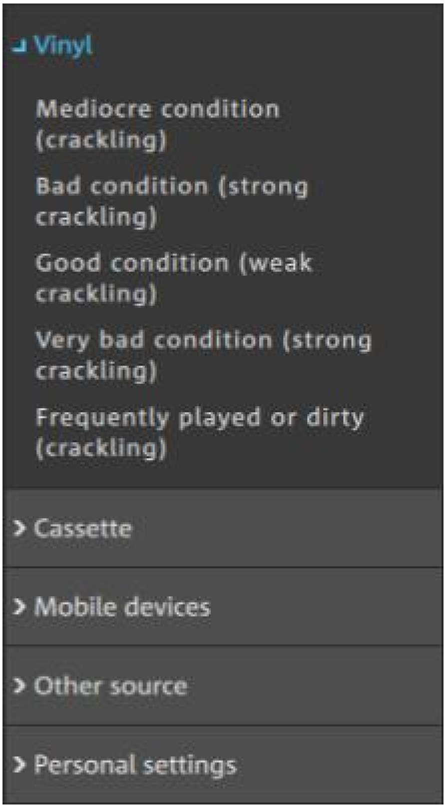
# Cleaning


Most cleaning functions are activated in the Cleaning FX section. Cleaning effects are used to remove annoying noises from the audio.

All effects in the "Cleaning" section are applied in realtime. You can switch them on and off during playback or change their parameters and hear the results immediately.

Additional cleaning functions can be found in the "Effects" menu (view page 136).

## Choose preset




The presets can be found to the left of the effect modules. Here there are presets for specific types of cases that are organized in categories. These presets always affect the entire page of effect modules and can also contain a combination of several effect modules.

Under My Settings (or in the "Edit" menu), you can save and load your favorite effect settings so they can be applied to other objects or in other projects.

Various presets (e.g. "Restore a poor quality record") are included and can be tried out right away.

Cleaning FX presets can be applied to individual objects or to the entire sound.

If the expanded user interface of an effect is open, the list contains presets for this effect.

The presets can be hidden using the arrow button.

# Using the effect modules

**On/Off:** Individual effects modules can be turned off and on using the buttons to the left.

**Control knobs:** Each feature has a knob that controls the intensity of the effect.

The effects include a range of useful presets which can be chosen from a drop-down list. Click on the arrow to access the feature you want. In most cases one preset is enough to achieve good results.

Audio perfectionists can perform further settings with the effects. To do so, you can open the detailed view of the effect using this button.

Hint: The order of the FX modules can be changed. Make a long mouse click on an effects module an pull it to another position. If there is not enough screen space and you had to scroll through the effects to get your favorite ones you can put these on top of the list.

# Detailed view of the effects


If the detailed view is open, the list of presets to the right includes presets for this effect so they correspond to the selection options in the drop-down menu on the effect modules. But there is also the "Own settings" area where the current effect settings can be saved as a presetting for later use. Here there is also the "Reset settings" entry that sets the effect back to its basic settings.

This switch can be used to switch the effect on and off.

Opens the help concerning the respective effect.


Changes to the next effect from the section (cleaning or mastering). So you can make changes in various effects without closing the detailed view.

Use this button to close the detailed view and to return to the overview. The effect remains active however.

The following two options are only available in the cleaning effects:


The "Inverse" button is used to switch over the playback. Instead of the filtering result, just that part of the audio signal that was filtered out by the cleaning effect is now played back. This allows more control of the effect: If you also listen to parts of the music track to the right of the filtered out disturbances, the effect is too strong.


Use this button to reset the realtime displays (spectrograms) in the effects.

# DeClicker/DeCrackler

This function removes crackling and clicking noises which are typical on scratched records. The DeCrackler was specially developed for removing uniform "crackling noises" from old records.

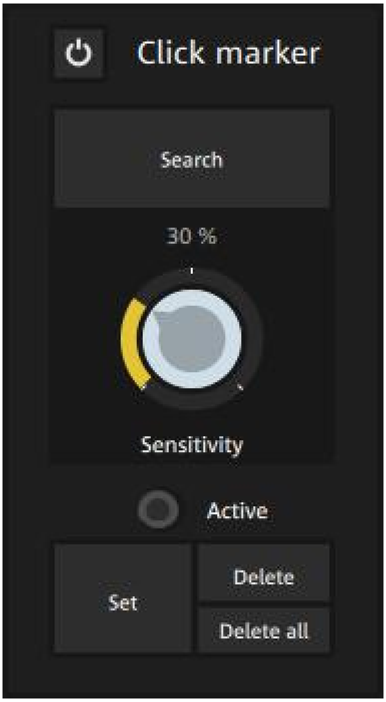

There is a waveform graphic in the middle of the effect dialog, which shows the crackling removed during playback (upper range) and crackling removed in real time (lower range). On the right and left of the graphic there is a slider for each of the two noise types, which is used to control the strength of the respective effect.

The "Inverse" button at the top of the dialog is used as an additional acoustic control. If this is activated, instead of the editing results, you hear only the removed clicks and crackling. This means you can judge whether too much of the original audio material is included in the filtered out noise and then reduce the strength of the effect accordingly.

## Click markers

Along with normal crackling on a vinyl recording, there can be some more noticeable crackling on a record which is scratched. If the DeClicker is set very high for those sections which are particularly noisy, it can lead to several components of the wanted signal being affected by the DeClicker if they exhibit similar characteristics as the crackling.

To avoid this, we recommend setting the DeClicker at a lower setting and removing the sharper crackling and clicks with the help of the Marker DeClicker. In this case, the crackling is marked with a click marker. Only at those points does the special marker DeClicker act more intrusively on the audio material; at the other points the "normal" DeClicker can be applied less strongly and without damaging the material accordingly.




You can use "Search" to look for especially strong clicks throughout the entire project and have them marked automatically. Use the "Sensitivity" slider to set the sensitivity of the search. The higher it is, the more clicks will be found.

But you can also set the click marker manually by placing the position line on a click and clicking on "Set". Use "Delete" or "Delete all" to remove the click markers again.

## DeClicker as object effect

If the clicking and crackling noises only occur in specific parts of the audio, you can use this function as an object effect.

Search for the noise interference in the track window. Once you have found a noise, cut the object just before it begins and once again where it stops (keyboard shortcut T) in order to make it into an individual object. Then activate the "Object" button in the cleaning area to use the DeClicker only on the area where the noise occurs.

**Tip:** If the DeClicker is not able to remove the crackling, you can also cut it out manually. To do this, you have to zoom in as much as possible on the object to cut out the crackling (preferably on a zero crossover - use the stereo display to get as close to a zero crossover as possible on both channels) and move the second object created back to first object (zero crossover). Then you can move the object ends of both objects sightly over each other to blend them together.

# DeNoiser

The DeNoiser removes persistent background noise like mains hum, hissing, noises from sound cards, disturbance from ground wires, interference from audio equipment with high-impedance outputs (e.g. turntables). Impact sound or rumbling can be removed effectively from LPs with the rumble filter.

The DeNoiser requires a short section from your music which contains a sample of the noise interference, usually from the start or end of the recording.

When the DeNoiser is activated in the project for the first time, a sample of the interference signal will be searched for immediately under the playback marker. This automatic noise sample also enables extremely good successes to be achieved without opening the DeNoiser dialog.

## Settings in the effect dialog.


**Noise sample:** If the automatic noise determination cannot find any suitable material, a noise sample can be selected from the list. There are several typical noises to choose from.

**Get:** Another better possibility is to to create a noise sample yourself. Click on "Get" to take a short sample as a noise sample from the current playback position. The playback marker can also be moved with the transport console when the dialog is open to search for a suitable position. If the playback position is between two objects, no noise sample can be obtained. The Play button can be used to preview the noise sample for test purposes.

**Wizard:** A wizard that helps in the creation of the noise sample is opened. The wizard primarily contains a search function that helps when searching for a suitable noise sample in the audio material. Various types of noise can be selected.

**Length:** The length can be set in ms if the "auto" button is switched off, otherwise the length of the noise sample will be determined automatically.

**Audio type:** Here you can set the type of audio material edited; the algorithm is adapted accordingly.

**Editing mode:** In **Broadband interference** mode the DeNoiser works in a mode optimized for removing noise. In the **low-frequency sound interference** mode, the DeNoiser works in a mode optimized for removing tonal disturbances. This includes feedback from power cables, mains hum, PC fans, video cameras or ventilation noises. Depending on the editing mode, one of the two following options are available:

- **Responsive:** The value for the noise level parameter is set automatically here by determining the level of the noise available in the signal. You can find more information about this option in the Dehisser section.
- **Maximum attenuation of tonal noise:** If this option is active, these types of disturbance are totally removed. The "Reduction" slider now only has an effect on any simultaneously available noise components. This makes sense because tonal disturbances are much more annoying than other noise - a small amount of "extra" noise in the recording may be allowed in order to avoid a greater loss of highs in the wanted signal.

**Noise level:** Here the application threshold of the function for noise removal must be set as precisely as possible. Values that are too low will result in insufficient noise dampening which causes artifacts such as interference or "twittering" (see below). High settings produce dull results – useful signals that sound similar to hissing noises are also filtered away. Take your time to find the best setting for the individual case.

**Reduction:** Here a balance between the original signal and suppressed signal can be set. It's often better to reduce interference signals in small increments, e. g. 3 to 6 dB, rather than as much as is possible to keep the sound "natural". For humming, it's best to apply complete removal.

**DeRumbling:** Here you activate a special filter for low-frequency rumbling noise. Examples of these noises are mechanical noises from old record players, wind and impact sounds in microphone recordings.

**Spectrogram:** The right spectrogram can be switched between a realtime display of the effect result and a static display of the used noise spectrum.

## Artifacts

If the settings are incorrect, the DeNoiser and DeHisser can leave behind a metallic chirping or twittering sound, the so-called artifact noise. The cause of this is the incomplete removal of the noise interference. The ear is quite sensitive to this sound because of its synthetic character. This problem, in practice, only occurs in especially difficult cases.

To achieve the best possible results, you should pay attention to the following notes:

- First, select a preset from the selection menu. In most cases the result is satisfactory.
- Be careful when "denoising" the effect: Less is more! The noise interference should no longer be audible, otherwise artifacts may be brought about.

It's best to remove any available DC component (see"Removing DC voltage" (view page 73)) from the audio material before using the DeNoiser or DeHisser.

# Dehisser

The DeHisser eliminates regular "white" noise typically produced by analog tape recordings, microphone preamplifiers, or AD transformers.


**Noise Level:** Here you should set the threshold of the DeHisser as precisely as possible. Setting the value too low will result in insufficient removal of the noise. Incomplete removal of noise produces artifacts and should be avoided. Setting the values too high leads to dull results – parts of the wanted signal that are similar to noise, such as the blow-off from wind instruments, are filtered out.

The setting doesn't cause any problems at a reduced noise level.

**Responsive:** The value for the noise level parameter is set automatically here, by determining the level of noise available in the signal. The setting on the noise level knob now has a relative effect, i. e. the used value results from the automatic system plus the setting of the noise level slider.

An advantage of this is that you no longer have to adjust the noise level manually and the values are adjusted for fluctuating noise levels, e. g. if you use different tracks with varying noise levels in one project.

If the noise volume is constant, you may be able to achieve better results by setting things up manually (responsive off). But the value for noise level has to be set exactly.

**Noise Reduction:** Here you can adjust the dampening of the hissing in decibels. It is often better to reduce noise in smaller increments, e. g. -3 to -6 dB, rather than as much as is possible to keep the sound natural.

**Audio type:** Here you can set the type of edited audio material, the algorithm is adapted accordingly.

# Declipper

If the input level of an audio recording is too high, distortion may result at the louder parts (the signal peaks). This digital distortion is also called "clipping". At the overmodulated area, the values that are too high are simply cut off and the typical, annoying crackling and distortions are heard.

MAGIX Audio Cleaning Lab has a special function for dealing with digital clipping.

Distorted sections are discovered and filtered out based on the material in the selected object. Lastly, the master volume of the material can be reduced so that the interpolated parts can be played back without distortion.

The declipping algorithm is particularly good for audio material with clearly audible clipping, e. g. distorted piano or vocals. The sound of distorted drumbeats on the other hand is hardly ever improved.


**Peak meters:** The peak meters show the input signal, the output signal and the strength of the declipping.

**Clipping level:** Here you can enter the level at which the algorithm will consider a sample to be clipped and correct it. This is important because different sound cards have different clipping characteristics.

**Get:** Here you can determine the clip level automatically.

**Output level:** The interpolated signal peaks cause a change to the total level that you have to offset with the output level slide in order to avoid renewed distortion. To do this, observe the peak meter at the top of the dialog.

For security, a limiter that reliably avoids clippings can be connected with the "**Output limiter**" option.

# Tempo/Resampling

This effect is only available as an object effect (view page 72).

The fader lets you change the playback speed of objects so that they are better aligned. The effect can be applied in two ways, either as resampling or as timestretching. You can change the mode in the preset list at the very bottom.

- **Resampling** mode can be used to change speed and pitch just like on a cassette. Use this mode to adjust LP recordings made at the incorrect speed.
- **Timestretching** mode applies a high-quality timestretching algorithm to keep the pitch constant in spite of speed changes. Use this mode to adjust the tempo of different tracks to match each other without influencing the pitch, e.g. for a DJ mix.

The effect is also available as a mouse mode (view page 147) for changing the tempo across a larger range of values.

## Resampling for incorrect record speeds

If you want to record a record that was recorded at 78 rpm, then you normally have a problem: These older shellac LPs aren't able to be played back by most turntables. With the help of this resampling technology, it's possible to playback the record at the incorrect speed, record it, and then correct the speed with a single click.

Different presets have been provided for this. The first number indicates the speed at which the record was played back, and the second shows the speed that it should be played at. For example, if an older 78 rpm shellac record was played at 33 rpms, then you would use the "33/78" preset.

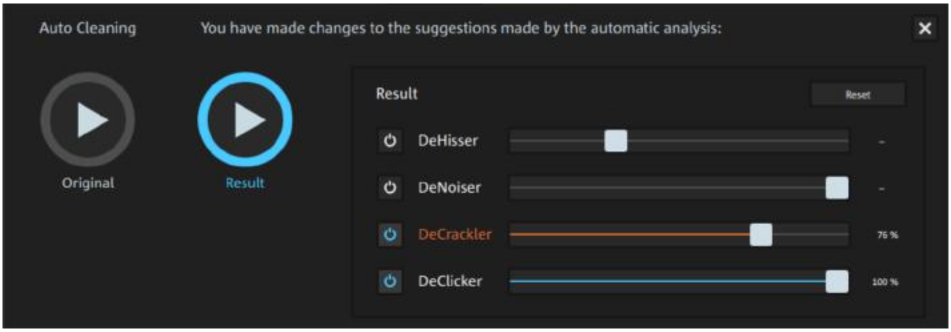
A second group of presets are for adjusting wave files with different sampling rates to the project. These are selected automatically when this sort of wave file is loaded into the project. The first number here is also the sample rate of the project (for playback, normally 44.1 kHz or CDs), and the second is the wave file (the target playback rate).

# Auto Cleaning

Auto Cleaning analyzes the audio material for disturbances and recommends suitable settings for the Dehisser, Denoiser, DeClicker and DeCrackler effects.


Start the analysis with this button.

The result of the analysis is shown and suitable effects with the right strenght are activated:



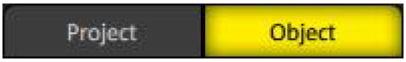

You can now change the strength of the single effects or deactivate them. Those changes from the original proposal of the analysis are marked red. A double click on a slider turns it back to its original proposed value, with "Reset" you go back to the whole analysis result.

For a comparison you can play back the original audio with "Original" and the processed audio with "Result".

Once you are happy with the iprovements, just close the Auto Cleaning with the close button at the top right, the last effect settings are kept.

# Object effects

All "Project" settings effects are also available as object effects and can be applied in addition to the project effects. This means you can remove noise from the entire project or a humming sound that only occurs at one point in a specific object.

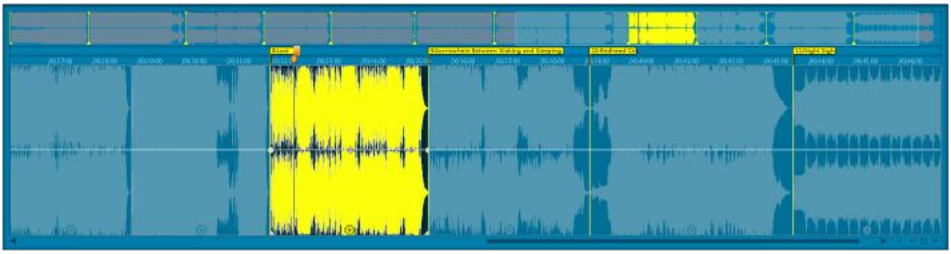



The object effects can be opened using the "Object" button located above the effects or directly on the objects in the track window using the cogwheel button. While the effects in the project section affect all objects in the master track, the object effects are only applied to the selected object.

You can create new objects by splitting objects (view page 57) into several smaller ones. Use the Scissors button or the "t" key to do this.

Tip: If there are already track markers in the project, e.g. ones that have been automatically set (view page 60) you can automatically split the object so that each track is a separate object. This can be done in the menu "CD "> „Split objects at track marker positions" (Keyboard shortcut: Ctrl + T).

As different objects can have distinct object effects, the indicator for object effect settings changes when you click on a different object. The selected object will then be noticeably emphasized to make it clear which object the current object effect settings are affecting.

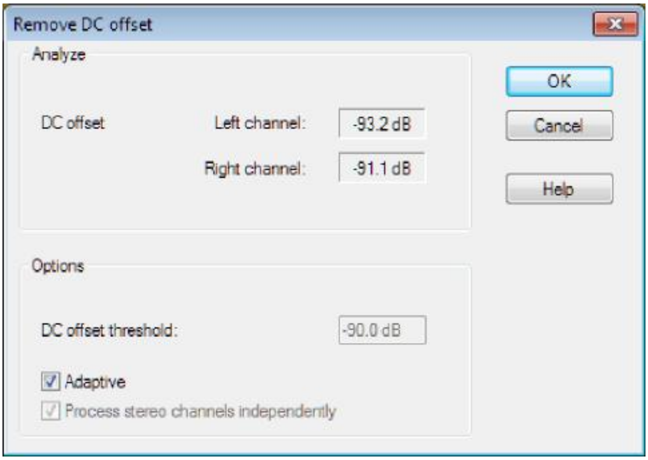



If there are active Object FX, the gear symbol is marked with "FX". In this way active object effects are identifiable even if project level effects are displayed.


If you select an object, the InfoBox (view page 37) displays which object effects are active.

# Remove DC offset

This function can only be opened using the "Effects" menu and can thus only be applied to selected objects. This can be useful if your sound card overlays your sample with a constant DC offset during recording, which leads to crackling during playback or editing. (This is pretty much always the case with recordings that use the integrated sound card of your PC.)




In (preset) **Responsive** mode, an entirely constant DC voltage cannot be assumed. A variable DC voltage value is also recorded. From a technical point of view, all frequencies below 10 Hz are regarded as DC voltage. If "Responsive" is deactivated, a constant DC voltage value is determined. Then you can also enter a minimum **DC offset threshold**, which indicates where DC offset removal will kick in. You can also edit stereo channels together to reduce processing time.

# Mastering


The mastering functions are activated in the Mastering section. Mastering effects are used for either improvement or a dedicated change of the sound of the audio.

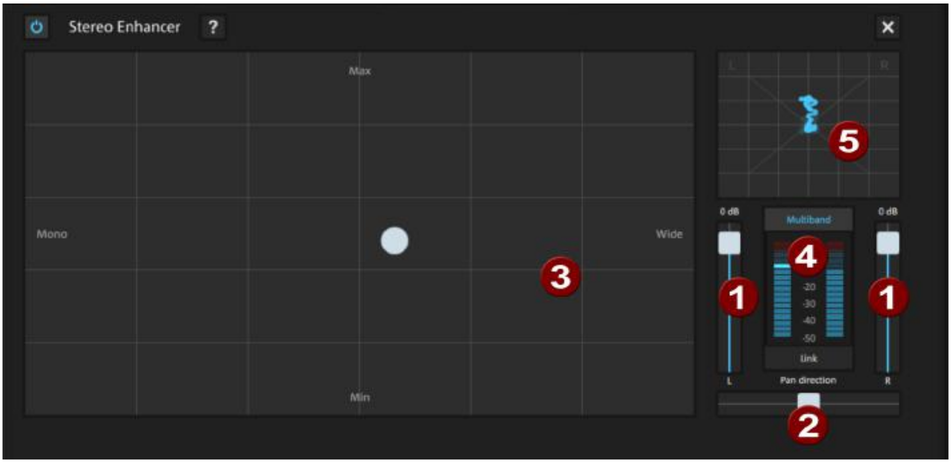
All mastering effects function in realtime which means that you can switch them on and off and change their parameters during playback and listen to the result of the changes immediately.

On information about the general usage of the effect modules, the project and object effects and the presets look above in the Cleaning Effects (view page 62) section!

## Stereo Enhancer

With the Stereo Enhancer you can determine the positioning of the audio material in the stereo panorama. If the stereo recordings sound unfocused and undifferentiated, an extension of the stereo base width can often provide better transparency.

Use the maximize function to move the echo and improve the stereo picture, for example, into the foreground.

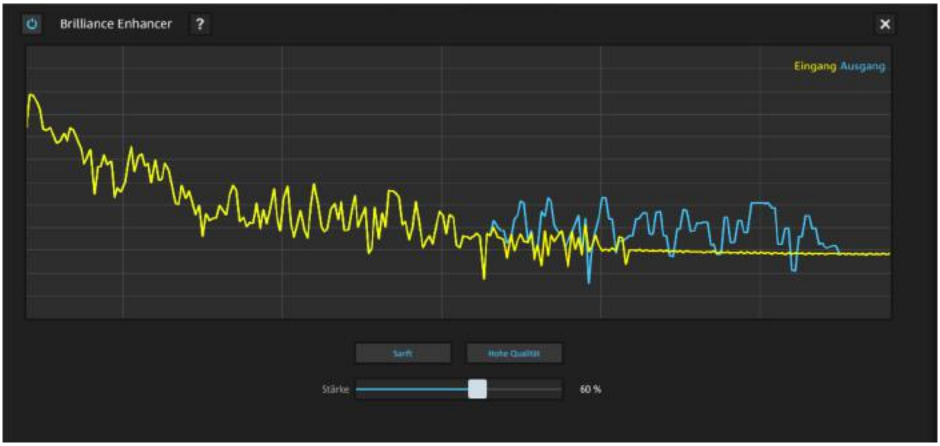


1. **Volume control:** Sets the volume, and thus the panorama as a whole, for the individual channels. The damping of left and right levels is displayed in dB underneath the controllers.
2. **Panorama:** The slider allows you to move the sound source that comes out of the middle into the stereo panorama. The signals at the outer edges of the sound picture remain unchanged.
3. **Base width/Maximize sensor field:** This sets the base width between "Mono" (completely to the left), unchanged base width ("Normal Stereo") and maximum base width ("Wide", completely to the right). Raising the base width (values over 100) diminishes the mono compatibility. This means that recordings edited this way sound hollow when listened to in mono.

   Maximize strengthens the spatial component of the recording, which also increases the stereo transparency without influencing the mono compatibility.
4. **Multiband:** This can be used to switch Stereo FX to Multiband mode. Stereo editing only applies to the middle frequency; the bass and highs remain unchanged.
5. **Stereo meter (correlation gauge):** This provides a graphical display of the phase relation of the audio signal. You can use it to check the alignment of the signal in the stereo panorama and the effect of the stereo enhancer. To maintain mono compatibility, the "cloud" shown should always be higher than it is wide.

# Brilliance enhancer

The brilliance enhancer is a high-end audio tool with which losses in high frequencies, which occur with MP3 compression or while recording older records, can be regained. As opposed to the equalizer, which only raises the available frequencies, the brilliance enhancer generates new overtones from the material available and really refreshes the sound.



**Soft:** This option reduces any distortions that occur. The entire sound appear less clear. The activation of "Soft" depends on your sound settings. If you prefer a brighter sound, you should deactivate the "Soft" option.

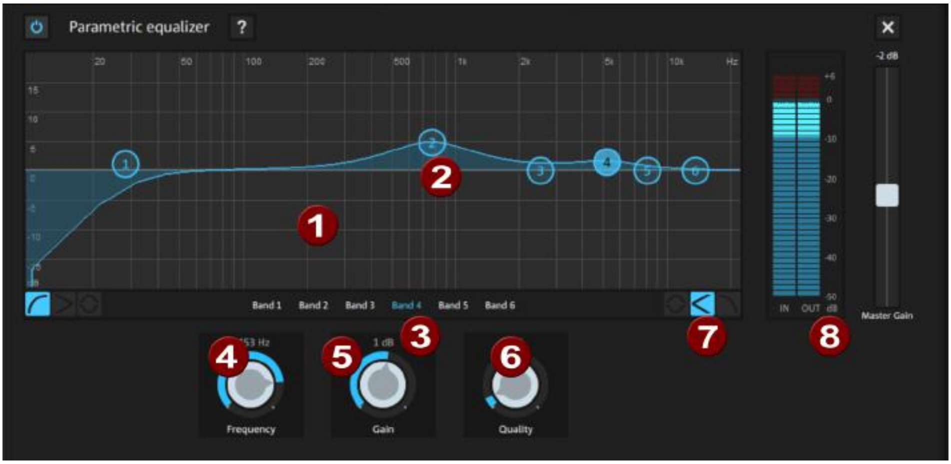
**High quality:** This option improves the results, but increases the computing power required. As a rule of thumb, you should decide as you go and rely on your own judgment.

**Strength:** This controls the strength of the effect.

# Parametric 6-band equalizer

The parametric equalizer has six filter bands that you can use to shape the sound of the music track. Each band is a filter with a typical "bell shape". Within a certain frequency range around an adjustable middle frequency you can increase or reduce the signal level gain by a specific amount (raising). The width of this frequency range is called bandwidth. The bandwidth is defined by the Q value. The higher the Q value, the narrower and steeper the filter curve.

You can influence the basic sound of the mix by boosting or cutting specific frequency ranges (low Q value) to give it more "depth" (lower mids 200-600 Hz) or more "air" (highs 10 khz). You can also decrease very specific frequencies (high Q-value) to remove unwanted noise.




1 **Sensor field**: The sensor field displays the resulting frequency response of the equalizer. The frequency is displayed horizontally and the increase or decrease of the respective frequency is displayed vertically.

2 The yellow bullets 1-6 symbolize the six frequency bands. You can move them around with the mouse until you find the frequency response you want. The bandwidth (Q value) can be adjusted using the mouse wheel.

3 The faders under the filter graphic display the parameters of the currently selected band. The rotary knobs can be used to adjust the values for each band more precisely:

4 **Frequency:** The center frequency of the single bands can be set between 10 Hz and 24 kHz with the frequency controller. Freely choosing the frequency enables multiple filters to be set to the same frequency in order to have a greater effect.

5 **Raising:** This controller allows you to raise or lower the filter. Setting the controller to 0 deactivates the frequency band so it does not use any processing time.

6 **Quality:** This sets the quality (bandwidth) of the individual filter.

7 Bands 1 and 6 include a special feature: They can be operated with three different filter curves.

 **Peaking:** This corresponds to the normal bell shape that is also used for the middle bands.

 **Shelving** (basic setting): From the set frequency, the frequencies will be gently raised or decreased.

 **High-pass or low-pass**: All higher or lower frequencies from the set frequency are filtered away.

8 You can control the output level of the equalizer with the peak meter. With the **Output Gain** beside it you can offset the level changes resulting from the EQ adjustments.

# Grafic EQ

The Grafic Equalizer subdivides the frequency spectrum into ten areas (‘bands’) and equips them with separate volume controls. This way, it is possible to create many impressive effects, from a simple bass boost up to complete distortion. Note: If low frequencies are boosted too much, the total level is heavily increased which may lead to distortion. In this case, the master volume should be decreased with the aid of the "Master Volume" slider in the main window.


**Bullets 1-10:** Each of the 10 frequency ranges can be separately increased or reduced with the 10 volume controls.

**A/B:** If you have selected a preset for the effect and make manual changes to it later, you can compare the original preset sound with the new settings by using the A/B button.

**Link:** Using this function randomly combines the frequency ranges with each other in order to avoid artificial-sounding overemphasis of an individual frequency range.

# Reverb


The reverb effect unit offers realistic reverb algorithms to add more depth to your recording. You can control the sound of the reverb effect with the following parameters:

**Room size:** Here you determine the size of the space (or, with plate and spring, the systems). The larger a space, the longer the sound travels between walls or objects. With some low "size" settings, you can also reduce the distance between the individual reflections. This allows resonance to develop (accentuated frequency ranges), which can sound oppressive if the reverberation time is too long.

**Tone quality:** Here you can influence the audio characteristic of the effect to a certain extent. The effect of this controller depends on the used preset. In rooms, "Tone quality" controls the dampening of the highs in the reverb (from dark to bright) as well as pre-filtering of the signal. With plate and spring presets, this fader determines the dampening of the basses as well.

**Reverberation time:** With this knob you can adjust the reverberation time and determine how much of it will be absorbed and, simultaneously, the reverb's decay.

**Pre-Delay:** The reverb portion ("mix") as well as the early reflections play a large role in the perception of sounds in the room. The time required for the perception of the early reflection is know as "Pre-Delay". For sound sources that are close, the reverb portion is usually low and the early reflections reach the ear noticeably later than the direct sound. In contrast, sound sources that are far away usually have a high reverb portion and the early reflections reach the ear almost simultaneously with the direct sound. The length of the pre-delay also influences the "sensed" distance of the sound source from the listener.

**Mix**: This controller sets the mix ratio between the original and the edited signal. For rooms, you can therefore move a signal further into the room by increasing the effect percentage.

**Presets:**

The presets represent the basic settings for the various room algorithms, which can be still be varied along with the other parameters. Hence, they are more than just simply parameter sets.

> Hint: In MAGIX Audio & Music Lab Premium you can use a second reverb effect, the VST plugin efx_reverb (view page 97)!

# Echo


"Delay" actual means a time delay. This effect acts like an echo, as the signal is reproduced after a delay.

**Delay:** This sets the time span between the individual echoes. The more the control is turned to the left, the faster the echoes will follow each other.

**Feedback:** This adjusts the amount of echo. Turn the dial completely to the left, there is no echo at all; turn it completely to the right and there are seemingly endless repetitions.

**Mix**: This fader determines how much of the unprocessed original sound (dry signal) is subjected to the echo (wet signal).

# SoundCloner

SoundCloner 2 analyzes the sound characteristics of songs and transfers them to other recordings. Allowing you to apply the sound of a 60s soul recording to a modern pop song for example. You can also use SoundCloner to analyze compilations of different songs and compare their sound.

SoundCloner features an FFT Filter (1024 - band equalizer) and a compressor. The filter curve and compression properties are automatically calculated and presented as a combination of the cloned sound and output sound.

As well as "Clone" presets the SoundCloner presets menu contains a few useful filter settings that provide a typical sound for a range of eras (70s, 80s, 90s etc.).

Filter curves can also be edited manually, SoundCloner can therefore also be used as a filter for creating strange effects. It can also prove more effective than the DeNoiser at removing noise in certain circumstances, e.g. for removing whistling in the background.

## Using the Sound Cloner

1. Load a song with a sound you really like.
2. Now change to "Object" on the effect page or click on the cog wheel icon next to the audio object to activate the object effects.
3. Open the Sound Cloner and click on "Get Sound". The sound characteristics are now calculated. After that, the sound will be available as a **SOUND CLONE** preset and can be saved permanently for later use by clicking "Save".
4. Activate the Sound Cloner on the object you want to apply sound to and click on "Apply sound clone". The Sound Cloner detects the sound and dynamic properties of the target object and calculates the filter curve and/or the dynamic setting, which resulted when the audio characteristics of SOUND CLONE and the edited object were combined.
5. The filter curve and the dynamics editing of Sound Cloner compare the audio characteristics of the target object to those of the analyzed object. Using the "Strength" slider under the filter graphic, you can regulate the intensity of the frequency adjustment.

**Note:** The Sound Cloner has to be used as an object effect because the final filter setting depends on the audio characteristics of the "cloned" object and of the target object.

However, if you have uploaded a record, you should only use the Sound Cloner on the master because a record has already been mastered. The frequency response and compression differ in fact from plate to plate, but the tracks on a record have already been synchronized with each other. Just think about recordings of classical concerts or conceptual albums, for example. In these cases different mastering of the tracks from each other would be rather distracting.

## Controls


1 **Get Sound:** This button analyzes the audio characteristics and makes the SOUND CLONE preset available.

2 **Save:** Save the SOUND CLONE preset under another name for later use.

3 **Apply Sound Clone:** The result of the current analysis, the SOUND CLONE preset is applied.

4 List of the five last presets used

5 **Original/Result:** Plays the edited/unedited object.

6 **Filter graphic:** This is the centerpiece of Sound Cloner, it is a filter drawn freehand that displays the frequency response, before and after applying the filter in realtime. (See below for instructions on how to use.)

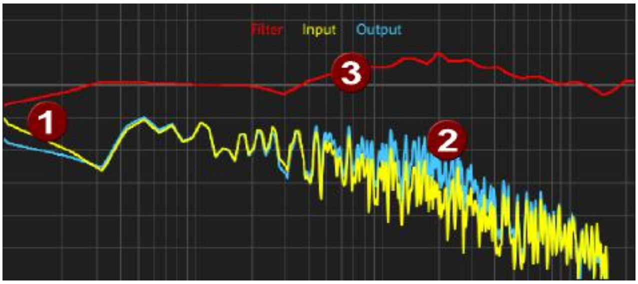
7 **Strength:** The strength slider allows you to enhance or diminish the differences in the curve movement. You can therefore control how much Sound Cloner adjusts the sound to the presets.

8 **Audio Type:** You can choose from a range of different audio types (speech, pop music etc.) to optimize the compressor's function.

9 **Compression:** This slider lets you change how much the preset's dynamic range should be adjusted. Dynamic range can only be reduced, so if the reference material has a higher dynamic range than the edited material, this slider won't have any effect.

10 **Level correction**: The slider allows you to balance out any volume changes caused by the compressor. The peak meter next to it displays the input and output levels.

## Filter graphic




The frequencies are listed in ascending order from left to right. The height of the curve represents the amount of a specific frequency in the entire sound. The yellow curve (**1**) shows the original frequency response, the blue curve (**2**) displays the corrected frequency response, i.e. the frequency response the spectrum has after applying the filter.

The red curve (**3**) is the filter curve, its height determines to what extent the respective frequency should be amplified or attenuated. When using Sound Cloner as a filter this curve is a combination of the desired frequency response ( = content of the SOUND CLONE analysis) and the current frequency response. The means the red curve will always look different regardless of the object when using the same preset.

The filter curve can be drawn using the mouse in the filter graphic. Draw or rotate a straight line by pressing and holding the Shift key. "Reset display" resets the red filter curve to neutral, i.e. to be on a straight line with the initial position.

## Removing audio distortions with Sound Cloner

Sometimes the DeNoiser isn't the right tool for removing certain kinds of noise interference. For example, if the noise doesn't occur anywhere "alone" in the recording it is hard to get a sample to use, or if the noise is particularly loud, the DeNoiser filters too little of the useful signal resulting in a hollow, artificial sound.

Some noise interference consists of a few frequencies, and often only one frequency (pure tone, such as the "pips" on the radio). A typical example of this is a constant, clearly-audible drone in old video recordings. The FFT filter is perfect for this kind of noise as it allows you to remove specific frequencies without affecting the rest of the recording.

Follow these instructions to do so:

1. Reset the filter curve ("Reset display" button in the top left of the dialog).
2. You will be able to spot the noise interference during playback as it is the only spike that remains constant in the otherwise constantly changing yellow filter curve.

3. Draw a horizontal line below the spike on the lower edge of the filter display, ideally it should be just long enough that the spike in the blue (resulting) filter curve disappears.
4. If the noise interference can still be heard, you can also remove the overtones of the signal, which are weaker peaks that can be found in the second, third, fourth multiples of the frequency etc.

# Video sound optimizer

There is a complex signal processing algorithm at work which combines different functions for emphasizing speech and reducing background noises (wind, rumbling, hissing).

# Dynamics


Dynamics is an automatic, dynamic volume controller: The volume of a loud section will be lowered and the volume of muffled sections will be raised.

Processing is carried out using a "look-ahead" method, similar to high-quality studio appliances. There are no peak overmodulations or other artifacts as the algorithm can never be 'surprised' by sudden level peaks.

**Threshold:** Set the volume threshold, below and above which compression is applied.

**Ratio:** This parameter controls the amount of compression.

**Attack:** Sets the algorithm's reaction speed to increasing sound levels. Short attack times can create an undesirable "pumping" sound, as the volume is quickly reduced or increased correspondingly.

**Release:** Sets the algorithm's reaction speed to falling sound levels.

**Gain**: The gain controller amplifies the compressed signal.

**A/B:** If you have selected a preset for the effect and make manual changes to it, you can compare the original preset sound with the new settings using the A/B button.

# MultiMax


MultiMax is a compressor with three independent frequency bands. The dynamics editing is carried out separately for each band.

The advantage of a multi-band compressor in comparison to a "normal" compressor is that the "pumping" tendency and other disturbing side effects are dramatically reduced during dynamics editing. For instance, it can prevent a bass level peak from "dragging down" the entire signal.

Multiband technology also lets you specifically edit individual frequency ranges.

**Setting the frequency bands:** The settings of the frequency bands are changed directly via the graphic. Simply click on the separator lines and move them.

**Bass/mid/high:** These knob controllers define the level of compression for each frequency band.

**Link bands:** When this button is activated and one fader is adjusted, all faders are changed in the same ratio. However, the way the dynamics are edited is not affected.

**Presets:** In MultiMax, you can use the presets to open 2 further special functions:

- **Cassette NR-B decoder**: MAGIX Audio Cleaning Lab simulates decoding of Dolby B + C noise reduction if a Dolby player is not available. Cassettes recorded with Dolby B or C sound more muffled and slurry if played back without corresponding Dolby.

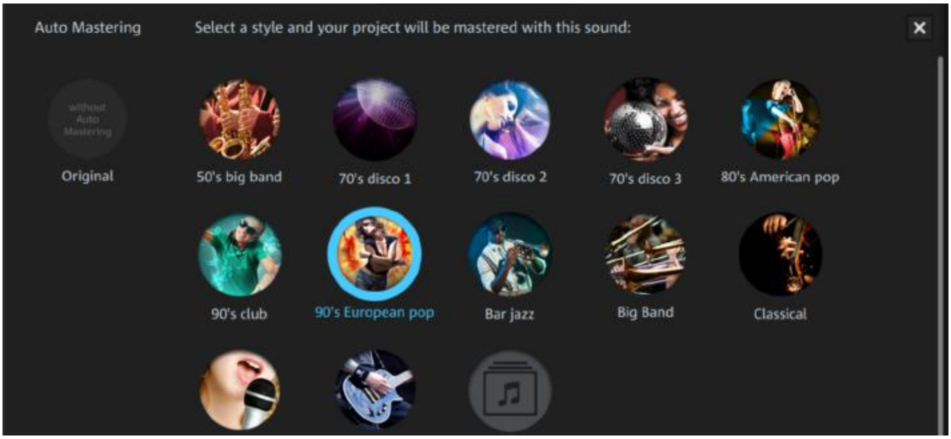
- **De-Esser:** These special presets help to remove overstressed hissing sounds from speech recordings.

# Auto mastering

The Auto Mastering feature enables you to automatically apply the sound of typical musical styles from the past and present (e.g. 70s disco, 90s club, jazz etc.). The sound of the source material is analyzed and appropriate equalizer and dynamic effects are applied.


The auto mastering wizards are activated by the corresponding button at top left above the presets.

Now choose the musical style.




MAGIX Audio Cleaning Lab analyzes the audio material in the project. The settings that are used to achieve the optimum sound for a particular musical style depend on the audio characteristics of the source material. If the audio material in all the objects is from the same source and are similar enough in sound, the wizard will apply the appropriate effects at the project level. If the objects differ too greatly, the effects are applied to each object separately. Audio & Music Lab automatically decides whether the auto mastering will be applied to individual objects or to the entire project.

Note: In MAGIX Audio Cleaning Lab the effects are always applied at the project level. If you want to work with separate objects, you can apply the Sound Cloner (view page 80) manually.


Selecting "Original" can be used for the comparison of the editing and the original.

# Plug-ins

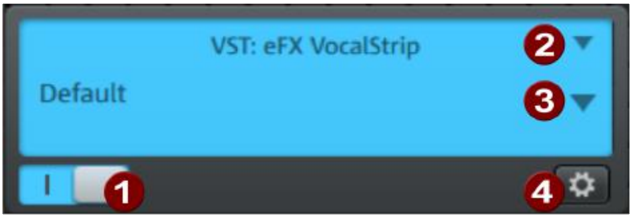
DirectX and VST-compatible plug-ins can be used for effects calculation in MAGIX Audio & Music Lab Premium. This makes it possible to use third party effects and dithering processors in addition to the effects that are already integrated into MAGIX Audio & Music Lab Premium.

Some of the included effects are loaded as plug-ins. These include:

- MAGIX essential FX (view page 89): DeEsser, Vocal Strip, Gate, Reverb, Tube Stage and Chorus Flanger
- Energizer
- am-track SE

> Note: Plug-ins are not available in MAGIX Audio Cleaning Lab but the DeEsser effect is included as a simple effect module.

Both the cleaning and the mastering sections have special effect modules where plug-ins can be loaded. The plug-in module has no effect level controller but consists of two selection lists.




1 **On/Off:** This button turns the plug-in effect on or off.

2 **Selection menu:** Select the plug-in from the selection menu on the right side of the module. For this, you will need to have plug-ins installed on your computer. The path to search for installed VST plug-ins can be set in the "Options" menu via -> "Path settings". All recognized plug-ins will be added to the menu list.

3 If the plug-in has presets, they will be included in this list. Many plug-ins also come with their own preset manager which can accessed through the plug-in interface.

4 **Edit:** The selected plug-in is opened in the VST Plug-in Editor (view page 88) in order to make specific effect settings.

# VST PlugIn Editor

The editor has two views: the so-called "GUI" (graphic user interface) of the plugin and the parameter view mode. The latter is either activated automatically if the VST plugin does not have its own GUI or can be used of the GUI of the plugin is too confusing or occupies too much space on the screen. The parameter view displays eight parameters of the plugin as sliders. In the Plugin menu you can switch between these views.

**Load/save patch/bank**: The instrument settings can be saved and loaded in the patch and bank formats typical for VST plugins (*.fxp and *.fxb respectively).

**Random parameters:** This function can be an important source of inspiration. However, before using it please save the current preset you've just created as this feature does not ask before it is applied.

**Program menu:** Here you can select the included plugins or the presets loaded via the plugin menu.

# essentialFX

The essentialFX is a collection of various studio effects with a standardized interface. The DeEsser (view page 93) and Gate are typical cleaning effects, the Vocal Strip (view page 92) and Tube Stage (view page 99) effects are used for sound optimization. The Chorus/Flanger is a special effect for distorting the sound.

> Note: The essentialFX were originally developed for the professional audio program MAGIX Samplitude. For this reason even the Help files are oriented towards professional users. But there's no need to worry because these plug-ins include a wide range of helpful presets. Simply try them out until you find the sound that's right for your project.

## MAGIX plug-ins

### Console

Some MAGIX plug-ins show a so-called "Console" at the top edge when they are opened – a display bar for managing presets with expanded settings options.


A menu containing available presets is located behind the display. To the right is a prev/next button, which lets you leaf through presets.

This button is used for saving presets. MAGIX plug-ins use a proprietary preset save format (*.fxml)


You can return the presets to their initial settings by clicking on the "Reset" button.

**Bypass switch**: Routes the signal directly to the output instead and bypasses processing. Internally, processing is continued so that you can toggle between processed and unprocessed material anytime.


**A/B comparison**: Very useful for trying out settings. The controller setting "A" memory is normally activated when the interface is opened.

As this is the initial status, "B" also contains the same settings. If you would like to experiment without losing the current setting, press the "B" buttons and try other settings. To transfer the values to "A", press the copy button between the two letters.


**"?" button:** This opens the online help for the plug-in.

## Parameter smoothing/ Controller knobs

Each plug-in offers soft controlling. Pot settings are softly faded internally from the old value to the new one. This is particularly noticeable when playing in "Live" mode. Due to performance reasons, switches (e.g. on/off) and some settings of VariVerb II that change delay times directly or indirectly are excluded from this action.

Use the mouse wheel to move the knobs (pots). A combination of the mouse wheel and "Shift" key slows down the increase or decrease by a factor of ten.
Double-clicking resets the knob to the original value.

## Vpot Controls

Some controllers snap to the middle setting, which may be useful for quickly returning to a "neutral" status. It is extremely difficult to adjust the fine settings within close proximity of this snap point. You can temporarily deactivate this snap mode by holding "Shift" before touching the controller.

Use the mouse wheel to adjust the controllers. A combination of the mouse wheel and "Shift" key reduces the increase/decrease by factor ten.

Apropos the controller movement, note that all plug-ins follow the host settings regarding linear or circular mouse movement. You can usually choose whether you prefer up/down or a circular movement to adjust the value.

# essentialFX Presets

By clicking on the tool symbol in the graphic display, you can open the presets.

Under "**Tweak"** you can find specific settings options for each effects device.

Under "**eFX Globals**" you can find settings options for graphic level display as well as mouse control.


## eFX Globals

These settings apply to all Essential FX.

**Metering:** Here you can set the meter display.

- **Decay Time** controls reaction speed.
- **Peak Hold** controls the hold of the highest peak.
- **Brightness** controls the display brightness.

**Mouse:** Here you can set the display reaction to mouse movements in the essential FX graphic display.

- **Linear mode** lets the faders move in response to vertical and horizontal mouse movements.
- **Circular mode** lets the faders be controlled by circular mouse movements.
- **Mode As Host** lets the faders move in response to the mouse exactly how mouse settings in MAGIX Audio & Music Lab Premium specify it.
- **[Shift] Factor** specifies the factor for fine adjustment of individual faders with the mouse with simultaneously held Shift key.

# Vocal Strip

This plug-in combines several components in one special tool which is ideal for working with speech or vocal tracks. Virtually all recurring voice signal edits are integrated in a compact, simple interface.


The signal flow is set by the e_FX_VocalStrip and is directly represented by the position of the fader.

## Vocal Strip Parameter

**IN / OUT:** Here you can set the input and output level.

- **highpass:** This fader determines the cut-off frequency for a steeply sloped highpass filter (24dB/octave) to remove part with low frequencies, such as rumbling or noises created when touching the microphone.
- **gate:** With gate you can attenuate the signals that fall below the threshold. Here, a smooth curve and a maximum 'softening' of up to 24dB is used to avoid hard transitions.
- **DeEsser:** The effect is very similar to that of the eFX_DeEsser. Here, however, the application frequency is set. The filter used in eFX_VocalStrip works in another range. The fader determines the degree of reduction.
- **compression:** For the most part, an eFX_Compressor component contributes to working parameters for optimizing voice recordings. The further the fader is turned the lower the threshold and higher the compression ratio. The Attack and Release times are selected according to the program.
- **tone:** This EQ corresponds principally to the filter network, which is also used in the eFX_TubeStage plug-in. With this you can simply and effectively calibrate the sonic balance of the voice signal, e. g. for better integration into the mix.

Tweak

- **DeEss DetFreq:** Determines the frequency of the filter used for detection, and filter blocking in the signal route. Typically the sibilants in speech or singing voices are in the 6 - 8 kHz range.
- **CompAutoMakeup:** The maximum level is retained while overall level is reduced.
- **Comp Attack**: Here you can determine how fast the compressor responds after reaching the threshold.
- **Comp Adapt.Rel.:** Rising values cause the compressor to work increasingly with "Adaptive Release". This means that the longer and harder the compressor intercedes upstream in the signal path, the longer is the resulting release phase.

**Note:** Detailed information about compressor-specific settings can be found under "essentialFX > Compressor > Compressor Parameter".

## DeEsser

This plug-in filters/attenuates sibilants in vocal recordings in a simple yet effective way. Since the underlying process operates on the basis of a dynamic filter, other signals are weakened such as cymbals in drum recordings or other signals in a similar frequency range.


In contrast to other devices of this type, the eFX_DeEsser works without adjustable thresholds. The plug-in constantly evaluates the input signal and recognizes signal peaks that are high above in the frequency range of the average levels. Through this comparative analysis with the wanted signal, a constant reduction of noisy sections can be adhered to, so a traditional threshold often requires adjustments for changing levels.

### DeEsser Parameters

**IN / OUT:** Here you can set the input and output level.

- **freq:** Determines the frequency of the filter used for detection, and filter blocking in the signal route. Typically the sibilants in speech or singing voices are in the 6 - 8 kHz range.
- **Speaker symbol (pre-listen):** This allows you to hear the "solo" filter frequency so you can locate noisy sections quickly and easily.
- **reduction:** Regulates the filter attenuation in the signal route.

#### Tweak

- **Filter Q**: Set the bandwidth of the individual filters between 0.10 (very wide bandwidth) and 2 (extremely narrow bandwidth).

## Gate

This plug-in can be used to remove noises in the pauses between audio material, e.g. a pause in speech during commentary. When the signal level falls below a certain threshold, the entire audio signal is faded out. The assumption is that there is no wanted signal present. The noise gate can be used in situations for original audio recordings with noise that is so strong that it cannot be removed using the DeNoiser without leaving the wanted signal muffled and hollow. For certain types of instrument recordings it is also not a good idea to remove all of the "dirtiness" from the sound, e.g. distorted guitar. With the gate you can at least clean up the sound during the pauses.


The operation is similar to a classic analog gate. Here emphasis was placed on a fast and accurate response, where the typical artifacts from digital gates such as rough, "fluttering" sounds are avoided. The essential FX Gate continuously scans the signal and automatically selects optimal settings according to the current values.

### Gate parameter

**IN / OUT:** Here you can set the input and output level.

**sidechain int/ext:** As with the essential FX Compressor an external control input can be used for triggering.

Note: Sidechaining is not available MAGIX Audio & Music Lab Premium.

**soft knee:** Normally the Gate has a hard characteristic i. e. below the threshold this signal is cut hard, in addition it is transmitted in an unchanged form. "Hard Knee" deals with abrupt transitions. "Soft Knee", however, allows the signal to be modified by "Gating" in the pass so that transitions can be softer and less detectable. This is particularly recommendable for work with acoustic instruments such as drums where the signal levels can fluctuate a lot.

**hold:** This three-way switch controls how long the gate process waits after passing through the attack phase before subsequently passing through the release phase. The Hold function is useful for signals that need a rather short release time, but shouldn't be flattened out.

- **sc filter:** Operation is the same as the essential FX Compressor.
- **threshold:** This fader defines the threshold beneath which the gate should be applied.
- **range:** Here you can set the gating strength. When it is set fully to the right, the signal beneath the threshold is cut out completely. Subtly attenuating the signal can remove background noises or breathing from vocal tracks. Where necessary the process can be attenuated even further by switching on the soft knee mode.
- **attack:** Regulates the attack time from the closed gate to the point at which signal is let through again.
- **release:** Sets the time that the gate requires to go from a normal state to gating.

**Tweak**

- **Look Ahead:** This control sets how long ahead the gate previews the signal. The audio signal will be delayed by this time.

## Chorus/Flanger

This plug-in offers a simple way to make signals sound more interesting, "spacier", thicker, etc. by modulating or delaying the pitch - the classical domain of application is for guitars, Hammond organs, electric pianos, or synths.

Chorus and flanger are two closely related effects, which are combined into a single plug-in. They normally differentiate in terms of delay time, type of modulation, and degree of internal feedback.


## Chorus flanger parameters

**IN / OUT:** Here you can set the input and output level.

**mode:**

- **Stereo chorus:** Compared to mono chorus, two copies of the original are created, modulated against each other in pitch, and then fed accordingly to the set mix ratio to the left and right output channel.

- **Mono Flanger & Stereo Flanger** In contrast to the Chorus Effects the lower delay periods and a slightly changed modulation are processed.

**- Ensemble:** This creates a denser chorus, similar to Boss/Roland CE-1: Instead of two voices, six are generated. Two internal sine LFOs for de-tuning, whereby for both LFOs the second and third voice phase length by 120°. This results in a denser-sounding effect that is also less warped.

- **Rate:** This specifies the speed of the modulation. Lower rates provide slight hovering effects, and high speeds produce a wobbling, typically distorted "underwater" sound.
- **Depth:** This parameter specifies the depth of the modulation, i. e. the maximum deviation of the modulation and the resulting pitch bending.
- **phase:** This fader moves the right channel's oscillator phasing relative to the left, wave is put back to the right. So that the Tremolo effect drifts apart in the stereo field with ever increasing values. At 180° both oscillators work in reverse, therefore the stereo effect is at its strongest.
- **Feedback:** This parameter defines the portion of the delay that is sent back to the input. Feedback causes the effects of modulation to be more drastic and cutting. Nullification of the feedback is set at the middle of the fader. Set to the right, the feedback is fed to the input equi-phasal; to the left, the feedback

occurs. Both variants may sound very different depending on the signal, since they prefer different frequency ranges for dissonance.
- **Mix:** Regulates the mix ratio of the original signal and the delayed portion.

Tweak

- **Low Cut:** This knob sets the filter frequency of a high-pass filter. Signal components below this frequency will be filtered out.

## Reverb

This involves a completely algorithmic reverb plug-in. Selectable algorithms produce thick and extremely transparent reverberation for all kinds of signals. Processing is therefore "true stereo", i. e. the selected algorithm includes the original localization of instruments in a stereo signal in the calculation of reverberation so that the original image will not be distorted.

"Reverb" features algorithms that are set up for natural quality and transparency. A signal treated with this reverb retains power at correct dosage in the overall context of the audio without perception of a conspicuous effect. The CPU load is also relatively low.


### Reverb parameters

**IN / OUT:** Here you can set the input and output level.

**Mode:** Select the desired algorithm here. The following options are available:

- **Plate**: Simulates a reverb plate. Very dense reflection pattern, penetrating but musically open "metallic" appeal. Very well suited to Drums and Vintage/special effects.

- **Room**: Small room reverb. Middle to high signal diffusion, low to middle reverb time. Well suited for adding a "dry", natural sound to instrumental, spoken, or vocal recordings.

- **Hall**: Mid to large hall. Low to mid diffusion, later response than a "Room". Ideal for simulating concert halls and larger soundscapes.

**- Plate II Classic:** Classic stereo plate reverb. The design is based on an algorithm that was frequently used in the 80's. This mode produces the typical "Cloud reverb", which is not very transparent, a bit sticky. Plate II Classic is a good fit for lead instruments and vocals.

- **Size:** The size of the room being simulated or of the reverb plate. This setting has a direct effect on the distance of the reflection and indirectly on the resonance behavior: smaller rooms/reverb plates possess a larger inclination for natural resonance than larger ones.
- **Time:** Length of reverberation time (in seconds). This value refers to the RT60 time, i. e. how long the reverberation requires for the level to fall 60 dB.
- **Damping:** Real rooms and reverb plates dampen certain frequencies according to their construction. This is most often audible in upper frequencies. Using this parameter you can set the cut-off frequency, above which the reverb will be particularly strong. In some cases, the "Time" parameter should be adjusted, since sometimes the subjectively perceived reverberation time will also be affected.
- **Modulation:** For several critical signals, the reverberation may produce disturbing resonance. This can be scattered by temporal modulation of the delay signal used in an algorithm. Excessive application may create an effect similar to chorus in tonal signals. The liveliness of the reverberations will be emphasized with lower values. In this case, the modulation applies a random pattern, the intensity of which may be visualized via the "MOD" display.
- **Mix:** This regulates the ratio of the unprocessed signal to the portion including reverberation.

### Tweak

- **Pre-delay:** This knob sets time in milliseconds between the direct signal and the appearance of early reflections. The late delay comes only following this time span.
- **Diffusion:** This knob simulates scatter on irregular walls in percent. Increasing values make the reverb sound denser.
- **Low Cut:** This knob sets the filter frequency of a high-pass filter. Signal components below this frequency will be filtered out.

# Tube Stage

Preamplifier are still used today, if the signal needs a warm, flattering sound. This is often not only about saturation or distortion artifacts, but rather more about getting a compact and more vivid sound picture. A tube stage makes things a little more dynamic, this is not due to the tubes itself, but rather the entire circuit, which interacts with the tube in a way that amplification levels don't do on a semiconducter basis. As with guitar amplifiers, the complexity increases with the number of stages.


Optionally inside the essentialFX_TubeStage plug-in, either one or two tube stages work.

Tube circuits, especially if they work in the border area and saturate/distort the signal with the **gain** control, react very sensitively to the input signal's spectral composition.

Two independent filter circuits accompany this plug-in's tube stage (s). The first (**pre-eq**) acts directly on the input signal. With this you can determine which frequency range should be principally edited. The second filter circuit (**post-eq**) is located behind the tube stage (or after the second) and determines the tonal balance of the output signal.

This enables you to create various sound nuances. For example, you can accentuate the high frequency range on the input side, for example, to create Exciter-/Enhancer effects, so that the signal appears more "tangible". To avoid overemphasizing this amplified range, a reverse process with the post-EQ can be subsequently made.

Both of the available tube stages can be globally switched in the A or AB operation. In the A-circuit the entire signal runs through, as is common with simple tube preamps, through a tube. However, since these can only edit a half-wave, the working point of the circuit is set approximately in the middle of the reference line. However neither halve will ever be treated equally, so that positive displacements are amplified differently to negative displacements. (so-called Class A operation). Thereby odd and even harmonics are created. The typical 'warm' tube sound to a large extent dates back to the characteristic overtone spectrum.

For each half-wave in class-AB operation, a separate tube is used, so that in principle there is a symmetrical reinforcement. The signal contains predominantly uneven harmonics. The sound image appears thinner but more transparent and solid. This is comparable to the sound, that tape machines create when they are slightly overmodulated.

## Tube Stage Parameter

**IN / OUT:** Here you can set the input and output level.

**stages:** In signal section **1** the signal only passes through the tube stages. In setting **2**, two stages are cascaded. This divides up the gain so that both stages are driven less 'hotly'. This increases the signal complexity and any tube artifacts.

**class A / AB** Optionally, a simple gain stage with asymmetric reference line (Class A) or symmetric AB mode. In the A-mode odd and even harmonics emerge, the sound image is similar to a 'warm' tube guitar amp. In the AB section only uneven harmonics are created. The sound image is somewhat 'colder', but more transparent with complex materials and can also be driven louder.

**oversampling:** If this switch is active, the virtual tube stages are controlled with one-to four-times the project sampling rate, i.e with an internal sampling rate of 176 - 192 kHz.

- **pre-eq:** This knob (called "Tilt" or "Level" on some devices) controls the effect of a soft, passive 6 dB filter for prefiltering signals before the (first) tube stage. If turned to the left the bass is emphasized and the highs are dampened. To the right the inverted filtering happens. With this filter the signal can be correspondingly processed before "Warming up" by selecting the prominent of desired section. Compared to a standard EQ the effect of this filter is more subtle, but it does possess a high 'musicality', by the internal circuitry and also because of the slightly different phase response.
- **gain:** This set the entire amplification factor. If two two tube stages are selected using 'stages', the available Gain is divided by two and is distributed equally over both stages.
- **post-eq:** Operation and function is like pre-eq, however, this filter is placed behind the (second) stage tube.

### Tweak

- **Tube Bias:** With this parameter you can control "Bias", or electric current voltage inside a tube.

# Energizer (plug-in)

The Energizer belongs to the so-called psycho-acoustic processors genre. These devices are used in the studio, acoustic irradiation, and restoration areas in order to enhance the sound.

An effect like this usually can only be partially achieved with standard methods like equalizing since an equalizer can only compile the frequency parts that are already in the signal. Additionally an excessive increase in the signal (of the highs) increases the chances of hissing.

With bass the problem often arises that for each device in an analog processing chain (for example, tapedeck, mixing desk, pre-amp) slightly delays the signal compared to the remaining spectrum. This type of phase lag cannot be restored with an EQ and increasing the depth often results in a washed-out and feebler sound.

Psycho-acoustic devices are based on our hearing's ability to perceive attributes like "freshness", "liveliness" and "naturalness" in a recording as a result of various parameters. One of them is the harmony spectrum. The mid and high frequency ranges of a loudly played instrument are richer than those of a quietly played instrument. The "Exciter" tries to imitate this property with artificial harmonies. However, this type of method cannot function statically as the noise and low-level signals will be influenced otherwise.

The order in which individual frequencies are heard is just as important for lively sound.

The psycho-acoustic method used in the Energizer is based on a combination of frequency-dependent phase correction, additional harmonics creation and recognition of so-called transients (short signal peaks).

The Energizer is subdivided into a bass and middle/high area. In both of these areas, the audio can be enhanced independently of one another.

## The parameters of the Energizer


The available effect presets cover typical areas of usage and are already set up as presets in order to, for example, format a CD for playback on the car radio, use as a soundtrack for your home entertainment system or for restoring distorted frequency response curves of old records. The Energizer can have a drastic effect on the sound even if only small changes are made to the parameters. So that you know which faders to change to get the audio results you want, the following section describes the available faders in detail.

**Low tune:** Here you can tune the bass processor to a specific input frequency (between 50 and 150 Hz). This is the preferred frequency at which, for example, a kick drum or an acoustic/electric bass is played. The phase position of the bass range can be influenced according to the set frequency, resulting in deeper sounds sounding more "succinct" and "broader".

**Low attack:** Using the transient recognition fader the attack behavior of the tuned range can be increased (fader to the right) or decreased (to the left). You can use this to create a "hard" or "dynamic" bass foundation or have a range sound "legato" or "soft.

**Low mix:** Here the processed bass signal is mixed with the unprocessed input signal. Please note that material that is already highly modulated may become overmodulated/distorted. For strong increases in the bass you should, if required, reduce the source material in order to have enough reserves. It is also recommended that you use the Audio Cleaning Lab Limiter as the next step.

**High tune:** This fader specifies the input frequency of the high tone circuitry (between 1 and 10 kHz). A part of the input signal is filtered and a phase lag is created depending on its frequencies. Simultaneously, the dynamic harmony is enhanced. When turned to the left you can influence the mids and highs of the signal so that, for example, the "articulation" of speech and instruments can be edited. The further up the frequency you go, the more the harmonies or bright sounds, like drum cymbals, are registered. You can use this to add "shine" or "silkiness" to your recordings.

**High stereo width:** Use this fader to specify whether the signal that's added to the high tone processor should be edited in mono, stereo or as a broadened signal if at all. This way you can specify whether the range should be compiled from the stereo mids of a recording or from the parts of the pages (for example, room information)

**High mix:** Here you can specify the amount of data from the high-frequency processor that should be mixed into the original sound.

Tip: At the beginning when using psycho-acoustic devices there is a danger of overkill. We recommend regularly toggling from unedited signals to processed ones. In order to make subtle changes in the sound it is often enough that the change be just about audible or that when switching off the effect, you get the feeling that there's something missing. If the effect sounds like it's in the foreground, this usually means that it's exaggerated too much.

# Analogue Modelling Suite: AM-Track SE


AM-Track SE is a purely analog compressor simulation. The tape simulation contained in the full version (Analog Modeling Suite AM-Track http://pro.magix.com/de/audio-plugins/analogue-modelling-suite.175.html) is not included. This is used primarily for so-called "tracking", i.e. editing individual channel strips or subgroup signals. The compression takes place in the "vintage" setting, whereas an additional "vca" setting is available in the full version. The plug-in recognizes the number of incoming signals and, if necessary, edits the signal in mono.

**Am-Track SE limitations** compared to the full version:

- No tape simulation
- No "VCA" mode in the compressor, only "Vintage" operation can be implemented along with the presets.
- Some expert compression settings (view page 107) are integrated in the interface, parameters: "ahead" (pre-delay) and "adapt release" (switchable release automation) are missing.

(Release automation is always activated in the SE version which corresponds to the set value of the mid position of the 'capacity' controller.)

Below, the full version of AM-Track is explained and its features compared to "normal" software compressors and the available parameters.

## Compressor Section

Two completely different compressors work in AM-Track, each with their own independent control and sound methods. You may be wondering why we mention sound when talking about a compressor, since compressors merely relate to control actions. This isn't as simple as the idea of "making loud quiet".

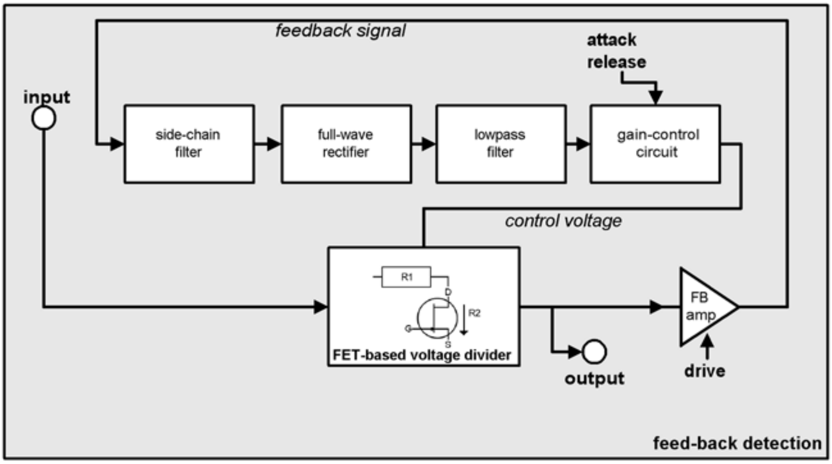
Various designs, algorithms, and topologies for solving the actual problem (dynamic reduction), which all have their own unique character, have come from the history of analog and digital signal processing. For example, pre-filtering in the detector circle and the type of detection have a large influence on the audio results. Plenty of hardware compressors have the same established VCAs (voltage controlled amplifier), but they all sound different and influence a signal, an entire production (or even a genre) with their "signature sound". We intend to provide you with acoustic variation via these dynamic tools in the digital world.

The two operating modes of the AM-Track may be selected using the switch "vca/vintage":

## VINTAGE mode


This mode appears as a preset on start-up of the AM-track. It has fewer parameters than the VCA mode and has audibly less of a "surgical" quality, but more of a trimmed sound character.

"Vintage" mode simulates a circuit design from the time when VCAs were not yet or could not be fully implemented. Instead, a FET (field effect transistor) was often used as a controllable resistor. Together with constant resistance at the circuit's gate, this forms a so-called voltage splitter, which is to say  it forms a resistance change at the FET (caused by a change in voltage at its gate) which results in a damping of the input signal. A very simple detector circuit is used to activate the FET which obtains its signal from the output of the compressor (behind the whole control circuit). For older designs, this feedback loop provides a stabilization of the work parameters and is one of the decisive factors for the often quoted soft and musical compression of exponents of this design, e.g. the Urei 1176 or 1178. The control circuit sees the layout of its previous work and oscillates to the signal.




The disadvantage is that the set time parameters for attack and release depend somewhat on the program. In some cases,  it's actually advantageous for vocals, bass, or even drums (e.g. subgroup, ambience, mics). You should rely completely on your ear for this.

Because of the feedback topology, the maximum gain reduction is usually lower than VCA devices with forwards detection, usually 20 dB. This way, there is almost always a level matching amplifier in the feedback loop. The "drive" controller of the AM-Track regulates the feedback amplification here. This can be so high that the detector may become saturated by a loud input signal, resulting in signal peaks being swallowed up. Simultaneously, the setting becomes more intense as quieter signals also start reaching the threshold. You can creatively implement this according to the situation to create complex signal compression which doesn't much sound like dynamic compression due to the transients that slip through and release at high "drive" levels.

The stated release of the signal, technically known as a ratio reduction, is also caused by the centerpiece of the circuitry: the FET. Level reduction works entirely as a function of its characteristic curve, resulting from the non-linear behavior of this element. The FET virtually comprises part of the input resistance of the compressor circuit. As a result, the input/output response curve does not create a plateau when "drive" is high, which would be the case for a reference line featuring a high ratio or even limiting. A saturated FET may no longer complete the job it was marked out to do, i.e. to keep its output at low Ohmic values. Once again, signal peaks pass through the entire circuit unaffected, but the average level could be severely compressed. From a technical point of view, the control process appears incomplete, but sounds pleasantly open and airy depending on its application.

The entire detection is dependent on the spectral balance in the virtual AM-track circuit, the highs are automatically compressed less so that even extreme settings sound less flat and more lively.

It's the same story with deep bass. On closer listening, you'll find that with strong compression the signal still retains its power which would otherwise be lost if the envelope were to follow shortly afterwards.

"Vintage" mode has another feature to offer: At the output of the compressor in the signal path, an emulation of a transformer-coupled matching-level amplifier can be found. This contributes to some subtle, non-linear distortions at high levels, but is very much frequency-dependent.

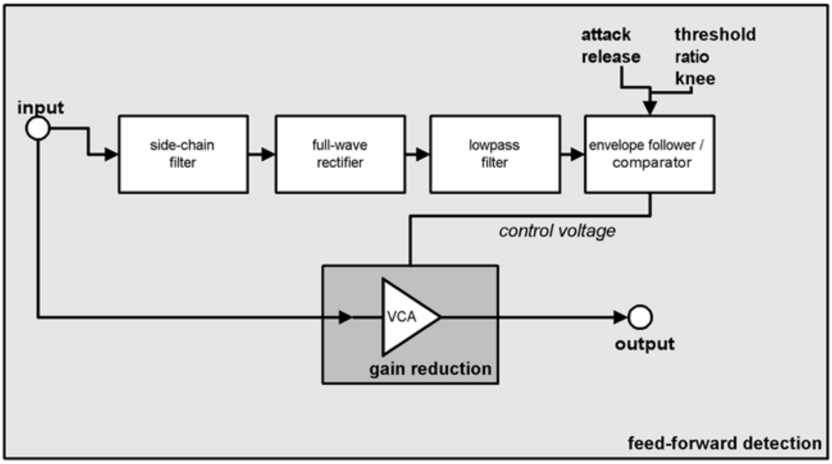
## VCA Mode


In VCA mode, the parameter selection and the circuitry design correspond to a modern compressor with a VCA element as the control circuit and a forward automatic gain control in the detector section ("feed-forward design", i.e. the controlling signal for level reduction is taken from the input signal).

The typical basic sound for this category is accurate, largely neutral and, in relation to the adjustable parameters, easily predictable.

In VCA mode, the control signal is accessed at the input where it firstly executes a controllable low-cut (which may be set up via "detector hp freq" in "Expert" mode). The filter makes sure that deep-frequency signals have less influence on the adjustment settings; this is a popular trick for more power, e.g., when using drums in a sub-group.

The filtered signal then arrives at the detector. With the forward gain control, previously set parameters apply fully and affect the adjustment settings immediately.




In contrast to this, there is a "feedback" method which provides a second compressor mode.

## Compression Parameters

### Vintage Mode

In this mode, you can intuitively (by ear) use the dynamic editing features with just three knobs. Do whatever you want, but keep in mind: less is sometimes more...

- **Drive**: You can use the "drive" potentiometer to control the amplification factor in the feedback loop, i.e. the signal strength which the detection circuit calculates. Furthermore, the internal "ratio" changes within a limit, the more "drive" there is, the higher the compression ratio.
- **Attack** and **release**: The same basic conditions as VCA mode apply here. However, not only do you change the actual control response time after detection, but the "temporal window" in the detector must be adjusted as well. Additionally, the feedback method does cause a certain amount of unpredictability. You should expect less control over the device in this mode, but more leniency on its part.

### VCA Mode

The regular set of parameters of dynamic compressors is available in this mode:

- **Threshold:** The threshold above which dynamic reduction begins.
- Check the threshold display if necessary (**thr**): If the input signal reaches the set threshold, the blue dash will move around the arrow symbols. If this dash moves upwards, the threshold is below the average level and compression is active. Inversely, if the dash moves below the marking, the input signal becomes too quiet to be able to reach the threshold; compression will be applied.
- **Ratio**: This ratio (1:n) specifies by which factor the signal should be reduced once the threshold has been reached. For example, if the threshold is set to -20 dB and the ratio to 1:4, an input signal of -10 dB will only be amplified by 2.5 dB (10 dB : 4 = 2.5 dB).
- **Attack**: This is the response time, e.g. how long the project takes to execute the required level reduction. Short attack times intercept level peaks, and longer ones let them through unimpeded (compression only starts past this value).
- **Release**: This is the time allocated to the circuit to reach the normal amplification factor.

**Note on attack & release**: In general, short attacks are used for moderate compression and making the transient response softer; longer times retain the "bite" of a specific instrument at larger compression rates or make the sound a bit snappier. With more difficult sources, like a very dynamic vocal track (ballads), for example, you can use a longer attack so that the project runs more smoothly and quietly; the release time may be trimmed audibly to match the pauses or the song speed. Shorter release time may be used for modern, aggressive "close up" vocals, e.g. when breathing sounds are an important stylistic device and the voice should sound very full and compact.

- **Knee**: Use this parameter to specify the shape of the characteristic around the threshold. A "hard knee" means that the transition of 1:1 amplification for level reduction occurs abruptly; a "soft knee" on the other hand starts much lower than the threshold and moves the characteristic softly into the reduction. A "hard" setting is useful for effect-filled, acoustic compression, e.g. individual drum tracks. A softer setting is useful for complex and sensitive sources like guitars, pianos, or vocals. The more complex the signal, the easier it will be to notice a difference. For less sensitive sources, this parameter is usually less important. Note that for "soft knee" settings, the "threshold" value will need to be re-adjusted, since the compression starts at a much lower level.

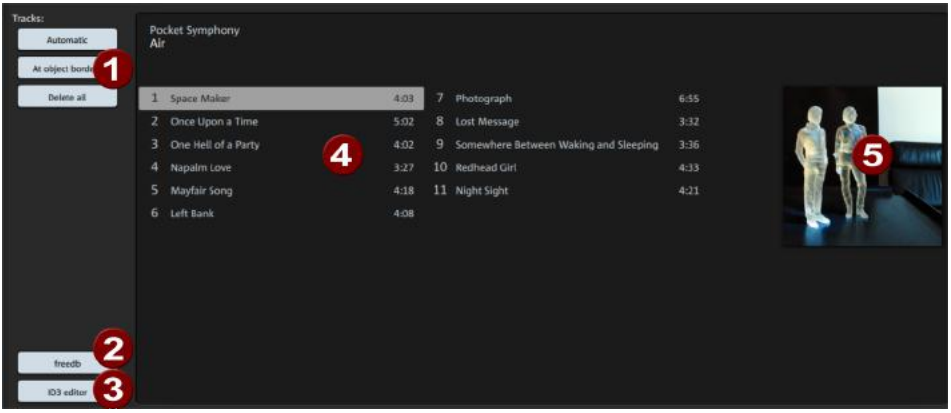
## Compression Expert Settings

Of course, you can efficiently compress a lot of data with AM-Track without having to press the "**Expert**" button or try out additional options. However, we have added a few "handy" parameters behind the front panel. This applies equally to both compression modes.


- **Look ahead**: AM-Track is always ahead of the signal. You can specify how many milliseconds you want to "look ahead". The audio signal path is delayed according to the signal route so that the detection circuit is fed first with the input signal (so-called "look-ahead delay"). You can increase the attack time and still avoid fast peaks. The latency compensation in the host program ensures that other tracks in the project are adjusted and that no time delay occurs. For percussive signals, you can even set the delay all the way to "0".
- **Detector hp filter**: This high-cut filter is positioned before the two compressors' detection circuit. You can use it to specifically exclude basses and mids from these rules. Complex signals with bass and hi information like a subgroup or complete mixdown produce fewer "pumping" artifacts. This is because low-frequency signals feature the most power and therefore always trigger regulation and modulate other frequency ranges in the volume
- **Auto makeup gain**: Normally, you have to continuously adjust level reduction to generate "compression" at the same maximum level. This is done by activating auto makeup gain. The volume difference expected from the set working parameters is determined and applied as an output factor after master regulation. If you prefer to adjust the "classic" level reduction and amplification manually, you can deactivate this function.
- **Adaptive release**: This is "semi-automatic", i.e. you can roughly adjust the release time, and AM-Track reduces it according to the current signal power from "a little (1%)" to "considerably slower (100%)". In "Vintage" mode, this regulation method is particularly intense, since it affects the feedback loop process. For instance, if you are editing vocal tracks or dense, complex material, it can sound "calmer" or more "musical" if adaptive release is activated.
- **Capacity**: Adjusting the "capacity" controller sets the time response of the "adaptive release". The greater the capacity, the more sluggish the release adjustment. You can therefore influence larger parts of the compensation response. For instance, if you want to use vocals that have been "moved forward", you should use a short release time (maybe 80-100 ms) and a greater value for semi-automatic (e.g. 80). Vice versa, you can reduce automatic feed by switching the relation (smaller capacity, generally greater release time).
- **Comp mix**: Parallel compression is a popular "studio trick", particularly with complex material. Adding the original signal retains the transients and spectral balance of the source. You can add compression by turning the mix controller. A mixed signal is particularly discreet, more transparent, and less "squishy" with vocals, whereby the compressed portion usually has a higher level reduction than without adding the original.

# Tracks

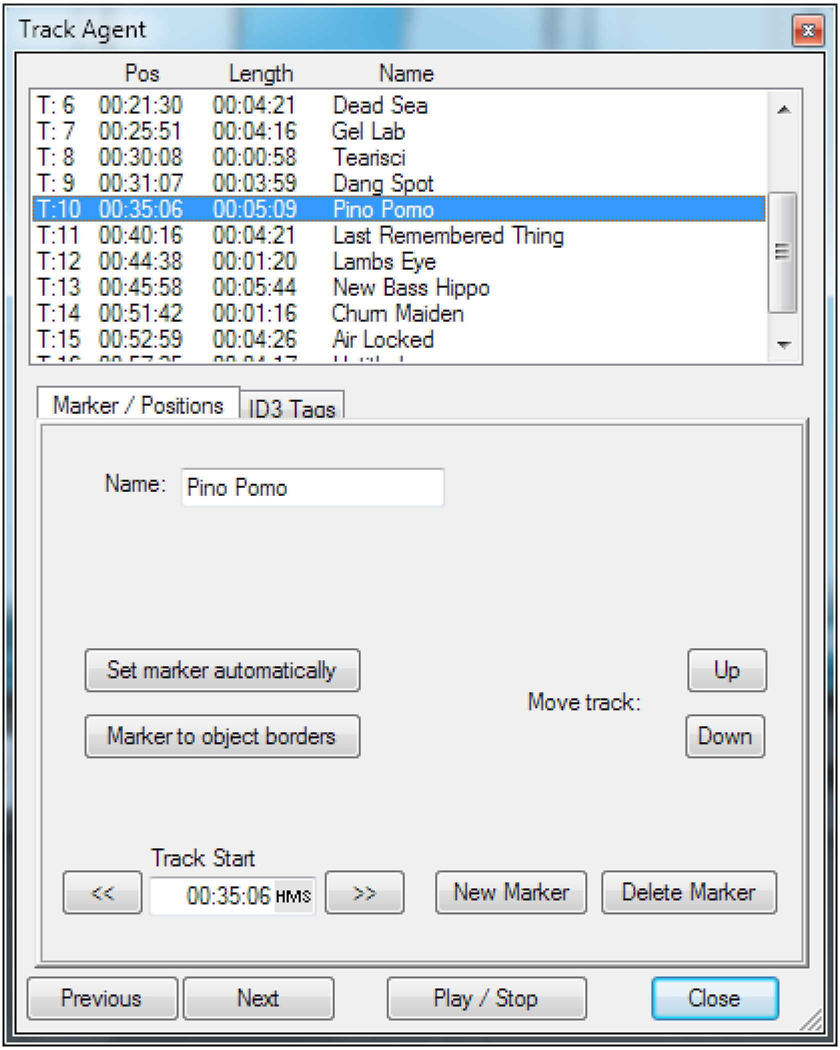
In the "Tracks" view you can check and edit the position and order of the tracks before export.




1. The "Tracks" buttons offer quick access to Automatic track marker recognition, setting track markers on all object edges or also deleting all markers.
2. **freedb**: Request for track information from the freedb Online CD Database (view page 143)
3. **ID3 Editor:** Opens the ID3 Editor (view page 110) for further editing of the track markers, track names, artists and other metadata (ID3 tags)
4. **Track list**: Here all of the tracks are listed by name and the name of the project (most often the album title) and artist is displayed above. To edit the names directly in the list, simply double-click on it.
5. **Add album cover:** Click on the cover area to add the selected cover to an album. If an album and artist have already been defined, an appropriate cover can be downloaded automatically from the Internet. But you can also load a file from the hard drive.

# Track Agent

In this dialog, all the CD tracks in the current project are displayed in a list. Every track can be given a name which is then shown in the master track.




The track list and the buttons underneath it are always visible. You can change the dialog view with the "Marker/Positions" and "ID3 Tags" tabs.

By choosing all tracks on the list, the corresponding objects are selected and the playback position placed at the track marker. All of the objects which belong to a track will also be selected and the playback position placed at the track marker.

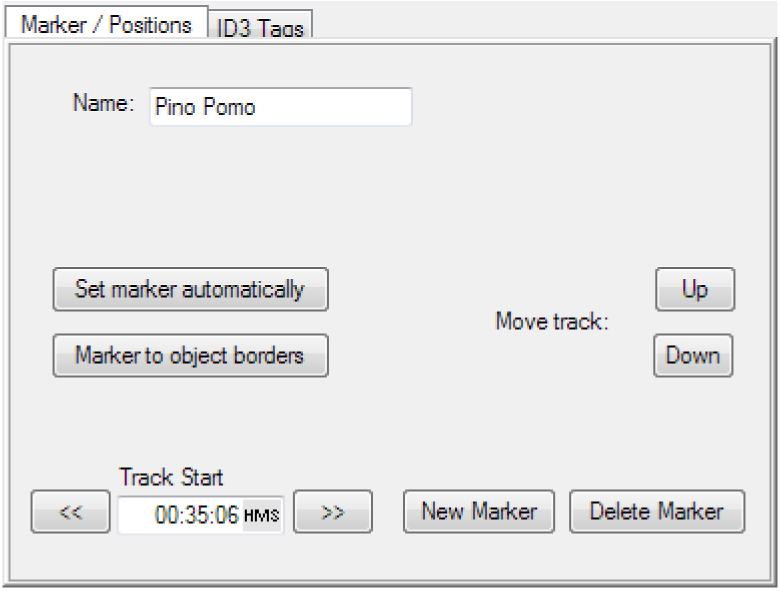
**Previous/Next**: Selects the previous or next track.

**Play/Stop:** Plays or stops the selected track.

**Close:** Closes the dialog and applies the changes that were made.

Keyboard shortcut: |

# Markers/Positions




You can edit the track markers in your project in the "Marker/Position" tab.

**Name:** The name of the track.

**Set markers automatically:** MAGIX Audio Cleaning Lab analyzes the audio material and sets the track markers automatically.

**Markers on object edges:** Automatically sets track markers on all object edges. This option only makes sense if you have already cut (view page 57) a recording into separate tracks.

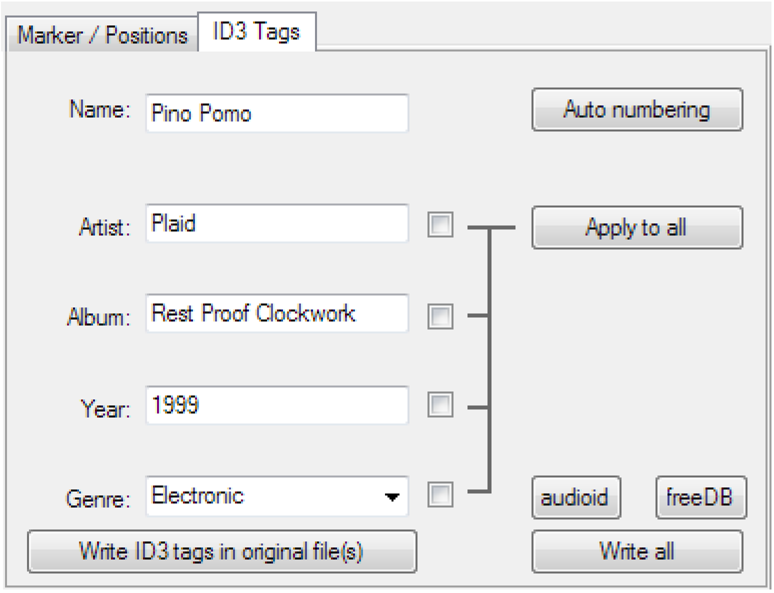
**Track Start:** Here you can set the exact starting point of the track. The two arrows change the position of the track marker. This means that the track remains in the same position and the same order but starts earlier or later.

**Move track:** The position on the CD of the track that is marked by the track marker can be changed by using the two buttons.

**New marker:** Sets a track marker at the current playback position.

**Delete marker:** Removes the selected track marker.

# ID3 Tags




In the "ID3 Tags" tab you can enter information about the artist, album, year and genre. This info will be included as ID3 tags when the MP3 is exported. The data is used by database and search functions in programs like MAGIX Music Manager.

**Track name/artist/album/year/genre:** More details about the tracks (ID3 tags).

**Auto-numbering:** Automatically numbers the tracks.

**Apply to all:** applies the respective entry to all tracks in the project. If you want to apply an entry simply tick the small box behind it. This way you can standardize information which is imported from different files, e.g. the spelling of an artist's name etc.

**freedb** loads the data for a complete CD or LP from a public database. For more about this see freedb (view page 143).

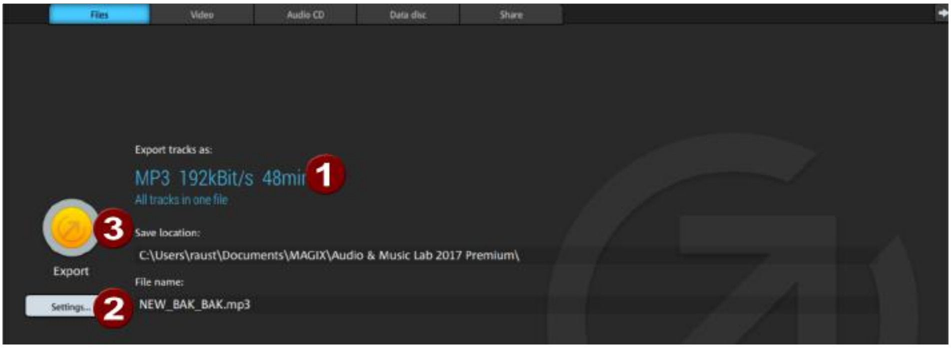
**Write ID3 tags in original file** The ID3 tags will be added to the MP3 file you loaded. "Write all" updates all the files in the project.

# Export


In the export section, you can either save the project as audio file(s) or burn it directly to an audio or data CD or shared on the Internet.

## Files

In the "Files" view you can save the tracks from the current project as audio files.




1. The file format, the total length of the project and the storage location are displayed here.
2. **Settings...** The default setting is MP3 files with a bit rate of 192 kbps saved in the project folder (view page 151). To select a different file format or storage location, open the advanced settings (view page 113). The changes made there will be applied to all subsequent exports.
3. **Export:** Starts the export process.

## Settings

In this dialog you can adjust the settings for file export, e.g. to change the save location or the file format.

In the upper part of the dialog you'll find a file explorer which you can use to set a new name or target folder for the audio file. Under „**Last storage location**" you can choose from a list of save locations that are already being used.

**File type:** Here you can set the format for the exported file. In the "Format settings..." you can access the configuration options for the format.

**Export:** Closes the settings dialog and starts the export process. The changes to the settings will be applied to any subsequent exports.

## Format

**Wave:** The audio material is exported as a standard Wave file. This is the conventional format for further use on Windows PCs. These files are not compressed and retain their full sound quality.

**FLAC**FLAC is the abbreviation for "Free Lossless Audio Codec". This is a freely savable format that can be used to compress your audio data to 50% of their original size. Unlike lossy compression methods like MP3 or OGG, the full sound quality is kept intact with FLAC.

**MP3**: MAGIX Audio Cleaning Lab contains a high-quality and extremely fast MP3 encoder. This can be used to save complete LPs including cleaning effects as MP3 files. You can also make an MP3 CD, and for that you can use the function "Burn data CD/DVD (view page 120)".

For good quality, we recommend a setting of at least 192 kbps. Sound quality will hardly be affected despite the compression. If you have memory to spare, full CD quality can be retained at 320 kbps – at approximately 1/3 of the original memory. This is ideal for building up a large high-quality music archive on your hard drive.

**AAC:** This is a modern competitor format to MP3, which is primarily used for portable music players (iPod, etc.).

> **Note**: For exporting in MP3 and AAC formats, you may need to activate your MP3 encoder. Simply go to "Help" >Activate additional functions.

**OGG**OGG Vorbis files have all of the important characteristics of MP3 files, except that they do not require any kind of licensing for their codecs. – They can be freely decoded and encoded. Not all portable devices support this format.

**AIFF:** The audio material is exported as an AIFF file. This is the most commonly used audio format for Apple™ computers.

**Windows Media**: Exports the arrangement as a WMA format file (Windows Media Audio).

## Format settings

**Output format:** Here you can set the output bitrate. The bit rate is the data stream during playback of audio data. It is given in kilobits per second (kbit/s or kbps) and also determined the file size. An MP3 file that is 3 minutes long and has a constant bit rate of 128 kbit/s is ca. 2.8 MB in size.

Common bit rates for music are 192 kBit for good quality, 256 kBit and more for excellent quality. For Internet streaming and speech recording (in mono), 128 kBit are enough.

**Encoder quality:** The included MP3 encoder can be operated in three "Gears": An especially quick ("Fast"), an especially powerful for high sound quality ("Highest"), which however requires more time, and a compromise between the two.

**Format:** Here you can set whether your MP3 file is exported in Stereo or Mono format.

**Variable bitrate:** "Use VBR" adjusts the bitrate of the audio material, which means that a lower bitrate will be used during quieter parts. Therefore, VBR files are smaller than files of comparable quality without VBR. Instead of a constant bitrate there is a quality setting. Not all playback programs can process VBR correctly, some will result in problems during title length display or when rewinding.

**ID3 editor:** Opens a dialog, where you can set ID3 meta data for files to be exported.

## Options

**Each track in its own file with names:** Exports each track (i.e. the area between one track marker and the next) as a separate audio file. In addition to the audio files, a playlist file (.m3u) will also be created. This file includes the names of all the audio files in the correct order. It also contains the name of the project; the list field allows different naming schemes for the audio files.

| Scheme | Example: |
|---|---|
| (File name)_(Track number) | CD_1.wav, CD_2.wav ,CD_3.wav |
| (File name)_(Track number) | 1 CD.wav, 2 CD.wav, 3 CD.wav |
| Track name: | AAA.wav, BBB.wav, CCC.wav |
| (Track number) (Track name) | 1 AAA.wav, 2 BBB.wav, 3 CCC.wav |

*File names for a project "CD.vip" with the tracks AAA, BBB, CCC*

**All tracks in one file:** Saves the project in one audio file with the same name as the project.

**File for track at playback position only:** Saves the track that is currently placed at the playback position, i.e. the project area between the last track marker before the position line and the following track marker. This way you can cut out parts of an audio file and save them separately by simply placing markers.

**Play back exported file(s):** After export the media player is opened to test the results of the export.

Keyboard shortcut: K

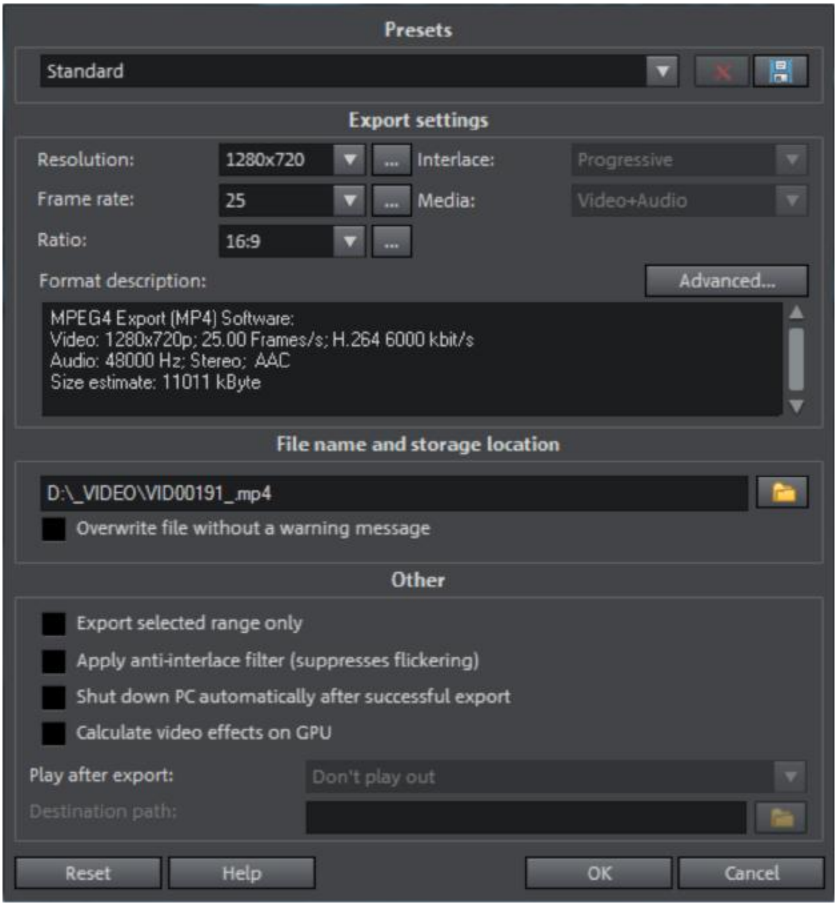
# Video

This option allows you to export projects as video files.

Note: The video export feature is not available MAGIX Audio Cleaning Lab.


1. Change video format here. If the audio track of a video that was loaded into MAGIX Audio Cleaning Lab has been edited, the format of the original will be discarded, in which case other properties of the original video will be preserved where possible.
2. **Settings...**: A settings dialog (view page 117) for the selected video format will open, this dialog allows you to change the properties of the exported video.
3. Once the export process has begun you will not be able to make any changes to settings.

# Advanced settings for video formats




*The dialog will look different, depending on which video format is selected.*

Here you can adjust the settings for the exported video precisely. The options available vary according to the selected format.

**Presets:** Under "Presets", you'll find typical settings for the selected format for the most important applications.

  Click "**Save**" to save your own settings and "**Delete**" to remove them from the list.

**Export settings:** You can specify general export parameters like resolution, aspect ratio and frame rate. Select the most frequently used values from the list fields; to set your own values click on the "**...**" The "**Advanced**" button reveals extra settings for the selected video format. (see below) **Format description** provides a summary of the most important core values of the exported video file and estimates the size of the finished file.

In the field below you can enter the file name for your exported file.

Use the folder symbol to select the folder into which you want to export. The dialog will remember the export path for future exports.

The option "**Overwrite file without confirmation**" option allows you to execute multiple exports in the same file.

Other

**Anti-interlace filter:** This option should only be activated for playback on a TV screen, since it is intended to reduce line flickering.

**Shut down PC after successful export:** Use this option to have the computer switch off automatically after lengthy export processes.

## Video as MPEG video...

MPEG stands for "Motion Picture Experts Group" and is a high-performance compression format for audio and video files.

Details on the settings of the MPEG encoder can be found in the program's Help file in the "MPEG Encoder Settings" appendix.

## Video as DV-AVI...

This option exports the video as a DV encoded AVI. The arrangement can be easily transferred to the digicam using a Firewire interface.

The dialog will provide further information on all available options. You can access it via **"Advanced"** in the export dialog. You will also be asked for which video standard you want to export the DV data - PAL (Europe) or NTSC (USA).

## Video as Quicktime Movie...

Exports the project in QuickTime Movie format. This format ensures the best possible compatibility on Apple computers.

Settings for video size, frame rate, and codec settings may also be specified here. The export dialog does not enable you to add comments, etc. to videos.

**Tip:**For Quicktime files (*.mov), the Quicktime library must be installed.

## Video as MAGIX video

Exports the project in MAGIX video format.

This format is used for video recording by MAGIX video software and is optimized for digitally editing high-quality video material.

## Video as MPEG-4 video

MPEG-4 is the most advanced video format available at the moment. Unlike other formats, this provides very high-quality pictures at the same file size.

## Windows Media Export

This is a universal audio/video format from Microsoft. Accordingly, the settings options in the "**Advanced**" dialog are very complex.

### Manual configuration

**Audio/video codec**: Various codecs corresponding to the various Windows Media versions (7, 8, 9) are possible. Should compatibility problems arise on playback, try an older codec with a lower version number.

**Bit rate mode:** Constant and variable bit rates are possible, most devices and streaming applications require constant bit rates. For VBR two pass modes the movie is compressed in two passes in order to optimally use the bandwidth for highly-compressed movies for the Internet.

**Bit rate/quality/audio format:** The bit rate is decisive in defining the display/audio quality. The higher this is set, the better the videos will look and the larger the files and the required encoding time will be. For variable bit rates, the bit rate is adapted dynamically to the requirements of the corresponding picture or sound material. Either the quality value may be set between 1-100 or, for two-pass encoding, an average or maximum bit rate. For audio, the bit rate is set additionally by the audio format.

**Import from system profile (export type):** For the most common methods (other than playback on mobile devices, with which you should use the supplied presets), like, for example, Internet streaming, Microsoft provides diverse system profiles to choose from. If you have the Windows Media Encoder 9 installed, which is available from Microsoft as a free download, you can edit the profiles or create your own. These can be loaded by pressing the "**Import from profile file**" button.

"**Clip info**" enables the title and artist names, copyright details, and descriptions to be entered.

# Audio CD

Using the "Audio CD" button in the export section you can burn your projects as exact copies on an audio CD.


1. **Capacity indicator:** Standard blank CDs can hold either 74 or 80 minutes of audio.
2. The **CD Title** is displayed here and can be changed if necessary. This CD title is also displayed on most CD or media players when the CD is inserted.
3. **Burn CD:** Starts the burning process. In the background a large wave file of the project is exported and can be burned to CD.
4. **Print CD cover:** Opens MAGIX Print Studio, an included printing program which you can use to print a CD cover. Track information is automatically transferred as a spreadsheet into the print project in MAGIX Print Studio.

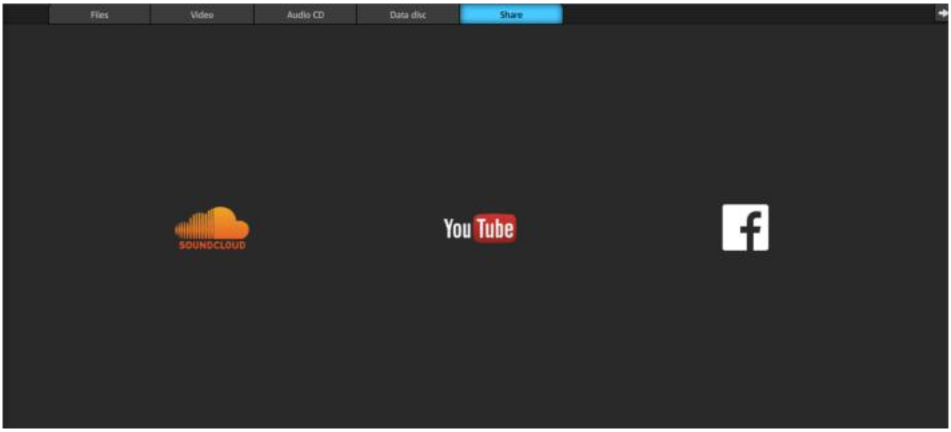
# Data disc

This option allows the project to be burned onto a data CD or DVD in MP3, OGG VORBIS or WMA format.

1. **MP3/WMA/OGG:** Here you can select the format. In most cases you can leave this set to MP3.
2. **Settings...:** Here you can change the format settings (view page 114) of each encoder.
3. Display on project length and bit rate
4. **Burn data** starts the burning process.
5. **Print CD cover:** Opens MAGIX Print Studio, an included printing program which you can use to print a CD cover. Track information is automatically transferred as a spreadsheet into the print project in MAGIX Print Studio.

# Share

In this view you can share your project on the Internet on services such as SoundCloud, YouTube and Facebook.




Simply click on the icon for the service you want to use.

## SoundCloud

SoundCloud is an exciting Internet community for musicians, record companies and anyone else who wants to distribute music and communicate via songs. After registering with SoundCloud, you can upload songs directly from MAGIX Audio Cleaning Lab to your personal profile. When this is done you can add your songs to your own SoundCloud player and integrate it into your website or share it on social networks such as Twitter or Facebook.

## YouTube

The command opens a dialog where you can set the name of the project for YouTube (default is the same as in MAGIX Audio Cleaning Lab), description, keywords (so-called "tags"), and the category for the video. After submitting this data by pressing "OK", the project is exported and uploaded to YouTube®. This is why you have to enter your YouTube® account access data (username, etc.) to connect to YouTube®. If you don't have an account open your browser and go to YouTube® to sign up for an account first.

If uploading was successful, your browser will open the info page for the video you just uploaded to YouTube® to double check your description and tags. If everything looks correct, just leave this page and the new video is already listed under your videos. YouTube® takes some time to process the video for online presentation, but after this period you and every web user around the world will be able to watch it.

## Facebook

The command opens a dialog where the project name (displayed according to the settings in MAGIX Audio Cleaning Lab), a description and search words (tags) can be entered. After confirming this data with "OK", the project will be exported as a video and uploaded to Facebook. For connecting and transmitting the video file, you must log in with your Facebook username and password. If you aren't registered on Facebook, first open your browser and create a Facebook account .

After a successful upload, your browser will open to show you your video's info page so you can check the entered data once again. If everything is as you want it, you can leave the page and the new video will now appear in the list of your own videos.

# File Menu

## New Project

Using this option you can set up a new MAGIX Audio Cleaning Lab project (view page 54).

Keyboard shortcut E

## Load project

Using this option you can load previously saved projects (view page 54).

Keyboard shortcut: O

## Save Project

The current project (view page 54) is stored under its given name. If there is no name chosen, the program opens a file requester, where the path and name can be determined.

Keyboard shortcut: S

## Save project as

It opens a file requester where you can determine the path and the name of the project (view page 54), under which it will be stored.

Keyboard shortcut: Shift + S

## Burn project backup onto CD/DVD / Burn data CD/DVD

In addition to the integrated audio CD burner routines, MAGIX Audio Cleaning Lab contains the CD burning program MAGIX Speed burnR. It can help you burn data CDs/DVDs, too.

**Burn project backup on CD/DVD:** The current project (including all audio files) can be burned directly to CD or DVD as a backup directly from the "File" menu in MAGIX Audio Cleaning Lab. Even extremely large files do not present a problem, since they are automatically distributed across several dics. The first of the backup discs contains a "Restore" program. This automatically begins the restore process to the hard disk as soon as it is placed in the drive.

**Burn data CD/DVD:** This command opens the MAGIX Speed burnR program with an empty burn queue for burning desired files to disc.

# Load audio file

Switch to the Import section - File tab (view page 41) to import audio files.

Keyboard shortcut: W

# Load video sound

MAGIX Audio & Music Lab Premium gives you the option of editing the audio tracks of video files like audio files. MAGIX Audio & Music Lab Premium can load video files in the following formats: DV-AVI, AVI (*.avi, suitable Codec, e.g. DivX must be installed), Quicktime (*.mov), MPEG1/2/4 (*.mp4,*.mv4, *.3gp), AVCHD (*.mt2) and Windows Media (*.wmv).

The audio track from the video is displayed as an audio object in the master track and can now be edited. The video is displayed in the video monitor during playback. A video preview strip (view page 28) is displayed in the the track window above the audio material.

**Note:** The video track of a video file is loaded for display in the monitor, but can't be edited (cut, for example) with MAGIX Audio & Music Lab Premium.

There are two other special video import options:

- **Show video image.**If you deactivate this option, the video track is not loaded and displayed in MAGIX Audio & Music Lab Premium. At the end, you can export (view page 113) only the audio.
- **Optimize video for smooth playback in video monitor.** Some video formats can be displayed directly in MAGIX Audio & Music Lab Premium, whereas others need to be converted into the internal MAGIX MXV format when opened. Opening them directly is faster, but places a larger demand on the system. If playback in the video monitor is jerky, you can use this option to convert the video into the resource-saving MXV format.

Keyboard shortcut: j

# Loading an audio CD

Switch to the Import section - CD (view page 42) tab to import audio CDs.

Keyboard shortcut: Shift + D

# Import DVD audio

This command imports the audio contents of a DVD into MAGIX Audio & Music Lab Premium.

**Note**: Importing the audio track(s) from a video DVD is not possible!

# Record

With this command you can open the MAGIX Audio Cleaning Lab record dialog. More information on this topic can be found in the chapter Record audio (view page 43).

Keyboard shortcut: R

# Export audio

Read the section "WAV and other file formats" (view page 113) in the "Export section" chapter.

# Export video

Please read more on this in the section "Video files" (view page 116) in the "Export" chapter!

# Internet

## Catooh – the Online Content Library

Catooh provides you with high-quality photos, videos, and music for every theme, expanded by intelligent iContent with professional Soundpools, DVD menu templates, and brilliant MAGIX ShowMaker styles to help you make your photo, video, and music projects reality. All of this is available directly from your MAGIX software.

Just choose "Share" from the menu "Catooh" to set up an Internet connection.

Browse through the thematically sorted categories or view the results directly by entering a keyword. After downloading, you can drag the objects from the Media Pool directly into your arrangement.

**Tip:** Read the introduction online http://rdir.magix.net/?page=JRF6LASAR2Z3!

### Open the Online Media Marketplace

This opens Catooh in a separate browser.

### Import media backup

iContent (for example, 3D transitions) which you buy and download from Catooh is stored directly in your central **My files\MAGIX Downloads\Backup** directory. If you have downloaded these files from other MAGIX programs, then you can use the command "Import media backup" to make them accessible for use in MAGIX Audio Cleaning Lab.

## Manage login details

These are options for managing user names (email addresses) and the associated passwords so that you are able to access your Online Services without having to enter the details each time.

**This information applies to all of my Online Services:** If this option is activated, then the account details you have entered will be applied to all Online Services. Deactivate this option if you have different details for individual services, then choose the corresponding service via "Select service and enter the associated login details.

# Delete old projects

The command "Delete old projects" is a convenient method for deleting old projects with all affiliated audio files.

**Project:** All of your most recently saved projects as well as all projects contained in the Project folder (view page 151) are listed here. You can choose one that should be deleted. Use "Search for project" to add other project files from any folder to the list.

Select the project file (.vip) that should be deleted by clicking on it. Of course, a project can not be opened if it is to be deleted.

The "Open in Windows Explorer" option opens an explorer window with the folder of the selected project. Here you can...no longer required files manually.

**Used files:** Lists the files used in the project that you want to delete. Use the small boxes to select the files you wish to delete.

With **"Delete"** you can delete the project (.vip) and the files selected for deleting in the file list.

When you select a project you wish to delete, some files in the list, i.e. files located in the project folder are selected already. They are recording files, files that are automatically produced when importing specific formats, and back up copies. In other words, data which is used only within one project of MAGIX Audio Cleaning Lab.

Files which are not located in the project folder, MP3 files from music collections or videos, for instance, are not selected, as you will probably use them in other projects or with other programs. You can select them if you're sure you don't need them anymore. In general, it's quite sensible to have the preset apply to the project folder because it allows you to simply choose a project at the top and to click "Delete" at the bottom to delete files which are no longer needed in a project.

# Exit

Closes MAGIX Audio Cleaning Lab.

# Edit Menu

## Undo

In the project you can undo the last changes you made. This way, it's no problem if you want to try out critical operations. If you don't like the result, you can always revert to the previous state using "Undo".

Keyboard shortcut: Ctrl + Z

## Redo

The “Redo” function undoes the previous “Undo” function.

Keyboard shortcut: Shift + Y

## Undo list

The last 20 editing steps are listed. This makes it easy to go back to a specific editing step.

## Set marker

With this command you set a marker into the track to mark a certain time position in the project. You can jump between the markers with the keyboard commands Alt+Arrow left/right.

Keyboard shortcut Alt+M

## Split

A selected object is split into two objects at the position line. This also works during playback.

Keyboard shortcut: T

## Remove object beginning

This command removes the part of an object beneath the position line, chronologically located before the position line. The audio material following it is moved forward to the initial position.

Keyboard shortcut: Z

# Remove object end

This command removes the part of an object beneath the position line, chronologically located after the position line. The audio material following it is moved closer.

Keyboard shortcut: U

# Cut

The selected object is cut out from the project and placed on the clipboard. It can then be reinserted elsewhere.

Keyboard shortcut: Shift + Del

# Copy

The selected object is copied from the project into the clipboard. It can then be re-inserted elsewhere.

Keyboard shortcut: C

# Paste

The content of the clipboard is inserted into the project at the position line.

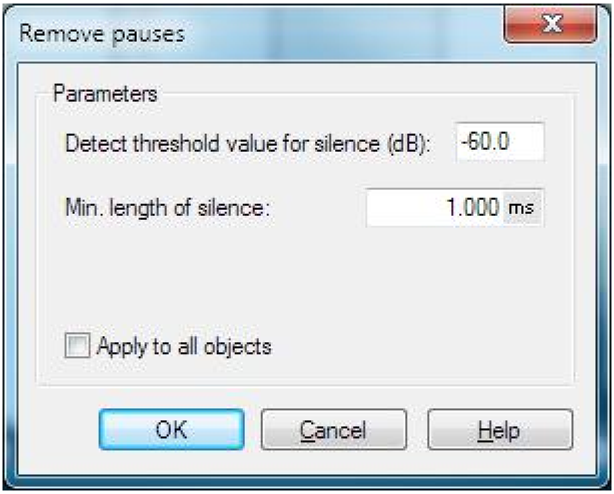
Keyboard shortcut: Ctrl + V

# Delete

The currently selected object will then be deleted from current project. The subsequent objects are moved forward so that there is no gap in the track.

Keyboard shortcut: Del

# Remove pauses




This function automatically removes sections without or with low audio level in the selected object. In the dialog, you can set a minimum time and threshold for recognition.

**Threshold detection for pauses (dB):** The threshold is set here. The value is specified in decibels, 0db corresponds to the maximum level. Typically you will enter negative values. The higher the numerical value, the lower the level and the less audio material is removed.

**Minimum pause length:** The minimum pause time is set here. If the function removes intended (musical) pauses in the music, increase the value.

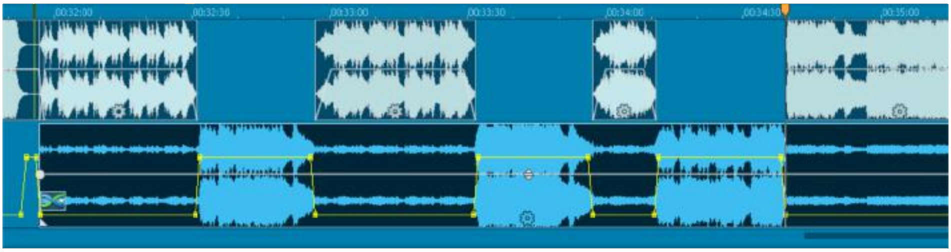
**Apply to all objects:** If this option is activated, the function will be applied to all audio objects in the project.

# Voice over

The "Voice over" effect creates a volume curve for automatically fading background music during spoken sections. To do this, proceed as follows:

1. Record speech.
2. Load your background music.
3. Move the object containing the background music to the second track below the spoken portion. Note: The second track can be opened with the "2" key.
4. Place your speech recording at the correct position, then cut and edit it with the object handles to remove undesired noise or mistakes.
5. Now open the voiceover dialog via "Edit" and activate the voiceover effect.

6. A volume curve (view page 146) is created on the second track which automatically fades the background music at the correct positions.



Use the faders to set by how much the volume of the background music should be reduced during speech passages and how quickly this should happen.

Use the selection list to apply the curve to the first track as well, should you have placed your background music there.

If you later want to move or shorten your voice recordings, click on the **"Update"** button to adjust the volume curve.

Note: The Voice Over feature is not available in MAGIX Audio Cleaning Lab.

# Batch conversion

Batch processing lets you automate work processes. You can extend a specific editing process from a single audio file to a list of files any size (i.e. the "batch"). The files are then executed automatically, over night, all day, or however you like.

Possible jobs may include:

- Normalization, volume adjustment, loudness
- Linear fading (in and out)
- All real-time cleaning and mastering effects
- Removing direct current
- Resampling/Timestretching (view page 138)
- Format conversions: Bit width (8/16/24-bit), sample rate, stereo/mono/left/right
- Save in all available export formats (view page 113).

Examples of use:

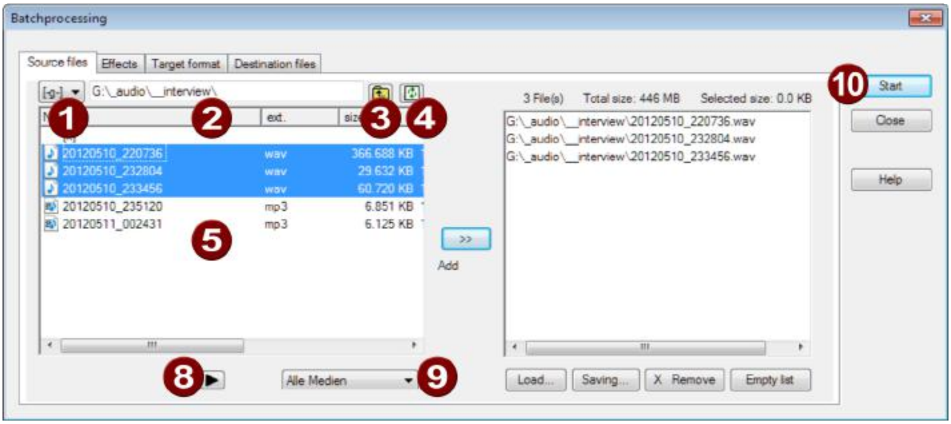
- Encoding a large number of wave files into MP3 while simultaneously adjusting volume and freshening up higher frequencies
- Loss-free conversion of wave files into FLAC to save up to 50% hard drive space
- Correcting multiple LP recordings with incorrect speeds
- Denoising original sound tracks

The process for batch conversion is as follows:

1. In the "Source file" tab, select the files for editing.
2. In the "Effects" tab, select the required editing processes.
3. In the "Time format" tab, select the output format.
4. In the "Target files" tab, specify where and under what names the files should be saved.

Note: The Batch processing feature is not available in MAGIX Audio Cleaning Lab.

## Source files

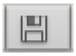



1. Drive
2. Path
3. Superior folder level
4. Refresh
5. File browser
6. Add to editing list
7. Editing list
8. Preview
9. File filter
10. Start editing

Create the list of files you want to edit. You will find a file browser on the right side. Select the files you would like to edit by clicking them; multiple selection is also possible (Ctrl + click for individual files, Shift + click for series, Ctrl + A to select everything). The view can be limited to certain audio formats via the file filter at the bottom. Every audio file can be pre-listened with the preview button.

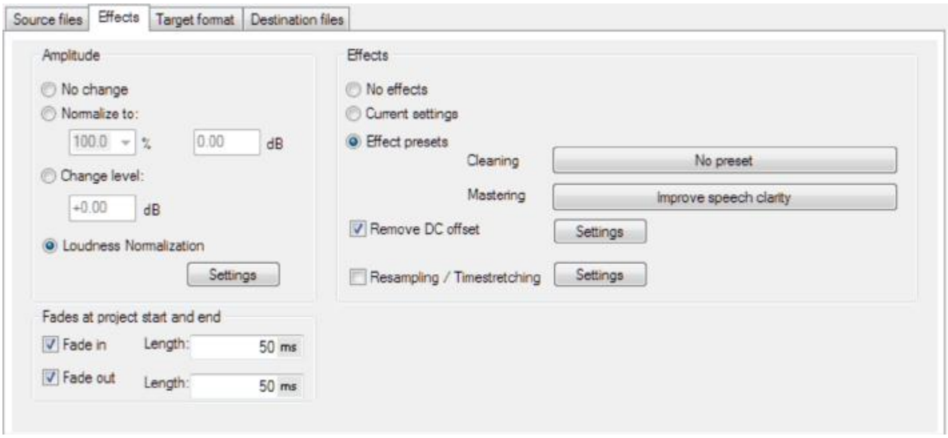
The "Add" button in the middle inserts all of the selected files into the editing list. If a folder has been selected, then all audio files in the folder, including all subfolders, will be added to the list.

 Loads a list (*.m3u format).

 Saves the list in *.m3u format for using your selection of files later.

"Remove" deletes all selected list entries. "Remove all" deletes the complete list.

# Effects




## Amplitude

**No changes**: The amplitude remains unchanged.

**Normalize/change level to**: You can normalize to a specific maximum value in %/dB or sink/raise the level by a certain value. Read more about this in the section "Edit menu -> Normalize object volume (view page 136)".

**Adjust loudness**: Loudness adjustment ensures a balanced average volume for each audio file. Read more about this in the section "Edit menu -> Adjust loudness".

## Fades at project start and end

Linear fades of any length can be added to the beginning and/or end of files.

## Effects

**No effects**: No effects editing takes place.

**Current settings/effects presets**: This section allows all cleaning and mastering effects in MAGIX Audio & Music Lab Premium to be used for batch conversion (including plug-ins). The batch processing dialog does not in fact offer direct access to the extensive settings options for the individual effects; instead, there are two choices:

- "Current settings": All effects settings in the currently loaded project will be applied (except object effects). First you can load one of the edited files into a new MAGIX Audio & Music Lab Premium project and make the required effects settings in real time there. Then you can open batch conversion and apply these settings to all of the other files.
- "Effects presets": You can select the cleaning or mastering effects presets included with the program or the ones you created yourself.

**Remove direct current**: Remove direct current noise from analog recordings; see "Remove direct current (view page 73)" in the "Edit" menu.

**Resampling/Timestretching**: Changes the playback tempo; see "Resampling/Timestretching (view page 138)" in the "Edit" menu.

## Target format

**File format**: All export formats (view page 113) available in MAGIX Audio & Music Lab Premium are available here with their associated format options.

**Stereo/Mono**: You can convert in stereo (mono sources feature the same signal on both channels), save one of the stereo channels as a mono file, or mix both stereo channels together to create a mono file.

**Bit resolution/sample rate**: The bit resolution can be changed to 8/16/24-bit, and the sample rate to 11,025, 22,050, 32,000, 44,100, or 48,000 Hz.

**No changes**: This means that the output format is applied unchanged.

## Destination files

There are several ways to save edited files:

**Replace source files**: The original files are replaced by the edited files. If the file is used in a project, then the project will be closed first.

**Save files in source directory with changed name/save files in following directory**: The edited file is saved in the source directory or in any chosen directory. The specified suffix/prefix is added to the file name. Optionally, you can delete the source files after editing them.

**Keep source directory structure**: This option saves all files including the source path. The folder structure is preserved if the source files originated from different folders.

# Load/Save realtime effects settings

Here you can save and load your favorite effect settings as "Mastering effect presets" or "Cleaning FX presets" so you can use them in other projects or objects.

FX presets can be applied to the "Object FX" page for individual objects or as project effects for the whole sound.

Since the available object effects are different from the project effects, some settings may be ignored.

# Apply all realtime effects

If the effects settings become too full to manage or you just want to ”summarize” your production, use this function to convert the entire audio arrangement into a single audio file. It will appear as a long object in a new project.

Once the effects have been added they will no longer require CPU power.

# Effects menu

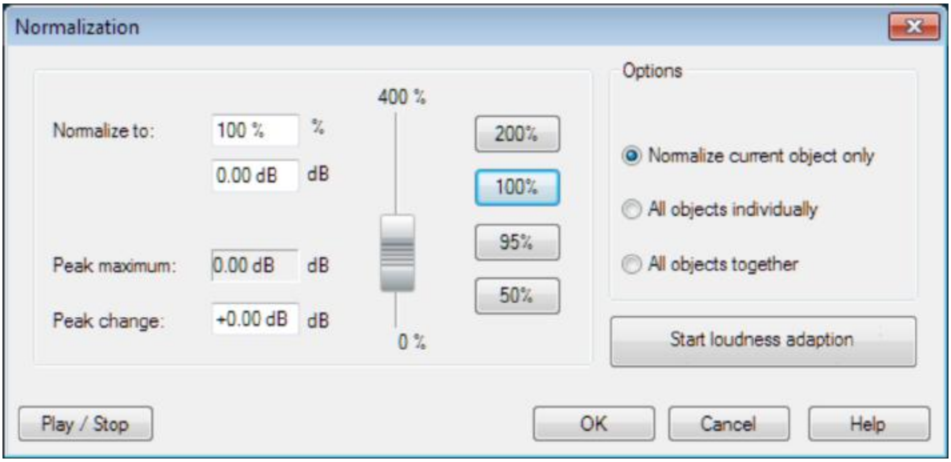
Here you can open all the settings dialogs for the "Cleaning" and "Mastering" effects as well as other effects. The effects are sorted categorically in submenus. If the cleaning or mastering section is switched to object mode, the effect will be loaded as an object effect (view page 72), which means it will only affect the selected object and not the entire audio material on the master track.

A description of the effects which are not explained here can be found in the sections "Cleaning effects" (view page 62) and "Mastering effects" (view page 74).

## Normalize object volume

This function raises the volume of an object to the maximum level without the material being clipped. This utilizes the dynamic range the best way possible. First the highest levels are detected, and then the object level is adjusted so that the max. level amounts to 0 dB, i.e. the maximum volume (or another value between 1% and 400%).

**Note**: If you experience very slight clipping during recording and then proceed to normalize the material, then you won't achieve the same quality as if you produce a correctly clipped recording! For example, if you only modulate half of the material, then your recording will have a quality of 15-bit samples – normalizing to 100% doesn't change anything.




**Normalize to**: Here you can set the value to which the audio material should be normalized by entering it into the input field, moving the fader, or selecting one of the presets (50, 95, 100 or 200%). The value will be shown in % and dB (100% = 0 dB = max.). Values above 0 dB bring about digital clipping.

**Maximum level**: Displays the highest detected peak in the selected range/object.

**Level change**: Displays the level change in dB, in accordance with the selected normalize level and the detected maximum level.

Different methods can be specified under "Selection":

**Normalize the selected object only**: Normalization is only applied to the selected object. This function can also be executed in "Object FX" mode by clicking the "Auto" button below the volume controller (always normalizes to 100%).

**Normalize all objects separately**: Each object in the project is normalized according to its own maximum (peak) level. The level ratios between the individual objects changes for this reason.

**Normalize all objects as a single unit**: The maximum level is detected for all objects in the project, and each object is normalized according to that value. The level ratios between the individual objects is preserved, but only the object that contains the maximum level is optimally clipped.

**Start loudness adjustment**: Starts normalization including the average loudness of objects, see Loudness.

Shortcut: N

# Loudness adjustment

This function unifies the volume of the individual tracks in the project. First all of the levels for every object are increased separately to the maximum without clipping the material (see Normalization (view page 136)). Depending on the musical production, however, each title may have a different volume at full level, since the relation between loud and quiet sequences within the track also influences how we perceive volume. In the second step, the average loudness of the song is determined and the object level is adapted accordingly.

Tracks with higher peak values but lower loudness may be normalized at a level above 0 dB (full clipping). To avoid overloads, the limiter is automatically activated (see MultiMax).

A target loudness can be given in dB. Since this is the average value, the loudness value is always less than 0dB; -15 is the preset.

The degree of adjustment decides how strictly the loudness normalization is applied. At 0%, no adjustment is made to the target value. At a value of 50%, the level is raised to half the difference between the detected loudness and the target value. Volume differences remain between the tracks in this case. At 100%, every track is raised to the target value regardless of its loudness. This is only recommended in rare cases because even in a single party mix, a dance hit will not have the same volume as a ballad.

**Tip**: Volume fluctuation within a song can be balanced with the MultiMax loudness presets.

Keyboard shortcut: Shift + N

# Isolate Stereo Channels

Displays a stereo file from two mono objects. The two mono objects are totally independent, are located one above the other on two tracks and can be worked on separately. This option is particularly suitable for removing undesirable audio interference that can only be heard on one channel.

# Switch channels

This function switches the left and right stereo channels.

This is useful for correcting recordings with switched channels. This function can be reversed if you don't re-select the range; opening it again will bring back the original material.

# Invert phase

The sample data is inverted along the amplitude axis, which means that negative values become positive and vice versa. This function allows you to adjust recordings with different phases to one another.

An incorrect phase occurs if during an analog recording parallel cables are switched. To correct such errors, some mixers have a switch that can be used to invert the phase of an input. If you press the switch by mistake, errors might occur as well.

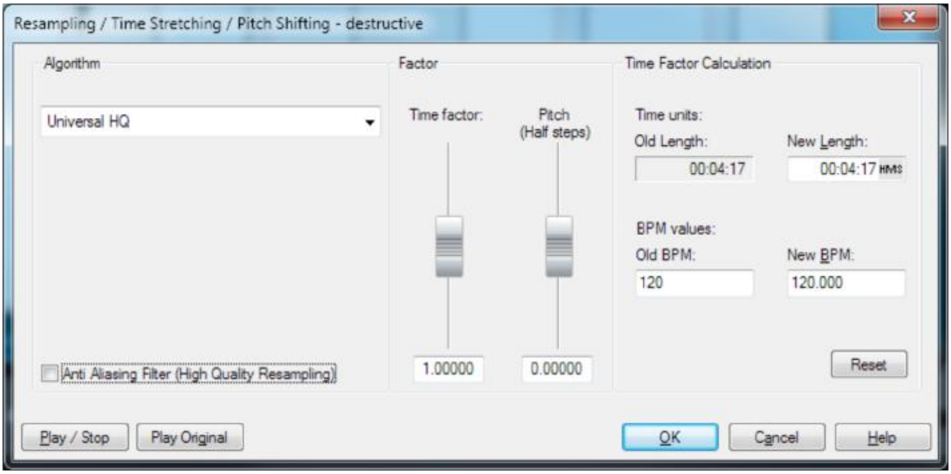
This function is reversible, so if you repeat it, the original signal is restored.

# Backwards

When Backwards is applied the sound file will be played in reverse. This way you can create very interesting effects, not to mention the “hidden messages” in many songs...

# Resampling/Timestretching

The pitchshifting/timestretching/resampling editor opens. This effect can change the tempo and pitch of the audio material separately. The effect is also available as a mouse mode or object effect. The dialog's advanced settings options and the pitch changing options are missing, however.




**Algorithm**: Selects the applied timestretching process.

**Time factor calculation**: All algorithms in this dialog apply a time factor as the input parameter. The input fields for the group "Time factor calculation" enable convenient detection of the time factor from the desired new length or a new tempo in BPM in relation to an old tempo (required beforehand).

**Pitch (semitones)**: For any algorithm except resampling, the pitch can also be set independent of the tempo. Use the pitch fader beside the factor fader to experiment.

**Play/Stop/Play orig.**: "Play/Stop" can be used to immediately control the result of the algorithm. "Play orig." plays the unedited material for comparison.

## Algorithms for timestretching/pitchshifting

- **Élastique Pro:**This algorithm is used by default and provides optimal results for most audio material.
- **Élastique Efficient:** This is a version of the algorithm that saves computer power and has reduced sound quality as a result.
- **Monophonic voice**: Timestretching and pitchshifting for vocal solos, speech, or solo instruments. The material must not contain background noise, and excessive reverb may also be detrimental to the use of this method. With suitable material the audio quality is very high. The "Correct formant factor" option preserves formants if pitches are changed. These are characteristic basic

frequencies of the voice that are independent of the pitch that is sung. In other words, the characteristic discoloration of pitch ("Mickey Mouse") effect does not occur in this case. The formants, however, can be shifted by +/- 12 half tones. This achieves suitable vocal distortions. Beat markers are not evaluated.

- **Resampling**: Pitch shift and tempo cannot be changed individually. This method requires considerably less CPU time. If the pitch is increased or the sample is shortened, then resampling is almost completely free of loss, and the sample material will suffer almost no damage. In other cases, resampling causes loss of overtones. For example, if the length of a 44.1 kHz sample is doubled, then the frequency level of the result will be limited to 11.025 kHz. The sound is the same as when the playback speed of a record player or tape recorder is changed.

# CD menu

The CD menu contains all special functions for audio CDs and CD mastering processes, e.g. setting CD tracks and pauses as well as the "Create CD" function.

## Set track marker

Allows you to set a track marker at the current location of the position line. All the following markers will automatically receive a corresponding number. Each CD track needs a track marker. The minimum length for a track is 4 seconds, whereas the maximum length of a track is only limited by the capacity of the CD.

Keyboard shortcut: M

## Set Pause marker

This function lets you set pause markers. At these points some CD-Players switch to absolute silence during playback until the next track marker appears. The CD-player displays a countdown for the next title.

Keyboard shortcut: Shift +M

## Set track marker automatically

Use this option to automatically detect pauses between the tracks and set track markers. The audio material is analyzed for quiet sections.

> For more detailed information about the functioning of automatic track marker recognition please refer to the chapter „Cutting and arranging objects“.

Keyboard shortcut: Ctrl + M

## Set track marker on the object edges

This function is used to set automatic track markers at the start of each object on the track. Before using the function, execute the "Remove all markers" command to delete any already existing track markers.

Keyboard shortcut: Ctrl + Shift + M

# Split objects at marker positions

This function will split all objects at the position of the track markers.

Keyboard shortcut: Ctrl + T

# Set auto pause length

Audio files that have been loaded successively into MAGIX Audio Cleaning Lab are arranged consecutively in the project. Between the tracks, a standard pause of 2 seconds is preset. In this dialog, the value can be modified.

# Remove markers

This function is used to delete existing markers. First put the position line at the same place as the marker (snaps to marker)

Keyboard shortcut: Del

# Delete all markers

This function removes all existing track and pause markers. This can be useful if you are going to use the "Set track markers automatically." function.

Keyboard shortcut: Ctrl+Del

# Delete CD track

This command deletes a selected track marker and the corresponding audio material from the position of the track marker to the next track marker. The audio material following it, is moved closer.

Keyboard shortcut: Alt + Del

# Create audio CD

This option does the same thing as the "Audio CD" button in the export section.

Keyboard shortcut: B

# Show CD-R drive information

This dialog shows you all available information on the active CD-writer. This includes the manufacturer, product name, product revision, cache and the features supported by the drive.

# Show CD-R disc information

Displays all available information on the CD inserted in the drive. The most important feature is the maximum length, which cannot be exceeded during production, for example: 74 minutes and 59 seconds.

# ID3 Editor

This opens the ID3 Editor (view page 110).

Keyboard shortcut: L

# Print CD cover

MAGIX Audio Cleaning Lab contains an easy-to-use CD print studio. Here, you can design and print not only simple track listings, but also sophisticated CD covers, CD booklets, and circular CD labels. The track information is automatically transferred from the playlist to the print studio.

# Get CD track information online (freedb)

You can retrieve the track information for imported CDs from the Internet using the Online freeDB Query function. This query is based on the exact combination of track lengths and the order of all tracks on an album. This also works if the tracks are loaded into the project individually (e.g. as MP3 files) and are in the right order. If the track lengths differ here by a few seconds from the exact track length, the correct CD should still be recognizable.

# Query album information recording online (freedb)

When recording cassettes or records onto your computer, one large file is created in which all tracks are arranged one after the other without track markers, much like a CD. You can, of course, use the function "Set track marker automatically" to analyze the audio material and, with the help of the pauses, separate the file into individual tracks. This does not work, however, when the tracks run into one another without a pause.

In this case you can ascertain the exact track division by querying the Online freedb CD database. To do so, proceed as follows:

1. Start up your Internet browser and go to the freedb search page by pressing the "Start Internet search" button.
2. Enter the name of the album or band into the search field. One or more albums, which match the search request, are listed. If you know that a certain album fits your recording, then click on "Details". It will display the CD track list you were looking for alongside other details.
3. Click on the link above the disc ID (a combination of 8-digits of numbers/letters, e.g. 7e120419). freedb data record   is displayed for this CD.
4. Copy the URL (Internet link) from the address bar of your browser onto the clipboard.
5. Change back to MAGIX Audio Cleaning Lab and enter the Internet link into the text field in the lower portion of the dialog. Then click on "Apply CD data". The CD tracks will be added to the project.

Warning: Sometimes the first track may begin very quietly (for instance an intro or applause in live recordings) and the start of the recording occurs too late. Due to this, it may happen that the length of the first track does not correspond to the track length suggested by the database. As a result, all track markers will be a little bit too far back. In this case, move the second track marker forward while holding down the Ctrl-key; all subsequent track markers will then be moved by the same amount and should be positioned appropriately at the start of each track.

# CD info options

Here you can start different advanced options for "freedb CD Database".

## freeDB > Submit CD to freeDB

You can add CDs in the online CD database. The enormous freedb project exists courtesy of the contributions made by worldwide users.

If you have a CD that is not in the database, you can enter the CD information.

- Make sure the right CD is in the drive.
- Select "Enter New CD in freedb "
- Enter the information, double-checking the details for possible errors.
- Press "OK"
- Within an hour or two, the new information will be online for everybody to access.

## freeDB user preferences

User info is entered here for MAGIX Audio Cleaning Lab to use when freedb is queried. Of course, using freedb is completely anonymous, but freedb often has to process several queries at once, so an i.d. is assigned to identify the user. If you have problems accessing info, maybe someone is using the same data; you can change the settings to avoid this problem.

## freeDB proxy options...

If you are having difficulties connecting to the freedb server, then you can choose another server from the list, or you can increase the "timeout" value. An increased work load causes the server to react slowly and a connection cannot be made properly.

## freeDB > Delete freeDB Cache

The freedb online database creates a cache on your hard disk, containing all data available via the freedb button. This allows you to access the data without having to go online. You can of course delete the cache, should it contain false data or when up-to-date current data is available.

# Options menu

## Mouse mode

"Mouse modes" are your tools for working in the track window of MAGIX Audio Cleaning Lab. For most processes the preset Edit mouse mode is sufficient. However, in the "Options" menu you'll find other modes for specific tasks. Depending on the mode, the function of mouse-clicks in the project changes. The current mouse mode is indicated by the appearance of the mouse pointer in the track window.

### Edit mouse mode

The Edit mouse mode is preset. You can take care of all important tasks with this mode. Select objects in the track window with a left-click. Selected objects can be moved using drag & drop. All subsequent objects are also moved so that no unwanted gaps develop later in the track.

In Edit mode you can use the 5 handles to fade or shorten all objects or to adjust the volume. Right-clicking on an object opens the context menu where you can select important editing options for the object.

Keyboard Shortcut: V

### Range Mode

In Range Mode you don't edit object with the mouse but ranges of audio. For more information, see Range Mode (view page 33) in the "Track window and constant control elements" chapter.

Note: This feature is not available in MAGIX Audio Cleaning Lab.

### Volume draw mode

In **Volume Draw mode** the volume curve can be "drawn". This way, you can create irregular volume progressions quickly.

To delete volume curve points, double-click on the corresponding point or click on a point in the Delete Object mode.

## Wave draw mode

Repair short distortions such as crackling directly in the wave form of the audio file by using the Wave drawing mode. These distortions usually only last a few sample values, so you can use the mouse and try to draw along the original waveform without the distortion.

There is an automatic zoom function in the wave form display when you switch into the Wave drawing mode, so sample values become visible.

**Warning:** Unlike other editing of the master track which only affects the project, this mode lets you work directly with the Wave audio file, i.e. with the recorded raw material. The latter is changed directly and permanently. Create a backup copy so you can undo changes. This automatically occurs when working with MP3 and other compressed formats, since such files have to be converted into the Wave format for this function.

## Resampling/Time stretching mouse mode

This mode lets you change the playback speed of objects with the mouse so that they are better aligned. This mouse mode can be found in the "Options" menu.

You can use this mouse mode to stretch or compress the object using the lower handle at the rear. The mouse pointer turns into a clock.

Resampling mode (preset) can be used to change speed and pitch just like on a tape, i.e. speed and pitch are changed together as one. If an object is compressed with the mouse, the speed and pitch increase just like a tape when it is played faster.

In the Time Stretch mode, the pitch remains unchanged if the object length and with it the speed are changed.

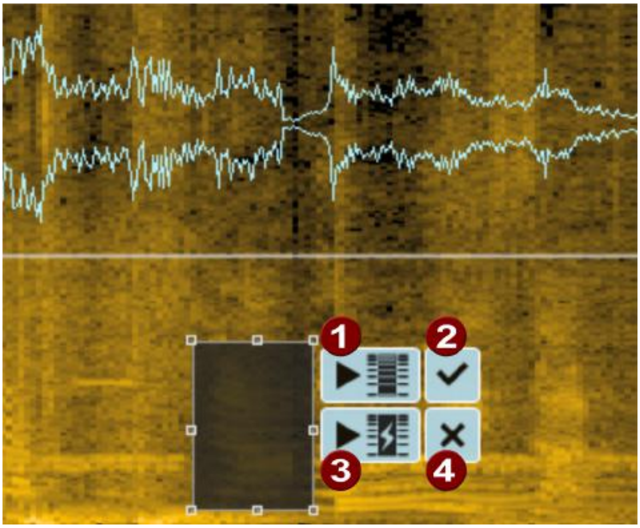
You can change modes by switching to the cleaning effects, selecting "Object" editing and then selecting time stretching from the tempo/resampling effects presets list.

## Edit spectrum directly

With the mouse mode "Edit spectrum directly" you can remove specific noises from the sound spectrum of the audio material.

The view of the master track changes to Spectral display (view page 32). You can create an area around the noise with your mouse. Its size is still adjustable afterwards by simply stretching the handles on the frame.

Four buttons are located at the frame of the disturbances.




1. Click the upper play button to hear the impact of the effect by playing the corresponding passage with filtering.
2. By clicking at the checkmark the editing will be calculated into the audio material instantly. A copy of your original material has been automatically saved beforehand so you can undo any editing (Ctrl + Z).
3. For comparison you can play the same passage without filtering with the lower play button.
4. By clicking the crossmark you delete the frame and cancel the filtering process.

Note: Spectral Edit mouse mode is not available in MAGIX Audio Cleaning Lab.

## Display volume curve

This option activates the volume curve.

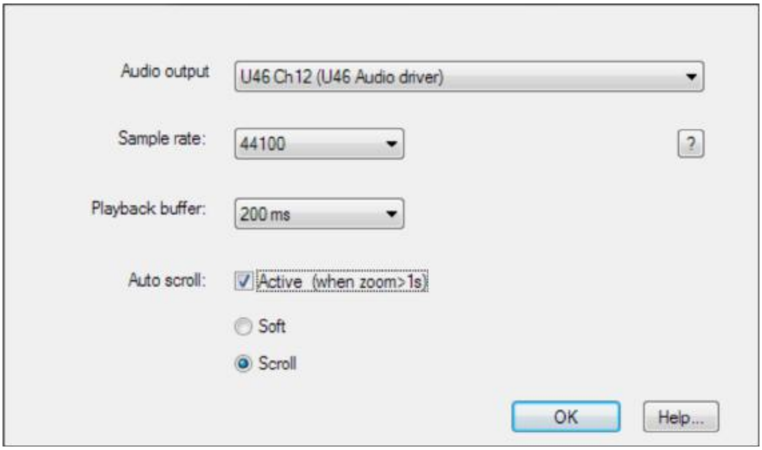
For more about this, please refer to the chapters "Cutting and arranging objects" (view page 54) and "Track window" (view page 26).

## Lock/Ripple/Freely moving objects

Options of the "Ripple" button. For more about this, please refer to the section Deleting and moving objects (view page 57).

# Playback parameters

This option opens the playback parameter window.




**Audio line-out:** Here you can specify the output of the sound card that will be used for playback. This is particularly important if you have a sound card with several outputs or if you have several sound cards in your computer. The lower entries in the list with the added prefix WASAPI use the more modern driver model of the same name.

**?** Here the supported audio formats of the sound card and the sound card driver information can be displayed. You can also toggle between driver types (MME and WDM). Adjust this setting only if you have problems with audio playback or recording. If WASAPI was selected for the audio line-out, the dialog also includes the option of switching the device mode between "Shared" and "Exclusive" besides the settings for the audio buffer. For recording the output of other programs (view page 51) (e.g. browser), "Shared" mode must be used.

**Sample rate:** This sets the sample rate of the project. For maximum sound quality, the sample rate of the project should agree with the sample rate of the recorded or imported material. 44 kHz is preset. If you are planning to edit video sound, you should change the sample rate before importing video files (whose audio track is mostly available with 48 kHz).

**Playback buffer:** If many effects are subject to dropouts during playback on older computers, you can increase the length of the playback buffer here. But this also increases the reaction time of the program when starting and stopping the playback or recording, and when changing the effect parameters.

**Autoscroll:** If active, the displayed section of the position line follows; "Soft" and "Scrolling" can be selected to choose between page-by-page scrolling and soft scrolling.

| | |
|---|---|
| Keyboard shortcut: | P |

# Units of measurement

This sub-menu allows you to determine different units for the timeline. You can choose between samples, milliseconds hour/minute/second and CD frames.

Keyboard shortcuts:

| | |
|---|---|
| Samples | Shift + 1 |
| Milliseconds | Shift + 2 |
| h:min:sec | Shift + 3 |
| Min:Sec:CD frames | Shift + 4 |

# Mouse snap active

When the mouse grid is switched on, the objects snap into place beside one another so that everything fits in seamlessly.

# Auto crossfade mode active

With every cut the two objects that are created are slighlty crossfaded in order to avoid crackling. This is referred to as "Auto crossfading" (for more info on crossfades please also see Crossfading objects (view page 55)). As all recorded and imported objects can be easily faded – which is not always desired – this option can be deactivated.

# Display values scale

The right edge of the track view shows a values scale. This indicates the level of waveforms in dB, and the spectral display indicates specific frequencies in Hz.

# Options for track marker recognition

For more detailed information about the functioning of automatic track marker recognition please refer to the chapter „Cutting and arranging objects“ (view page 54).

**Settings for recordings of vinyl/tape/speech and audio books/digital:** These are presets for various types of media. If you make adjustments to the following parameters, these changes will be saved in the presets.

**Minimum pause length:** The higher this value is set, the fewer the number of pauses that will be detected. If the value is set too low, it can also happen that very short breaks during a song will be recognized as pauses.

**Minimum track length:** A new marker will only be set when the time between the last track marker and the pause is at least this value. For example, if you want to burn a CD of pop songs from a cassette, you can set the track length to around 2 minutes. However, if you are working with a cassette with very short audio segments, you should set this value accordingly.

**Maximum (Minimal) level for pauses:** MAGIX Audio Cleaning Lab searches for logical volume levels during pause recognition. If too many (an therefore incorrect) pauses are found, you should move both sliders further to the right. If not enough pauses are marked, you should move the sliders to the left.

**Recognition of cassette or LP sides:** In some cases you may want to record both sides of a record one after the other without having to interrupt the recording process. The result is usually a very long audio file with a very quiet section in the middle. MAGIX Audio Cleaning Lab then only finds one pause, recognizes the situation, splits the recording into two objects at this position and reattempts the process using these two objects. In the best case, the noises created when the record or cassette is turned around are also removed.

# Path settings

In this dialog, you can set the memory path for your recorded audio files (New projects) as well as the search path for VST plug-ins.

# View Menu

## Display 2 tracks / Display 4 tracks

Switches the track view to two or four tracks. For more information, see Number of tracks (view page 32) button.

Note: The Additional tracks are not available in MAGIX Audio Cleaning Lab.

Keyboard shortcut: 2

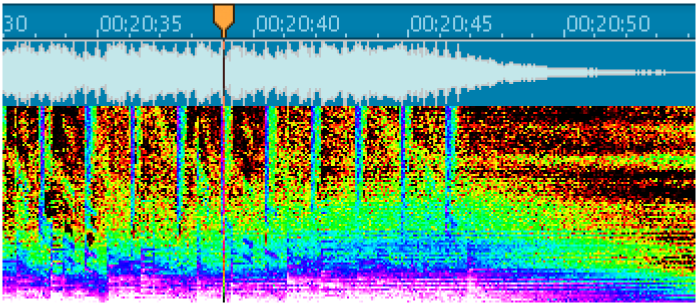
## Stereo display

Using this option you can switch the view of the wave shape, which splits up the material between the two stereo channels. This view is useful to visually control the material in the stereo panorama or to find zero-crossings for cutting operations.

Keyboard shortcut: Tab

## Spectral display

Spectral display equates the X axis (horizontal) to the time dimension. In contrast to the waveform display, whereby the height of the waveform only indicates the total level of the signal, the spectral display shows the level of each of the audio signal's individual frequencies. The actual level of each frequency is indicated by the color of the points in the spectrogram image.




The whole color spectrum is used to display the volume of individuals frequencies. Pink indicates loud sounds in a frequency range, green indicates the areas with middle volume and red the very quiet sounds (in the preset color palette). Black is used for silence and white for maximum volume. Other different color palettes can be selected in the menu "Options -> Spectral display".

The spectral display is significantly more processor-heavy than the normal waveform display, so redrawing after a section changes is always slightly delayed. This delay is increased the further the zoom is extended, since MAGIX Audio & Music Lab Premium needs to include more and more data for calculation of the display. For this reason, spectral display is only available from a certain zoom level.

Spectral display of the audio enables specific disturbances in the audio material to be detected. Crackling can be recognized by vertical lines across the entire frequency spectrum; continuous disturbing sounds can be detected by horizontal lines.

This display also makes it easier to find sections in a song quickly, since instrumental changes can be clearly seen in the spectrum. On the other hand, the waveform display will not indicate changes if the volume level does not fluctuate.

Note: The spectral display is not available in MAGIX Audio Cleaning Lab.

# Overview mode

Use the "Overview mode" in the "Options" menu to show or hide the overview track (view page 28).

# Maximize Upper/Lower/Info area

Maximize quickly the according screen areas, for more info see Changing the size of sections in the program interface (view page 40).

| | | |
|---|---|---|
| Keyboard shortcut: | Maximize upper section | F5 |
| | Maximize lower section | F6 |
| | Maximize Info area | F7 |

# Share menu

The Share menu provides access to online social networks as well as transfer functions to other MAGIX programs.

Here you'll find options for uploading individual objects in the Arranger or files from the Media Pool as well as the entire arrangement, as audio or video. You can also transfer your arrangement to another MAGIX program (if it is installed) in order to, for example, use it as background music for your slideshow.

## Use as background music

Converts the project into MP3 and forwards it directly to a MAGIX video editing program (e.g. MAGIX Movie Edit Pro deluxe MX) where it can be used as background music.

**Note:** This function is only available if you have installed an appropriate MAGIX program.

## Add to music collection

Converts your arrangement into MP3 format and sends it directly to a MAGIX music management program (e.g. MP3 deluxe MX), where it is added to an existing music collection.

**Note:** This function is only available if you have installed an appropriate MAGIX program.

## Publish online

Publishes the project or single tracks form it on various online platforms. For more information please refer to Export section > Share (view page 121).

# Help menu

## Help

Open the program's help file. The help file provides explanations of all functions in the program and step-by-step instructions.

Keyboard shortcut: F1

## Display tips

Determines whether the **tooltips** are displayed or not. If activated, a small help window will be displayed as soon as you hold the mouse over a button for a while.

## Tutorial videos...

This plays a tutorial video for MAGIX Audio Cleaning Lab. An Internet connection is required.

## magix.info

**Share your knowledge & get answers** This MAGIX service offers you and the MAGIX Community at large a central platform for exchanging knowledge, photos, videos, and music, for discussing and evaluating, for communicating with one another, for presenting yourself, and networking with other members.

You will find the sections "Questions & Answers", "Show & Discuss", "Online Training", and "Chat" where you can play an active or passive role in the MAGIX Community. For this there's not only the www.magix.info portal, but magix.info directly in your product.

**Ask questions online**

If you have questions or problems with your MAGIX program or you're looking for tips & tricks concerning multimedia, then you're at the right place.

**Display all questions & answers...**

Use the questions which other community members have asked and the collective knowledge of the answers to improve your own working methods, to solve problems, and to get to know the details of MAGIX Audio Cleaning Lab.

**Launch the online training center**

Learn not only theoretically, but also practically what the program is capable of: "learning by doing", quickly and uncomplicated.

**Let your friends view the contents of your MAGIX.info screen over the Internet on their PCs.**

Ideal for viewing photos and videos together or for receiving support for questions arising during your creative workflow while using the program (a licensed client will be installed for this purpose).

# Register online

This option opens the MAGIX homepage for online registration where you can register as a MAGIX user. Registration grants you access to the MAGIX support website where various program updates and help programs can be downloaded.

# Update online

Update your program to the newest version by downloading any available patches from the Internet.

# Deactivate Program

This menu item deactivates MAGIX Audio Cleaning Lab with immediate effect. After deactivation, it is possible to immediately install and activate MAGIX Audio Cleaning Lab on another computer

# About MAGIX Audio Cleaning Lab

Copyright notices and version numbers are displayed.

# Restore original program behavior

This command resets all settings of the program to the delivery status.

# Language

Here you can switch the language of the program interface. After changing the language, the program is restarted. Note that the help and manual continue to be available only in the language in which the program was installed. For a completely different language version, you have to reinstall the Audio & Music Lab. Choose the required language using the flag at the lower edge of the start window of the installation program.

# Keyboard layout and mouse-wheel support

## Keyboard layout

### Sections

| | |
|---|---|
| Import | Ctrl + 1 |
| Cleaning | Ctrl + 2 |
| Mastering | Ctrl + 3 |
| Tracks | Ctrl + 4 |
| Export | Ctrl + 5 |

### Menu functions

| | |
|---|---|
| New project | E |
| Load project | O |
| Save project | S, Ctrl + S |
| Save project as | Shift + S |
| Load audio file | W |
| Load video file | J |
| Loading an audio CD | Shift + D |
| Recording | Shift + R |
| Export audio (Wave) | K |
| Close | Alt + F4 |
| Undo | Alt + Back, Ctrl + Z |
| Redo | Ctrl + Y |
| Set marker | Alt + M |
| Split | T |
| Remove object start | Z |
| Remove an object end | U |
| Cut | Shift + Del, Ctrl + X |
| Copy | Strg + Ins, Ctrl + C |
| Insert | Shift + Ins, Ctrl + V |
| Remove (delete) | Del |

| | |
|---|---|
| Batch processing | Shift + B |
| Normalize object volume | N |
| Loudness normalization | Shift + N |
| Set track marker | M |
| Set pause marker | Shift + M |
| Set track marker automatically | Ctrl + M |
| Set track marker on object edges | Ctrl + Shift + M |
| Split objects at track marker positions | Ctrl + T |
| Remove all markers | Ctrl + Del |
| Delete CD title | Alt + Del |
| Create audio CD | B |
| ID3 Editor | L |
| Edit mouse mode | V |
| Playback parameters | P |
| Measurement units | Shift + 1...4 |
| Mouse snap active | Ctrl + R |
| Display values scale | #, ' |
| Display 2 tracks | 2 |
| Stereo display | Tab |
| Maximize upper section | F5 |
| maximize lower sectio | F6 |
| Maximize Info area | F7 |
| Help | F1 |
| Search | Ctrl + F |

**Transport functions**

| | |
|---|---|
| Playback/Stop | Space bar |
| Pause | , |
| "Emergency" stop during playback | Esc |
| Back to the start | Home |
| To the end | End |

| | |
|---|---|
| Playback/Stop | Space bar |
| Fast forward (rewind) | Arrow left (right) |
| Playback position to next (previous) track marker | Alt + arrow left (right) |
| Playback position to the next (previous) object edge | Shift + Alt + arrow left (right) |

Keyboard shortcuts for zooming the display are outlined in the chapter **Zoom commands** (view page 29).

# Mouse wheel support

| | |
|---|---|
| Click on the middle mouse button | Start/Stop playback |
| Wheel | Horizontal scrolling |
| + Ctrl | Horizontal zooming |
| + Shift | Vertical zooming |

# If you still have questions

## Tips for Program Help

When the program is open, you can access the Help feature by pressing F1 on your keyboard. The program Help contains tips on how to use the program and lots of additional information. Many important terms are indicated in the text in italics. Simply click on these terms for a more detailed explanation of what they mean.

**Context Help:** Press the "F1" key on your keyboard at any point in the open program and Help will open with the matching help topic.

**Search function:** This feature can be used to look for specific words in the Help section. Enter either the individual word or use logical operators (e. g. OR, AND, NEAR) to refine your search if you have several search words.

- **OR** (between two words): all topics which contain both words or one of the words will be listed.
- **AND** (between two words): only those topics will be listed which contain both words.
- **NEAR** (between two words): only those topics will be listed which contain both words. A maximum of 6 additional words are allowed between two search words.
- **NOT** (before a word): topics which contain this word will **not** be listed.

**Print:** The print function can be used to print out specific Help topics or entire topic sections. The print button is located at the very top of the toolbar in the Help window.

## System requirements

### Operating system:

Microsoft Windows Vista | 7 | 8 | 8.1 | 10

### Minimum configuration:

- 1 GHz processor or higher
- RAM: 2 GB
- Hard drive memory: 1 GB
- Graphics card resolution 1024 x 768
- 16-bit sound card
- CD-ROM drive
- Internet connection required for activating and validating the program*

*Software can also be used offline. Please note, however, that the software requires an Internet connection at least once a month to validate your license.

Optional:

- Burn CDs/DVDs with CD/DVD±R(W) recorder

# Uninstalling the program

If you would like to uninstall MAGIX Audio Cleaning Lab, go to the Control Panel and select "Software" or "Programs and Features".

# Serial number

A serial number is included with each product. This serial number is required for software activation. Please store this number in a safe place.

## What does a serial number do?

With a serial number your program license is clearly assigned to you and only you. This allows you to use the free customer service via email.

Serial numbers also help protect against software piracy. This makes it possible for us to offer our customers the most value for their money.

## Where is the serial number located?

The serial number is on the insert card in the program box.

If you have purchased the download version, you will receive a confirmation email containing the serial number that you can use to activate the program. This will be sent immediately following purchase to the email address you provided.

## When is the serial number needed?

The serial number is required when you start the program the first time, as well as for program registration.

# MPEG-4 encoder settings

MPEG-4 is actually a collection of highly efficient codecs for video and audio based on the MPEG standard. Compared to MPEG-2, the same quality can be achieved with smaller file size.

**Recommendation:** Check the presets that the program offers to find the right preset for each application and the right playback device. These presets are optimized for typical applications.

The advanced settings in **"Advanced"** are divided into "Video", "Audio", and "Multiplexer".

Under **"Video“** there is a choice between "MPEG-4 Simple" (MPEG-4 Visual/MPEG-4 Part 2) (view page 168)and "MPEG-4 H.264" (AVC/MPEG-4 Part 10) (view page 163).

Depending on the purpose of application of the material to be exported, either one of these encoder settings can be selected for compressing the video material. Refer to the manual for the playback device to find out which encoder is supported.

Under **"Audio"**, "AAC" (view page 171) is preset. If you have chosen an AVCHD transport stream from the presets, "AVCHD audio track" will be available.

Under **"Multiplexer"** you can adjust certain presets, e.g. container format and streaming properties. This is only recommended for experts.

The option "Export as website" also creates an HTML page in an integrated Flash player that can play back the video created. Read the topic "Embed Flash videos into your own web site".

**Tip**: First, check if there is a suitable preset in the export dialog for your purposes. Advanced users can make specific adjustments to these three areas using the advanced settings.

# Video codec

## MPEG-4 H.264

The MPEG-4 H.264/AVC codec is suitable for all types of material; however, it requires relatively higher CPU power for later decoding.

In the advanced encoder settings of the AVC encoder the "Generic" options in "Main Settings" are mainly interesting.

The AVC preset and video format can be adjusted here. For instance, if DVD quality is desired you should select "DVD". The video format should be selected specific to the country so that the material can be played on the devices most commonly available in these countries. For instance, you should select PAL for Germany, SECAM for France and NTSC for the US.

### MVC

MVC: Multiview Video Coding (MVC) is an expansion for stereoscopic applications. Activate this to export MVC files.

**Note:** MVC is only available in the Video deluxe Pro versions.

### 2-pass

The video to be exported will be encoded twice. The first run-through serves to calculate the streaming rate for each section (for video this is per frame). The second run-through is when the actual video is created with the streaming rates that were calculated in the first run-through. The result is an improved image quality, but encoding takes place approximately twice as long.

### Smart Render and Smart Copy

**Smart Render** lowers the demand on the encoder for AVCHD material. In the production of AVCHD files, only those parts of the movie that were changed in the program (e.g. by video cleaning or effects) are re-encoded. Please note: The AVCHD files contained in the movie **must** have the same format, i.e. the bit rates (variable or constant), audio formats, image resolutions and video formats must match.

The Smart Rendering special mode **„Smart Copy“** enables AVCHD material to be transferred without having to encode it for the target medium which greatly increases the encoding speed. The video material cannot appear to have been altered in any way, only hard cuts (without fades) are permitted. These won't be executed precisely to the frame, but rather take place at the next GOP borders. For this reason, cuts should be set somewhat more generously.

## Hardware acceleration - CUDA, Quick Sync Video, OpenCL

MAGIX Audio & Music Lab Premium supports hardware acceleration for AVCHD exports as well as AVCHD and Blu-ray discs. This can noticeably speed up the encoding process.

The hardware acceleration that is applied depends on the encoder being used and the hardware built into the system.

- **Quick Sync Video:** Intel processors starting with „Sandy Bridge“ when using a standard MPEG-4 encoder (view page 162) or a Main Concept MPEG-4 encoder (view page 162).
- **CUDA:** NVidia graphic cards when using a Main Concept-MPEG-4 encoder (view page 162).
- **OpenCL:** AMD graphic cards when using a Main Concept MPEG-4 encoders (view page 162).

**Note:** Ask your graphics card manufacturer whether your graphics card supports hardware acceleration. Make sure that all the necessary drivers are installed.

If your system does support hardware acceleration, you can activate it in the "Encoder Settings" dialog under "Advanced..." > "Hardware acceleration".

## Advanced video settings

### Generic

### AVC preset

This is where the actual video stream that will be exported is selected.

- **Baseline**: According to ISO/ICE 11172-1/2 standard
- **Main**: Corresponds with ISO/ICE 13818-1/2 standard
- **High:** High Profile
- **SVCD**: Corresponds with MPEG-2 Super VideoCD
- **D1**: Corresponds with MPEG-2 DVD
- **DVD**: DVD video
- **Blu-ray:** Blu-ray-Disc
- **Blu-ray HD:** Blu-ray-Disc in High Definition
- **Sony PSP**: Sony PSP-compatible format
- **HD 1280x720p:** High Profile with a resolution of 1280x720p (progressive)
- **HD 1440 x 1080i**: High profile with a resolution of 1440 x 1080i (interlaced)
- **Apple iPod**: Apple iPod-compatible stream.

## Profiles

### Profiles

Profiles define the encoder properties that are supported.

- **Baseline profile**: The basis of applications with limited computing performance, especially for video conferences or videos on mobile telephones.
- **Main profile**: This profile was originally intended for the broadcast industry and for backup purposes. It has become less popular since the development of the "High Profile" for this purpose.
- **High profile**: This profile is used for broadcast and backup applications, and it is also used sometimes in the HDTV industry (**H**igh **D**efinition **T**ele**v**ision). This profile is used for HD-DVD and Blu-ray Discs, for example.

### Level

H.264 defines different levels. The level determines which bit rate and resolution are possible for the video.

With the "Auto level" option, the encoder determines the level automatically on the basis of the resolution of "AVC preset" video formats, the set bit rate and the profile.

## Frame type

"Picture type" specifies which parts of a frame should be used as the basis for the encoding:

- **Progressive Frame:** A single image from a video sequence, also called a fullscreen.
- **Interlaced Field:** This is half an image. Two of these are combined to produce a frame. Read more about this explanation regarding "Interlace (view page 175)".
- **Interlace Frame:** The encoder creates a „Frame field" which is the basis for the encoding.

## Field order

**Note**: This parameter is only available if the setting "Field" is selected for "Frame type".

In case of interlaced streams, the half-image sequence is set. Read more about this explanation regarding "Interlace (view page 175)".

## Slice count

A frame can be divided into multiple slices for encoding. Specify the maximum number of slices are permitted. If set to "0", then the number will be determined automatically.

## Rate control

The bit rate indicates how much data per second is saved in the video (playback speed). This makes the bit rate the deciding parameter with regard to the file size and quality of the video to be encoded. A higher bit rate means more quality but also a larger file size.

## Mode

- **Constant bit rate**: The constant bit rate should only be used if the device used to play the video supports constant bit rates.
- **Constant quantizer**: In this mode, a fixed colour quantization is used for the macro blocks. Under Advanced settings, a value between 1 and 32 can be set independent of the respective frame (I-Frame (view page 175), P-Frame, or B-Frame (view page 176)). The higher the value, the stronger the quantization: small values produce qualitatively high-quality images and the data rate increases, and larger values produce a reduction in data, but the quality suffers.
- **Variable bit rate**: The bit rate varies. For faster movements in the video, the bit rate increases, and for still images or slow pans, a lower bit rate is sufficient for creating the video in constant quality.

## Pass

Here you can choose whether you want to manually perform one-pass or multi-pass encodings (2-pass, multi-pass)

**Note:** The 2-pass option in the main dialog of the MPEG-4 export is recommended. This automatically performs two encoder passes.

If you want to perform 2-pass encoding manually, you must select "Multi-pass analysis" in the first pass and "Multi-pass encode" in the second pass.

- **Single pass**: The encoder process takes place without prior analysis. This requires the least amount of time.
- **Multi-pass analysis:** Analysis data is calculated during encoding and used for multi-pass encoding.
- **Multi-pass encode:** Assumes that an analysis has already been performed The analysis data is used during encoding to optimize the results.

## Bit rate (Bits/s)

- In "**Constant bit rate**" mode: Exactly those values entered are applied to be able to calculate the size of the video precisely.
- In "**Variable bit rate**" mode: The values entered here are applied to the video as an average as a guideline. The size of the video to be exported can only be approximated.

## HSS rate

This is the maximum bit rate that should be present in the video stream, i.e. maximum number of bits that may be transferred to the decoder.

**Note**: This option is only available in "Variable bit rate" mode. "Use HRD" must also be activated.

## Aspect ratio

In the film industry, this is an indication of the ratio between width and height of a rectangle, monitor, or screen.

There are 3 different sizes available:

- **Picture Aspect Ratio** (also **D**isplay **A**spect **R**atio, **DAR**)**:** This indicates the desired aspect ratio of the video to be exported. Here are some examples of typical aspect ratios: at home **4:3**, **16:9** (typical for TV sets) or **16:10** (widescreen-flatscreens, widescreen notebooks), **3:2** for 35mm films and photos. In cinemas you mostly find **1.85:1**.
- **Pixel Aspect Ratio (PAR, pixel aspect ratio**: Indicates the aspect ratio of individual pixels. The majority of computer monitors have quadratic pixels (PAR=**1:1**), for analog television monitors (PAL at 4:3) **128:117**.
- **Sample Aspect Ratio (SAR, also Storage Aspect Ratio)**: Aspect ratio of the saved resolution (number of pixels), e.g. 720:576 at PAL. It also calculates picture aspect ratio and pixel aspect ratio: **SAR = DAR / PAR**.

**Note**: In the standard case, the "Aspect ratio" remains set the way it is. You should only change the settings if the resulting video is exported distorted or stretched or if you need to correct the video because it is in the wrong aspect ratio.

## GOP structure

## Max GOP length

Determines the maximum GOP (view page 174) length. High values mean improved compression. Lower values create stronger security protection and enable improved access to individual frames for processing the video.

## Max b-frames count

The maximum number of b-frames (view page 176). Several cases of application, e.g. video conferences require "no b-frames" in order to achieve the shortest possible reaction times during transfer.

### Scene change detection

If this option is activated the scenes will be detected during encoding, thus allowing you to insert an I frame (view page 175) after a scene change.

## MPEG-4 simple

If MPEG-4 H.264 cannot be played back on your device, use MPEG-4 Simple.

> **Note:** We recommend that only advanced users make adjustments to the advanced settings. You can use the technical specifications of your playback device to help with this.

### Advanced video settings

### Generic

### MPEG-4 preset

Different presets located within the encoder.

**(A)SP@L0-L5**: (Advanced) Simple Profile in Level 0-5

**(Q)CIF (Common Intermediate Format):** CIF is a video format produced as soon as 1990 with the video compression format H.261. At that time, the format was used for video telephone conferences.
The "Q" in QCIF stands for "Quarter", and since resolution is halved in terms of height and width compared to CIF, the entire size is only a quarter of CIF.
QCIF was popular with mobile telephone manufacturers, since the resolution of 176 x 144 pixels was sensible for the first affordable SmartPhones (144 x 176).

**(Half)D1:** D1 corresponds with MPEG-2 DVD. HalfD1 has exactly half of the entire number of pixels, meaning that the pixel number of the height and weight is 2/3 of D1.

**720p:** Video stream with a resolution of 1280 x 720p (progressive).

**Apple iPod**: Apple iPod-compatible stream.

**Sony PSP**: Sony PSP-compatibler stream.

### Profile/Level

**Profiles:** Profiles define the encoder properties that are supported.

**Level:** The level determines which bit rate and resolution are possible for the video.

## Picture type

"Picture type" specifies which parts of a frame should be used as the basis for the encoding:

- **Frame**: A frame is a single image from a video sequence, also called a full image.
- **Field**: A half-image, two of which combine to produce a frame. Read more about this explanation regarding "Interlace (view page 175)".

## Field order

**Note**: This parameter is only available if the setting "Field" is selected for "Frame type".

In case of interlaced streams, the half-image sequence is set. Read more about this explanation regarding "Interlace (view page 175)".

## Slice count

A frame can be divided into multiple slices for encoding. Specify the maximum number of slices are permitted. If set to "0", then the number will be determined automatically.

## Rate control

The bit rate indicates how much data per second is saved in the video (playback speed). This makes the bit rate the deciding parameter with regard to the file size and quality of the video to be encoded. A higher bit rate means more quality but also a larger file size.

## Mode

- **Constant bit rate**: The constant bit rate should only be used if the device used to play the video supports constant bit rates.
- **Variable bit rate**: The bit rate varies. For faster movements in the video, the bit rate increases, and for still images or slow pans, a lower bit rate is sufficient for creating the video in constant quality.
- **Constant quality**: Similar to the "Variable bit rate" mode, the bit rate varies according to the video material. The quality depends on the selected profile and can be changed.
- **Constant quantizer**: In this mode, a fixed colour quantization is used for the macro blocks. Under Advanced settings, a value between 1 and 32 can be set independent of the respective frame (I-Frame (view page 175), P-Frame, or B-Frame (view page 176)). The higher the value, the stronger the quantization: small values produce qualitatively high-quality images and the data rate increases, and larger values produce a reduction in data, but the quality suffers.

## Bit rate (Bits/s)

- In "**Constant bit rate**" mode: Exactly those values entered are applied to be able to calculate the size of the video precisely.
- In "**Variable bit rate**" mode: The values entered here are applied to the video as an average as a guideline. The size of the video to be exported can only be approximated.

## Max. rate

This is the maximum bit rate that should be present in the video stream, i.e. maximum number of bits that may be transferred to the decoder.

**Note**: This option is only available in "Variable bit rate" mode.

## Pixel aspect ratio

Specifies the page ratio of the individual image points (pixels).

**Meaning**: Different television norms and the standard pixel ratio. Select a setting and the results are displayed as "X" and "Y".

**X/Y**: The actual pixel ratio. If under "Meaning" the setting "Custom" is selected, then a custom ratio can be set.

## GOP structure

## Max key interval

Determines the maximum GOP (view page 174) length. High values mean improved compression. Lower values create stronger security protection and enable improved access to individual frames for processing the video.

## B-frames count

The number of B-Frames (view page 176). Several applications, e.g. video conferences, require a setting of "0" for this, i.e. no B-Frames, in order to enable the shortest possible reaction times for transfer.

## Scene change detection

If this option is activated the scenes will be detected during encoding, thus allowing you to insert an I frame (view page 175) after a scene change.

# Audio codec:

Under "Audio", "AAC" (view page 171) is preset. If you have chosen an AVCHD transport stream from the presets, "AVCHD audio track" will be available.

## AAC

AAC was developed by MPEG, the Moving Picture Experts Group, as an audio data compression process that was specified as a further development of MPEG-2 Multichannel in the MPEG-2 standard.

AAC is equally suitable for encoding general audio information and not especially optimized for certain types of audio material.

AAC audio can be encoded with a sample rate of 8000, 16000, 24000, 32000 or 48000 Hz in either mono or stereo. By default, the sound is set to 48000 Hz stereo. The higher the sample rate is, the larger the resulting file and higher the audio quality. You can use the technical specifications of your playback device to help with this.

Advanced audio settings:

- The **bit rate** can be set between 6 and 512 kbit/s. 160 kbits/s is active by default. The higher the value is, the larger the resulting file and higher the audio quality. After a certain limit, additional improvements to audio quality will not be perceived. Bit rates under 64 kb/s are not recommended.
- As an **MPEG version**, set MPEG-4 or the older, proven MPEG-2 format.
- For the **File Header Type**, choose either RAW or ADTS. The "Header" indicates an explanatory head for the beginning of the file segment, which in fact takes up extra space, but is required for decoding under circumstances.
  - **RAW** indicates material which does not include a file header in audio format. The audio material is therefore transferred directly without any special additional information (raw).
    This requires that decoding routines are able to process the material without the explanatory file header. Especially in case "exotic" sample rates are set, this can lead to problems during RAW encoding.
  - **ADTS** indicates a file header type which contains information for encoded audio material. In case of doubt, select this file header type, since fewer problems can be expected in this case.

Profiles:

- **Low complexity**: Data is present in a form that hinders different decoding algorithms (noise replacement), but enables others (temporal adjustment noise formation).

**Note**: For example, Apple iPod requires "low complexity encoding". However, you don't need to worry if you select the right preset for Apple iPod in the export dialog.

### AVCHD audio track

If you have chosen an AVCHD, AVCHD transport stream, or Blu-ray (H.264) from the presets, "AVCHD audio track" will be available.

**Note:** 5.1 surround is only available in MAGIX Audio & Music Lab Premium.

## Multiplexer

The Multiplexer combines audio and video streams so they can be played back together on playback devices.

### Output format

**MPEG-4 file**: This is an MPEG standard (ISO/IEC-14496) with the original goal of supporting devices with less computing performance. Currently, MPEG-4 has reached a wide bandwidth of application, from HD video to support for mobile telephones.

**JPEG2000 file:** DCI (**D**igital **C**inema **I**nitiative) has been replaced by the JPEG2000 format for encoding movies. The current distribution and presentation of films has been taken over by digital projectors that play back **high-resolution Mj2 streams** in outstanding image and sound quality.

**3GPP file**: A standard supported by plenty GSM and UMTS mobile telephones. 3GPP is very similar to the MPEG-4 standard, but also supports formats that are not permitted by MPEG-4.

### Streamable format

Activate this option if you would like to create files that can be integrated into and played back on websites. This way the file can be played back without having to be completely preloaded.

**Note:** This setting is recommended if you choose the option "Export as website".

### For Sony PSP

Switch on this option if the video should be played back with the Sony PSP.

### For iPod

Switch on this option, if the video should be played back with the Apple iPod.

# MPEG glossary

## Motion estimation

Motion estimation is a further element for reducing data used in MPEG encoding.

Motion estimation also occurs in the B and P frames. The image difference that still exist after prediction (view page 177) are examined. Complex algorithms are used to search for an original occurrence of the macro block in the reference frame of each macro block of the P or B frame (these are units of 2x2 blocks specially combined for this purpose), which have been moved either by movement or by camera pan. They can then be left out in the P and B frame. Only the information by how far and to where the macro block has been moved is saved instead. This vector is called the motion detector.

In the General encoder settings, you can specify the quality of the final MPEG video. This factor also influences the time required for encoding. The longer it takes, the better the quality.

## Bit rate

MPEG is a format used for storage and transferring. With older formats (e.g. AVI) you could predict that 20 seconds of movie would result in 20 MB of data. The file size is this a direct measurement of quality.

This is different for MPEG: The amount of data available can be used differently for different display modes. 20 MB can be 4 seconds of DVD Video or 5 minutes Internet streaming in thumbnail format. The quality of an MPEG video is measured by the width of the created data stream, the bit rate. This is the amount of the transmitted data per time unit; it is stated in kBit/s or bit per second.

> Bits, not bytes are used, since the data word width has to address the transmission restrictions.

The file size can be calculated from the average bit rate, if its length is known:

F = (BRV + BRA) * t

| | | | |
|---|---|---|---|
| F=File size | BRV=<br>Video bit rate | BRA=<br>Audio bit rate | t=Length in s |

## Block

For almost all image file editing techniques the image is subdivided into 8 x 8 pixel blocks (image points). This should be noted if you would like to used user-defined image resolutions (width/height), and they should always be a multiple of 8.

# Chroma format

The color value of each image point consists of the color values for the primary colors red, green, and blue (RGB), and for traditional and technical reasons it is transformed into one brightness value (Y = 0.299*R = 0.587*G + 0.114*B) and two color difference values (U = R - Y, V = G -Y).

The Y value alone produces the black and white picture. These signal components allow brightness and color information to be handled separately. The first data reduction occurs when single rows comprising a picture are read. Because the human eye has a lower color resolution than a brightness resolution, the color components are recorded only for every other point of a row (4:2:2) for each four pixels grouped (4:1:0), i.e. color signal under-reading.

**4:2:2** This corresponds to the established TV standard. One piece of color information is transmitted per row for two pixels which corresponds to a 2/3 compression of the output data.

**4:1:0** This is the color coding used for DVDs and most other consumer video applications. For each 4 pixels grouped together on two rows, one unit of color information is saved. This corresponds to a output data compression of 1/2.

# Field

A half-image, i.e. two halves which combine to produce a frame (see de-interlacing (view page 175)).

# Frame

A frame is a single image from a video sequence which also called a full image. PAL video, for example, contains 25 frames per second, NTSC 29.97 frames. Video recordings, with the exception of computer animations and still frames, don't contain full images. Instead, they have double numbers of half-images (fields) which are transmitted in an interlaced state. However, we still refer to frames, since many predecessors of MPEG compression are based on such frames. Video editing literature usually refers to frames.

# GOP

**G**roup **o**f **P**ictures: The sequence of I frames and the P and B frames that belong to them.

e.g. I B B P B B P B B I ...

(This GOP has a length of 9, with 2 P frames and 2 B frames)

I frames contain the entire image information of a frame, while P and B have part of the information. So-called prediction (view page 177) and movement approximation are methods used for reduction.

The combination P B B is called a subgroup.

I frames must appear in regular intervals in the data stream for image and sound to be synchronized. Between the I frames only a limited count of P and B frames is allowed. This explains a few things: Since P and B frames contain only differential information, these differences will be larger with time, since more and more changes takes place from frame to frame. A large count does not make much sense, since GOP has a maximum length of 15 (4P, 2B) in PAL and 18 (5P, 2B) in NTSC. (More than 2 B frames between P frames is not allowed).

In a **closed GOP**, B frames of the last subgroup may contain only backward predictions or references to the preceding P frame, but no references to the following I frame, since it belongs to the next GOP.

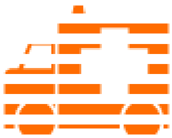
# I frames

Intra-frames: In these pictures, the entire image information of a frame is saved and only information from this frame is used ("intra-frame encoded"). In contrast to the I frame, P and B frames save only the differences between the current frame, and preceding and/or following frame are also found in MPEG video (P frame = "predicted frame", B frame = "bidirectional frame", see Prediction (view page 177)).

# Interlace

For historical reasons, pictures in a movie are always recorded and transmitted in the form of two fields, first the lines with even numbers and then those with odd numbers. These fields are alternatively displayed with a double-frame rate. The (lazy) eye of the viewer or the processing of the TV tube puts the two frames together to form one.

The output image



First field

Second field

You normally don't have to worry about field processing. The video material goes through the entire processing chain as fields and is exported again as fields or burned onto DVD or shown on TV when played back on a DVD as a full picture. Only in certain rare conditions is it necessary to delve deeper into this process. Two problems can occur:

### Interlace artifacts

To be displayed on a computer monitor the two fields must be combined to form a full screen.

These two fields are not the same, since two fields are created during the recording (between which a 1/50 of a second gap is evident). Moving objects can therefore produce artifacts at the vertical edges.

Typical interlacing errors

You can use so-called "de-interlacing" to avoid these artifacts. A picture in between the two fields is created (interpolated). So if you want to create stationary pictures from movies, then you should definitely use a de-interlace filter.

In the system settings (File menu-> Program settings) you can set the preview monitor display to use hardware de-interlacing during video recordings, for the video recorder, and for display in the arranger.

### Incorrect field rate

If you move around the series of fields in a movie data stream you can see strong jitter and flicker effects. Picture objects move in a backward movement - two steps forwards, one back - since a delayed field is shown before the previous one. This can happen in the processing chain if you export video material improperly with the wrong field order and then import it into different material.

We use MXV or MPEG "Top field first" format for all analog recordings ("odd" in other programs). DV-AVI on the other hand is saved with "Bottom Field First".

You can correct the field series for each video object in its object settings. See: Menu -> Effects -> Object properties

## P frames and B frames

P frames save only the difference between the current picture and the preceding I frame. The "P" comes from the term "prediction" which describes this process.

B frames save the differences between the current picture and the I or P frame preceding and following. This includes the information that was the same before and remained the same after the current frame. Both directions are analyzed (indicates the "B" in the name, i.e. "bidirectional-predicted"). You can read more under prediction (view page 177).

# Prediction

Prediction is a method of data reduction used by the MPEG format. The image elements already known from the previous or following frames are removed from the data stream.

## How does it work?

The encoder has a precisely defined GOP, for example IBBPBBPBB. This sequence is transmitted together with the encoder, which always knows exactly which kind of frame comes next. I, P, and B frames are differentiated.

> Hint: When we talk about pictures, we mean frames of the video output, and I, P an dB frames are the frames of the encoded video. Just as in movement approximation, blocks (8x8 pixels) are united into macroblocks (16x16 pixels) during prediction.

The first frame is always the I frame. It is completely encoded from the first picture. Afterwards, the 4th picture is analyzed for the creation of the first P frame. (As already said, the encoder, and later the decoder, will know that two B frames belong between them.) This image will also be completely encoded, and afterwards all macroblocks that haven't changed in comparison to the I frame will be deleted. They will be replaced by corresponding references for the decoder that tell it "you already know what should be shown here, and you can get it from the last I frame".

Now, the 2nd will be completely encoded, and all macroblocks identical to the first I frame **and** the following P frame will be removed. References to previous frames are called **backward predictions**, and references to following frames are called **forward predictions**. The third picture will be edited in exactly the same fashion.

The fourth picture we have already explained, and now we need the next P frame, or picture number 7. Pictures 5 and 6 are B frames again, which are compared to P frames to both sides of them (picture 4 and 7); these are followed by the last two B frames. These have a special place, since in closed GOPs, they may contain only backward predictions, and no references to the next I frame, because it belongs to the next GOP.

Something else: Since the decoder is no prophet, the P frames are always transmitted before the B frames! The GOP explained above will be encoded and transmitted in the order it is written.

| | | |
|---|---|---|
| Original GOP | $I_0\ B_{01}\ B_{02}\ P_{01}\ B_{11}\ B_{12}\ P_{02}\ B_{21}\ B_{22\ I1}$ | |
| Data stream | $I_0\ P_{01}\ B_{01}\ B_{02}\ P_{02}\ B_{11}\ B_{12}\ B_{21}\ B_{22\ I1}$ ... | for closed GOPs |
| | $I_0\ P_{01}\ B_{01}\ B_{02}\ P_{02}\ B_{11}\ B_{12}\ I_1\ B_{21\ B22}\ P_{11}$... | For open GOPs |

Due to this nested structure, it is easy to see that during direct editing of MPEG material, complicated computations have to take place! These are made easier using a **frame table**. A frame table contains a list, where the information of every frame in the data stream is found, identifying the type of frame it is.

Using Movement prediction (view page 173) P and B frames are likewise reduced.

# Quantization scaling

The single pictures in MPEG are saved using a compression method comparable to JPEG with bitmaps and associated with quality loss. For this single images are divided into 8 x 8 blocks (view page 173).

Each one of these blocks is then transformed into an 8 x 8 matrix (a table with rows and columns) using a **DCT** (discreet cosinus transformation) mathematical method. Each of these values is produced using all 64 individual pixels of the block, but the values in the matrix are ordered in such a way that the image information is ordered according to its importance.

This matrix will then be multiplied by another matrix, i.e. the **quantization matrix**. Exactly how and why this matrix must be created is the biggest secret of encoder programmers, since this determines the quality of the whole encoding process. What is known is that the result should contain as many zeros as possible! These zeros correspond to the "unimportant" image elements mentioned and will not be transmitted in the data stream.

Depending on the encoder parameters regarding the target bit rate, fewer or more values of the matrix will be declared unimportant by dividing the quantization matrix by the **quantization scaling factor**. Since only whole numbers are used, a division can produce a zero is the remainder is discarded.

This factor is also a direct measure of the sought image quality of the MPEG data stream, since the "Q" in "Q" factor stands for quantization and quality.

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