SOUND FORGE

SOUND FORGE AUDIO STUDIO

English language manual

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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.
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  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.
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  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

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

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

Europe
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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

What is Sound Forge Audio Studio 12?

Sound Forge Audio Studio 12 is an audio editing program used for recording, editing, exporting and converting audio files. For example, you can digitize, repair and store your old records and cassettes. Or you can produce your own podcasts and optimize the sound in your videos. Produce audio material in 32-bit/384 kHz studio quality and apply more than 20 professional filters and effects.

Record your audio material directly from the turntables, cassette decks or microphone. Cut, edit and optimize your records and sound. Then export your material conveniently as MP3, FLAC or burn to a CD.

Features

- 64-bit architecture for more power and performance
- VST-3 interfaces for external audio plug-ins
- Non-destructive Slice Editing
- Spectral Cleaning: spectrogram noise removal that does not audibly affect the wanted signal
- Convert and save audio in all standard formats
- Burn Red Book Standard audio CDs
- iZotope Ozone Elements (plug-in for preset-based mastering)

Quick Start

In this chapter we will introduce all of the important features in Sound Forge Audio Studio 12. You don't need any prior knowledge for this; just some time and hard drive space.

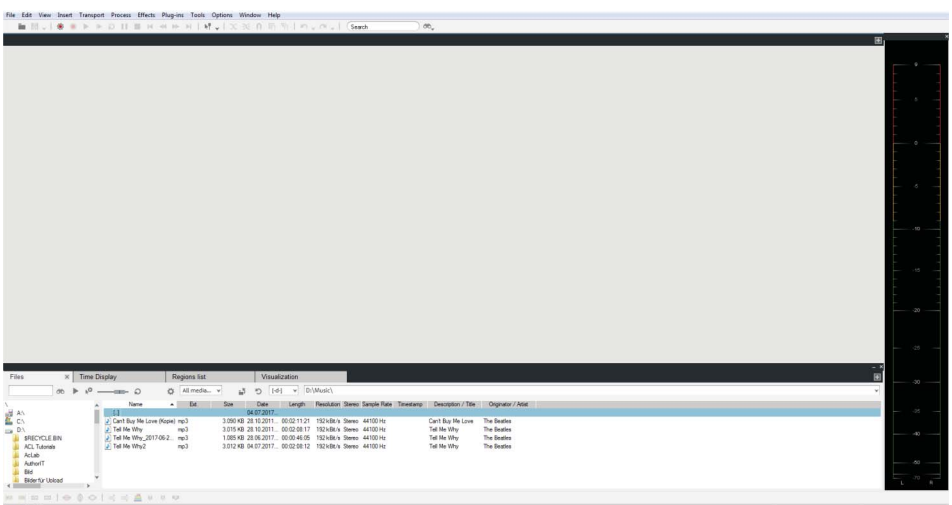
- With Sound Forge Audio Studio 12 you can load audio material and clean it, edit it, and export (view page 9) it using a wide range of special features.
- Analog material, such as records or tapes, must be digitized first so that it can be processed by the computer. Podcasts can be recorded and processed directly in the program. To do this, use the recording feature in Sound Forge Audio Studio 12 (view page 20).
- You can load multiple files at once to burn them to an audio CD or to create an MP3 playlist for export (view page 24).
- Load videos and optimize the audio tracks (view page 31).

We'll explain all of these applications here step by step.

Audio cleaning

Load and play audio files

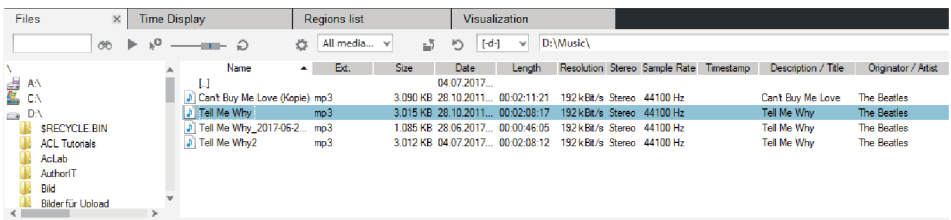
After starting the program, Sound Forge Audio Studio 12 displays an empty project window.



There are two ways to load an audio file:

- Via File > Open (view page 36) (Shortcut: Ctrl + O)
- Via the Explorer (view page 60) in the lower third of the program window

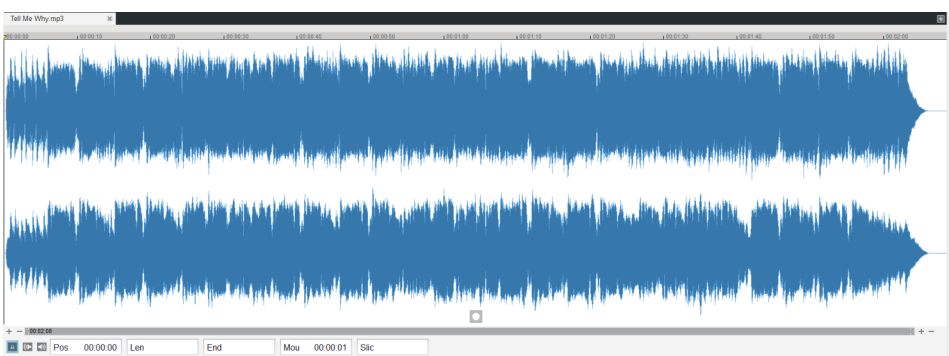
The Explorer is a file manager and works in a similar way to the File Explorer in Windows. You can search all drives and folders on your computer and then select and load a file.



Ctrl + mouse click opens several objects. For a group of files, click on the first file and then on the left file with Shift + click.

Drag & drop the selected files to the upper section.

After import the file will appear as a waveform in the upper section of the program window. In the case of several files, a tab will open for each file. To load several files to the same window, use the other Import method (via "File" > "Open" (view page 36).



The waveform provides you with an image of your audio material.

Some noises can be recognized straightaway. Crackling, for example, creates a clearly identifiable peak in the waveform.

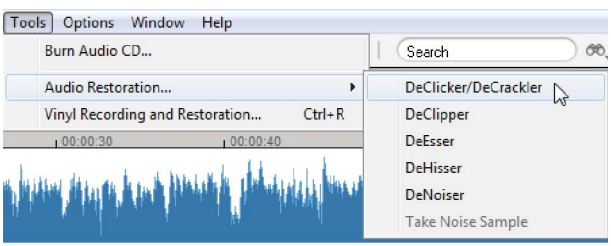
In the toolbar above, you'll find the transport controls for playback, stop and moving the playback marker.

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.

Drag a selection

You'll find most effects for editing in the Effects menu. Special functions for noise removal are in "Extras" > "Audio restoration".



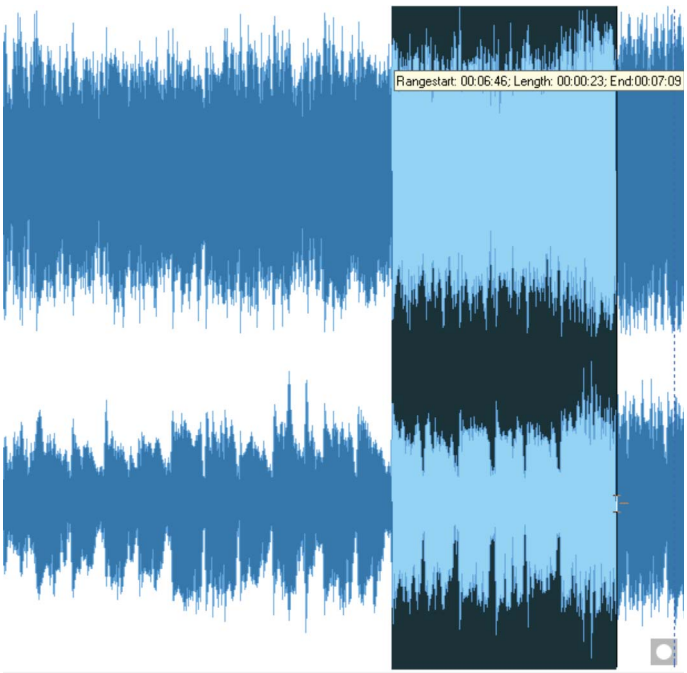
Before applying a restoration effect (or other editing effect), decide what kind of editing you want to undertake. The most important question is: Do I want to edit just a certain section of the material - or the entire audio file?

For instance, there is an option for filtering out constant noise in a cassette recording using the DeHisser. This kind of noise usually exists through the length of the whole file. So here, you would use an editing function for the entire audio file. In this case you open and apply the editing effect directly.

Other noise, for instance a one-off crackling noise or a sudden rumbling only affect a certain section of the material, so to edit this you need to drag out a selection.

Click in the waveform using the mouse and drag & drop the data you want to edit out. You can either select an individual channel or both stereo channels together.

Drag a rough selection around the passage you want to edit.



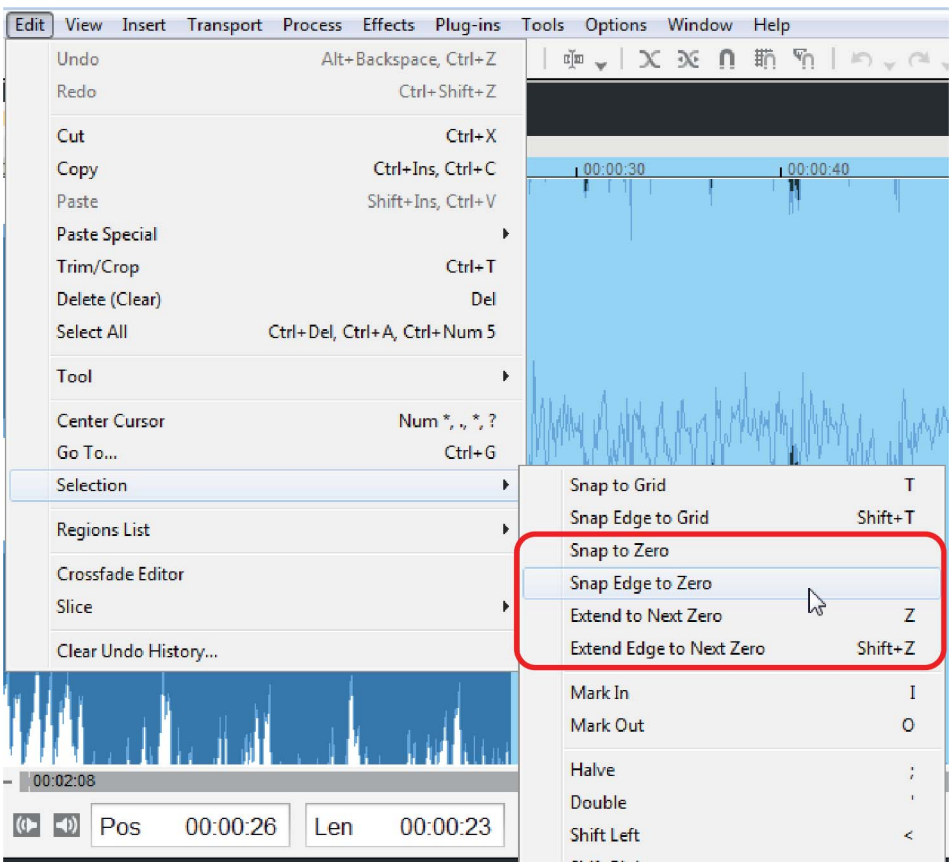
You often need to make a very exact selection in order to choose only the passage where the noise is located. Sometimes, it's necessary to meet the zero crossover exactly in order to avoid crackling.

Tip: Use the Soft Cut mode to automatically create a small dissolve to prevent any crackling at the edges.

- Left and right below in the waveform window you'll see the Zoom buttons with "**+**" and "**-**" symbols. Zoom using these buttons into the display. You can also zoom using the mouse wheel if you're using a mouse. You can zoom in as far as the sample level, until the oscillation and zero crossover can be seen clearly.

- Move the mouse pointer to the borders of the selection until it becomes a double arrow, then drag the border to the right location.

- To set zero crossover precisely, you can also use the command under "Edit" > "Selection".



Tip: For more information about options in this submenu, go to the Menu chapter (view page 86).

- To deselect the current selection, click in the timeline.

Cleaning functions

There are many different types of noise. Let's start with constant humming or irritating hissing. These kinds of noise are usually caused by tape recorders, record players or substandard microphones and are audible throughout the complete track.

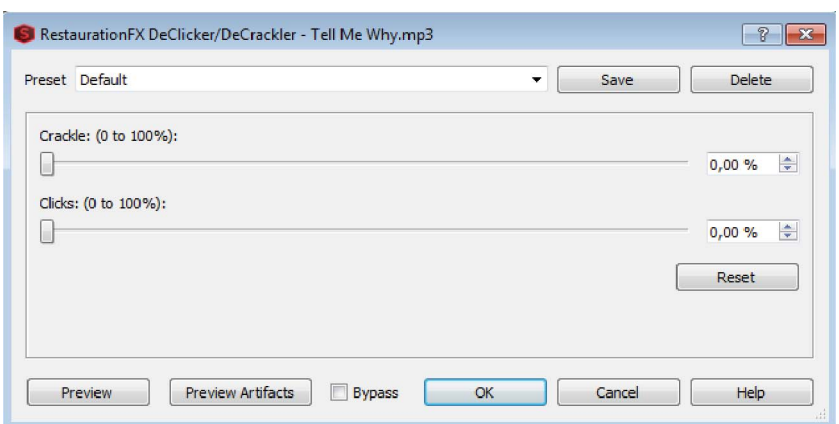
Follow the next steps to get rid of it:

- Remove the stretched sleection if necessary so that the effect can be applied to the entire track. (Alternatively, select the entire track by pressing Ctrl + A.)
- Select the most suitable restoration function for the noise on your track.

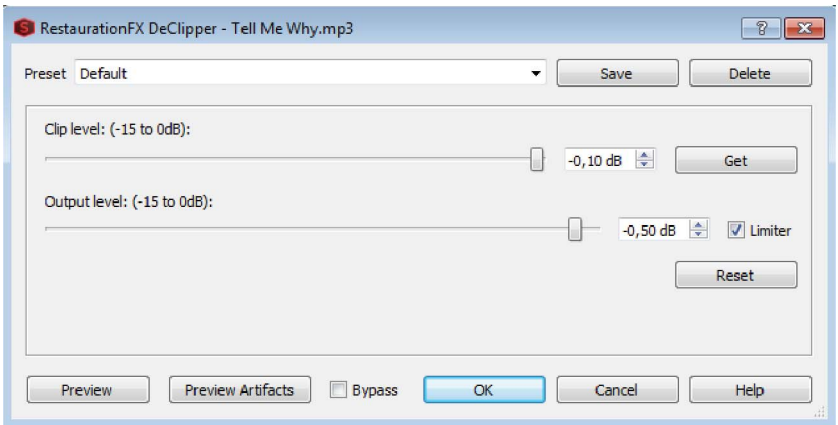
It's not always possible to know which function is best for your needs, so try out a few different options first. You can undo edits with Ctrl + Z.

Here's a basic overview, however:

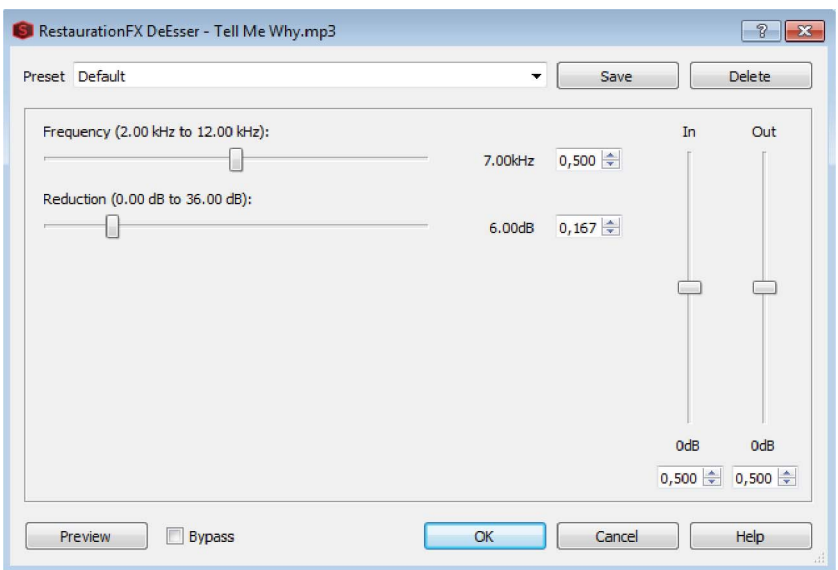
- The **DeClicker/DeCrackler** allows you to remove crackling noises from old records.



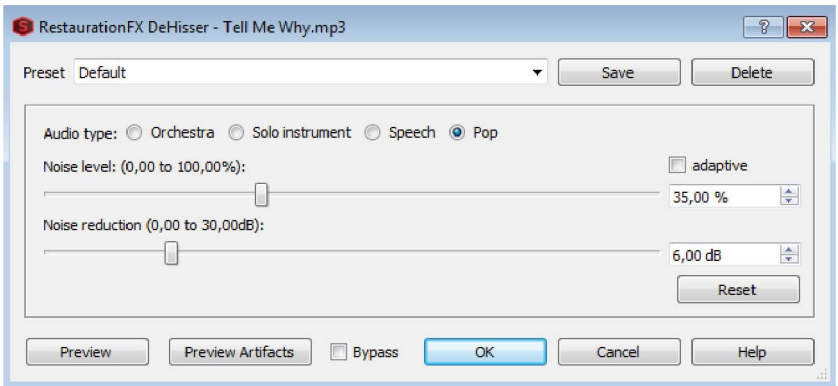
- For more severe crackling, use the **DeClipper**.



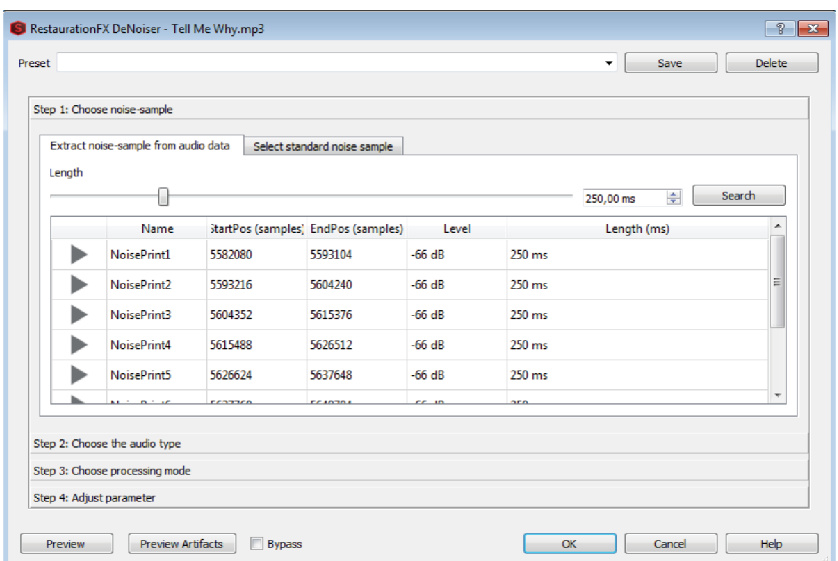
- The **DeEsser** is used for editing vocal recordings: more precisely, for reducing sibilants resulting from the consonant S.



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



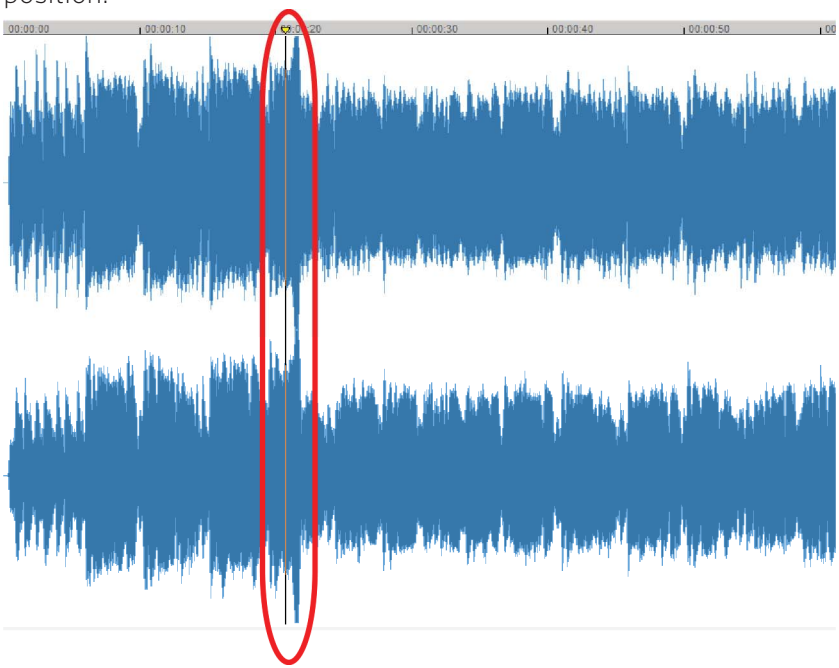
- The **DeNoiser** removes noise like mains hum, hissing, ground wire noise, interference from audio equipment with high-impedance outputs (e.g. turntables), low frequency impact sound or rumbling sounds from vinyl. It works using a noise sample. First, find a sound sample in the dialog that contains the noise and then in the second step filter the interference frequency out of the material.



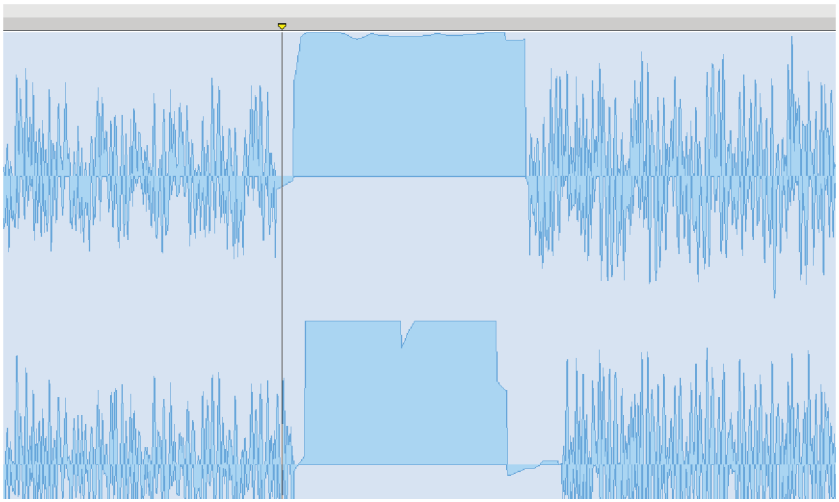
Retouching short noises such as clicks or pops

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

First, search for a section in the material where you hear crackling and set the playback marker before that position.



- Zoom in so that you can see more details. You can use the zoom buttons in the bottom right corner of the track window.



You can now edit this position using a pencil tool.

Select the pencil tool under "Menu" > "Tool".

The mouse pointer will now turn into a pencil which you can use to draw directly into the waveform. When you click, the zoom level increases automatically, and you can draw.

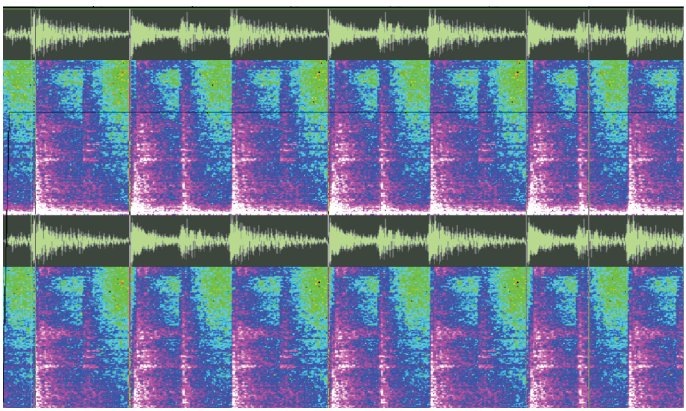
The goal here is to try to turn the flat form of the crackling into more of a peak and to move it slightly downwards. The tool is especially suitable for very short instances of crackling.

If you want to completely remove the sound from a certain spot, you can also select the spot and use the "Mute" feature in the "Process" menu.

Spectral Editing

Other kinds of noise are hard to see in the waveform, but are clearly audible during playback. These include things like a single cough or a passing car — anything that isn't particularly loud, but still negatively impacts the audio.

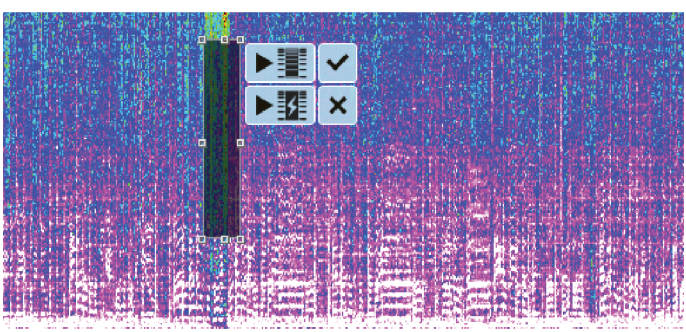
To deal with these noises, select the "Spectral Editing" tool. This will switch the waveform display to the spectral display.



The spectrogram displays the frequency proportions in a time curve. The volume of frequencies is visualized by a color code. A continuous sound is displayed by a pattern consisting of horizontal lines, which correspond to the sound components or overtones of the sound. A distortion with an impulse quality is seen as a vertical line.

Place the playback marker at the position of the noise and, if necessary, zoom into the display a bit.

With the mouse, make a rough selection in the spectral display of the area containing the noise.



- The upper Play button plays the corresponding part with effects applied. You will hear the results of the editing immediately.
- Clicking the checkbox will include the edits in the audio material.
- For comparison's sake, you can use the lower Play button to listen to the same part without any effects.
- Click the X symbol to cancel the edits. If you do this, no filters will be applied.

We use an interpolation technique to prevent there being any audible gaps: parts of the wanted signal around the interference are calculated at the highlighted section.

Save or render audio files or burn to CD

At the end of the editing process, there are several options for what you can do with your material:

- **Overwrite the original file**: Select "File" > "Save As" and overwrite the file in its original folder.
- **Store the modified file as a copy**: Select the option "Save" or "Save as" in the File menu and give the file a different name. You now have the original data saved as a backup. When saving normally, a different file name combining will be created automatically which comprises the original name and the current date.
- **Save all open files**: You can use the "Save all" command in the File menu to avoid having to saving each file individually.
- **Save the file in a different file format (convert/export)**: From the File menu, choose Save As and then select a different file format in the dialog. The encoder settings for the selected format can be accessed via the Format Settings. The available formats are WAV, MP3, MP3 with an external encoder, WMA (Windows Media Audio), AAC, FLAC, OGG (Ogg Vorbis) and AIFF.
- **Burn audio CD**: To do this use the option for burning CDs in the Extras (view page 24) menu.

Record vinyl/cassettes/podcasts

In the previous chapter we showed you how to improve audio material with the range of tools and restoration functions found in Sound Forge Audio Studio 12. We also learned about the functions that are suited for vinyl recordings (DeCrackler) or cassette recordings (DeHisser).

Here. we'll expand on these topics and explain how to digitize your vinyl or cassette collection, and illustrate how to record a podcast or speech.

Basic knowledge about recording with the PC

The record feature converts analog audio signals – records, cassettes, sounds, speech – into digital data, which can be saved on the PC and edited with Sound Forge Audio Studio 12.

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 as 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.

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

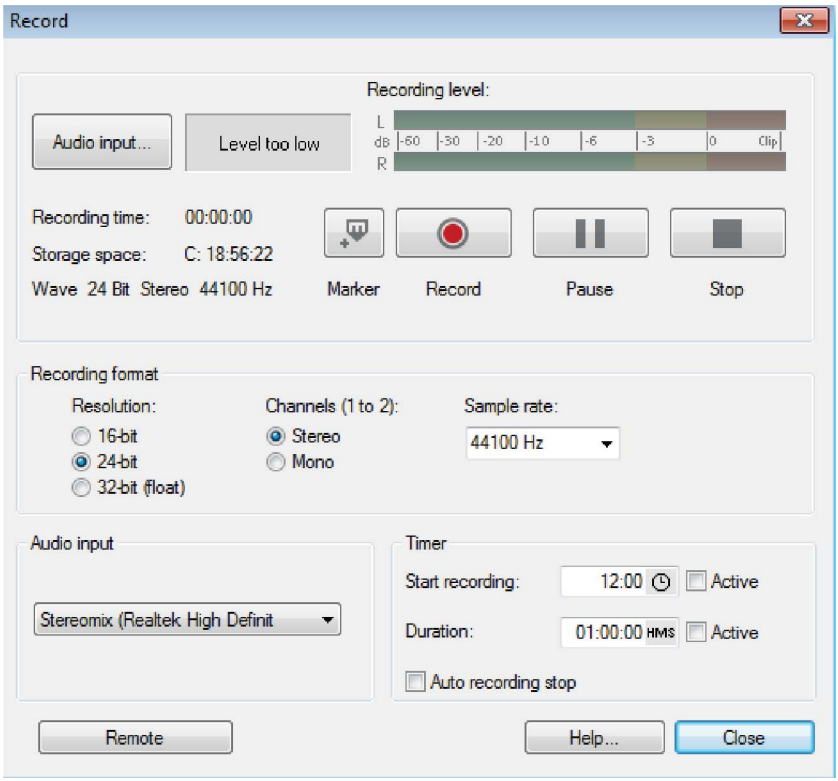
have to be connected to the computer's line-in. Hi-Fi amplifiers usually have RCA (composite) connectors and the computer has mini stereo connectors. You need to then get a suitable cable with two small RCA (composite) connectors and a mini stereo jack.

- 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).

Audio Setup

Before starting recording, view the options in the "Audio Setup" dialog.

This dialog can be accessed via the toolbar.



Do a test recording or speak into the microphone to see whether the level display moves up. If in doubt, click the "audio input" button to open the input and level automation (view page 40) for automatic selection of the correct input signal and level.

Additional Steps

The actual recording process can be started by pressing the large "Record" button in the dialog or via the separate "Record" button in the toolbar.

After recording, we recommend the following actions:

- Restore the recorded material (view page 9)
- Cutting: Precisely align the beginning and end of the overall recording or of each part (view page 24).
- Export the file or burn (view page 20) it to disc.

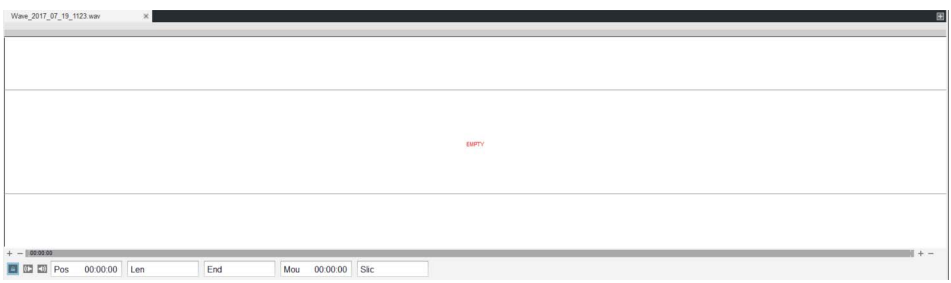
Burn audio CD/Export MP3 song list

Next, we want to edit several songs or audio files together and then burn them to audio CD or export as an MP3 song list.

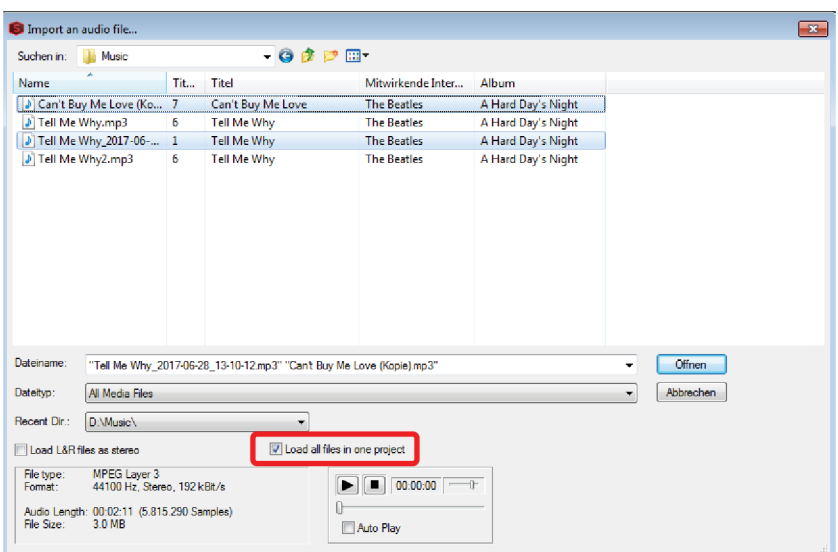
Group audio files

First group all of the songs or audio files you want to include on the CD or export.

Create a new, empty audio file in "File" > "New". Then add the audio files that you want to burn or export together. Transfer the presets from the settings dialog and click OK.



If files are in the form of separate audio files, select "File" > "Open" and activate the option "Load all files in one project".



Use the Ctrl key to select files in sequence and click "Open".

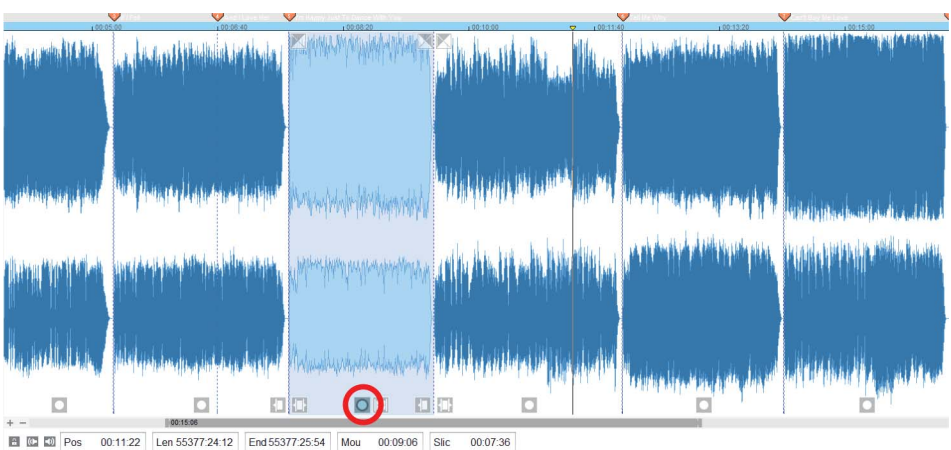
- To attach more files, select "File" > "Open and append".
- To transfer a certain range from another, open the file, drag a selection and copy it to the clipboard using Ctrl + C. Then, open the burn file, place the playback marker at the position where the new material should be inserted. and add it using Ctrl + V.
- To move an audio file to another position, the simplest method is by using the CD Index manager (view page 29).
- If you want to remove a specific section, drag out the selection and click the Del key. This deletes the section and brings the material behind it to the front.

Slice Editing

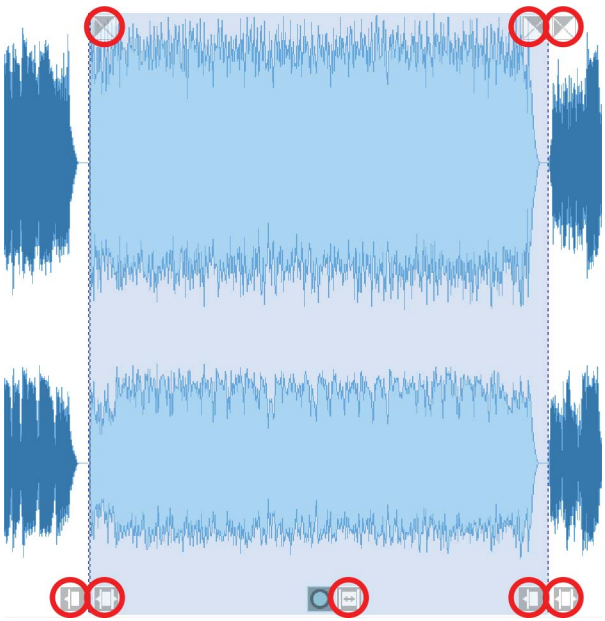
Sometimes it is necessary retrospectively trim the beginning parts of a song or vocal recording. Especially with sections of audio which have been pasted in, crackling can arise at the edges, which needs to be removed.

This is where slices come in. When pasting material into an audio file, or during selection editing, slices are created automatically. Slices are like virtual selections whose borders and contents can be moved at a later point in time.

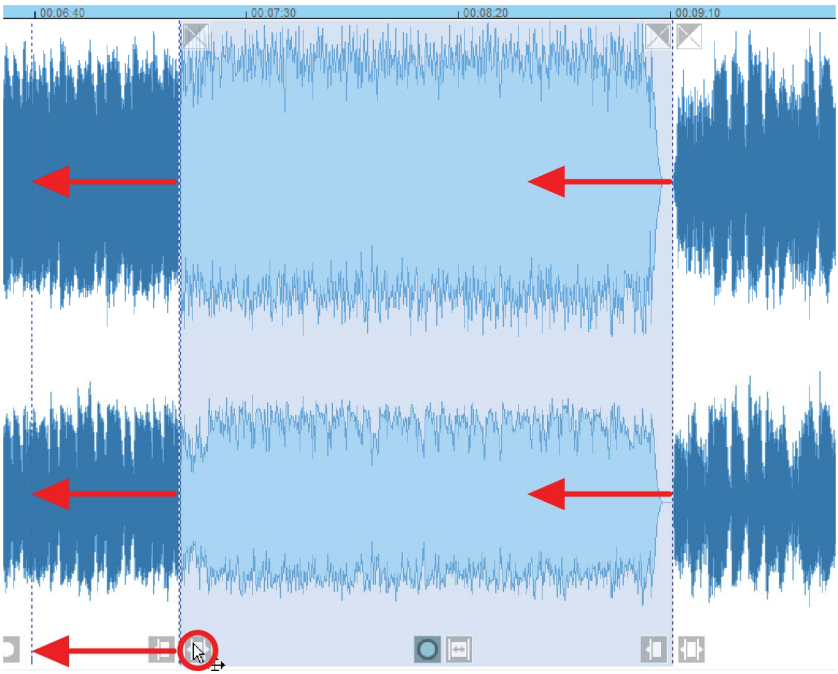
To select a slice and active slice editing, click the slice selection button.



The slice will now appear in a different color to show that it has been selected. At the same time, editing buttons will appear.



The lower buttons can be used to change the borders of the slice in different ways. The material can be moved over the neighboring slice like a sheet of paper. The slices that follow are moved along with it.



Dragging in the opposite direction puts the border back where it was, and the overlapped material is pulled out from the slice in front of it.

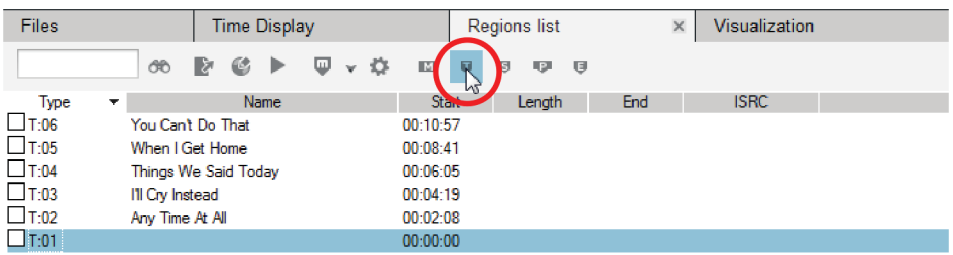
The button in the middle can be used to move the contents of the slice, if the slice represents a section of a longer audio file.

The two upper buttons are used to create fade-ins and fade-outs.

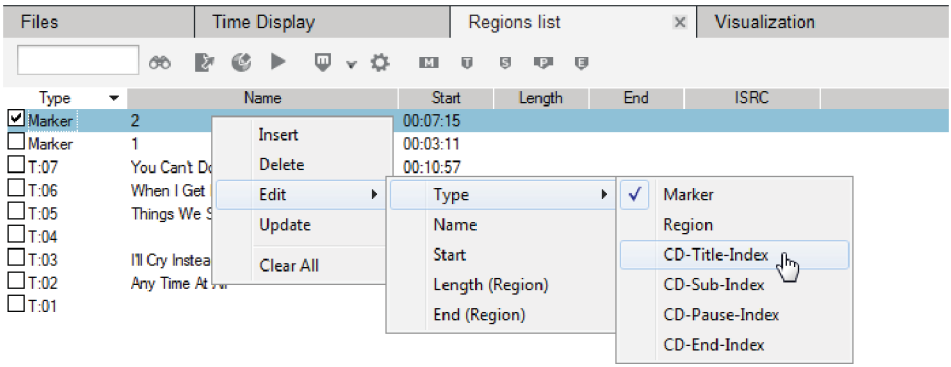
Set Track Marker

You may have noticed when combining audio files that a marker is automatically set for each file in the timeline above. These markers are CD track markers or "CD Track Index" markers

Additional track markers can also be set "by hand". To do this, simply set the position line at the beginning of a new track and click the Track Marker button in the Regions List.



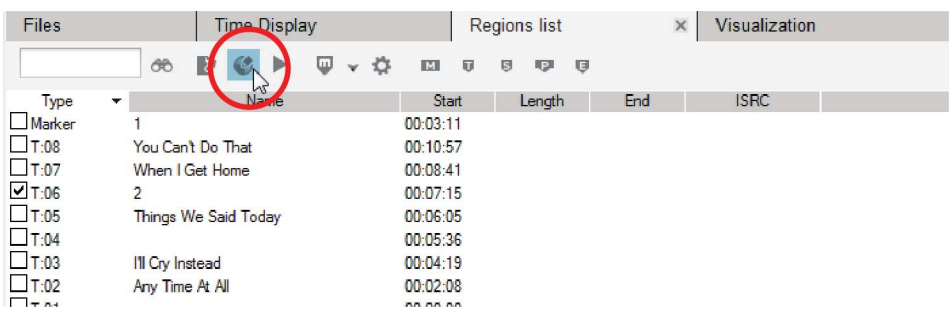
You can also redefine each normal marker as a CD track marker (option: "CD Track Index") in the Regions List context menu.



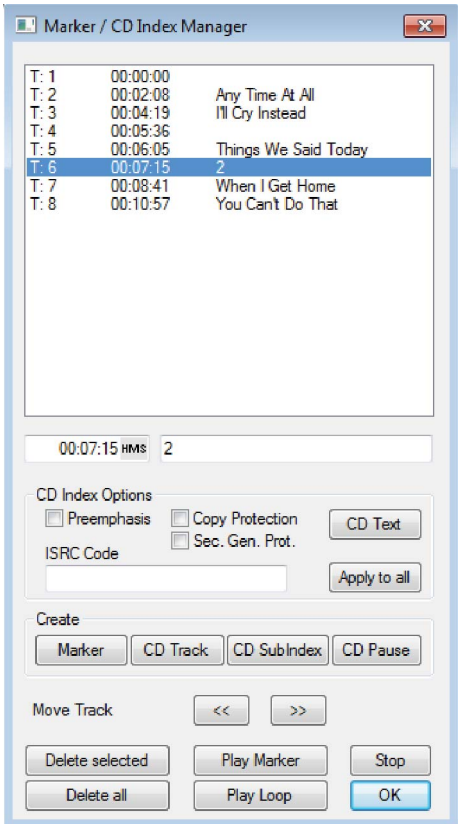
You can click on track markers in the timeline and drag & drop them to new positions or even delete them.

Checking and Moving Tracks

Before burning the audio to CD, it is best to check once more that all tracks and track markers are in the right places. To do this, open the CD Index Manager from the Regions List.



The CD Index Manager displays all track markers in a clearly arranged list. If you have also created CD subindex markers and pause markers, you can also check those here.



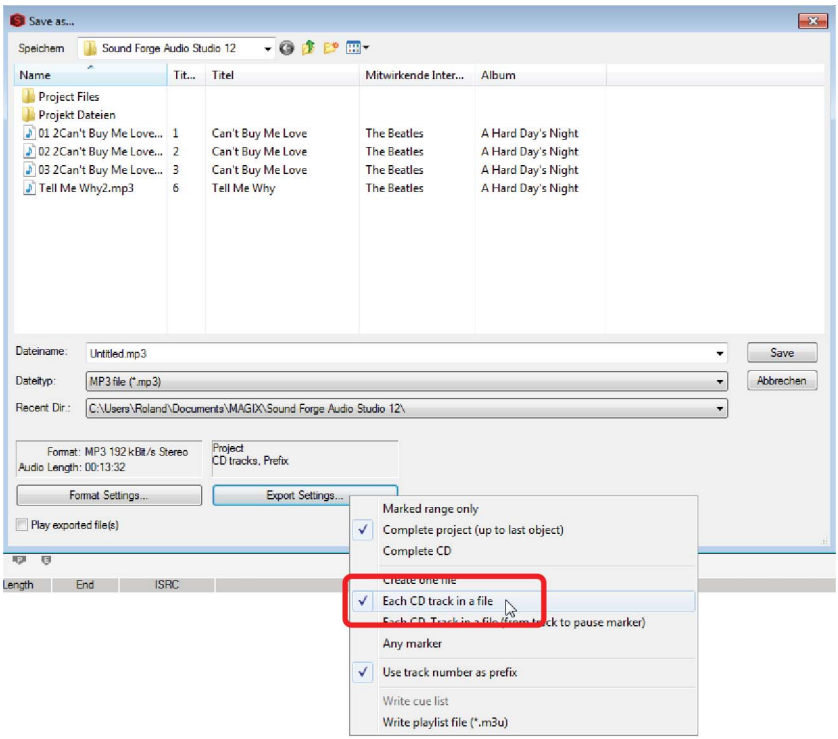
The CD Index Manager lets you easily move tracks by clicking on a track marker and repositioning the track in the arrangement with the double arrow keys (<< and >>). A few seconds is all it takes to get all your tracks in the right order.

Burn audio CD or export file list

At the end of a mixed audio file, burn all the tracks together to an audio CD or export individual tracks as a file list.

- To burn, select "Extras" > "Burn Audio CD".

- To export, select "File"> "Save As" and open the "Export Settings" first. Each track can be saved to a separate file.



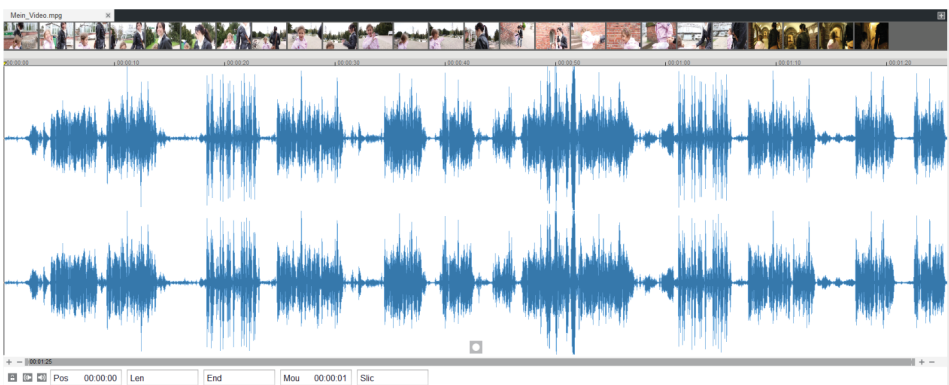
In the "Format Setting", you can set other formats (e.g. MP3), then specify a folder for the track and a name (e.g. album name) and click "Save".

Video Sound Editing

Sound Forge Audio Studio 12 can also be used to edit the audio tracks of videos. Video files are loaded exactly like audio files:

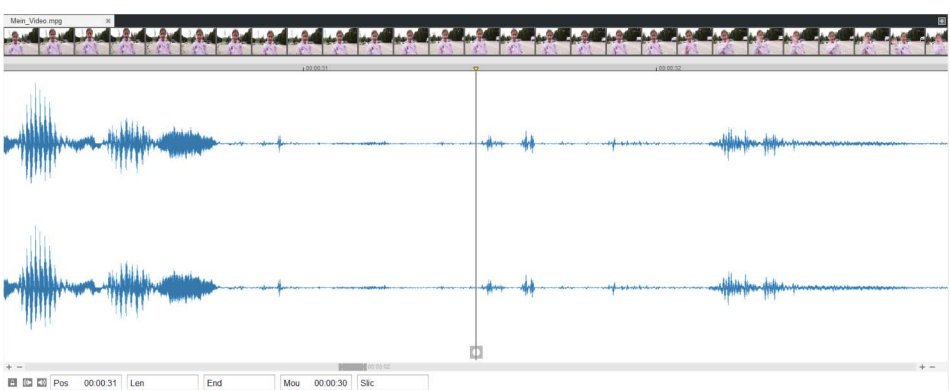
- By going to "File" and selecting "Open" (view page 36) (Shortcut key: Ctrl + O)
- By using the Explorer (view page 60) in the bottom third section of the program window.

After video files are loaded, the picture track appears above the audio track waveform.



The filmstrip provides a visual orientation of how the video progresses.

If you zoom into the audio track, the picture display will also zoom in down to the frame level.

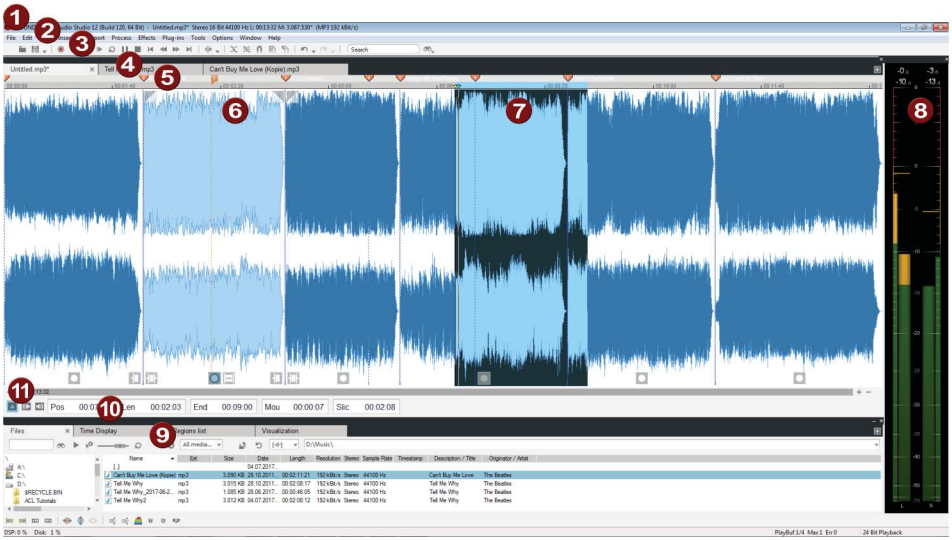


The audio track can now be edited (view page 9) exactly like an audio file.

You can also convert it directly into an audio file, which can be useful when making music videos, for example. To do this, go to "File" > "Remove video". This removes the picture track and changes the file format in the file tab from the original video format (e.g. *.mpg) to the standard *.wav audio format.

After editing, you should save the file by going to "File" > "Save". This will create a new video file which contains the edited audio track.

Program interface overview



1. **InfoBar:** At the very top, you'll find the Sound Forge Audio Studio 12 name and version number, as well as the name and properties of the currently loaded audio file.
2. **Menu Bar:** Here you'll find all the menus available in Sound Forge Audio Studio 12.
3. The **Upper Toolbar** contains the most important buttons for saving, loading, playback and editing. Below, there is a second toolbar. You can add other buttons by using the "Edit Toolbar" command in the context menu.
4. **Timeline**: Right click to adjust the time ruler. You can also set markers and regions within the audio file here.
5. **File Tab:** Here you can switch back and forth between loaded audio files.
6. **Active Slice:** Slices are separate sections within a file. Slice borders can be freely moved (view page 52).
7. **Selection:** Playback or editing area, e.g. for effects or cut editing (view page 42).
8. **Peak Meter:** Shows the peak control of the audio material during playback.
9. **Manager**: Here you'll find the different managers (view page 59) for file and marker management, for visualization and for time display.
10. **Position Fields:** Shows the positions of the playback marker, range boundaries, the mouse and the active slice. You can double-click to edit

these values numerically. You can right-click to change the number of fields and the type of displayed time positions.

11 **Channel selection:** The channel lock all the way to the left ensures that the channel selection is maintained for editing. The other two buttons are used to remove the left or right channel from the selection and mute it.

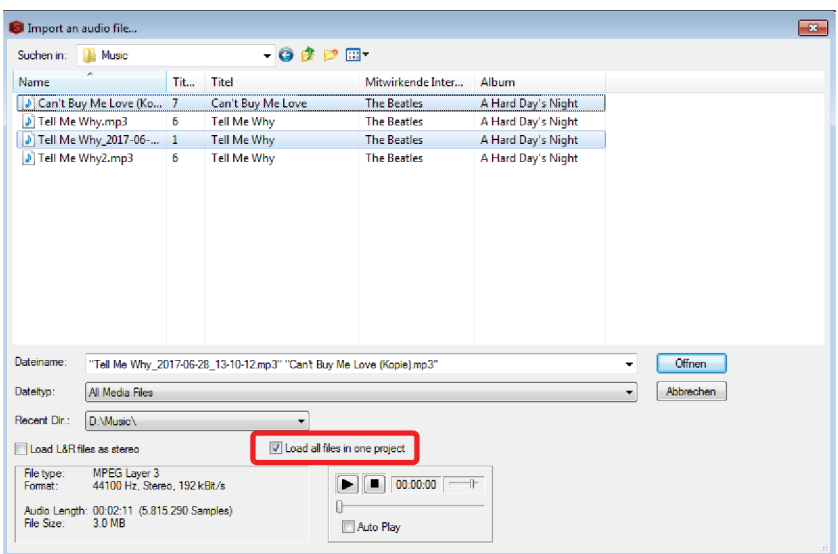
Working with Sound Forge Audio Studio

Loading audio files

Load files: Load new audio files either via the menu "File" > "Open" or via the Explorer in the lower section of the interface.

Waveform window: The audio file is opened in a separate window and displayed as a waveform (as long as it is not attached to an existing audio file). In the title bar of the audio file window, you will see the name of the file, the bit resolution, the length of the sample and the resulting required memory.

Load several files: To load several files together (for instance to burn to CD), tick the option "Load all files in one project" in the "Load audio file" dialog. Then mark the files in sequence with Ctrl + mouse click and click "Open". To load a whole file list, select the first and the last file in the list with Shift + mouse click.



Add audio files from existing audio files: The File menu includes the option to add a second audio file to the currently opened file ("File" > "Open and append"). This adds new material to the end of the existing file and sets a CD track marker.

Drag & drop to create new audio file: To quickly create an audio file as a copy of part of a range or the entire audio file, drag the selected range in your audio file out of the editing window and drop it on a free space in the program background.

Load L&R files as stereo files: Sometimes stereo recording channels lie in two separate files - L&R (= left and right) files. This kinds of files can be automatically loaded as normal stereo wave files. To do so activate the "Load L&R files as stereo" checkbox in the "Load audio file" dialog.

Record audio

Basic knowledge about recording with the PC

The record feature converts analog audio signals – records, cassettes, sounds, speech – into digital data, which can be saved on the PC and edited with Sound Forge Audio Studio 12.

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 as 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.

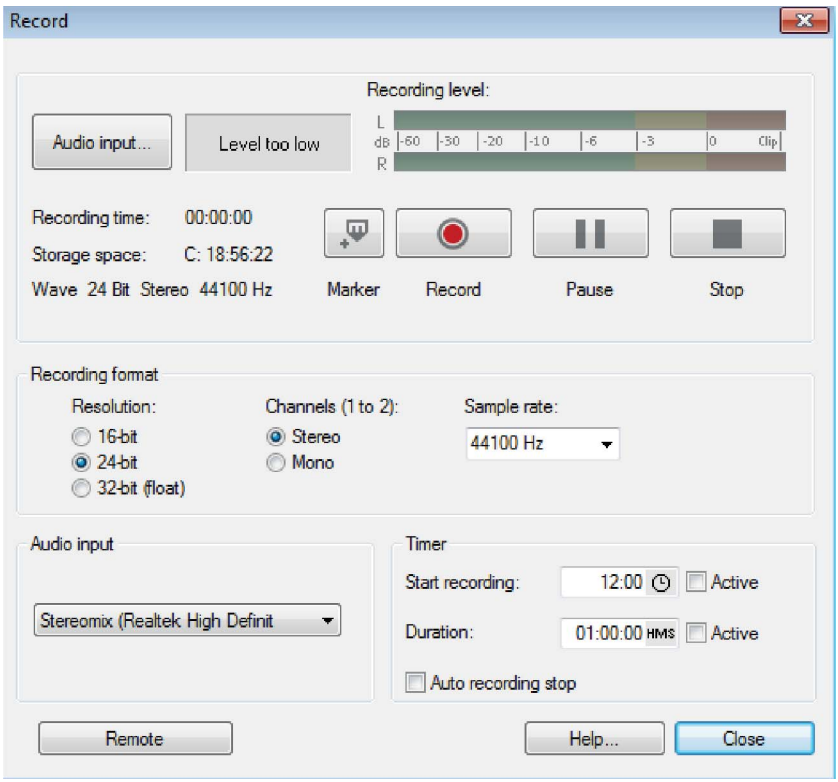
Connecting the recording sources

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 Dialog

Access this dialog via the toolbar or "Transport" > "Record..."



Audio input: Opens the input and level automation (view page 40) 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) .

Marker: Even during recording, you can set track markers by clicking the corresponding button in the recording dialog.

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

Note: The dialog can not be closed whilst recording is in progress to avoid accidentally stopping the recording.

Pause: Stops the recording. Click the button again to resume.

Stop: This button ends recording.

Recording format:
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 41) 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.

Audio line-in: 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.

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.

Remote: The Sound Forge Audio Studio program interface is hidden until this dialog.

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

Close: Closes the recording dialog.

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.

24-bit Audio Support

Note: 24-bit recording is not available in .

Not only can audio data be recorded in Sound Forge Audio Studio 12 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 Sound Forge Audio Studio 12 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.

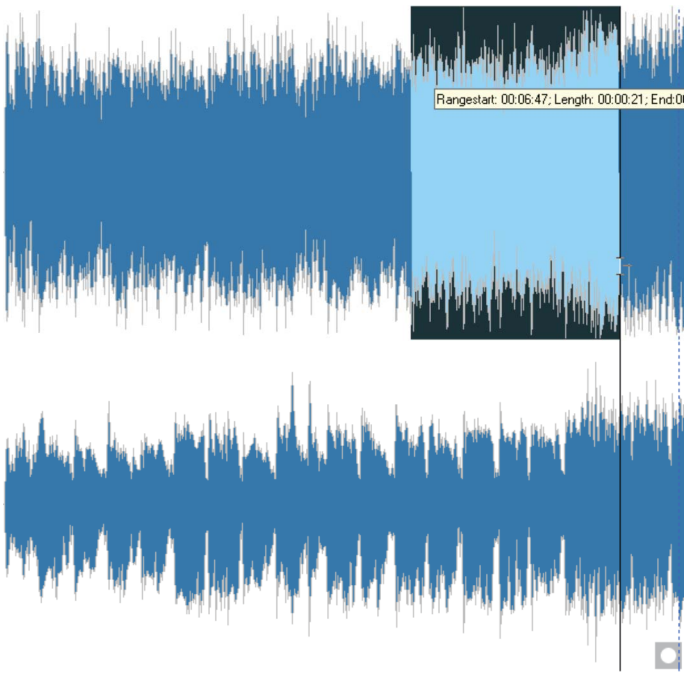
Compared with 16-bit recordings, 24-bit recordings take up double the memory on the hard drive (by saving them as 32-bit float data), but considering the capacity of modern hard drives this is a good compromise for the increase in quality.

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.

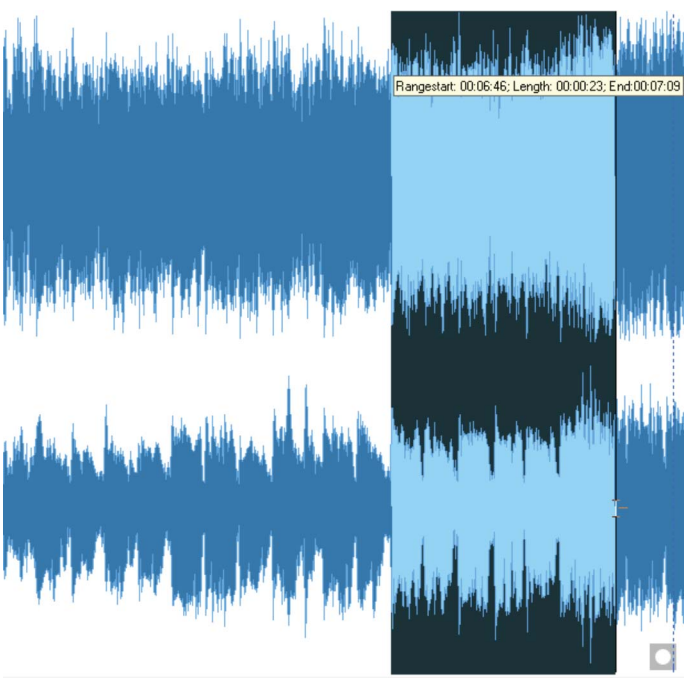
Selection

A selection is a defined section of audio data that can be used as playback range or for editing. Material within the selection can be cut or copied and pasted in another position or you can add effects or plug-ins to it.

For stereo material, the selection may include either one channel or both channels simultaneously.



To select both channels, drag & drop the mouse pointer a little above or below the channel border.



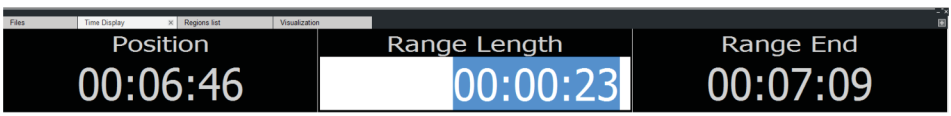
You can also use the grid feature when creating a selection ("Edit" > "Selection" > "Align to Grid").

Lock channel selection: The lock channel symbol on the far left in the upper section of the manager ensures that only channels switched to audible can be selected for editing. This avoids the possibility of working on just one channel by accident.

Change selection borders: Place the mouse pointer on the edge of the selection bar until it turns into a double arrow. You can now reset the selection borders by clicking and dragging in the horizontal direction.

Move selection: Click inside an existing selection on the timeline, hold the mouse button down and move the selection horizontally.

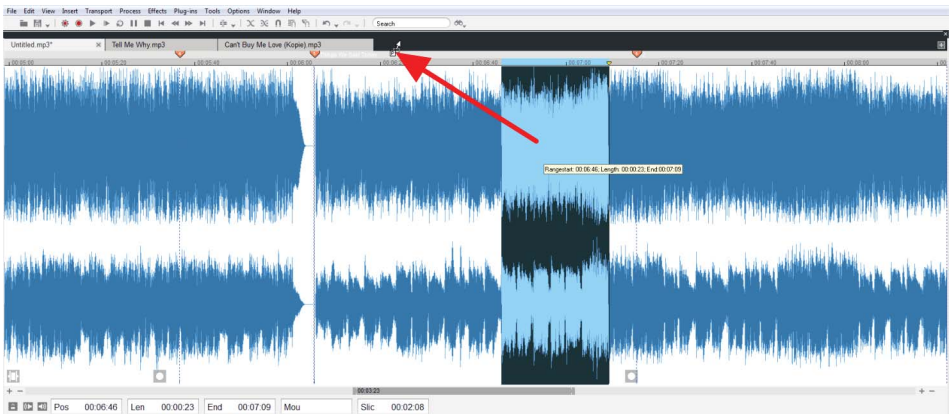
Edit selection numerically: The current selection will be displayed numerically in the "time display". Double click in the display to edit values.



Change selection: There are a range of menu commands and their keyboard shortcuts under "Edit" > "Selection" for editing the selection further.

Loop selection: The selection can also be played back in a loop. To do this, press the Q key. Pressing this button again switches back to normal playback.

Extract selection as audio file: Drag & drop the selection up to the line with the file tabs. This will enter the selection into a separate audio file.



Alternatively, copy the selection to the clipboard (Ctrl + C), and then select "Edit" > Paste Special" > "Paste to New".

Zooming

Depending on the editing operations you want to perform, you may want to view the entire file or only a small section. The zoom level can be changed for each data window.

Mouse wheel: Move the mouse wheel up and down to zoom in and out. Holding down the Ctrl button at the same time changes the size of the waveform display itself. This is useful for seeing very quiet material.

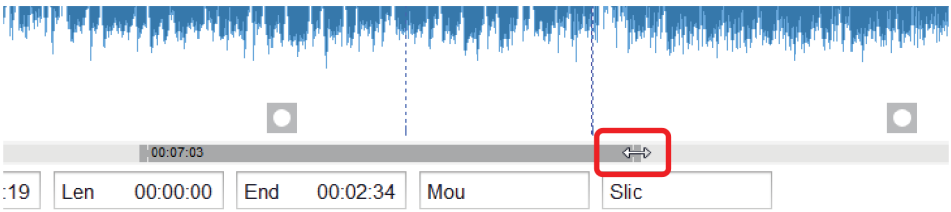
Magnify Tool: You can also use the magnify tool in the editing menu for zooming.

Zooming with the keyboard: "Up arrow" zooms into the current range, "down arrow" zooms out again.

Zooming with buttons: In the lower right and left corners of the waveform window are a "**+**" and "**-**" button, respectively, for zooming in and out.

Zooming with the scroll bar: You can also zoom with the horizontal scroll bar. To do this, move the mouse to one end of the scroll bar until the mouse pointer

turns into a double arrow. You can now drag the scroll bar to make it bigger or push it to make it smaller. The displayed section also changes size.



Zoom with vertical mouse dragging: An elegant method of zooming, which does need some practice and skill, is to left-click on the timeline, hold down the mouse button and drag the mouse vertically.

By dragging the mouse up, you can zoom out of the wave form and you can zoom in with the reverse action. This allows you to position the playback marker precisely "in one go".

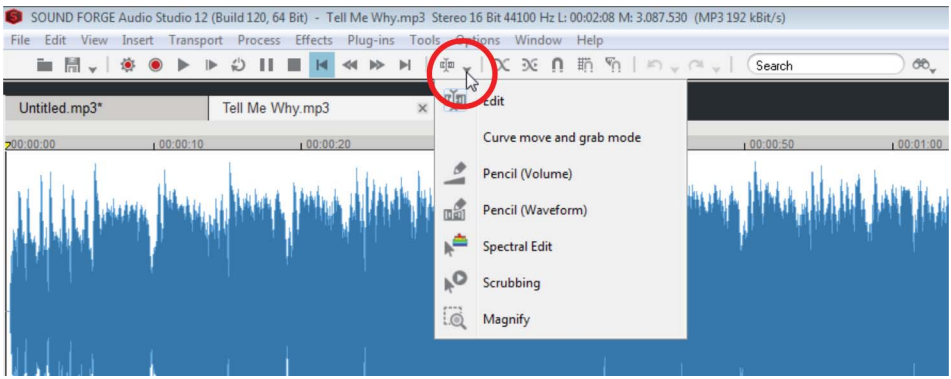
1. Click on the the marker bar to roughly position the playback marker.
2. To zoom into the waveform, hold down the mouse button and move the mouse downwards.
3. Correct the mouse position by moving it horizontally. Dragging up zooms back out.

> Tip: If you press the Ctrl key before letting go of the mouse button, you will return to the original zoom level.

Deactivate this function at any time in the system settings (Y key > Keyboard, Menu & Mouse > Mouse > Disable zoom with vertical mouse dragging on the timeline").

Edit tools

Tools can be switched via the toolbar.



Alternatively, choose the tool you need from the Edit > Tool submenu.

Edit Tool

The Edit Tool is used for selecting data and moving the playback marker.

Further information about this tool can be found in the Menu chapter (view page 83).

Magnify Tool

The magnify tool lets you zoom into and out of the waveform.

More information on this is available in the Menu Chapter (view page 83).

Pencil (Waveform)

Use the Pencil tool to edit the waveform by drawing on it. If the audio file contains a short crackling sound, for example, zoom into the crackling and redraw the waveform as a lower peak.

More information on this is available in the Menu Chapter (view page 83).

Spectral Editing

With the Spectral Mode you can remove noise from wave forms. The spectral display of a wave form shows a graphic of the different frequency proportions in a time curve. This makes it possible to filter out targeted frequencies.

More information on this is available in the Menu Chapter (view page 83).

Preview

This tool enables you to preview and control the playback speed by dragging and dropping (scrubbing). The project is played forwards or backwards, depending on the drag direction.

More information on this is available in the Menu Chapter (view page 84).

Cut, copy and paste audio material

To add audio material to a position within an audio file, first choose the audio data that you want to add.

- Drag a selection (view page 42) using the mouse or select the entire data using Ctrl + A.
- Copy the audio data to the clipboard (Ctrl + C or "Edit > Copy").
- Then open the file into which you want to insert the material.
- Set the playback cursor to the position where it should be inserted.

Tip: The Home key moves the playback marker right to the start, the End key right to the end. Ctrl + G open to "Go to" dialog, enabling you to move the playback marker to a certain position. It's recommended to set markers (view page 55) for all important positions and navigate them via the Regions List (view page 64).

- Use Ctrl + V (or select Paste from the Edit menu) to insert a copy of the clipboard contents at the current cursor position. Audio material following it will be moved to the background.

Pasting into a stereo file will insert data to all channels, as the channels in a stereo file must always be equal in length.

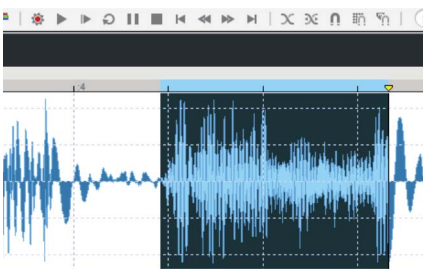
Further pasting options are available in "Edit" > "Paste Special": "Mixing" mixes existing audio material with pasted audio material, "Overwrite" overwrites the material rather than moving it.

From the Edit menu, choose Trim/Crop to remove all data from the audio file except the current selection.

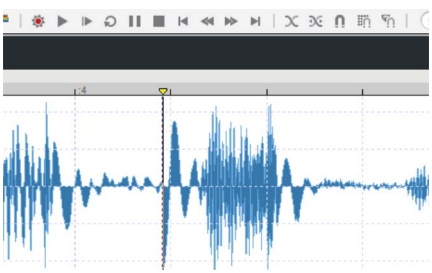
Soft Cut

This button in the upper toolbar is used to activate the Soft Cut mode.

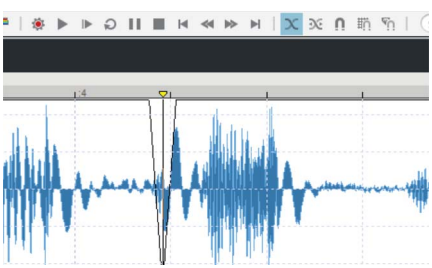
During all copy & paste operations, deleting or effects editing, short crossfades are added to the edges between the pasted or editing audio material.



A range in the selected audio file was selected and then deleted here



Without Soft Cut, a hard transition is made at the cut border, which produces a clear crackling sound.



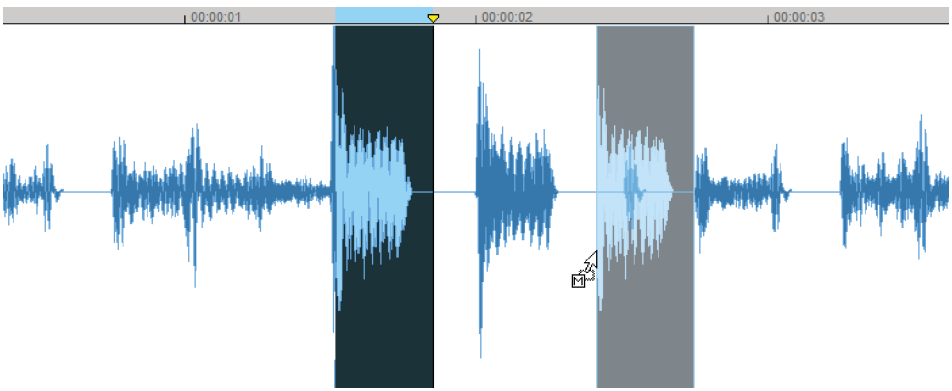
With Soft Cut activated, a crossfade is applied to the transition, and the crackling is harder to hear.

With Slice Editing (view page 52) and the Crossfade Editor (view page 90), you can adjusts the transitions in much more detail!

Copy and paste by dragging and dropping

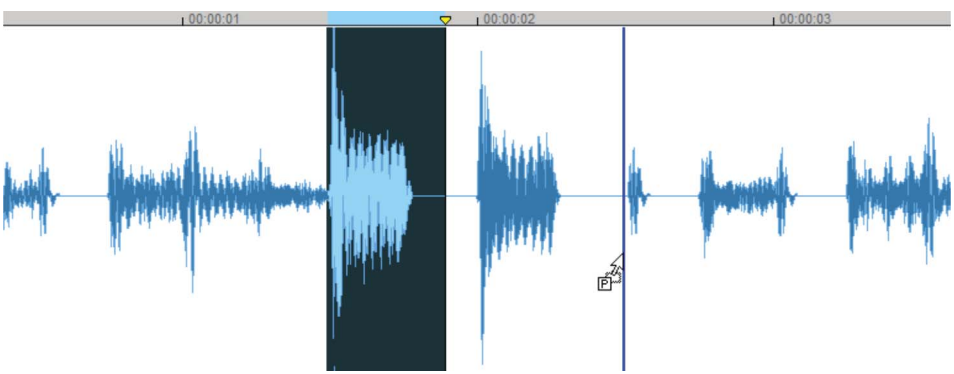
It is also possible to copy and paste by dragging and dropping. To copy a selection by means of drag & drop, drag **upward** with the mouse **before** moving it to the left or the right. This action must be performed in this way, otherwise the selection will be changed. Now, keep holding the left mouse button down while dragging the selection to a new location to be pasted.

Drag Mix: This is the standard mode. Releasing the mouse button will mix (view page 81) the audio material. You can drag the selection to the same data window or a different one.



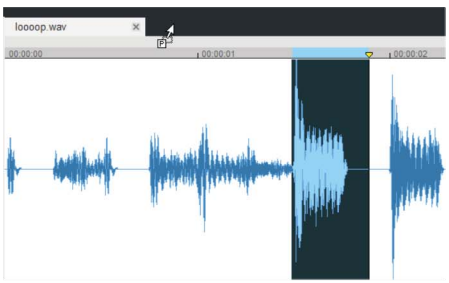
The selection will appear transparent, and the mouse pointer will display an M for "mixing".

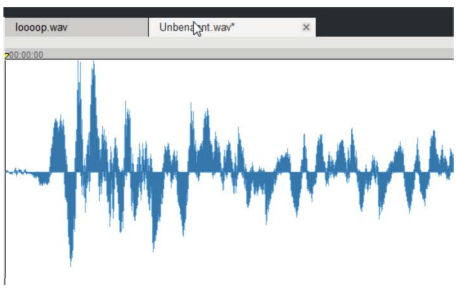
Drag Paste: Hold down the Ctrl key while dragging the selection upward and then release to paste the audio material. The audio material located behind the pasted material will be moved further back.



The blue line will shows the paste position, and the mouse point will display a P for "paste".

Insert into New File: Drag the selection to an empty spot above next to the tab with the file name in order to create a new file from the selection.





Tip: Additionally clicking the right mouse button while dragging will switch between "mixing" and "pasting".

Processes, Effects & Plug-Ins

Sound Forge Audio Studio 12 includes many processes and effects that you can use to manipulate the audio data in your media files. Processes are available in the "Process (view page 111)" menu, effects are in the "Effects (view page 119)" menu. You can also use audio plug-ins by going to the "Plug-ins (view page 138)" menu. Sound Forge Audio Studio 12 supports DirectX and VST effects from third parties.

- First select the data you want to edit by making a drag (view page 42) selection.

If no data is selected, processing will be applied to the entire file.

Most functions can be applied to the right or left channel or all channels. However, since the channels in a stereo file must be equal in length, functions that affect the length of the data cannot be performed on individual channels. These functions include "Resample" and "Time Stretch".

- Go to the "Process", "Edit" or "Plug-in" menu, select a command and adjust the settings in the dialog.

The dialog for each included effect contains a list of presets at the top for the effect. If you would like to save an effects setting for later use, click the entry

field, enter a name and click "Save". The effects setting will then be added to the list.

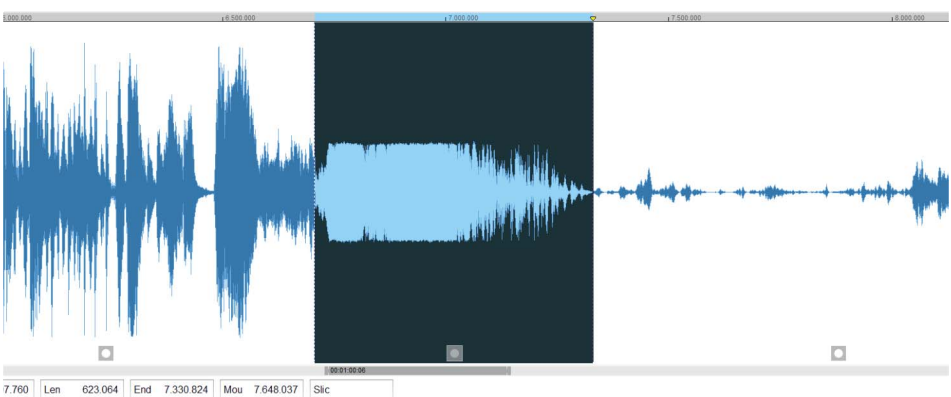
Click the "Preview" button to hear the effects of your processing settings. Checking the "Bypass" box lets you listen to the unedited original material for comparison.

- Click the "OK" button to start processing.

During processing, a progress meter is displayed at the bottom of the data window. You can cancel the operation at any time by clicking the "Cancel" button to the left of the status display, or you can press the "Escape" key.

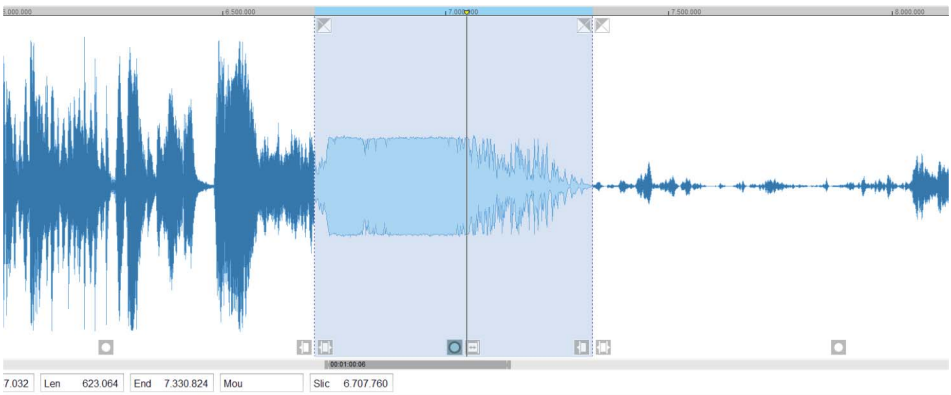
Slices

Once you have pasted audio material into an audio file, or have changed a selection within the audio file, the file will be divided into different sections at the edges of these edits. In Sound Forge Audio Studio 12, these sections are called "slices".

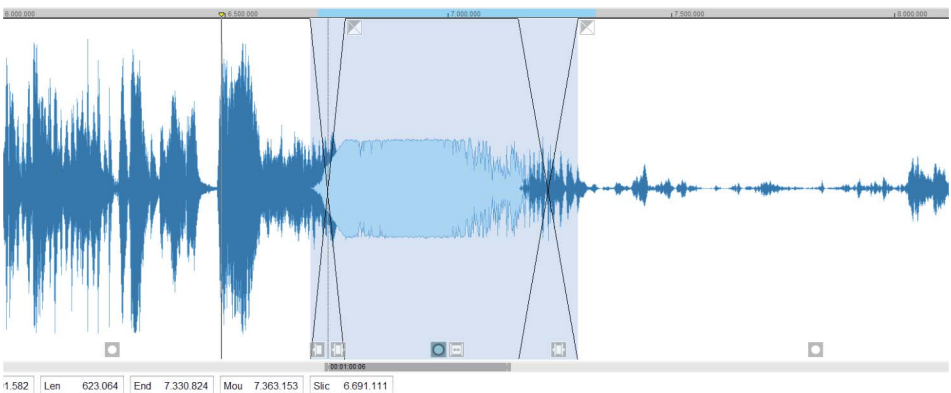


In this example, the volume of a selected range has been modified. As a result, the file has been divided into three slices.

The slice icon can be used to select a slice for editing. Selected slices appear in a different color.



Selecting a slice opens various handles which can be used to further edit the slice. You can also fade slices in and out; the result is shown on top of the output material. Slices can be moved on top of one another, and the slice borders can be changed. If a slice represents a section of an audio file, the audio material represented by the slice can also be moved within the slice.

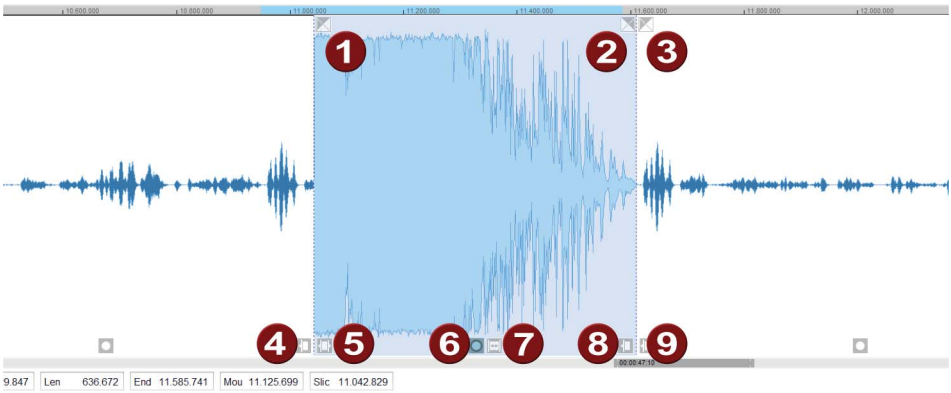


Crossfades have been added to the slice borders.

This can be done using keyboard commands, menu options or even the mouse. To edit with the mouse, use the handles on the selected slice. The menu commands and shortcut keys can be found under "Edit" > "Slice".

Slice Handles

The handles that appear on a selected slice have the following functions



1  **Fade-in:** This handle creates fade-ins. To do this, drag the icon slightly to the right.

2  **Fade-out:** This handle creates fade-outs. To do this, drag the icon slightly to the left.

3  **Fade-in/Crossfade (next Slice):** If the slice is not the first slice in the audio file, then the slice will be crossfaded with the previous slice instead of being faded in.

> Tip: Right-clicking a handle opens a context menu where you can select various curve forms. The Crossfade Editor (view page 90) lets you edit crossfades in more detail.

4  **Slice border:** This handle lets you correct the front border of the slice. The previous slice's rear border is moved accordingly.

5  **Move slice**: Moves the slice (and all following slices) as a whole. The previous slice is correspondingly shortened or lengthened.

6  **Slice selection:** This icon lets you select a slice and makes all other slice handles visible.

7  **Move audio material:** Moves the slice's audio material along with all the audio material and slice borders that follow it. (In other words: The beginning of the slice is shortened or lengthened; the slice border stays in the same position).

8  **Slice border (end):** This handle is used to correct the rear border of a slice and the front border of the next slice. In the case of the last slice, the audio file is correspondingly shortened or lengthened.

9  **Move slice (next slice):** See 5

Notes

Sometimes crossfades can be made, or audio material or slice borders moved, up to a certain point — or not at all. This is because slices can only refer back to the audio material from which they were originally created.

Crossfades are made symmetrically by default. This means that the middle point of the crossfade is placed at the slice border and requires that there is audio material both in front of and behind the slide border. If the slice is part of a larger audio file, then there aren't any problems. This won't work, however, if the slice was created as a result of audio editing, or if it comes from another audio file or the clipboard, and is located at the beginning or end of the audio file. Be aware of the following:

- If you want to apply an effect to a partial range in an audio file and have it fade into the surrounding audio material, be sure to select a somewhat larger range for the effect. Move the slice borders somewhat inward along with the following effect in order to apply the crossfades.
- Alternatively, you can click on the fade handle to the right and select "Fade inside" from the context menu. This sets the crossfade to only use audio material within the slice borders.
- If you want to fade a slice in, but do not want it to crossfade with the previous slice, deactivate "Crossface active" in the context menu.

Markers and Regions

Markers and regions serve as reference points along the timeline. Markers are individual points which can be directly jumped to. Regions are specific ranges between two markers.

Markers

Markers are reference points you can place throughout a file (e.g. to mark positions for editing).

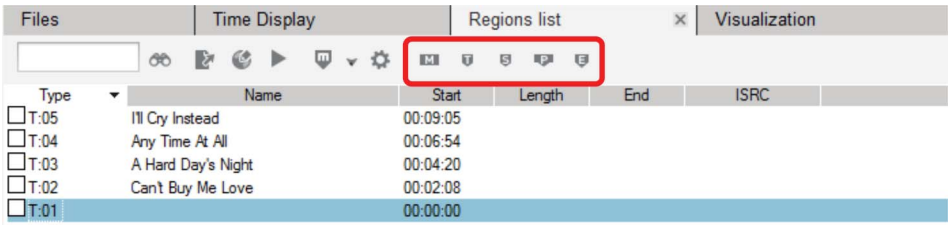
Inserting Markers

To set a normal playback marker, proceed as follows:

- Set the playback marker where you'd like to place a marker.

- Press the "M" key (you can do this during playback as well).

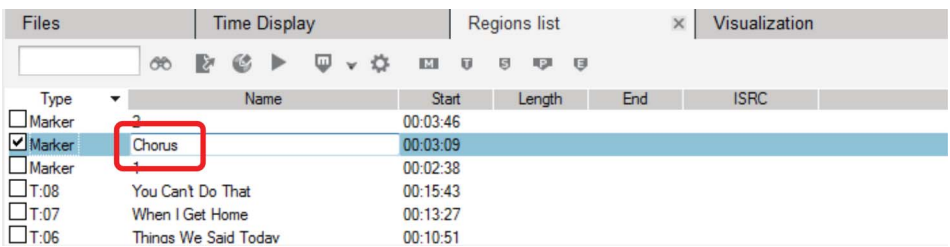
To place other markers (CD Track Marker, CD Subindex Marker, CD Pause Marker and CD End Marker), use the icon buttons in the Regions List.



Naming Markers

You can assign any name you want to an existing marker:

- To do this, double-click the name column in the Regions List (view page 64) and enter a name in the text field.



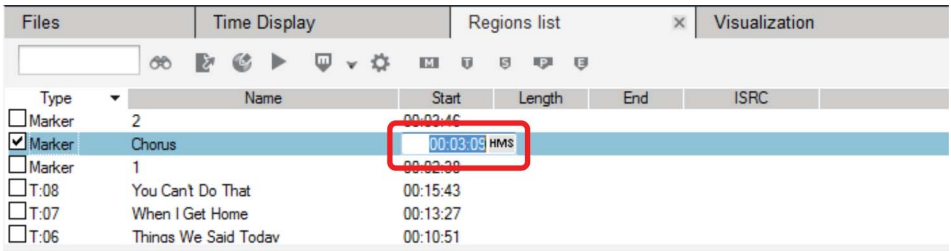
Or right-click the marker in the Regions List and go to "Edit" > "Name" in the context menu.

Changing Marker Types

All markers can be used as CD track markers or region markers. To do this, right-click the marker in the Regions List (view page 64) and go to "Edit" > "Type" in the context menu. In the submenu, select the marker type you want.

Moving Markers

Drag and drop the marker to a new position or double-click the column in the Regions List (view page 64) to enter a new position numerically.



Or set the play cursor to the position you want, right-click the marker on the timeline or in the Regions List, and select "Upgrade" in the context menu.

Deleting Markers

Click the marker in the Regions List (view page 64) and press the Del key.

Regions

Regions can be used to save different selection ranges (view page 42) for later opening.

Creating Regions

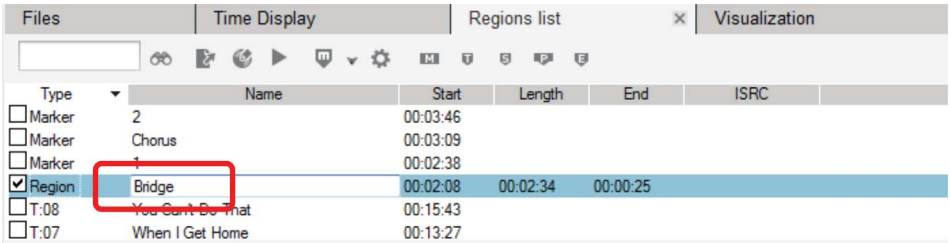
Make a selection (view page 42) and press the R key. Alternatively, you can select the "Region" option in the "Paste" menu.

Doing this saves the selection as a region and adds it to the Regions List (view page 64) with a consecutive number. Numbered region markers are placed at the beginning and end of the saved selection.

Name Region

You can assign any name you want to an existing region:

- To do this, double-click the region in the name column in the Regions List (view page 64) and enter a name in the text field.



Or right-click the region in the Regions List and go to "Edit" > "Name" in the context menu.

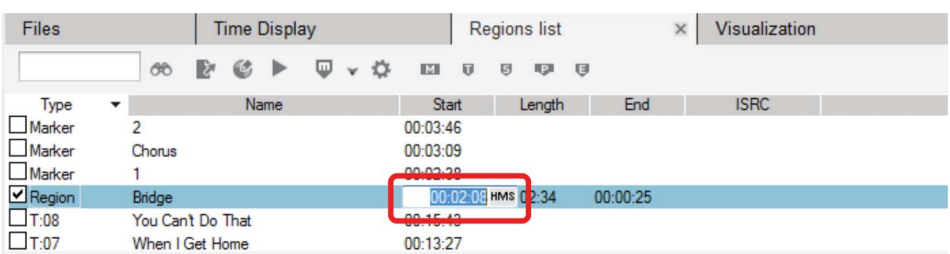
Select Region

Double-click on the region in the Regions List (view page 64). This selects the region and stretches the saved selection.

Moving Regions

Drag either region marker using the mouse to move the region border and change the region's size.

Or double-click the "Start" or "End" column in the Regions List (view page 64) to enter a new position numerically.



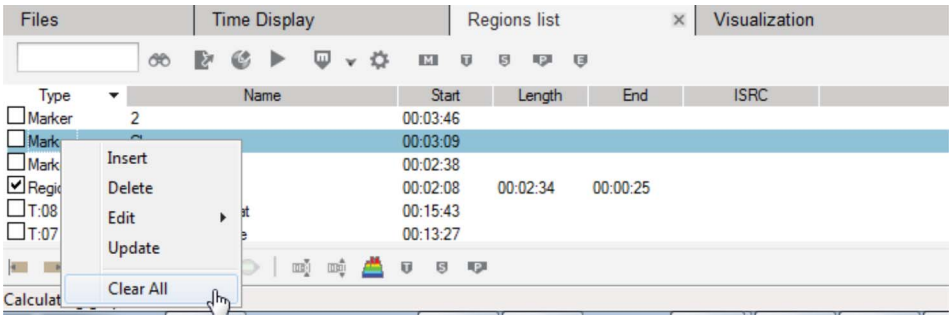
Or select a new range, right-click on a region marker on the timeline or in the region list, and select "Upgrade" in the context menu.

Deleting Regions

Click the region in the Regions List (view page 64) and press the Del key.

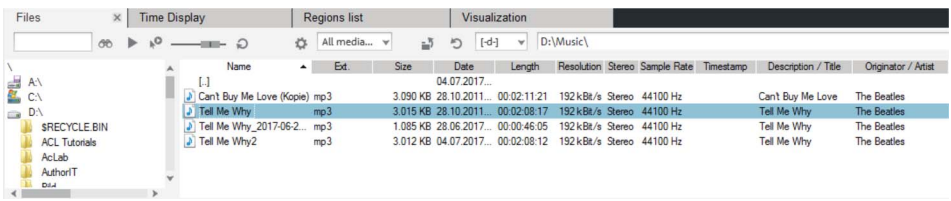
Delete all markers and regions

Right-click an entry in the Regions List (view page 64) and choose Delete All from the shortcut menu.



Managers

The managers are located in the bottom third section of the program window.

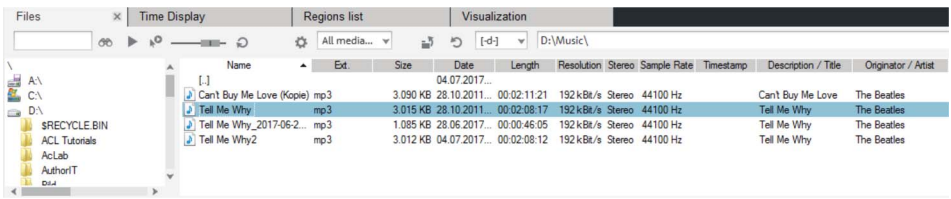


There are four preset managers available with their own tabs: "Files", "Time Display", "Region List" and "Visualization". Additional windows can be opened by using the + button on the right.

Hide Manager: You can click in the tab bar to hide the manager to create more space for the wave form display. Clicking on the tab bar again reopens the manager window.

Explorer

Sound Forge Audio Studio 12 contains a file explorer for previewing and quickly loading files. The integrated display filter limits the files shown to media formats which can be loaded into the project.



Preview audio files

A selected audio file may be previewed using the arrow.

If the "Autoplay" box is activated, every selected audio file will be automatically played.

A complete list can be previewed using the cursor keys. Clicking the play button again stops playback. The fader symbol may be used to control volume.

Note: To preview certain media data, e. g. in WMA format, these files must be converted to wave format. Conversion is carried out automatically in the background.

Organization

The file manager includes a directory tree that behaves similarly to Windows Explorer. The following file information can be displayed in the file list:

- Name
- Extension
- Size
- Date
- Length
- Resolution in bits
- Stereo/Mono

- Sample rate
- Timestamp
- Description / Title
- Originator / Artist
- Path (opened with right click)
- BPM (opened with right click; automatically specified based on loop length)

A drive selection menu found in the toolbar lets you quickly switch between all data storage devices connected to the workstation.

The "Search" option lets you search the current folder for specific files. Enter a search term into the box and click on the binoculars. Found directories or files are highlighted. You can find and highlight several entries that meet this search criterion by entering one or several first letters.

On the right side of the File Manager you can add or remove the current folder in your list of favorites. Favorites can be opened by clicking on them in the list.

Underneath the favorites list, a selection menu can be found with a list of recently used paths. Here you have the option of switching to the current project folder.

Load files from Explorer window

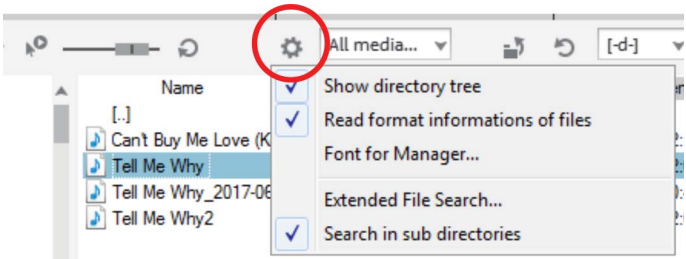
Each file can be loaded from the Explorer to the waveform display window via drag & drop or the Enter key. Doing so creates a new waveform display window.

If you select several fiels in the Explorer and load them via drag & drop, they will all be loaded in separate waveform display windows.

Use "File" > "Open" (view page 36) to hang several files in a row in order to burn them to CD.

Options

You can open the Explorer options by clicking the gear icon.



Display Folder Structure: Shows/hides the directory tree on the left side.

Read File Format Information: Turns the automatic audio file format recognition on/off.

Manager Font: Changes the font in the explorer.

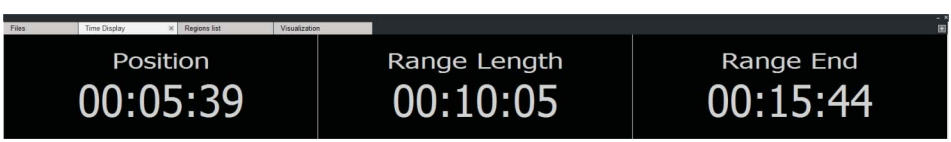
Extended File Search: Enables more advanced searches. Additional criteria are available for extended file searching, e.g. Date of last file change between, Date of file creation between, Files with audio length between, and Files with sample rate of, File name, Project comment, Description / Title and Originator / Artist:

Search in Subdirectories: Limits the search to the current folder without subdirectories.

Time Display

This function shows a zoomable time display. This makes it possible to maintain a good overview of the project even at longer distances, which can be useful in some situations.

The time display is opened with three display fields, one for position, one for range length and one for the range end.



n

In the context menu of a field (right-click) you can insert additional fields ("Insert field..."),remove existing fields ("Remove field") or select the time position to be displayed in the field ("Current field...").

Additional Settings Options in the Context Menu

Font...:	This is the font type for the display
Measurement units:	The type of measurement unit used in the display
Current field...:	In this submenu you can adjust the display size and assign individual color settings to each field.
Move field forward / Move field back	This function can be used to change the order of the fields.
New time display:	This opens a new time display in a separate window. This may be necessary when the fields in the dialog can no longer be displayed properly.

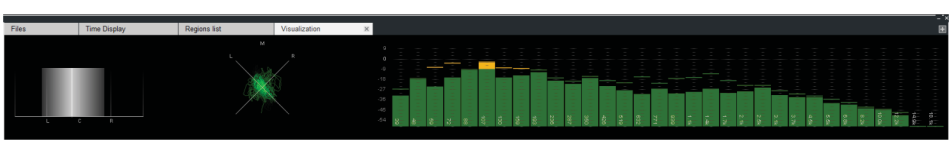
The following measurement units are available: samples, milliseconds, hour/min/sec, SMPTE (project frame rate), SMPTE (individual frame rate), SMPTE/milliseconds, bars, CD MSF, feet & frames 16 mm (40 fpf), feet & frames 35 mm (16 fpf), independent time format.

Select the option "Independent Time Format" to enable the use of several different measurement units.

Most sizes can be selected and edited directly in the time display by double-clicking on them.

Visualization

The visualization displays the audio material in graphic form. You can integrate the visualization into the Docker (view page 69) or open it in its own window.



Normally the visualization is divided into several display windows. In the context menu (right-click) you can choose between the following display options:

- Peak Meter (peak meter display)

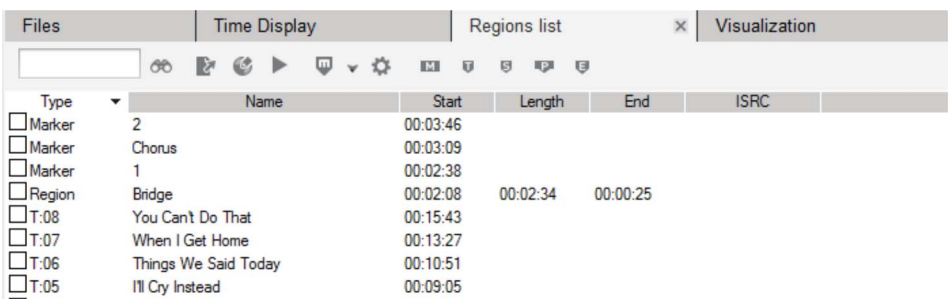
- Phase oscilloscope (vectorscope)
- Correlation Meter
- Direction meter
- Spectroscope
- Spectrogram
- Bit meter
- Oscilloscope
- Tuner

More information about the individual visualization windows is available under "View" > "Visualization (view page 94)"

Regions List

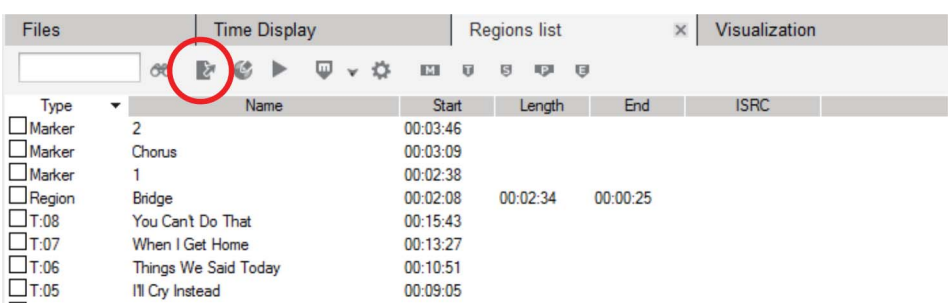
In the Regions List you'll find a list of your markers, regions and CD markers. All markers and regions can be directly jumped to.

- To jump to a marker, place a check in the selection box.



Export Text

Regions List information can be exported as a text file. Click the "Export text" button in the Regions List toolbar to do this.



The Windows text editor opens with an excerpt from the Regions List. The following information will be saved:

- Project name
- Marker position in project
- Marker type
- Marker name
- ISRC (International Standard Recording Code) for CD markers

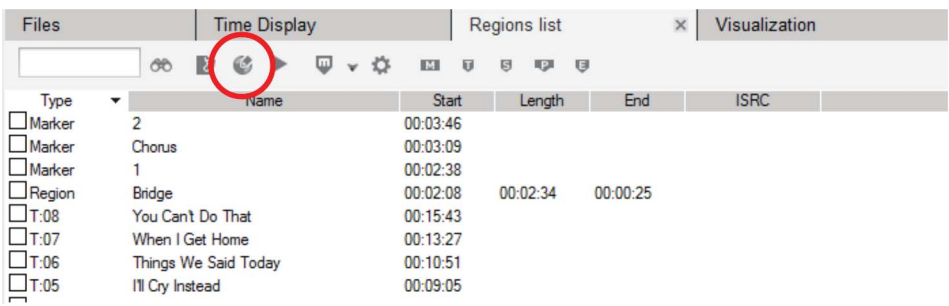
You can find this file in the project folder (Projectname.txt).

Search feature

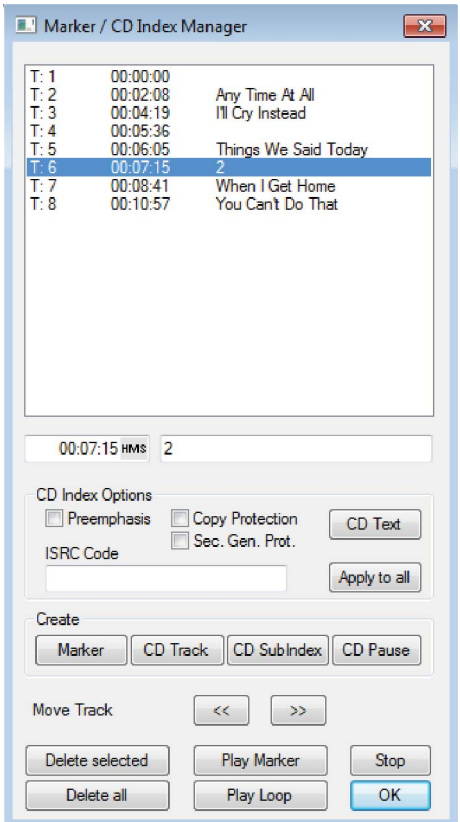
As in other file managers, the Regions List also provides a search option which allows you to search for specific markers in the current window. Enter a search term into the box and click on the binoculars button.

CD Index Manager

The third button in the Regiosn List toolbar open the CD Index Manager.



This manager is important for checking the track marker for a file that will be burned to CD and also for rearranging tracks or the selection fast.



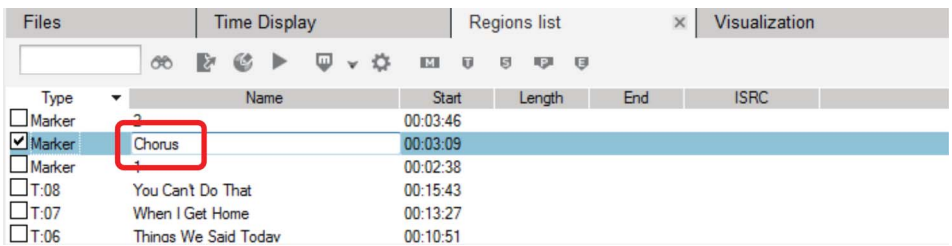
This dialog displays all the CD tracks and sub-indices in the current project in a list. If you select the markers, you can change their numerical position in the list via the corresponding time input field and name or rename them.

CD Index Settings: Here, you can set several options for the individual CD tracks, including **Copy protection (SCMS)** ,**Pre-emphasis**, **Second generation protection**, as well as **ISRC code:**These settings can be transferred to all indices.
Pre-emphasis means that high frequencies are accentuated before A/D conversion when creating a CD. Even during A/D conversion high frequency quantization noises will emerge. Now the Audio signal with high accentuation will be written to the CD. When playing the CD player itself will use "Emphasis Bit", which means that the frequency accentuation (highs) will be curbed. This guarantees a 'true to original' sound, likewise Pre-emphasis leads to a reduction in quantization noises.

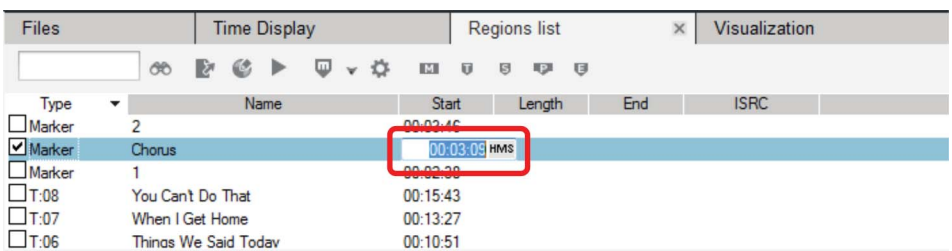
In practice Pre-emphasis is rarely used owing to the fact that the signal noise gap in the CD creation process already large enough (16 Bit) that quantization noises are negligible anyway.
"CD text" opens a separate dialog field for entering CD text information.
You can set **new markers, CD tracks, CD sub-indices, and CD pauses** by clicking on the corresponding buttons.
Move the track within the list using the **double arrow buttons**.
"Delete selected" Deletes the selected markers.
"Delete all markers" Deletes all markers.
"Play marker" starts playback from the position of the selected marker.
"Play loop" plays a loop around the marker.
"Stop" stops playback.
"OK" applies your settings.

Changing and Deleting Markers

- Each marker can be given a name by double-clicking the "Name" column and entering one.



- To change marker positions, double-click the "Start" column and enter a new position. Or simply drag the marker to a new position in the timeline.



- Markers can be redefined in the context menu at any time, e.g. in order to change them to CD track markers (option: "CD Track Index").
- To delete a marker, select it in the Regions List and press "Del".

Editing Toolbars

Sound Forge Audio Studio 12 offers two toolbars that contain a pre-configured selection of available icon buttons.

You can customize the toolbars to your requirements by adding icons, changing their positions, removing the ones you don't need, or changing the arrangement of buttons. To do this, right-click to open the context menu of any icon and select the option "Edit Toolbar".

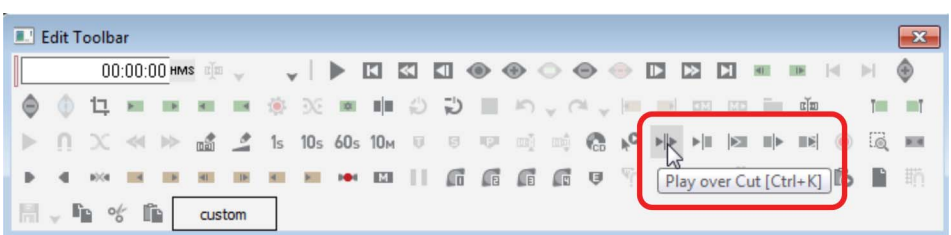
This opens the "Edit Toolbar" window which contains all of the available icons.

Icons that are already on the toolbar are grayed out. The others can be added by dragging & dropping them onto one of the toolbars.

To remove an icon from a toolbar simply drag it off. To change the position of the icons simply drag & drop them onto new positions on the toolbar.

Note: When the "Customize Toolbar" window is open, clicking on the icons has no effect. This helps to prevent unwanted changes to the project.

The easiest way to find out the functions for these button is via the tooltips. A tooltip is displayed when you move the mouse pointer and hold it over a button in the dialog.

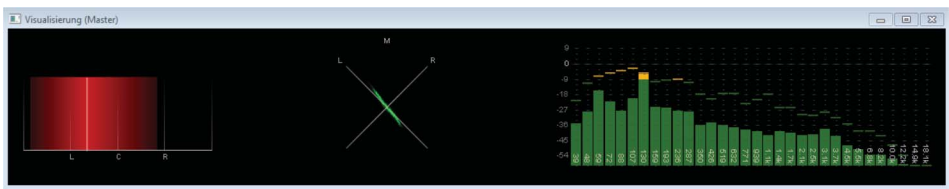


Modify user interface

You can modify the Sound Forge Audio Studio 12 interface and customize it for your needs. For example, you could place the waveform display for a file in fullscreen on a second monitor and arrange the other windows on another monitor.

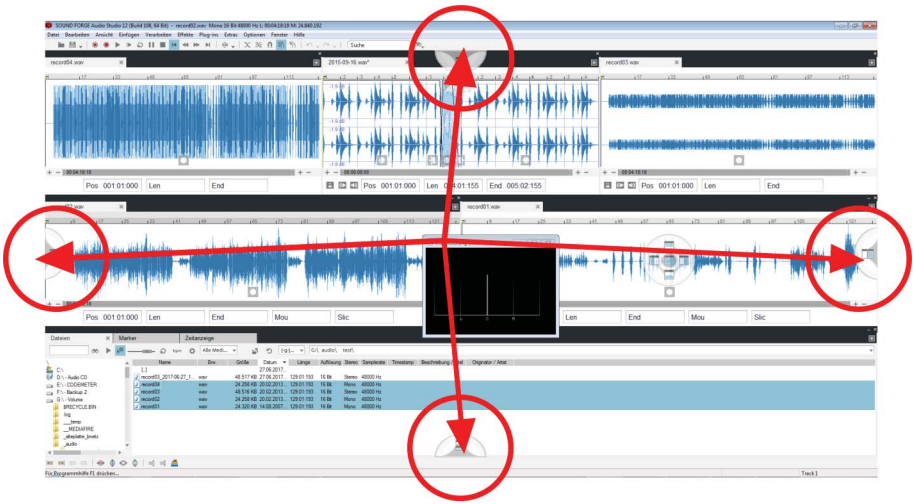
The Docking feature helps you undock sections of the interface and dock them to other areas or position them freely.

To undock a window click on the title bar and drag the window out of the interface. In an undocked state, the corresponding dialog appears with a conventional title bar and can be positioned freely.



Undocked windows can be docked to the areas provided in the VIP by double-clicking the title bar or by clicking the title bar and moving with the "Ctrl" key pressed. When moving windows, an arrow symbol appears on the interface. If you move an undocked window with the mouse onto one of the arrow symbols, the window will be docked there.

Docking position options for the visualization window can be seen here:



Menu Reference

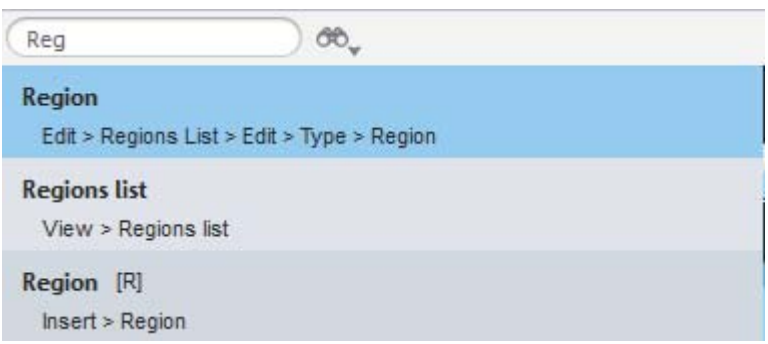
Extensive information on the individual menu points of Sound Forge Audio Studio 12 can be found in the following sections.

The menus in Sound Forge Audio Studio 12 can also be individually customized, i. e. menu items can be hidden or shown. "File" > "Program Preferences" > "Edit Keyboards Shortcuts and Menu..." (view page 157).

Searching for menu commands and help topics

Sound Forge Audio Studio 12 provides a search field for finding menu commands and help topics.

Enter a character combination into the search field which is relevant to a search term. Immediately up to five commands from the main menu will be displayed that contain the desired character combination.



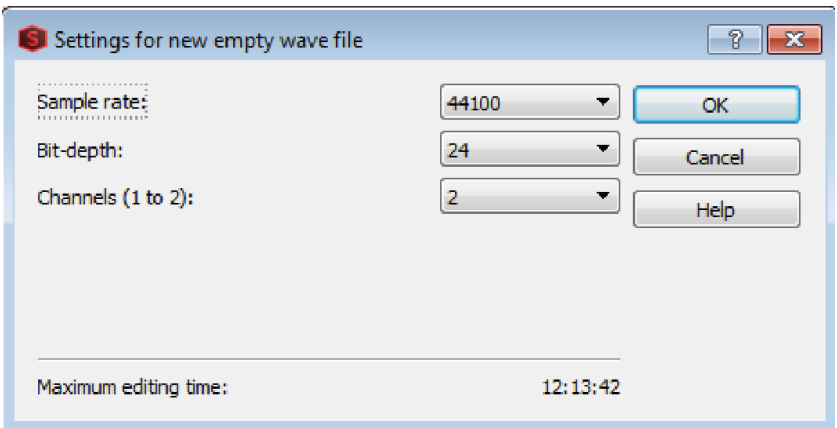
All results that are listed can be directly selected. The commands in the upper section are listed immediately. Clicking on the small arrow next to the search button opens a list of recently performed commands.

Shortcut: Ctrl + F

File Menu

New...

Creates a new, empty audio file.



In the dialog, the sample rate, bit depth and number of channels can be set. You can now paste audio material to this file via the clipboard, the recording, or the menu command "Open and append".

Note: Pasted audio material is automatically converted to the format of the audio file to which it is pasted. If, for example, you paste mono audio material into a stereo file, it will appear there in stereo (both channels will have identical contents). The same applies for bit depth and sample rate.

Open...

You can open various file types here and load them into Sound Forge Audio Studio 12. The following formats are supported and read directly by Sound Forge Audio Studio 12:

Wave files (.wav), MP3/MPEG files (.mp3, .mpg, .mus), QuickTime files (.aif), MS Audio files (.asf, .wma), Ogg Vorbis (.ogg), Sound Designer II, FLAC (.flac), video files (.mp4, .mov, .avi), and playlists (.m3u, .cue).

You can also load several files simultaneously. To do this, expand the selection via "Ctrl + click" (just like with Windows Explorer), or select a range of files via "Shift + click".

The option "Load all files in one project" opens all selected files as one large file with the individual files being arranged one after the other. CD track markers are set at the start positions of each individual file and assigned the same name as the file.

Clicking the payback button enables each audio file to be listened to in advance. If "Auto Play" is active, this will happen automatically when you make a selection.

Open and append...

Works like "Open file", but the file is attached to the end of a file which is already open. A CD Track Index is generated at the same time. This enables you to create a master file for burning CDs.

Close

Closes the current file. If changes have been made, you will be asked if you want to keep them.

Save

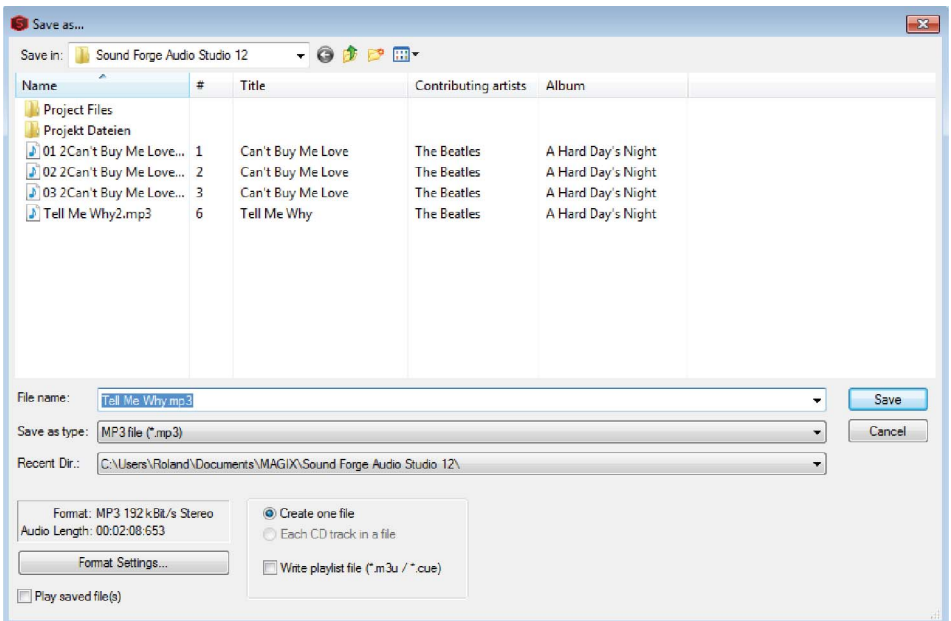
Saves the current file in the current file format. All changes that have been made to the file are final and can no longer be undone.

Save as...

Saves the file under a new name and/or a different format. The original file is kept. It's also possible to split the file into multiple files.

Tip: Use this feature to convert files to other formats!

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(s). 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.

 Tip: You can quickly save files in specific formats by using the save button in the menu.

Format settings: Here you can access the configuration options for the format.

Export settings: Various options are available for selection under the button (see below)

The changes to the settings will be applied to any subsequent exports.

Play back exported file: After saving, the media player is opened to test the results of the export.

Export settings

Only Selected Range: Exports the selected range. If you go to "Save as..." when a range is selected, this option is already selected for you.
Complete Project: Exports the entire file.
Complete CD: Exports the range between the first CD track marker and the CD track end marker.

Create a File: Saves an audio file under the name entered in the dialog.
Each CD Track in a File: Creates one file per set CD track marker, beginning at one track marker and ending at the next. The files are named according to the names of the track markers. When creating a playlist (see below), the selected file name is used as the name for the playlist.
Each CD Track in a File: The individual tracks end at the next pause marker, not the next track marker.
Any Marker: Creates one file for each set marker (CD track, marker, region) starting at one marker and ending at the next. The region start is used for regions.

Apply Track Numbers: Exported files can be optionally numbered to make it easier to order them alphabetically in Explorer.

Write Playlist File (*.cue): Provides the option of saving the positions of the track markers to a cue list when you create a file.
Write Playlist File (*.m3u): When creating multiple individual files, provides the option of creating a playlist file (.m3u) in which the names of the files are listed in the correct order.

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.

FLACFLAC 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: Sound Forge Audio Studio 12 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".

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.).

OGGOGG 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.

Save all

All open files are saved.

Extract audio from CD...

This function allows the import of audio data from CD/DVD drives. The data is imported digitally which eliminates any loss in sound quality. CD tracks can be created in all supported audio formats during import.

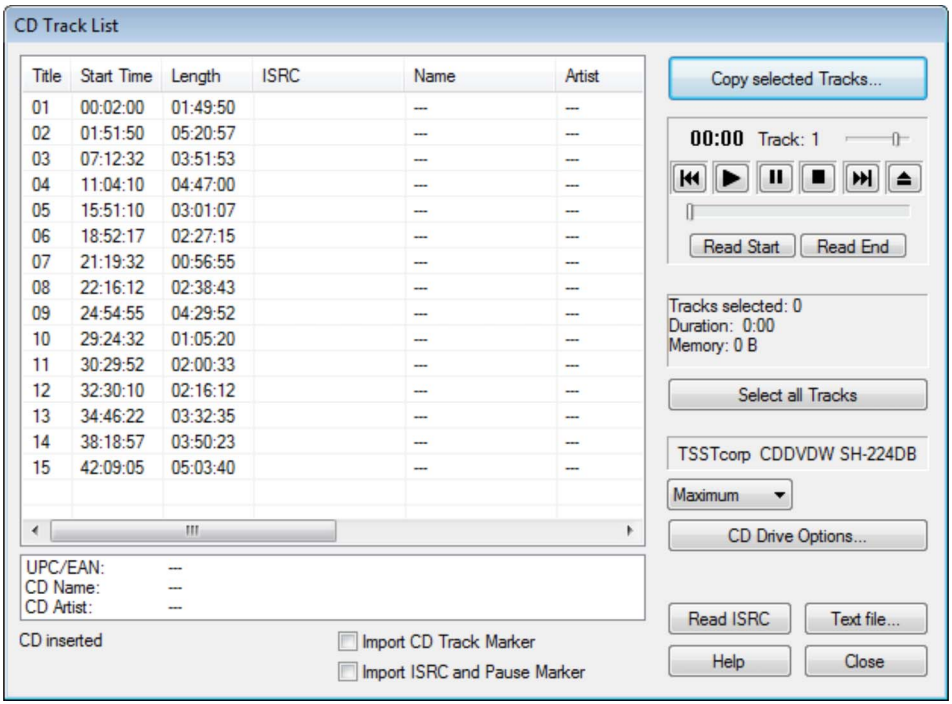
To import audio tracks in Sound Forge Audio Studio 12, follow these steps:

1. Open the "Extract audio from CD..." dialog in the "File" menu.
2. If you have more than one drive installed, click the button "Drive options" to open the drive list dialog. Select the desired CD-ROM drive and then close the dialog by clicking "OK".

> **Note:** The name of the currently selected CD-ROM drive is displayed above the button "Drive Options".

3. Select a title. For several titles use the Shift or Ctrl keys, for all titles click "Mark all".
4. Click on "Copy selected CD tracks..."
5. Select a file name for the created audio file. Under "File type and "Format Settings", you can select another file type as .wav in order to convert CD tracks to formats such as MP3 during import.
6. By clicking "OK", the audio material is copied onto your hard drive from the CD. A progress bar indicates the status.
7. Close the track and drive lists. The audio files containing the audio material selected from CD are now opened in Sound Forge Audio Studio.

Track List Dialog



Copy Selected CD Tracks: Opens a save dialog where you can assign a name to the resulting audio file. You can also selected a file type other than .wav under "File type" and "Format settings" in order to convert the CD tracks to MP3s, for example, during import. **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. Click "OK" to start the audio copy process.

Volume controller: Control the playback volume for the digital preview function of the CD tracks here.

Reverse: Jumps to the previous track.
Play: Starts audio playback of the first selected track in the list.
Pause: Stops playback; start again later at the same position by pressing "Resume".
Stop: Stops playback.
Forward: Jumps to the next track.
Eject: This ejects the CD from the drive.

Read Start / Read End: This defines the read start and read end of the CD track. Isolate the start and end points in the CD track progress display by dragging with the mouse.

Select all tracks: Select all tracks in order to copy the entire CD. Track selection is possible by pressing the Shift key and arrow keys. "Ctrl + mouse click" allows multiple tracks to be selected.

freeDB Title Info: Clicking on this button takes you to the freeDB database where you can display the title information for the selected track.

Drive Selection: Here you can select the disc drive to import from.

Read ISRC: This option reads the ISRC (International Standard Recording Code) for the inserted CD. This is a 12-digit ID number that provides specific information like the label's country of origin and company number, the release year, and a sequential title number. The ISRC code is entered into the subcode when the CD is pre-mastered.

Text file: Exports all of the information listed as a text file.

Import CD Track Marker: If this option is activated, track markers will be automatically placed at the beginning of the imported tracks.

Apply ISRC and Pause Indices: If you activate this option, you'll see the ISRC of the imported track in the CD track marker of the Regions List even if you haven't activated the option "Read ISRC". In addition, CD pause indices are also read and listed in the Regions List.

Attach Video.../Remove Video...

You can create new media files by attaching video to an existing audio file or removing the video stream from a video file.

If an audio file window is active, you can add a video file to the audio by using this menu item.

- If the added video file already has an audio stream, it is discarded.
- The length of the audio file determines the total length. If the attached video is longer, it is shortened; if it is shorter, black frames are added for the rest of the audio file.

If a video file window is active, the menu command changes to "Remove video...". This lets you extract the audio component from a video by loading the video in Sound Forge Audio Studio and then removing the video portion.

Exit

Exits Sound Forge Audio Studio 12.

Keyboard shortcut: Alt + F4

Edit Menu

Undo

Undo and redo give you the freedom to experiment with your media. Edit to your heart's content. If you change your mind, you can always undo your changes. If you change your mind again, you can redo the undone edits.

Shortcut key: Ctrl + Z, Alt + Backspace

Restore

"Redo" recalls the step that was undone immediately before.

Shortcut: Ctrl + Shift + Z

Cut

Copies audio material in the selected range to the clipboard and deletes it from the project. Audio material following it will be moved forward.

Shortcut: Ctrl + X

Copy

Copies the audio material in the selected range to the clipboard. Please note that any previous contents in the clipboard will be deleted when you do this.

Shortcut key: Ctrl + C, Ctrl + Insert

Insert

Inserts the contents of the clipboard at the position of the mouse cursor; existing audio material is shifted back.

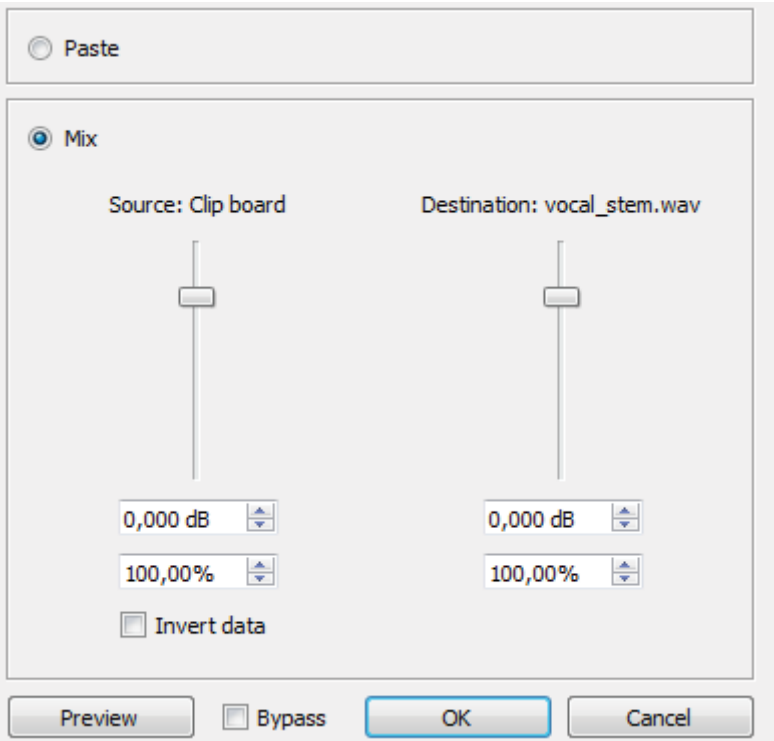
Shortcut: Ctrl + V, Shift + Insert

Paste Special

The "Insert contents" submenu contains other options for inserting the contents of the clipboard.

Mix

Mixes the contents of the clipboard with the audio material at the cursor position.



The mix ratio can be set in the dialog. The "Paste" option corresponds to the normal paste command (Ctrl+X).

Shortcut key: Shift + M

Overwrite

Overwrites the audio file at the cursor position with the contents of the clipboard; as opposed to normal pasting, the existing content is not shifted back.

Paste to New

Opens the contents of the clipboard in a new file window.

Keyboard shortcut: Ctrl + E

Trim/Crop

Removes all audio material outside of the selection.

Shortcut key: Shift + T

Delete (Clear)

Deletes the selected audio material. Audio material that comes after the deleted material is pushed forward.

Shortcut key: Del

Select all

The entire audio material is selected.

Shortcut: Ctrl + Del, Ctrl + A, Ctrl + Num 5

Tool

Here you can select a tool to work with in the file window. For most tasks, the preset Edit tool is sufficient. However, there are other modes available for specific tasks. Depending on the mode, the function of mouse-clicks in the file window changes. The current tool is indicated by the appearance of the mouse pointer in the track window.

Tip: Right-clicking any tool (except for the magnify tool, see below) opens a context menu with commands from the editing and view menus.

Edit

The **Edit** tool is the preset tool. Use the Edit tool to select data and position the cursor.

Position the cursor by clicking in the waveform. Click and drag to select a range in the audio material for editing. Double-click in the waveform to select the entire waveform.

For more information about selecting ranges , please read the Selection (view page 42) chapter.

Shortcut: Ctrl + D

Pencil (Waveform)

Repair short distortions such as crackling directly in the wave form of the audio file by using the Waveform Pencil.

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. As soon as you click in the waveform display with the pencil, it will automatically zoom in to display individual sample values.

Tip: For standard vinyl record crackling, there is the special Declicker/Decrackler tool located under Extras > Audio Restoration

Zoom In

Use the right mouse button to zoom out of the project, and click with the left button to zoom into it.

In this mode you can also draw a rectangle around the area you want to zoom in on.

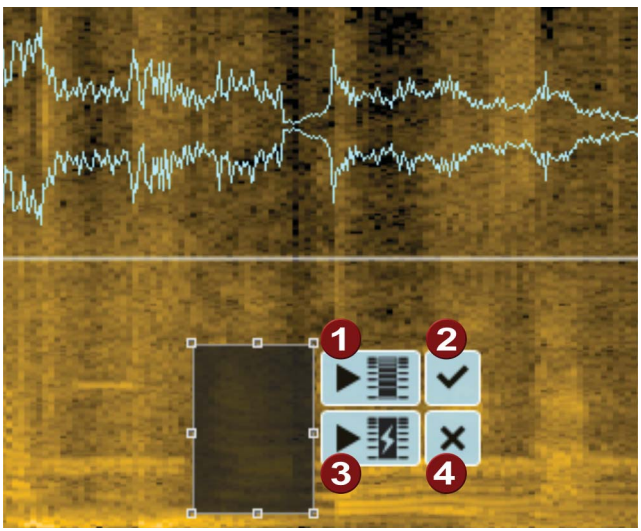
Tip: The Zoom In tool can also be temporarily activated by holding the Z key.

Spectral Editing

You can remove individual noises within the sound spectrum of the audio material with the help of the "Spectral Editing" mouse mode.

The view of the data window will change to spectral display (view page 106). 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 around the disturbances:



1. The upper Play button plays the corresponding part with effects applied; your edits can be heard immediately.
2. Clicking the checkbox will include the edits in the audio material.
3. For comparison's sake, you can use the lower Play button to listen to the same part without any effects.
4. Click the X symbol to cancel the edits. If you do this, no filters will be applied.

Preview

By holding down the mouse button, the project can be previewed at the point where the cursor is positioned.

The cursor follows the movements of the mouse. This tool is especially suited to searching for specific parts in audio material. 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.

There are different scrubbing modes which you can set in the playback options (view page 154) (Options > Settings > System Options > Playback).

Tip: The preview tool can also be temporarily activated by holding the Z key.

Center Cursor

Places the cursor in the center of the file window.

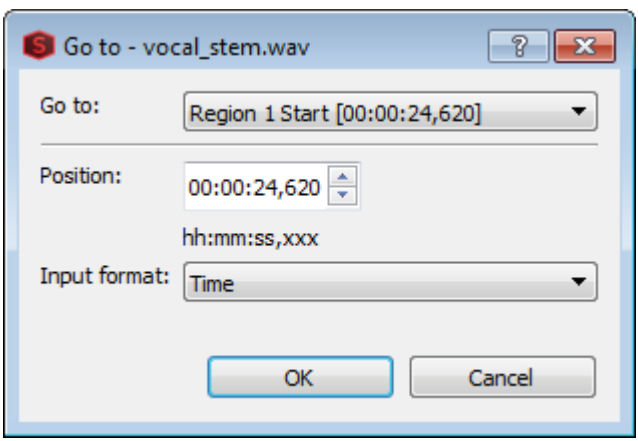
This command does not actually move the cursor to a new position in the data window; it simply redraws the display so that you will see the areas of the sound file equally on either side of the cursor. If the cursor does not seem to center, the cursor may be very close to the beginning or end of the sound file (and therefore cannot be centered), or you may be zoomed out so the entire waveform is already visible.

Tip: If you have a range of data selected, you can switch the cursor between selection points using the Home and End keys.

Shortcut: ., Num *

Go To...

This dialog lets you move the cursor to a specified position.



If necessary, the view will adjust to make the cursor visible. The (preset) option "User-defined" lets you enter a specific position numerically. The input format can be set to time or samples.

Under "Go to" you'll also find the predefined targets File Start, File End, Set Marker (if applicable), Region Start Point and Region End Point, as well as the play cursor (the latter merely moves the view).

Keyboard shortcut: Shift + G

Selection

The Selection submenu contains a number of commands for editing the selection using shortcuts.

Snap to grid

Start and end of the selection will be moved to the nearest snap points.

Shortcut: T

Snap Edge to Grid

Moves only the range edge where the mouse cursor is located to the next snap point.

Tip: Pressing the Num5 key makes the cursor switch range edges.

Note: Depending on whether you make a range selection from the left to the right, or the other way around, determines if the cursor will be at the beginning or end of the range. When you move one of the range edges with the mouse, it always jumps to the end of the range.

Shortcut: Shift + T

Snap to Zero

Start and end of the selection will be moved to the nearest zero crossovers.

Note: This is done to prevent crackling during editing, which results from jumps in value during an audio material sequence.

Snap Edge to Zero

Moves only the range edge where the mouse cursor is located to the next zero crossover.

Extend to Next Zero

The start and end of the selection will be extended to the next zero crossover, meaning that the start will be shifted to the next zero crossover in front of its initial position and the end shifted to the next zero crossover behind it (the end of the wave). The function is almost identical to "Snap to Zero", but also ensures that none of the original selection is lost during alignment.

Shortcut: Z

Extend Edge to Next Zero

Extends only the range edge where the mouse cursor is located to the next zero crossover. (Depending on whether the cursor is at the beginning or end of the range, it will move the edge backwards or forward, respectively)

Shortcut: Shift + Z

Mark In/Out

This sets in and out points like those in cutting programs. Select "Mark In" or click I to set the start of the range at the cursor position. Select "Mark Out" or press O to set the end of the range.

Note: When O is set before I or has not been set, I = File start. For analog, when I is set behind O, O = File end. You can use commands individually to define a range from the file start to the cursor, or from the cursor to the file end.

Shortcut: Mark In I
Mark Out O

Halve/Double

Cuts the range length in half or doubles it.

Shortcut key: Halve ;
Double '

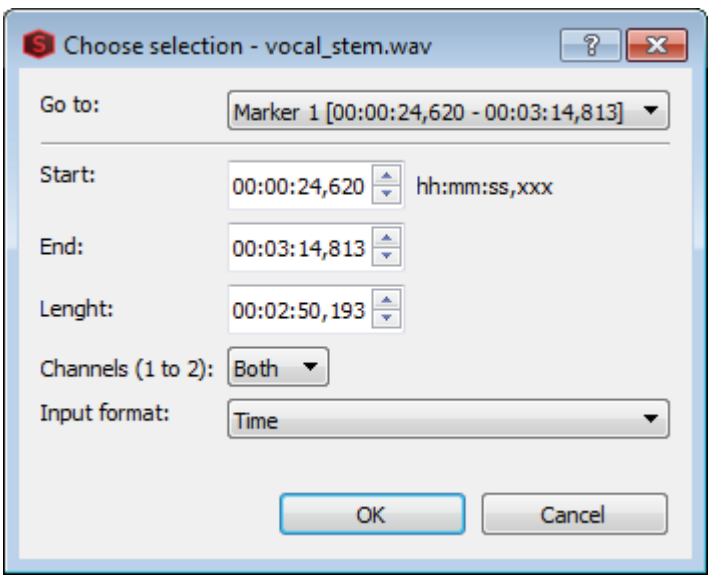
Move left/right

Moves the range to the left or right at a distance equal to the length of the range.

Shortcut key:	Move range left	<
	Move range right	>

Set...

In this dialog, you can set the selection to a specific range.



The (preset) option "User-defined" lets you enter a start and end position numerically. The input format can be set to time or samples.

Under "Go to" you'll find the saved regions as well as the specified ranges Cursor to File Start, Cursor to File End and All Sample Data. Under "Channels" you can limit the selection to individual channels.

Keyboard shortcut: Ctrl + Shift + D

Switch

Simply clicking inside the file window will relocate the cursor. Doing this will cancel any existing range selection. If this happens by accident, you can use the "Switch" command or the S key to restore the previous range selection (the light blue selection on the timeline).

Note: Using the "Switch" command will also move the curser back to the range edge it was on before. Using "Switch" again will deactivate the range again, but the cursor will remain at its (new) position.

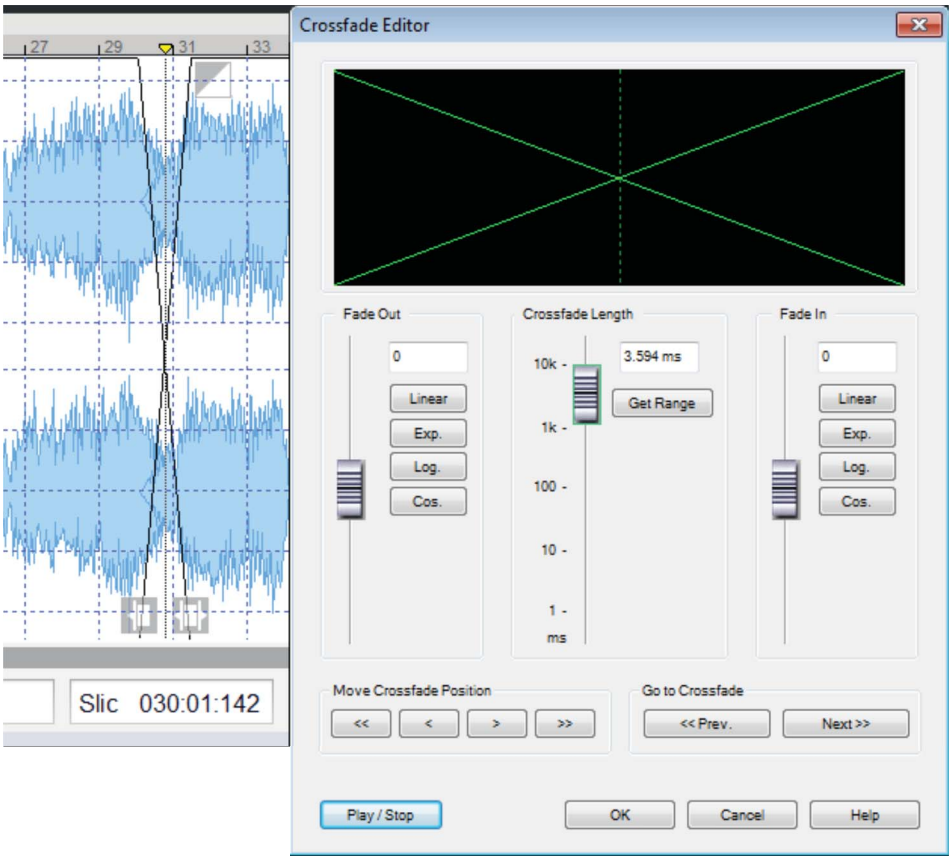
Shortcut key: S, #

Regions List

This submenu contains commands for editing markers and regions in the Regions List (view page 64). This data can also be edited directly in the file window or the Regions List; however, the commands here can be assigned shortcut keys (view page 157).

Crossfade Edit

Editing audio data creates slices (view page 52) which conatin borders that fade into each other. The shape, length and position for a crossfade can be set in the crossfade editor.



Crossfade length: The length of the crossfade can be set using the fader. The crossfade length can also be defined by a selected range in the audio file. Before openign the crossfade editor, mark a range and clicking on the "Get range" button in the crossfade editor.

Fade In/Fade Out: There are lots of different curve types available for the crossfade:

- Linear
- Exponential
- Logarithmic

- Cosine

The faders can still be used to change the curves in the corresponding available range.

Move crossfade position: The crossfade will be moved together with all the following audio material. When you use the < > keys step size 1 is used, with << >> step size 2. These step sizes can be set under "Edit" > "Slice" > "Move Step Settings..." .

Go to crossfade: Switch to the previous or next crossfade using this button.

Play/Stop: This button starts playback in the selected range or at the cursor position.

Slice

This submenu contains commands for easily managing slice positions and crossfades (view page 52).

Split at playback marker

The audio material will be sliced in two at the marker. In the current range selection, the audio material is sliced to both range edges, creating 3 slices.

Shortcut: Shift + S

Select Slice under Playback Marker/Range

Selects the slice under the playback marker or under a selected range for editing.

Select previous/next slice

Selects the previous/next slice.

Remove slice selection

Cancels the slice selection.

Play Slice

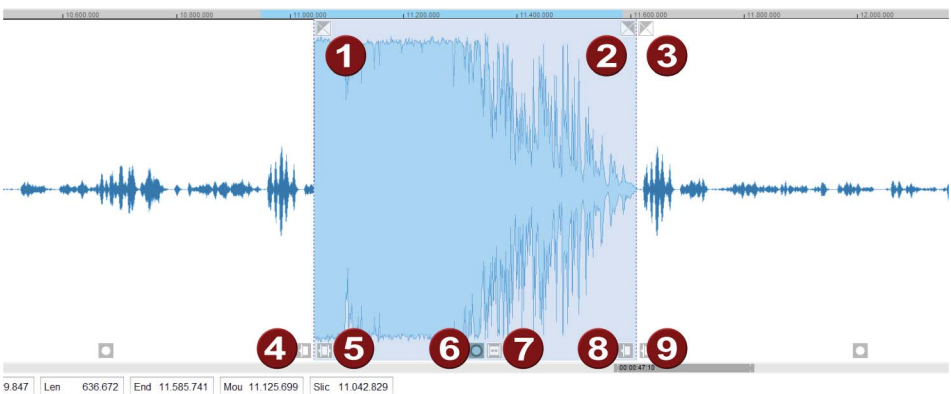
Places the cursor at the beginning of the selected slice and plays the selected slice.

Shortcut: Ctrl + Spacebar

Move slice left/move slice right

The following commands are used to edit slice and fade positions with the keyboard. They correspond exactly to the edits performed with the slice handles (view page 54). Each movement is available in a smaller and a larger step size. The command for the second step size is located in the "Step Size 2" submenu. You can set the step sizes in the "Slice-Move Step Settings..." dialog.

Note: For more information, read the chapter entitled Slices (view page 52).



Command	corresponds to slice handle
Slice to left/right	5
Right slice to left/right	9 (right)
Slice start to left/right	4
Slice end to left/right	8 (right)
Fade-in handle to left/right	1
Fade-out handle to left/right	2
Slice content to left/right	7

Clear Undo History

Recent edits are saved and can be undone step by step by using the undo feature. This feature can be used to delete the list.

> Tip: You can change the number of saved undo steps in the program settings (view page 156).

View Menu

In the View menu, you can adjust the zoom level of the active file window and open various windows in the docking area.

Horizontal zoom (time)

Zoom in Full

Sets the zoom level to maximum in order to show individual sample values.

Zoom Out Full

Zooms all the way out in order to show the entire file in the window.

> Tip: You can also double-click the scroll bar in the data window to do this.

Normal/Selection

"Selection" zooms the data window into the currently selected range. "Normal" switches back to the last zoom level.

Shortcut key:	Selection	Ctrl + Up Arrow, Ctrl + Q
	Normal	Ctrl + Down Arrow

Vertical zoom (sample level)

Zoom Out Full

Zooms all the way out of the waveform so that the entire waveform can be seen.

Explorer

Shows or hides the explorer (view page 60).

Shortcut: Alt + 1

Video Preview

Shows/hides the video preview window.

When you load a video file, the corresponding video picture will appear in the video preview window. Right-clicking the video window lets you toggle the aspect ratio between 4:3 and 16:9, select a range of video sizes and switch to fullscreen mode.

Tip: If video playback is jerky, decrease the size of the video window.

Shortcut key: Alt + 3

Time Display

Shows the time display (view page 62).

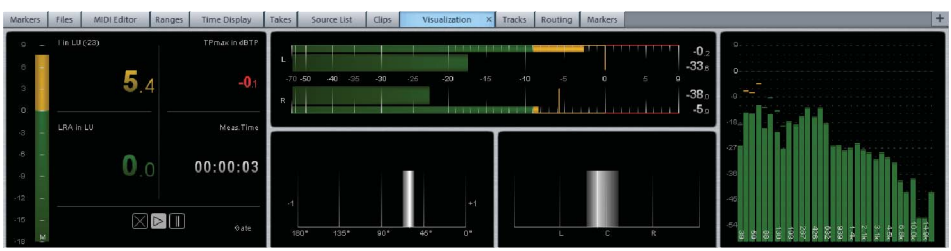
Shortcut key: Alt + 4

Visualization

Shows/hides a visualization window.

Shortcut key: Alt + 5

The visualization screen displays the audio material graphically. The interface allows you to design multi-visualizations individually. You can integrate the visualization interface into the Docker (view page 69) or open it in its own window.



Normally the visualization is divided into several display windows. You can choose from the following display options for each window: **Peak meter** (control display), **Phase Oscilloscope (vectorscope)**, **Correlation meter**, **Direction meter**, **Spectroscope**, **Spectrogram, Bit meter**, **Oscilloscope** and **Tuner**.

The visualization display can be adapted easily to your personal presets. To do this **right-click on one of the visualization windows** to open up a context menu where you can change the window settings. The size of the window can be changed by dragging its edges.

Device: Here you can set which output the visualization should refer to. With this option you can determine which device should be displayed if multiple audio devices are available.

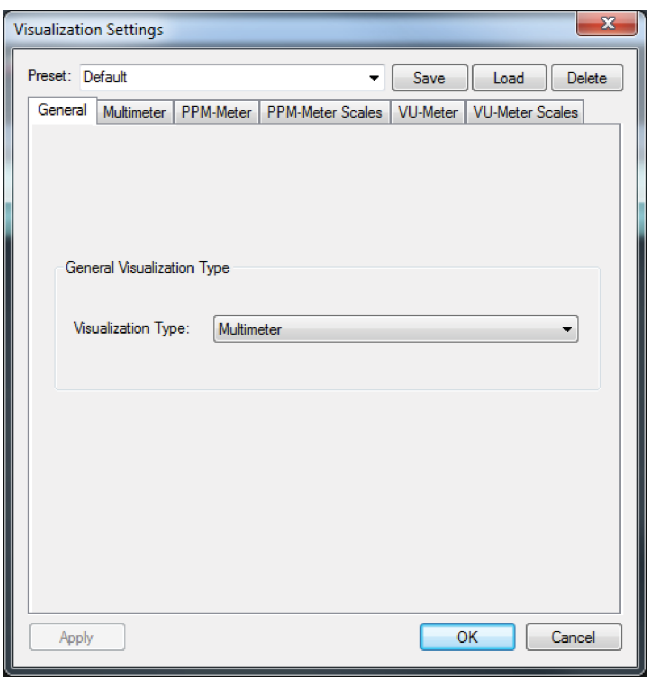
Visualization layout: Here you can load the preset **Layouts** and save your customized **Layouts**.

Reset this visualization (double-click): This resets the display of the current visualization.

Reset (All): This resets all the displays of the opened visualization.

Presets: Here you can choose from various presets for the visualization display. If you have already saved your own custom presets, they will also be displayed here.

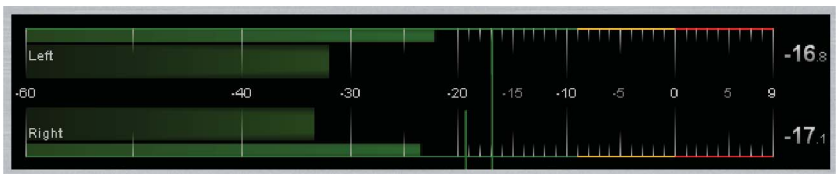
Settings...: Click here to open the visualization settings dialog.



In the "Instrument" field, look for the desired display instrument. According to the selection a range of presets is available in the preset list. Confirm your selection with "Apply".

You can also load or delete the settings using the corresponding buttons.

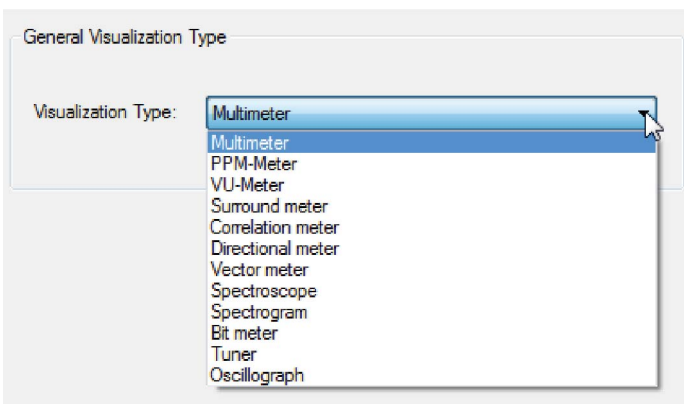
Peak Meter



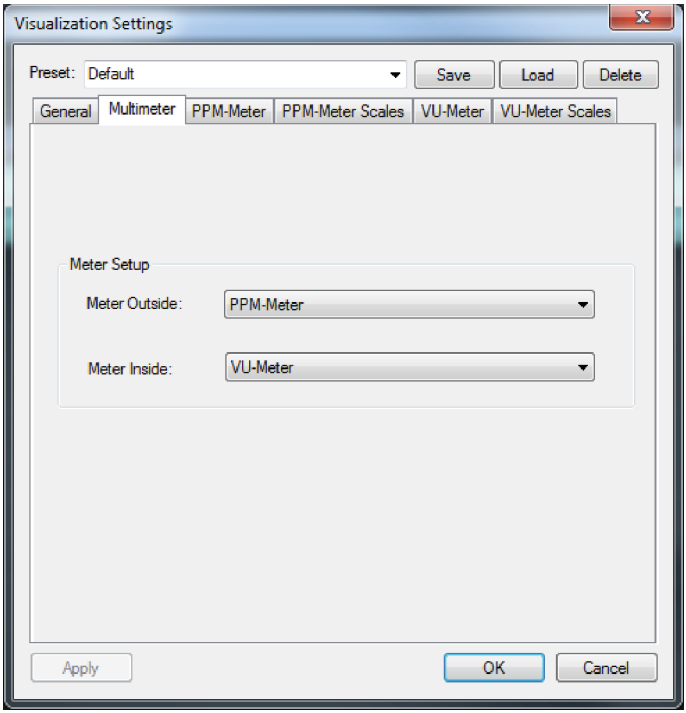
The (multi-) peak meter (Instrument multimeter) displays the volume during playback in dB. The thinner, outer bars show the **Program Peak Meter (PPM Meter)** and the thicker, inner bars display the **VU meter**, which you might be familiar with from analog equipment.

Both meter displays are based on a standardized peak meter with precisely defined display characteristics. The PPM meter displays the peaks of an audio signal, whereas the VU meter displays the metered values over a specific metering time period.

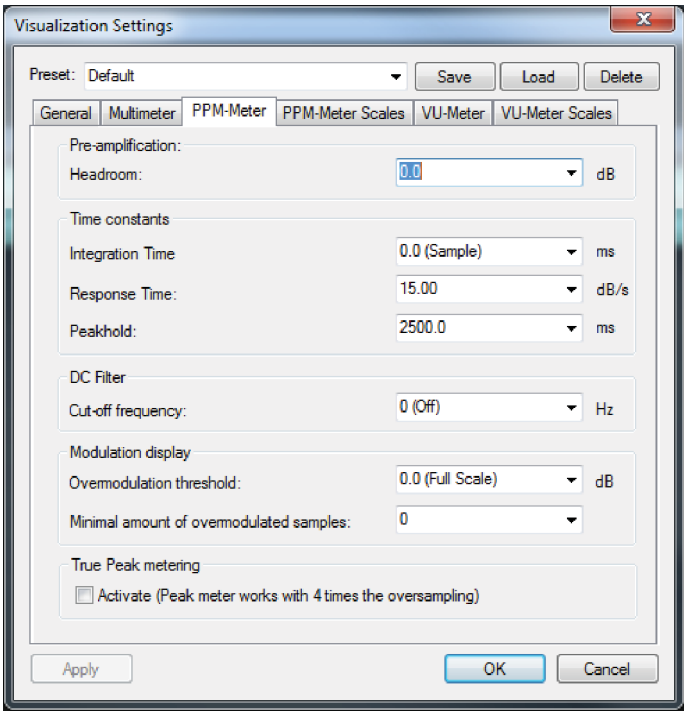
If you only want to see either the PPM display or VU display, select the corresponding display instrument.



You can also determine the display configuration for **Multimeter**.



PPM-Meter



Headroom (dB): With this you can specify a level offset which will be globally added before measuring. This is due to the fact that different systems are calibrated differently. A value of 9.0 (IRT) e.g. ensures that the level is calculated with + 9 dB to make sure that you have 9 dB more headroom available if your playback device is accordingly calibrated.

Integration time - PPM (Mms): This constant ensures that the peak meter reaction time is delayed by the set value, so the display doesn't appear quite as fast for individual level peaks. The created lag is modeled on the analog behavior of old, traditional peak meters.

Release time (dB/s): The lower the value, the slower the display bars of the peak meter will move when descending. A typical value would be e. g. 13.3 which corresponds to a release time of 1.5 seconds at 20 dB.

Peak hold (ms): With this value you can specify how long the level peaks should remain.

Cut-off frequency (Hz): With this high-pass filter you can filter out the DC component so that it doesn't affect peak meter measuring.

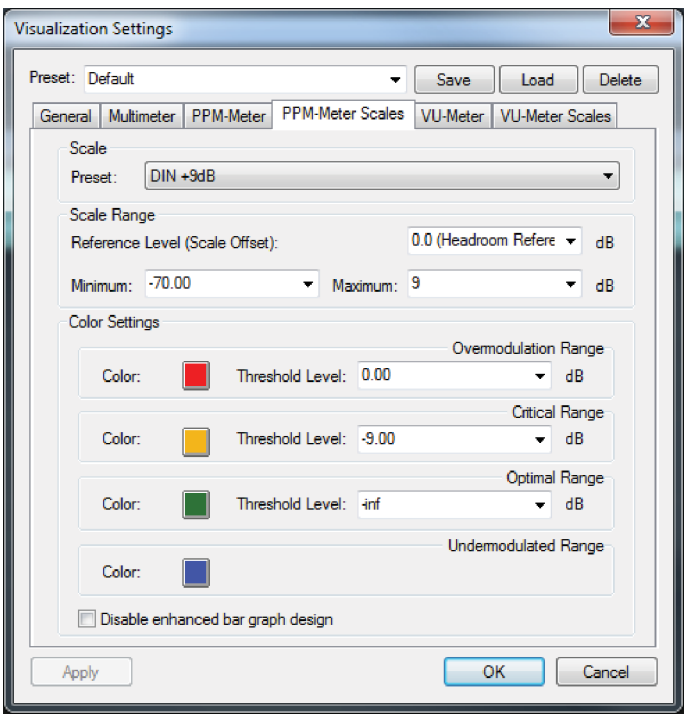
Clipping threshold (dB): This values indicates from which dB value clipping is displayed, i .e. when the display hits the red.

Minimum number of clipping samples: This value represents the number of consecutive samples that are allowed to exceed the clipping threshold before the display hits red.

Activate True Peak Measuring: In True Peak mode peak meter measuring is performed with fourfold oversampling.

PPM-Meter Scale

You can choose different display options on this page.

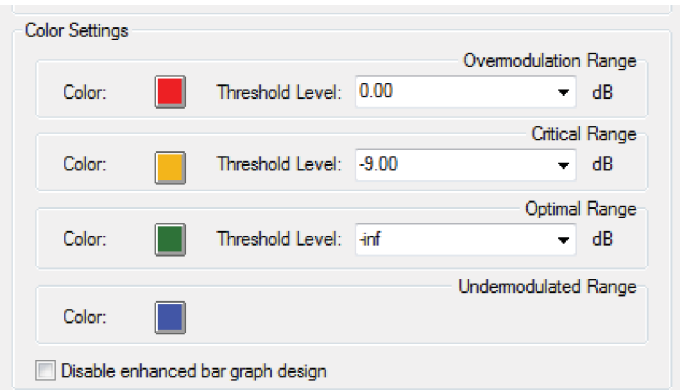


Scale: Select from a range or different scale display settings. In the "Presets" you will find a selection of peak meters with different scales and display characteristics based on the standards in various European countries.

Reference level (Scale offset): Enter the value for the scale offset and reference level here (PPM Meter > Headroom). The scale offset will be added to the reference level.

Minimum/Maximum: Here you can set the minimum and maximum display values. You can restrict or expand the value range of the scale.

Color settings: Enter the colors and thresholds for clipping, critical, optimal and weak ranges.



Note: If you define a threshold for the optimal range which is higher than the reference level minimum, a defined color will be displayed for the weak range when it falls below the set threshold level.

VU Meter

As with the PPM meter, if you open the tab you will have the following parameters available: **Headroom**, **Integration Time**, **Peak Hold** and DC Filter **Cut-off**.

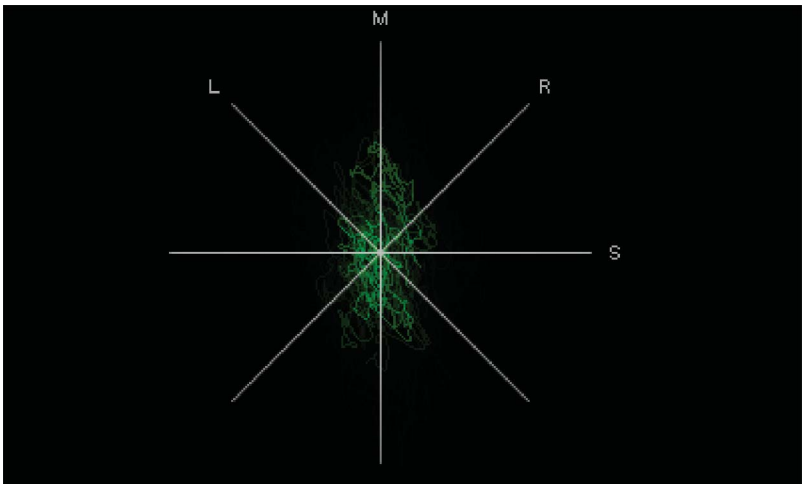
If you check the option **+3dB IEC**, The set headroom under "PPM Meter" - standardized according to DIN IEC 60268 - will be increased by 3 dB.

You can also choose between the current peak hold display and the RMS value display.

VU Meter Scale

As with the PPM-meter scale, here you can select the display options and change the color codes for the bar graph, which employs the same scale representation as the PPM meter.

L/R Oscilloscope (phase correlation):

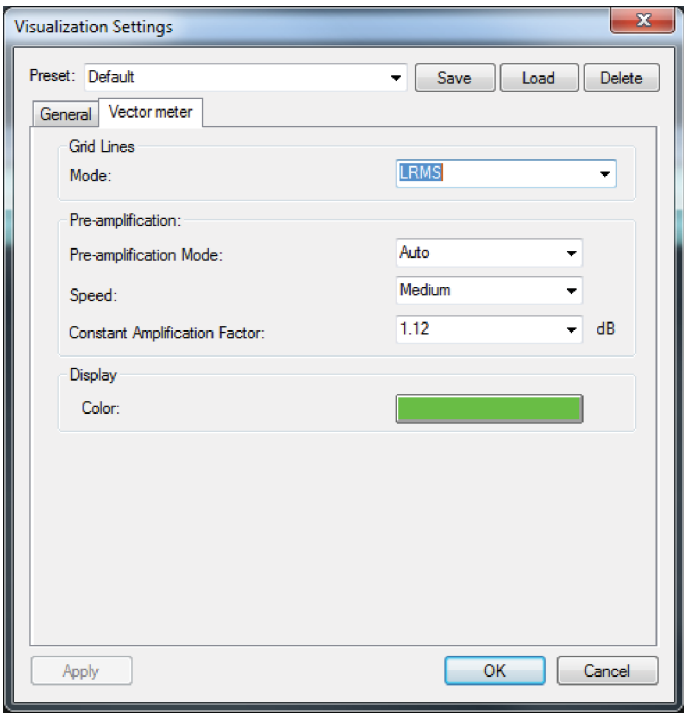


Phase correlation provides information on the distribution of the stereo image in your recording. A mono recording would be displayed as a vertical bar in this view. A song produced in stereo is by contrast shown as "diffuse ball," because multiple instruments were assigned to the mix in different panoramic positions.

The wider the display, the wider the stereo field of the recording. Please note, that a broadening of the display implies more deletions, and in turn that the signal is not as mono compatible.

If the signal display tends to be diagonal the stereo mix is not balanced. A channel would be accordingly louder than the other.

In the settings dialog you can access the various grid settings. You can choose between Mid/Side, L/R, Hybrid and a mode for measuring signals.

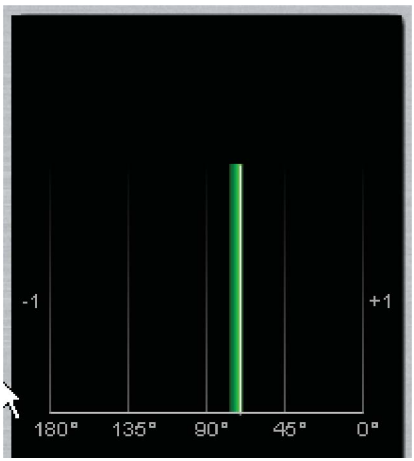


The image is correspondingly increased in pre-amplification mode, to ensure readability. You can also switch off this mode and set a constant amplification factor instead.

Using the "Speed" function you can set the rewind speed at which the curve is dimmed.

Finally you can set the display color.

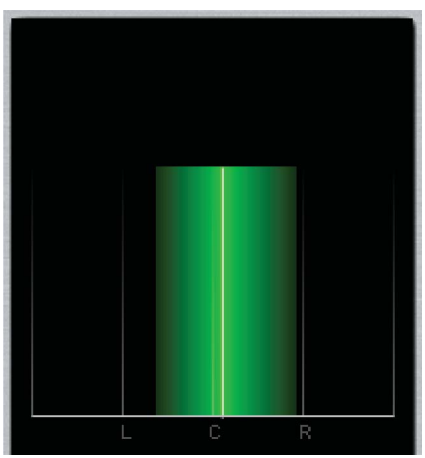
Correlation meter



With the correlation meter you can read phase offsets between the two stereo channels, and examine how much has been deleted. If the signal display is in the left, red area between 90° and 180° the signal will no longer be reproduced properly over a mono receiver.

In the settings dialog you can select the colors for Mono and Stereo and mono compatible ranges, threshold values for the ranges can be set too.

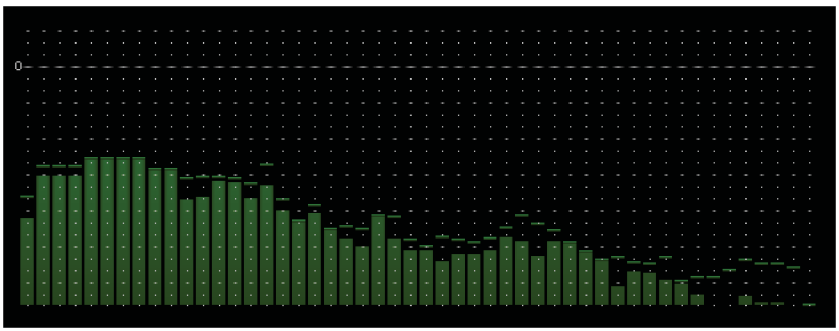
Direction meter



The direction meter displays the signal's detection direction. The width corresponds to the correlation measurement.

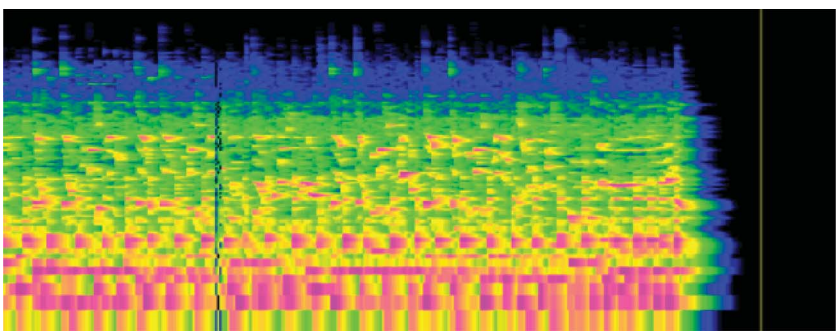
In the settings dialog you can select the colors for Mono and Stereo and mono compatible ranges, threshold values can be set too.

Spectroscope



In the Spectroscope the signal is divided up into individual frequency ranges. 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 displayed as frequency proportions in a time curve. The volume of frequencies is visualized by its brightness.

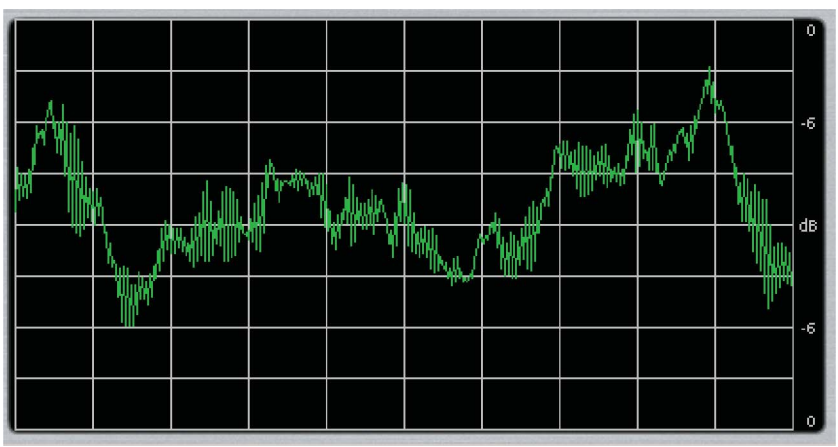
The spectrogram is designed to be able to immediately detect noise in your recordings. Audible distortion noises are mostly louder than the music, and are usually limited to a certain frequency spectrum. They are highlighted with colors in the Spectrogram.

In Sound Forge Audio Studio 12 you can remove unwanted noise with the help of Spectral cleaning.

Bit Meter

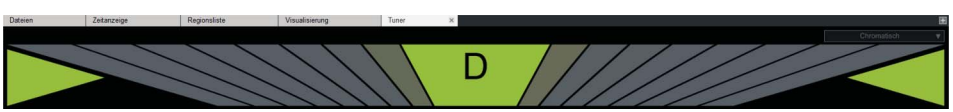
The bit meter shows you the rate at which the signal is being calculated at and which maximum editing rate is possible.

Oscilloscope:



The oscilloscope displays the signal amplitude over the time curve.

Tuner



The tuner shows the respective pitch for the signal. Use the visualization to tune your guitar or another instrument.

In the middle you will see the current pitch. The calibration marks show the deviation between the played pitch and the exact pitch. Red triangles to the

right and left show the direction the strings should be tuned. If the tuning is exact, the triangles turn green.

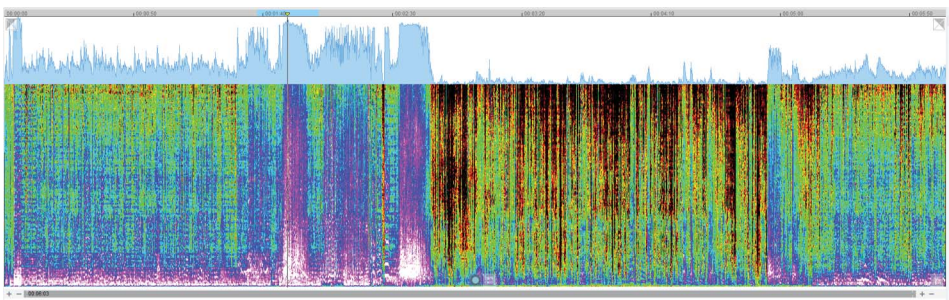
In the default "chromatic" mode, the tuner tries to assign each played string to the chromatic half-tone steps. You can select other conventional tunings for guitars, basses and ukuleles from the list. Then, only the notes relevant in each corresponding tuning will be recognized.

Regions List

Shows the Regions List (view page 64).

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. Different color palettes can be selected in the menu "Options" > "Spectral display".

Spectral display of the audio enables specific disturbances in the audio to be detected. Clicking is shown as vertical lines across the entire frequency spectrum; continuous disturbing sounds are shown as 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.

Insert Menu

Markers

A marker is added at the cursor position. The markers are numbered consecutively by default; double-clicking on the markers allows you to give them custom names. Markers can be moved with the mouse.

For existing selections, the marker is placed on the same region edge as the cursor.

> Tip: The Num5 key switches whether cursor is placed at the beginning or end of the selection.

Double-clicking between two markers selects the duration as a range.

Markers can be further edited in the Regions List (view page 64).

Shortcut key: M

Region

Region markers are placed at the edges of the selected range. Regions are numbered consecutively by default (markers and regions are numbered together); double-clicking on the markers allows you to give them custom names. All markers can also be moved using the mouse.

Double-clicking between two markers selects the duration as a range.

Regions can be further edited in the Regions List (view page 64).

Shortcut key: R

Set CD title index

Use this function to set a track marker at the position of the playback marker (index makers). The numbers of previously set markers are adjusted automatically. Each title of a CD requires a track marker, which is typically placed shortly before the track starts.

To manage the markers or rename them, use the "CD title/index manager (view page 65)".

Set CD sub index

Use this function to set a sub index marker. The numbers of previously set sub index markers are adjusted automatically.

Sub index markers are not essential when creating a CD but are useful for marking sections within an audio track.

Set CD Pause Index

The CD pause index is a special Sub index (Index 0). Choose this option to place a pause marker at the current play cursor position.

Pause markers allow CD players to switch their output to absolute silence until the next Track Marker. and to count backwards to the start of the next track.

Set CD end index

This marker indicates the end of the CD. Setting an end marker is not necessary - if one is missing, the end of the file will be used as the end of the CD.

You can use the CD end marker to burn a file partially to CD. Remove all track markers prior to the first track to be burned and then set the end marker after the last track to be burned.

Silence...

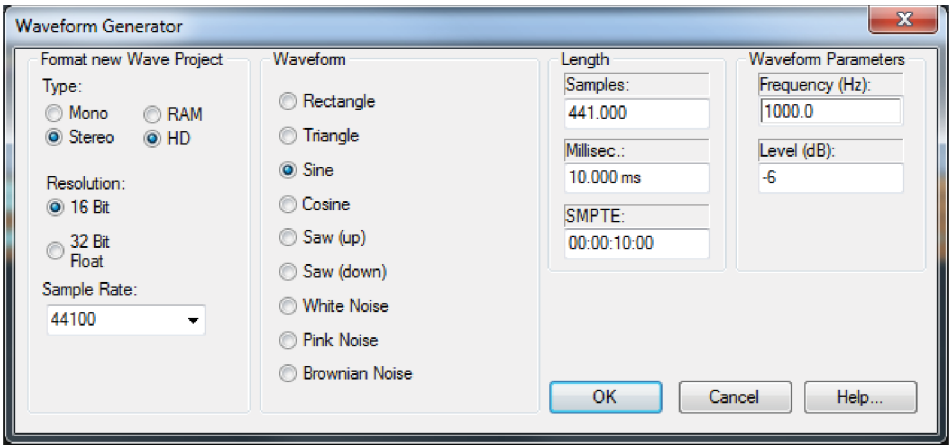
This command inserts silence at the current playback marker or at the beginning of the selected range; the audio material that follows is shifted back.

If a range is selected, its length is adopted as a guideline. The selection remains intact even after the operation. It is possible to change the unit in the input field.

Synthesis

Simple

This dialog contains a powerful generator of (several) test tones.



Select RAM or HD as the **type** in mono or stereo at a **resolution** of 16-bit or 32-bit float.

The following **sample rates** are available: 22050, 32000, 44100, 48000, 88200, 96000, 176400, 192000, and 384000.

As a **waveform** you have the choice between rectangle, triangle, sine, cosine, sawtooth (upwards), sawtooth (downwards), white noise, pink noise, and brown noise.

You can specify the **length** in samples, milliseconds, or SMPTE code.

The frequency (Hz) and volume (dB) are the final two **waveform parameters** at your disposal. The created signal can be further edited as an audio file or a virtual object.

Transport Menu

Record...

This command starts a recording in a new file.

More information on recording can be found in the chapter Record audio (view page 37).

Play

Start playback from the cursor position, by selecting from the start of the selection range.

Shortcut: Spacebar, F12

Playback in selection/loop

Start playback from the cursor position, by selecting from the start of the selection range. Where a selection is available, this is played back in a loop. If the cursor moves to a looped area, this will also be played back in a loop.

Pause

Stops playback; the cursor remains at the current playback position.

Shortcut key: Enter

Stop

Stops playback and places the cursor back at its starting position.

Shortcut key: Spacebar, F12

Tip: To switch the stop and pause functions of the spacebar and enter key, activate the option "Spacebar and F12 Play/Pause instead of Play/Stop" in the playback settings (view page 154).

Go to Start

Sets the play cursor at the beginning of the file.

Shortcut: Ctrl + Home

Go to End

Sets the play cursor at the end of the file.

Shortcut: Ctrl + End

Preview Cut/Cursor Start (In Point)

Plays a short duration up to the beginning of the selection.

Shortcut: Ctrl + Shift + K

Play from cut/cursor start (In Point)

A short duration from the start of the segment will be played.

Preview Cut/Cursor End (Out Point)

Plays a short duration up to the end of the selection.

Play from cut/cursor end (Out Point)

A short duration from the end of the segment will be played.

Preview Cut/Cursor

Plays a short duration up to the beginning of the selection and directly after the end of the selection. This allows you to hear what it would sound like if the selection were deleted.

Shortcut key: Ctrl +K

Playback Settings

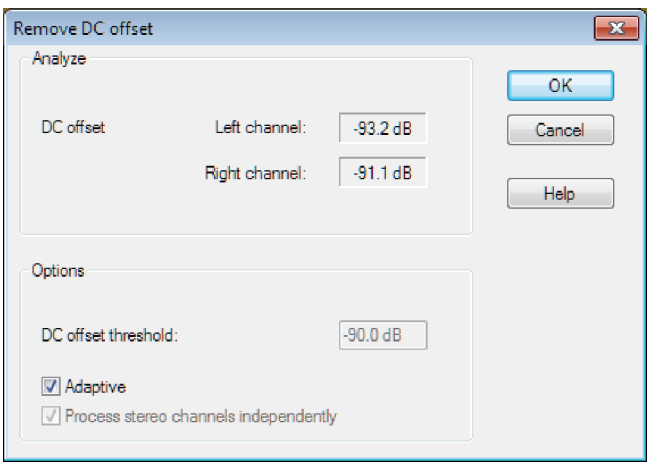
See "Options" > "Settings..." > "Playback" (view page 154)

Process Menu

Contains general audio processing features.

DC Offset...

This function is 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 **Adaptive** mode (preset) a variable, rather than constant, DC voltage is emanated. On technical terms, all frequencies under 10 Hz are used as DC. If "Adaptive" is deactivated, a constant DC is emanated. You can enter a minimum **DC offset threshold**, which indicates where DC offset removal will kick in. You can also edit stereo channels together to reduce computing time.

Fade...

This feature allows you to fade selected sample ranges in an audio file in or out. The amplitude from the start value at the beginning or the range to the end value at the end of the range will be changed.

The "Fade In" and "Fade Out" menu commands are shortcuts that allow you to quickly set linear fade-ins and fade-outs from 0% to 100% of the volume (or the other way around). "Graphic" opens a dialog where you can select any start and end values you want and switch the curve progression between logarithmic, linear and exponential as much as you want.

Tip: You can use the fade handles on slices (view page 52) and the Crossfade Editor (view page 90) to easily make crossfades and change them at a later point in time.

Invert/Flip

This function inverts the sample data within the selected range along the amplitude axis, i. e. the phase is inverted. Negative values become positive and vice versa.

This allows you to match recordings with different phases to one another. This feature is reversible and can be applied to both channels in an audio file, or to just the left or just the right channel.

Mute

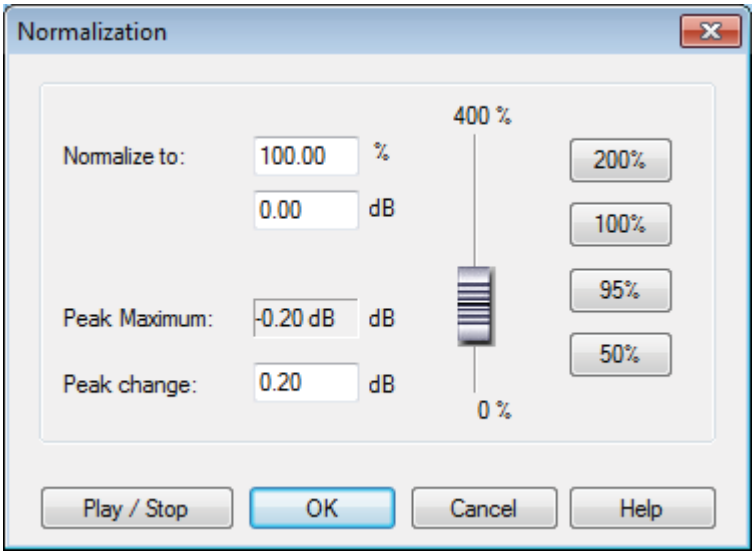
The data in the selected range will be deleted. The data behind this will not be moved, silence will be added behind the affected range. The length of the file stays unchanged.

Normalize

Use this feature to change the amplitude of the sample data. It will be amplified in such a way that the highest amplitude that appears in the range is set to 100% (or another value between 1% and 400%) of the value range. The maximum value will first be determined and calculated using the selected percentage. All values are then weighted by the new factor.

With this feature you can set samples to maximum gain or even overmodulate them for targeted sound editing. Note that the noise level also rises during normalizing.

This feature is especially useful before converting high bit resolutions to lower ones. This guarantees that the low dynamic range is made full use of at lower resolutions.



Note: If you experience very slight clipping during recording and then proceed to normalize the material, you won't achieve the same quality as when you produce an overmodulated 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% will not 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 can be entered in % or dB. 100% = 0 dB = full level. Values above 0 dB will cause digital clipping.

Maximum Level: Displays the highest peak in the selected range.

Peak Change: Displays the level change in dB, in accordance with the selected normalize level and the detected peak maximum.

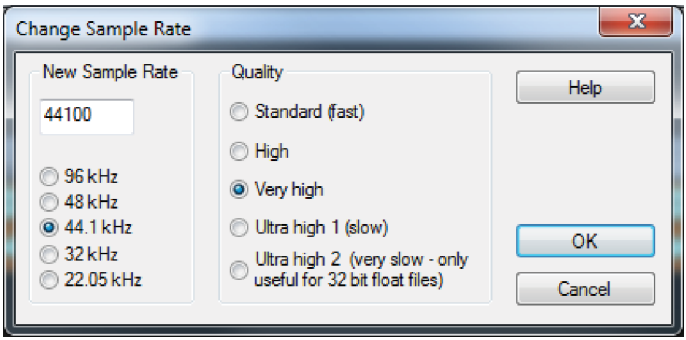
Normalize (Quick Access)

Quick normalization to 100% without having to open the dialog.

Shortcut key:	Normalize	Shift + N
	Quick Access	N

Resample...

This feature lets you change the sample rate of an audio file.



If the sample rate is increased, the change is made almost completely without loss. However, the required disk space does increase.

If you reduce the sample rate, you'll lose overtones. Cutting the sample rate of a 44.1 kHz sample in half to 22.05 kHz limits the frequency response of the resulting signal to 11.025 kHz. The frequency response always corresponds to half of the sample rate.

Click "OK" to enter the file name for the new project.

Quality

Here you can specify, what quality Anti-aliasing filtering or Reconstruction filtering should be used when increasing or decreasing the sampling rate.

Standard: At this quality level, a faster interpolation algorithm is used, which is also applied during realtime resampling.

High: In addition to the "standard" quality level, a filter is set to dampen the alias frequencies or to improve the reconstruction (interpolation). When using extreme conversion ratios, such as converting from a 48 kHz signal to a 22.05 kHz signal, this setting may improve the results.

Very high: Here the linear-phase **UTR – "Ultra Transparent Resampling"** algorithm is used. This is a relatively new resampling algorithm for considerably faster and even better linear-phase, high-quality resampling. The speed increase permits the use (even in real time) of high quality settings (object resampling) that were previously only available for offline processing in

previous versions. The quality of offline resampling also benefits from the new algorithm, since calculation in higher quality stages is now even more precise.

Extremely high 1: In addition to the "Very high" quality level, this algorithm works at a higher filter setting. Doing this increases CPU load dramatically.

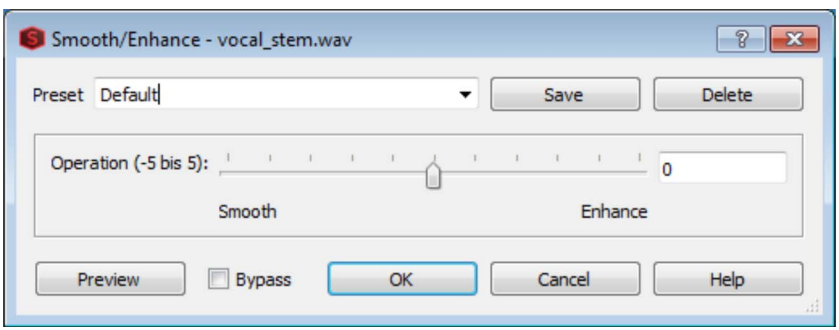
Extremely high 2: In addition to the "Very high" quality level, this algorithm works at a considerably higher filter setting. This setting is only meant for use with 32-bit float files because the maximum possible artifacts from aliasing are below 96 dB.

Reverse Play

With this feature, the sample data in an audio file within the selected range along the timeline is reversed so that it can be played from the end to the beginning.

Smooth/enhance...

This feature allows you to add or remove high-frequency content to/from an audio file.



Drag the Operation slider toward "Smooth" or "Enhance"

The "Smooth" feature will smooth fast-changing transients in a sound and is useful for removing glitches.

The "Enhance" feature boosts the very highest frequencies (close to the Nyquist frequency, i.e. 1/2*sample rate) in the sound file, making the sound file sound more vivid. It is useful for compensating for the effects of downsampling or for bringing out very fast transients in a sound file.

Note: Since the Enhance feature boosts frequencies near the Nyquist frequency, the current sample rate will determine what frequencies will be affected. At 44,100 Hz, the effect can be very subtle.

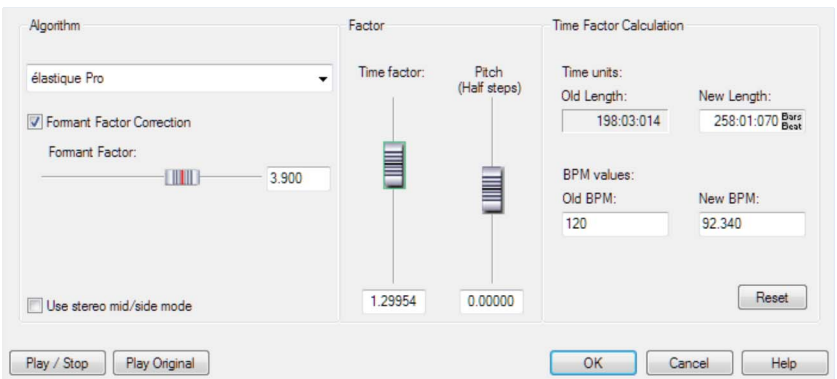
Swap Channels

This feature lets you swap the right and left channels on stereo samples in order to correct channel swapping that may have occurred during recording.

This feature can be reversed, i. e. if you don't re-select the range, opening it again will restore the original material.

Time Stretch...

The pitch shifting/time stretching/resampling editor opens. This effect can change the tempo and pitch of the audio material separately.



Algorithm: Selects the applied time stretching 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 time factor to experiment.

Preview/Original: With "Preview" you can immediately control the result of the algorithm. "Original 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.

Volume

The Volume dialog allows you to change the volume of a selection.

Attention: Excessive volume increases may cause distortion. You can also use the Normalize dialog to maximize the volume of a sound file. This will calculate the maximum level increase that does not result in clipping.

Drag the Gain fader to adjust the volume of the selection:

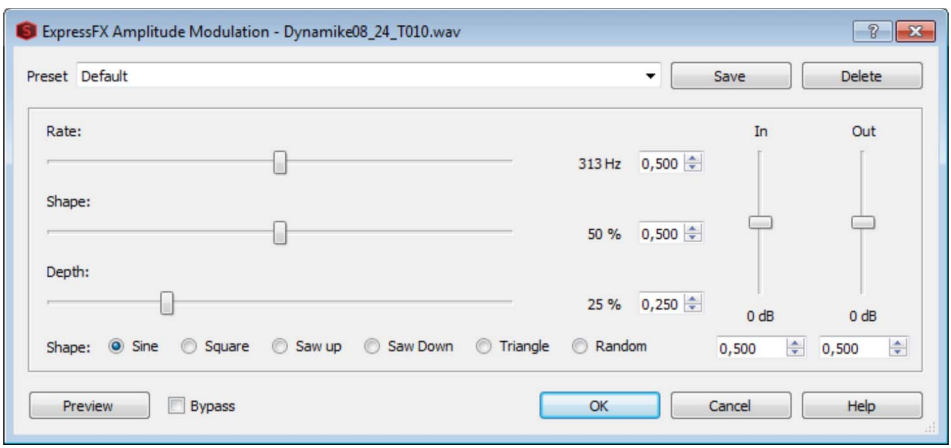
- Positive values boost the volume.
- Negative values attenuate the volume.
- A value of negative infinity (-Inf.) mutes the selection (0%).

Effects Menu

This menu contains all the effects included with the program that can be used to modify sound.

Amplitude Modulation...

Use the command "Amplitude Modulation" to modulate volume for the input signal with a waveform. Different waveforms are possible: Sine, Square, Sawtooth, Triangle and Random. The frequency of the modulation can be adjusted to create effects within the range varying from a slow tremolo to unusual sound distortions.



Rate: Sets frequency for the Modulator waveform that will be applied to the input signal. To achieve a slow tremolo, use a low frequency. Higher frequencies will create extreme sound distortion. As frequencies exceed 20 Hz, modulation is audible not as a change in amplitude, but as additional frequency side bands.

Shape: Distorts the Modulator waveform. E.g. Pulse width will be used for Square.

Depth: Controls Modulation strength, i.e. the strength of the waveform on the amplitude for the signal.

Shape Selection: Choose from a range of modulation waveforms.

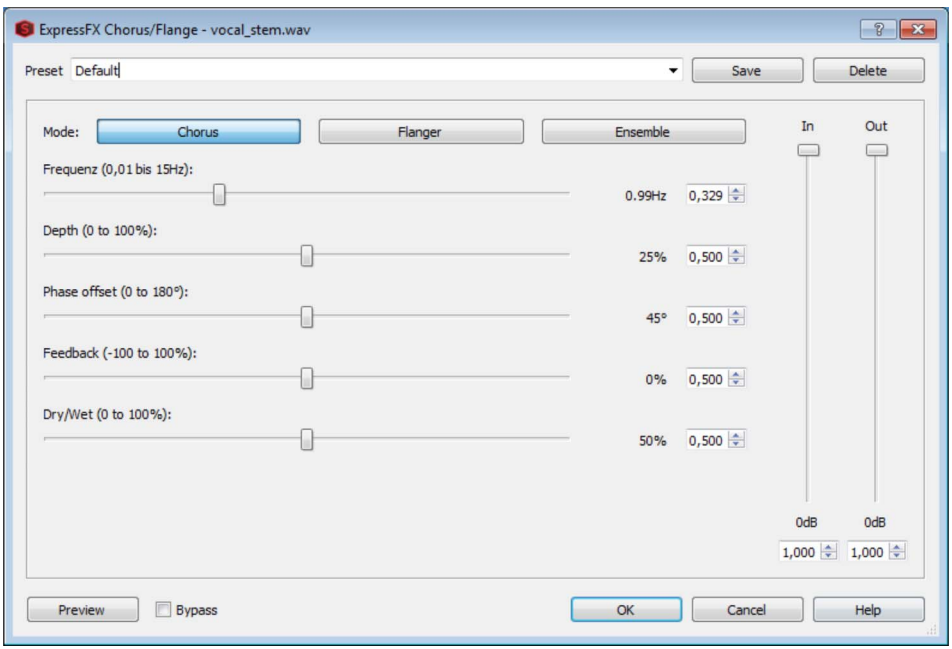
In/Out: Controls input and output level for the effect.

Note: As frequencies exceed 20 Hz, modulation is audible not as a change in amplitude, but as additional frequency side bands.

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.



Mode:

- **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.
- **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.

Frequency: 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 offset: 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.

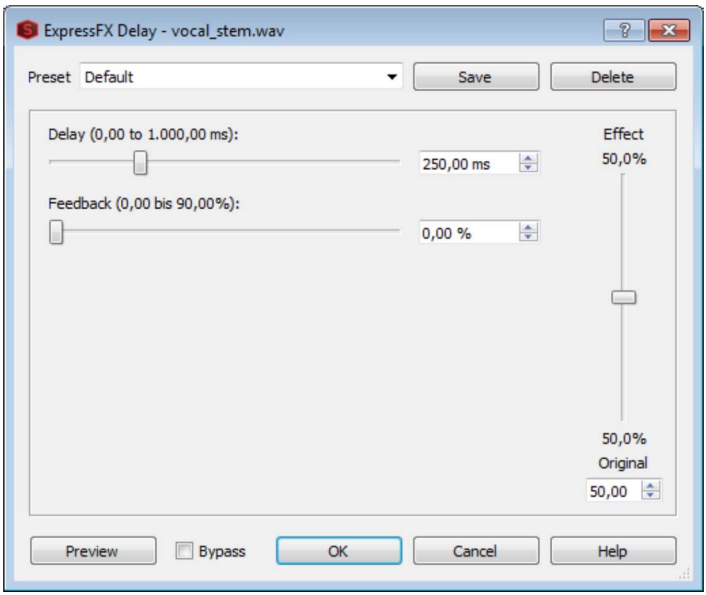
Feedback: This parameter defines the portion of the delay that is sent back to the input. Feedback causes the effects of modulation to be more drastic and cutting. Nullification of the feedback is set at the middle of the fader. Set to the right, the feedback is fed to the input equi-phasal; to the left, the feedback occurs. Both variants may sound very different depending on the signal, since they prefer different frequency ranges for dissonance.

Dry/Wet: Regulates the mix ratio of the original signal and the delayed portion.

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

Delay...

You can use the Delay effect to create subtle ambiance, slap-back echoes, and interesting effects.



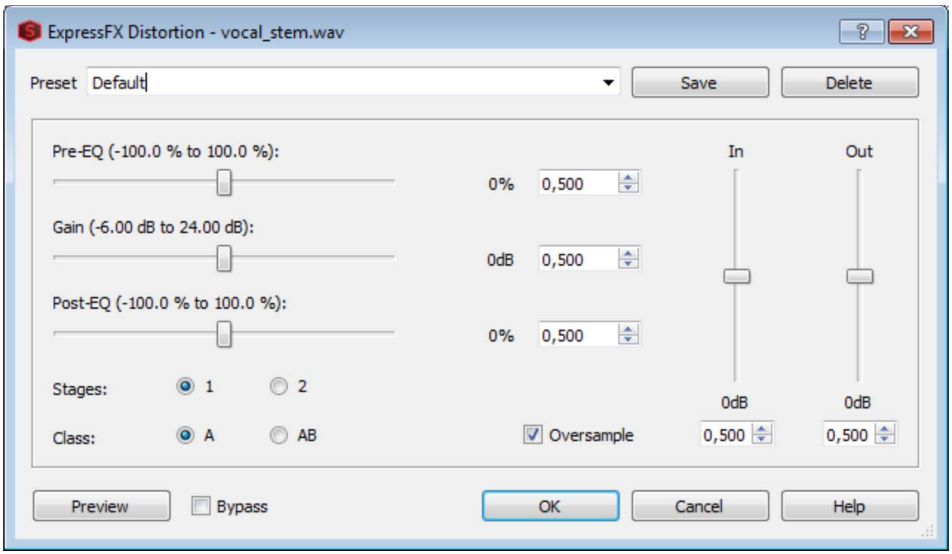
Delay: This fader sets the delay between single echoes or the original signal and the first echo in milliseconds.

Feedback: Sends a portion of the output of the right channels back into the input of the delay line, which create repeating echoes. The higher the percentage value, the longer the length of the repetitions.

Effect/Original: Set the mix ratio from the original signal to the effect signal in percentage here.

Distortion...

Distortion is the virtual simulation of a tube amp and can be used to create pleasant and "warm" sounding distortion.



Input 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 sets 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.

Output EQ: Operation and function is like Input EQ, however, this filter is placed behind the (second) tube stage.

Stages: In signal section **1** the signal only passes through the tube stage. In the **second** stage two stages are cascaded. This divides up the available gain. This increases the signal complexity and any tube artifacts.

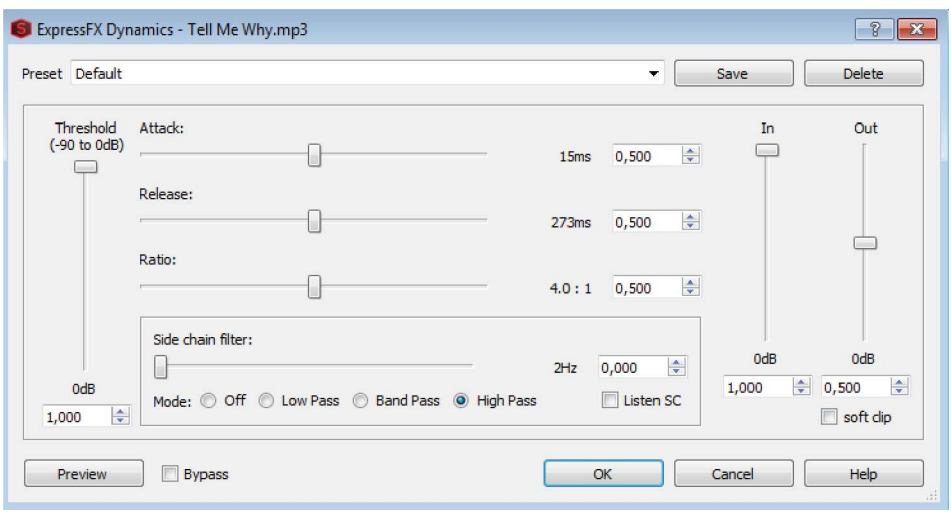
Class A/AB: Optionally, a simple gain stage with asymmetric reference line (Class A) or symmetric AB mode. In the A-mode odd and even harmonics emerge, the sound image is similar to a 'warm' tube guitar amp. In the AB section only uneven harmonics are created. The sound image is somewhat 'colder', but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

Oversample: 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.

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

Dynamics...

With Dynamics you can reduce the dynamic of your audio signal and, hence, the differences between quiet and loud signals. This allows you to turn up the overall signal higher without exceeding the available headroom and causing unwanted distortion. The result is more presence and punch in your audio signal.



Threshold: This sets the response threshold for regulation. For example, -20dB means that the input signal is only compressed once it reaches -20dB; below this level, no change or hardly any change will take place. Note that this plug-in deals with the threshold window independent of the program. It can be the case that a small regularization has already taken place at -25 dB. This so-called soft-knee characteristic ensures a soft, musical compression process.

Attack / Release: This parameter specifies how quickly the compressor responds when the threshold is reached (Attack) or how quickly the signal applies normal amplitude once the signal drops below this level (Release). The transient time settings can be defined in a wide range. Note that due to the adaptable plug-in leveling, actual times may differ. This semi-automatic process favors quick adjustment without causing dreaded artifacts (rough sound image with rushed regulation, overly low/inefficient compression with long time constants).

Ratio: This parameter regulates the compression ratio. For example, a value of 10:1 means that when the threshold level is reached, the input level is increased by 10dB, while the output is only 1dB louder. A lower compression, e. g. 2:1, is recommended for subtle compression, while 50:1 produces a very hard limiting that cannot be made inaudible by even the most transparent compressor.

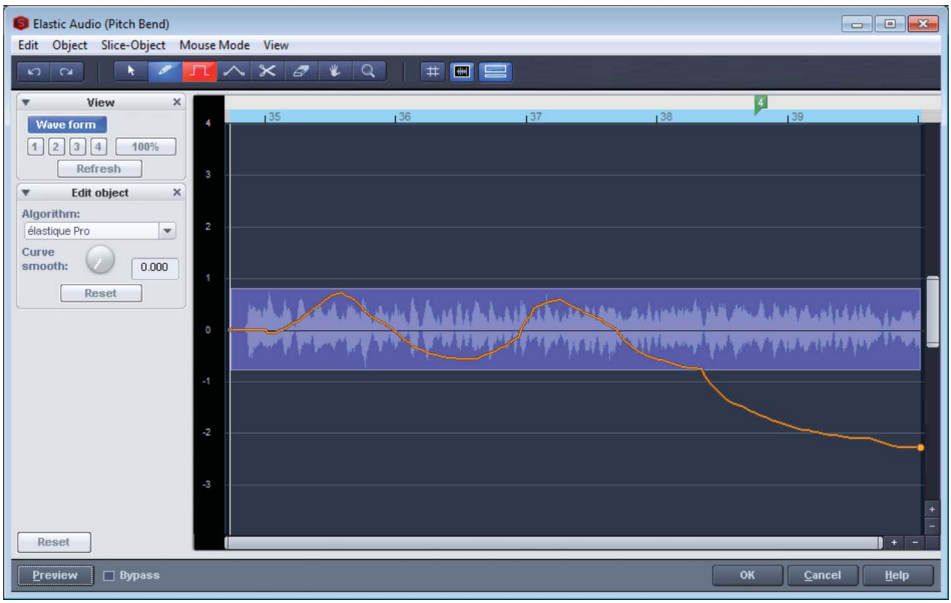
Side chain filter: This fader determines the application/middle frequencies which are used to filter the control signal for level detection. The filter works selectively as a high-pass, band-pass or low-pass filter. In many cases, weighting of the detection is good for optimizing the leveling process. Thus, for example, "high-pass" filtering enables complex sources (drums or sum signals) to achieve consistent regulation without typical pumping artifacts, since the process is mainly activated by mids and highs. The properties of the detector circuit can be drastically changed by means of the filter type switch. Listen SC is used to listen to the detector signal.

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

soft clip: Soft clip circuit is switched on directly at the output in order to catch any clipping.

Elastic Audio (Pitch Bend)...

Elastic Audio is a specialized editor which lets you change the pitch of audio material. The pitch envelope can be drawn, allowing you to edit the temporal progression of the pitch changes in detail.



Editing can be done with a free-hand curve, quantized "step" curves, or using the Curve Bend tool. The “curve smooth” tool enables the equalization of the automation curve. This smoothes pitch curve value changes that are too steep during playback.

Graphical display overview

Y-axis: Displays the pitch shift in semitone steps (within the range of -24 to +24 semitones).

Orange curve: Editable pitch and automation curve as relative detuning of the original note.

Dark blue line: 0-line as a reference for the edited pitch curve.

Elastic Audio Control Elements

View

Waveform: This turns the waveform display on and off.

Zoom buttons: Here you can save the zoom depth and position of the current window view. Buttons 1-4 or numbers 4, 5, and 6 on the number pad can be used to save presets. These correspond to the zoom buttons 1, 2 and 3. There is no assignment to the number pad for the fourth zoom button.

Save zoom levels:

Shortcut: Ctrl + Numeric keypad 4

Ctrl + Numeric keypad 5

Ctrl + Numeric keypad 6

The numbers 4, 5, and 6 on the number pad can be used for the first three zoom buttons.

Get zoom level:

Shortcut: Numeric keypad 4

Numeric keypad 5

Numeric keypad 6

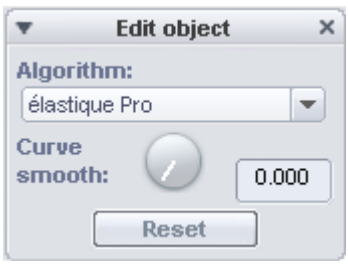
Clicking on the zoom button while holding down the Shift key overwrites the last zoom level.

Clicking on the button while holding down the Ctrl key opens only the horizontal position of the respective zoom level.

Zoom 100%: Horizontally zooms the entire audio material of the currently loaded track.

Refresh: This updates the graphical display of the material. Resulting changes to length are adapted.

Edit object



Algorithm: You can choose between the modes élastique Pro, élastique Efficient, Resample and Monophic Voice.

> Detailed information about these algorithms is provided under "Process" > "Time Stretch...".

Curve smoothing: This parameter brings about the smoothing of the given pitch curve using a time constant (in ms). When smoothing to a large extent, even jump-like pitch shifts are changed to a "glissando".

Reset: Here you can undo all changes made to automation curves or the tuning of selected objects.

Toolbar

Several tools are available for processing the pitch envelope. It is possible to select different tools for the left and right mouse buttons. The tool assigned to the left mouse button turns blue; the function assigned to the right button is marked in red. To make an assignment, click with the corresponding mouse button on the desired console button. Only in the case of the zoom tool are both mouse keys automatically assigned functions.

Selection Tool (arrow): Use this tool to move slice objects (see "Cut") vertically. This modifies the pitch of slice objects as a whole. Slice objects and curve handles can also be selected, and a slice object lasso can be dragged out by holding down the left mouse button and drawing a rectangle around the objects you want to select. Multiple selection is possible with the Ctrl or Shift key.

Freehand drawing function: You can use the Pencil Tool to draw the pitch curve freely. If you press the Shift key, a straight line is drawn from the starting position to the current position of the mouse. If the Ctrl key is pressed, the slice objects are combined when you draw.

Pencil Tool for Quantized Drawing: Quantized drawing means that the drawn line is vertically gridded at half steps.

Drawing a straight line using the Shift key and compiling slice objects with the Ctrl key is also possible in quantized draw mode.

Rubber Handle Tool: You can use this tool to bend the pitch envelope between two neighboring handles. Here the range between the curve handles is moved, the handles themselves remain unchanged. The curve is bent inwards at the handle point. In combination with moving the curve handles at the Slice Object's boundaries you can bend the tone envelope and thereby maintain the microtonal structures (vibrato).

Double-clicking an edited position like this opens a dialog box in which you can see the pitch changes in cents and adjust them.

Cut: Use this tool to manually split the audio material into slice objects. To recompile slice objects, use the pencil tool to drag along the slice while holding down the "Ctrl" key.

Eraser: Use the eraser to reset the output value of the orange curve. The pitch envelope will correspond to its original curve again.

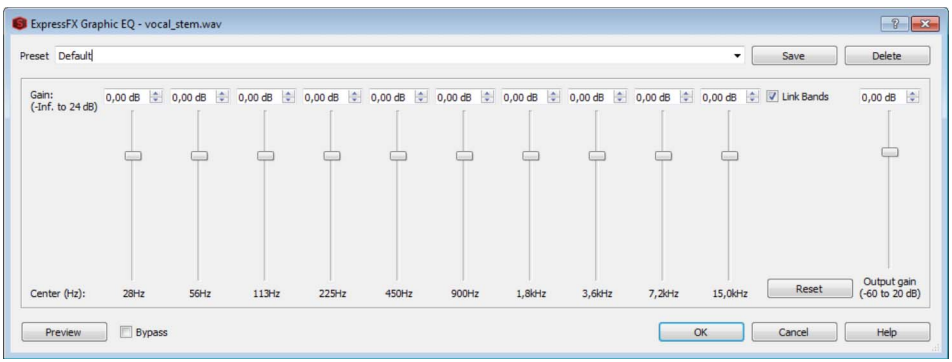
Navigation Tool: Use the Navigation Tool to move the visible selection vertically and horizontally.

Zoom Tool: Click with the left mouse button to zoom into the display; zoom out with the right mouse button. By clicking and dragging you can stretch a range which will then be displayed as a zoom range.

EQ

Graphic...

The graphic equalizer subdivides the frequency spectrum into ten areas (bands) and equips them with separate volume controls. This makes it possible to create many impressive effects, from a simple boosting of the bass to complete distortion. Note: If low frequencies are boosted too much, the overall sound level is heavily increased, which may cause distortion. In this event, adjust the overall volume downward by using the "Output Gain" controller.



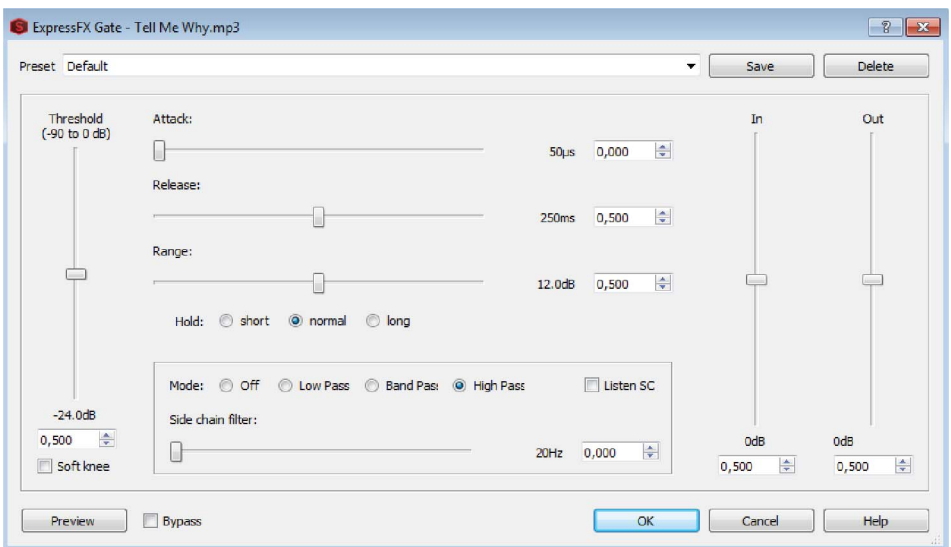
Band coupling: Using this feature randomly combines the frequency ranges with each other in order to avoid artificial-sounding overemphasis of an individual frequency range.

Simple...

The simple version of the EQ divides the frequency spectrum into only 3 bands. Using the "EQ/Original" controller, you can set the mix to the editing signal or the unedited signal.

Gate...

This effect 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 Gate is operated in much the same way as the Dynamics effect.

Threshold: This fader defines the threshold beneath which the gate should be applied.

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 recommended for work with acoustic instruments such as drums where the signal levels can fluctuate a lot.

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.

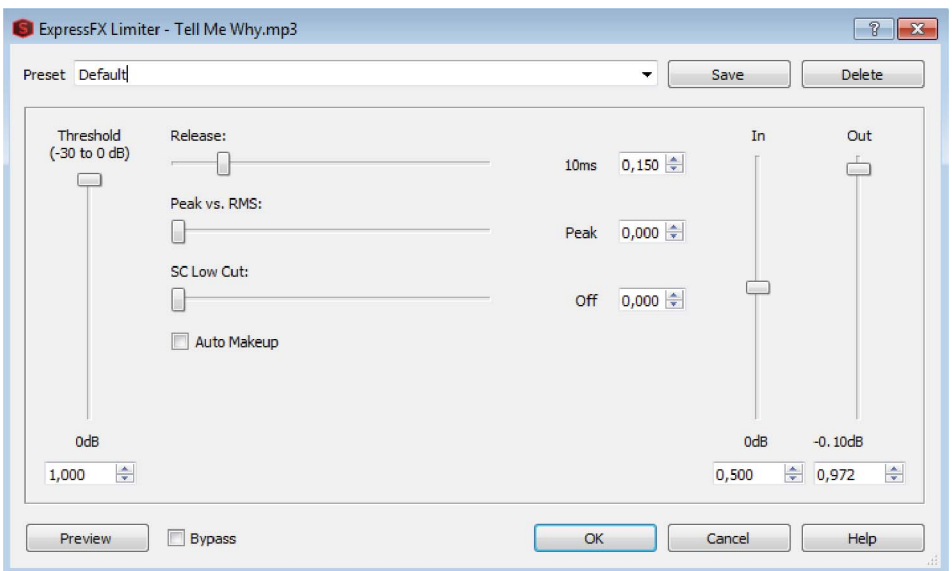
Range: Here you can set the gating strength. When it is set fully to the right, the signal beneath the threshold is cut out completely. Subtly attenuating the signal can remove background noises or breathing from vocal tracks. Where necessary, the process can be attenuated even further by switching on the soft knee mode.

Side chain filter: Operation is the same as the dynamics effects (view page 124).

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

Limiter...

This effect is a simple yet effective tool for increasing the loudness of your audio signal. This creates a compressed, but loud, signal without allowing the defined output volume to be exceeded. The limiter operates with a soft curve and an adaptive regulator that ensures that the sound remains absolutely musical.



Threshold: This sets the threshold; the effect will be applied above this level.

Release: This sets the time frame between the point when the signal drops below the threshold and the complete reduction of the effect.

Clip Gain: This sets the amplification factor.

Peak vs. RMS: Here you can choose whether to display the eFX_Limiter peak levels (slider completely to the left) or the RMS levels (slider completely to the right). You can also set the display between peak level and RMS values to any ratio you like.

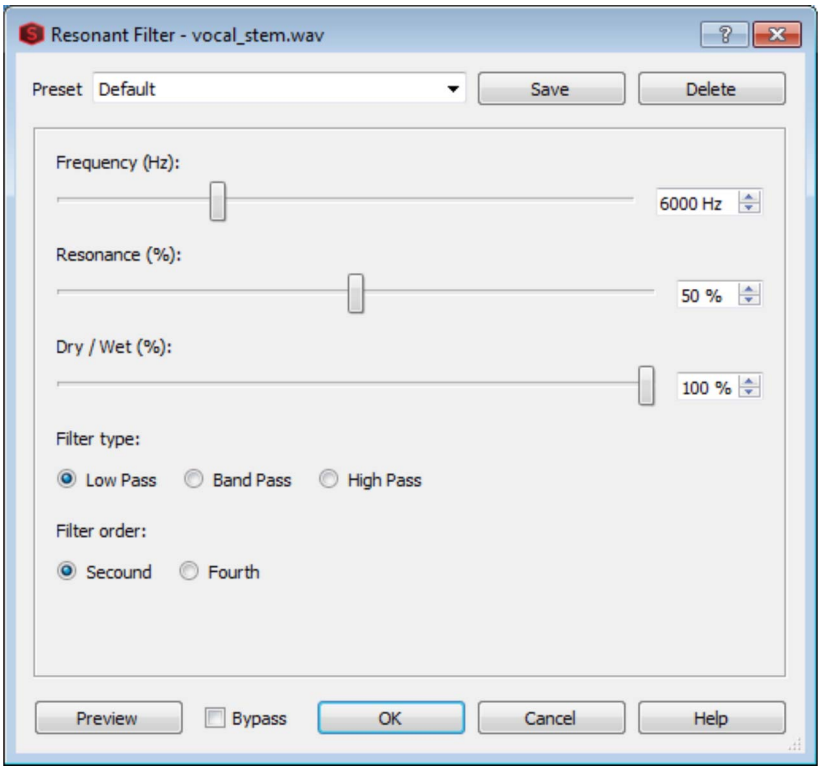
SC Low-Cut: This sets the filter frequency of a high-pass filter for the internal sidechain signal. Signal components below this frequency will be filtered out.

Auto Makeup Gain: Here you can set the automatic tracking of the output amplification during level reduction in order to maintain compression at the same maximum level. 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 level reduction and amplification manually, you can deactivate this feature.

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

Resonant Filter...

The Resonant Filter plug-in makes it possible to restrict the range of a sound using low-pass, band-pass, or high-pass filtering, and then boost and add oscillation to the resonant frequency.



Drag the **Frequency** slider to set the frequency at which the filter oscillates (rings). Depending on how the **Filter Type** radio buttons are set, frequencies below this frequency will be played (Low Pass), the frequency band will be played (Band Pass), or frequencies above this frequency will be played (High Pass).

Drag the **Resonance** slider to set the level of the effect.

Drag the **Dry/Wet** slider to set the balance of the processed and unprocessed signals mixed into the output.

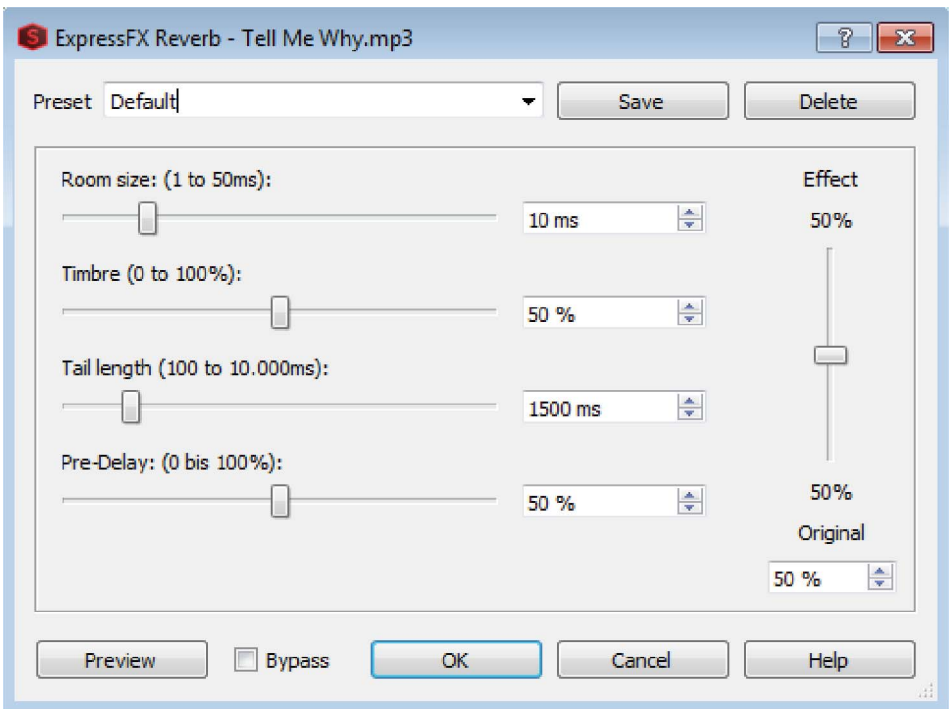
Filter order: "Second" applies a second-order filter (12 dB per octave) for low-pass and high-pass filters. "Fourth" applies two second-order filters in series

(24 dB per octave). This setting will produce higher peaks and a more pronounced effect.

Attention: Higher resonance values can quickly lead to clipping. Lower the volume, if needed, before using the effect!

Reverb...

The reverb effect offers realistic reverb algorithms to add more room depth to your recording.



You can control the sound of the reverb effect using the following parameters:

Room Size: Defines the size of the room (or the system for the plate and spring). The larger a room, 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 reverb sustain is too long.

Tone Quality: You can manipulate the sound characteristics of the effect to an 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.

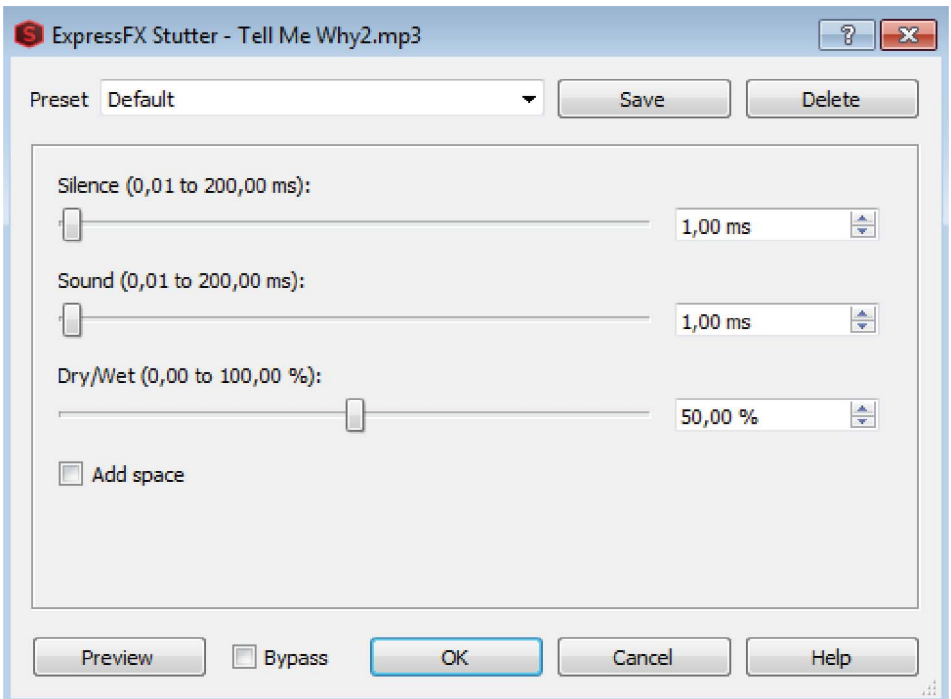
Pre-Delay: The reverb portion ("Mix") and the early reflections play a big role in the spatial perception of the sound. Here the time required for the perception of the early reflection is know as "pre-delay". For sound sources that are close, the reverb portion is usually low, and the early reflections reach the ear noticeably later than the direct sound. By 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 can be used to influence the perceived "distance" between the sound source and the listener.

Effect/Original: This controller sets the mix ratio between the original and the edited signal. For rooms, you can move a signal further into the room by increasing the effect share.

Note: The presets contain basic settings for different room algorithms which can be varied with the parameters above. The presets need to be used in order to achieve all the reverb effect variations.

Stutter...

The Stutter effect works by periodically muting the sound signal. Using longer Silence and Sound times produces a noticeable stuttering effect. As the period decreases, the stutter becomes unnoticeable, and instead adds overtones to the sound. Very short periods produce a pitching effect.



Silence: Set the duration of the muting that is applied periodically to the signal.

Sound: Set the duration of the sound output between periods of muting.

Dry/Wet: Set the balance of the processed and unprocessed signals mixed into the output. Select 100% for an audible stutter effect. For more subtle sound effects with very short periods, use this feature to control the intensity of the effect.

If you're working with a stereo file, select the Channel Offset check box to cause the stutter to alternate between the left and right channels, which serves to widen the sound.

Plug-Ins Menu

All installed plug-ins are sorted in this menu, along with all the plug-ins found by Sound Forge Audio Studio 12.

VST plug-ins can be used for effects calculation in Sound Forge Audio Studio 12. This allows you to use almost any effect algorithms and VST instruments from third-party developers in addition to the effects included in Sound Forge Audio Studio 12.

Installing Plug-Ins

Install every VST plug-in according to the instructions provided by the manufacturer. VST instruments and VST effects are not added to the Windows Registry; rather, they must be installed in a specific folder.

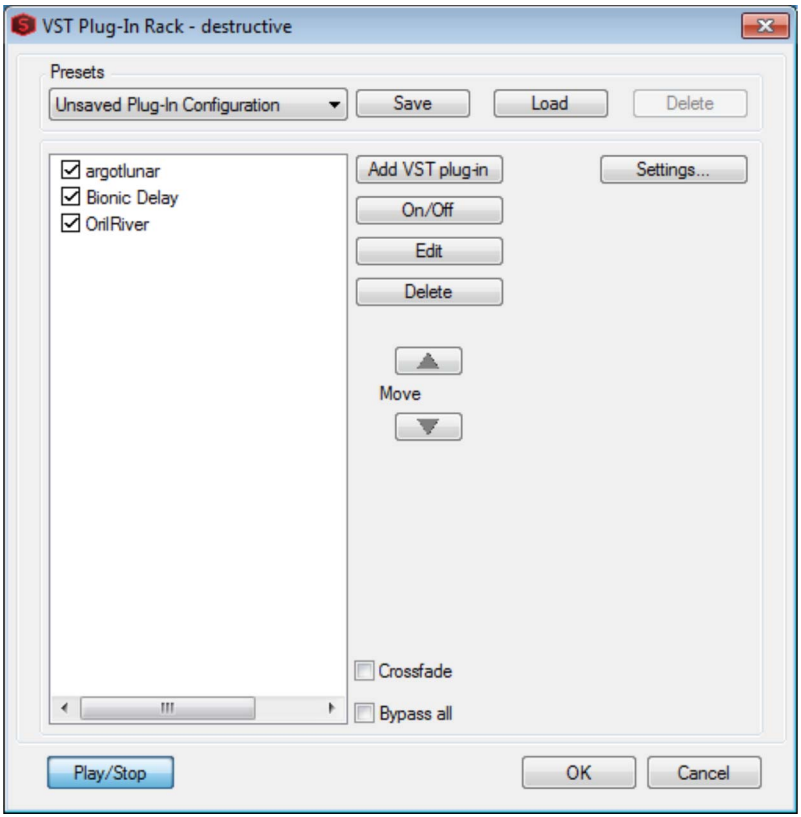
Following installation, have Sound Forge Audio Studio 12 scan for new plug-ins (see below).

Sound Forge Audio Studio 12 will search for installed VST instruments in the "VST plug-ins" program subfolder. You can either install your VST instruments in this folder or use any other path to an already existing folder. If you already have VST plug-ins installed on your system, it is best to use the existing folder. You can enter the path in the subdialog "Effects" > "VST" (view page 160) in the program settings ("Options" > "Settings...").

Following the scan, all the plug-ins that were found that can be used in Sound Forge Audio Studio are displayed in the menu. VST3 plug-ins are sometimes assigned a category by the manufacturer. This category is detected by the program and used to categorize the plug-ins in submenus at the very top of the plug-ins menu. All other plug-ins are listed after in the VST-FX menu. If there are VST plug-in subfolders in the main VST folder (e. g. "Equalizer", "Filters", "Modulation", etc.), these will be displayed there as submenus.

VST Plug-In Rack

The plug-in rack lets you apply multiple plug-in effects at once and save these combined plug-ins and their respective settings as a preset.



Add VST plug-in: Opens the plug-in selection menu.

On/Off: The respective activated effect or the activated plug-in can be switched on or off by clicking on this checkbox in front of the effect/plug-in.

Edit: Opens the dialog of the selected effect/plug-in. Another way to open an effect or plug-in dialog is to right-click on the name of the effect or plug-in.

Delete: Removes the plug-in.

Move: Change the position of an effect or plug-in with the up/down arrows.

Crossfade: Activates the Auto Crossfade mode. Short crossfades are applied between the effects editing and the rest of the audio material.

Bypass all: Temporarily deactivates all effects.

Tip: The rack can be closed intermittently. The plug-ins are kept in place for when you want to use the rack again. The rack is also good for temporarily saving the settings of a specific plug-in if you want to use it more than once in different areas of the project.

Scan VST Folder for New Plug-Ins

When you start Sound Forge Audio Studio 12 for the first time after installing new plug-ins, you can run a plug-in scan here. The program scans for new plug-ins in the directory specified under "Options" > "Settings..." > "Effects" > "VST" (view page 160) and adds them to the menu.

In the settings you can also use the option "Scan user and system VST folders automatically for new plugins" to run an automatic scan for plug-ins each time the program is started. This can slow down the program start considerably, especially if you have a lot of plug-ins installed.

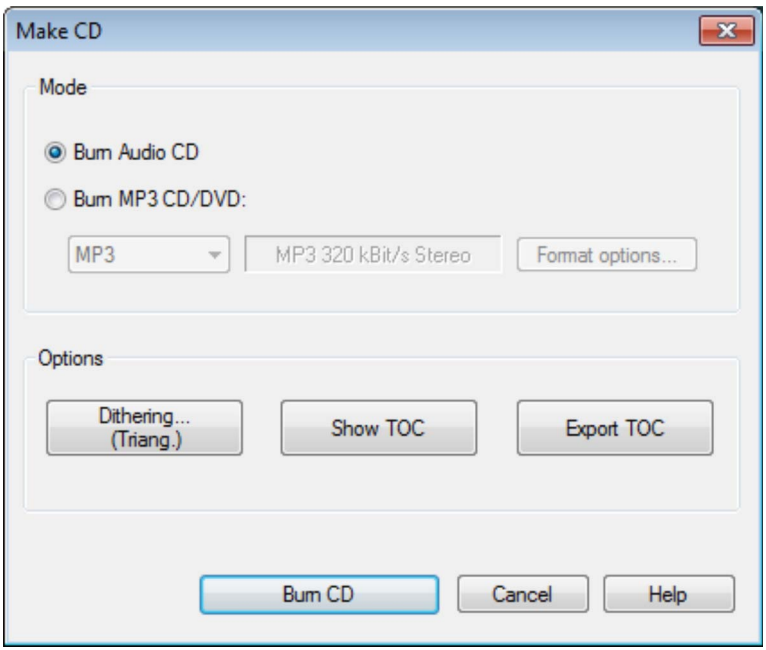
VST Setup...

Opens the program settings under Effects > VST (view page 160).

Extras Menu

Burn audio CD

This menu item lets you burn the current audio file to CD. The right CD track marker must be set first. More information can be found in the section Region List (view page 64).



"On the fly": Use this mode if you want to write a CD directly from the project. Here all necessary calculations are made in real-time during the burning process.

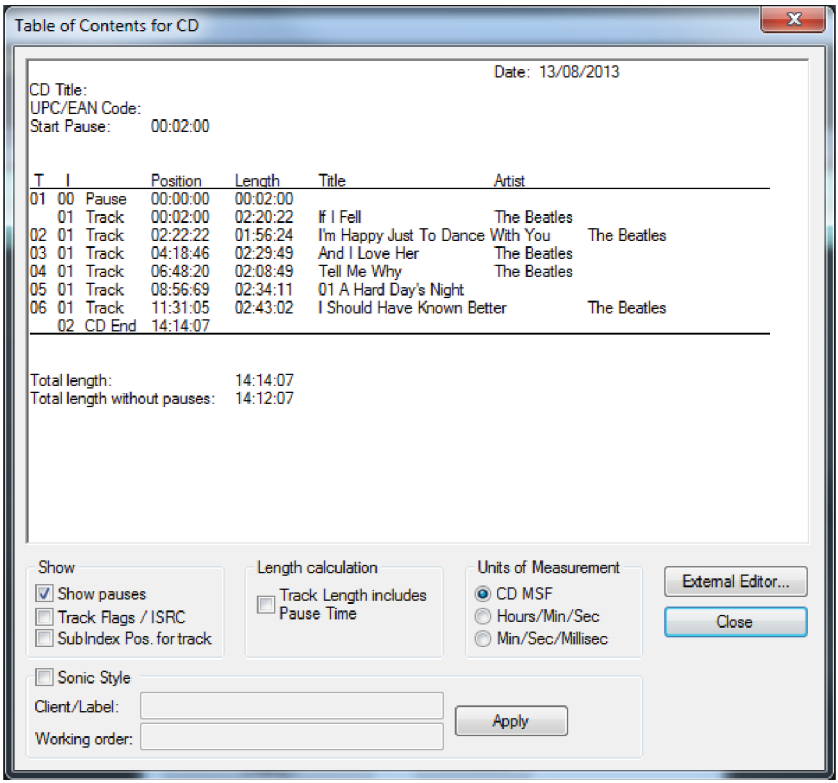
Generate a complete new file for the whole CD: Use this function to create the CD in case the computer is not fast enough to burn on-the-fly. This mode calculates your project including all effects and creates a new file. Make sure there is enough space on your hard disk for this file (ca. 700 MB for a complete CD).

Burn MP3 CD/DVD: All tracks are exported as individual MP3 files according to the selected format options and then loaded into the MAGIX Speed burnR tool. Here you can load more files or begin burning a data CD.

Dithering: Detailed information about dithering options can be found in "Options > Preferences > "Dithering" (view page 160).

Show TOC...

This button leads to the TOC (table of contents) dialog.



Under "Display" you can select what extra information should be displayed.

- Pause
- Track Flags / ISRC
- Sub-index position in the Track

Under "Length calculation" you can select the "CD track length includes pauses" checkbox. This means that the time span between CD pause index and CD track index is factored into the length of the track.

The following measurement units are available: "CD MSF", "Hours/Min/Sec" and "Min/Sec/Millisec".

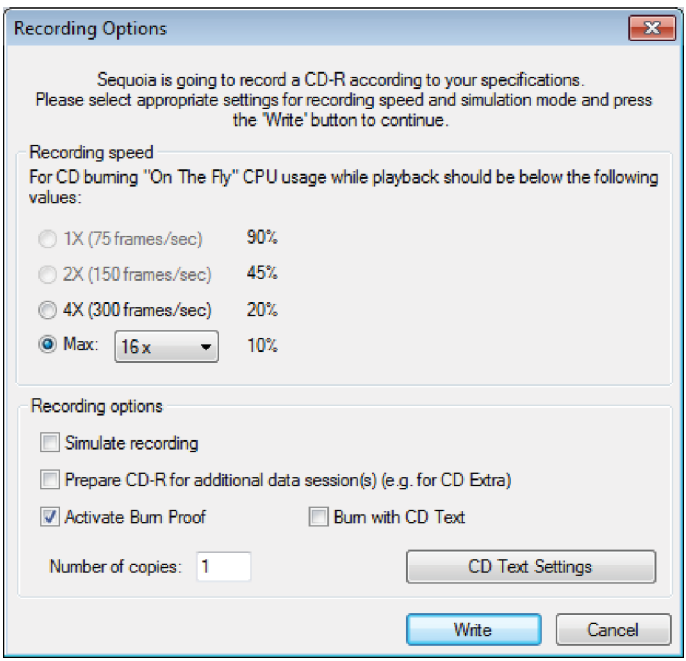
Using the "External editor" button you can call up a text editor of your choice, in which the TOC information should be opened.

"Sonic Style" displays the TOC information in another layout. You can enter a working advice for the pressing plant in "Client/Label" field as well as the "Working order" field. With the "Adopt" button, this information will be included in the CD contents.

TOC Export

Clicking on the „TOC Export“ button saves the information from the CD content folder in a *.toc file (view page 142).

Burn CD



CD-R Recording Options

Simulate recording: Use this option to test the various write speeds.

Prepare CD-R for additional data session(s): The audio CD is not finalized and data may be added to it by an external burn program.

Note: The non-final audio CD should not be used as a multi-session disc. Use it as a blank disc with less capacity.

Activate Burn Proof: This option activates a quantization check after the burning process.

Burn with CD Text: CD text information is saved in the CD audio format for export via the "Load audio CD track(s)" function.

CD Text Settings: This opens the CD-Text/MP3 ID editor.

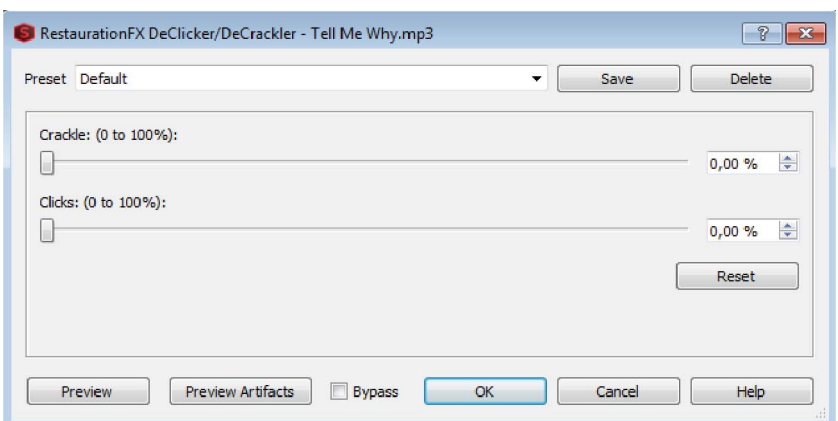
Note: Windows Media Player (up to version 10) cannot analyze CD-Text information.

Audio restoration

This submenu contains special effects for audio restoration.

DeClicker/DeCrackler

This functions removes crackling and clicking noises which are typical on scratched records. The DeCrackler has been specifically developed to remove crackling noises from old records.

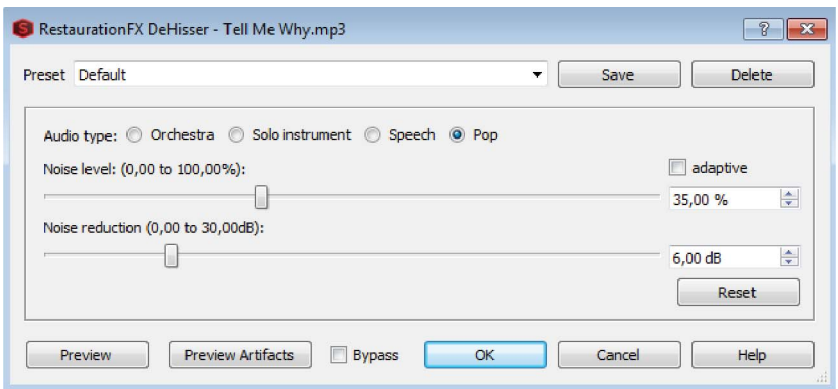


For "crackling" and "clicking" noise types, there is a fader for controlling the strength of the effect.

You can also use the button "Preview Artifacts" for help with acoustic editing. When this is activated, you can hear the crackling or clicking noise that you have removed, not the edited audio. You can then decide whether the noise that has been filtered out contains too much of the original material, and go back and decrease the effect strength.

DeHisser

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



Audio type: Here you can set the type of edited audio material, the algorithm will be customized accordingly.

Noise Level: Here you should set the threshold of the DeHisser as precisely as possible. Setting the value too low will result in only partial removal of the hissing. Incomplete removal of hissing produces artefacts 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 hissing level.

Adaptive: The value for the noise level parameter is set automatically by determining the level of the hissing present in the signal. The setting on the noise level has a relative effect, i.e. the used value will be derived from the automatic setting and the setting of the noise level fader.

An advantage of this 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 (adaptive off). The value for noise level must be set exactly.

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

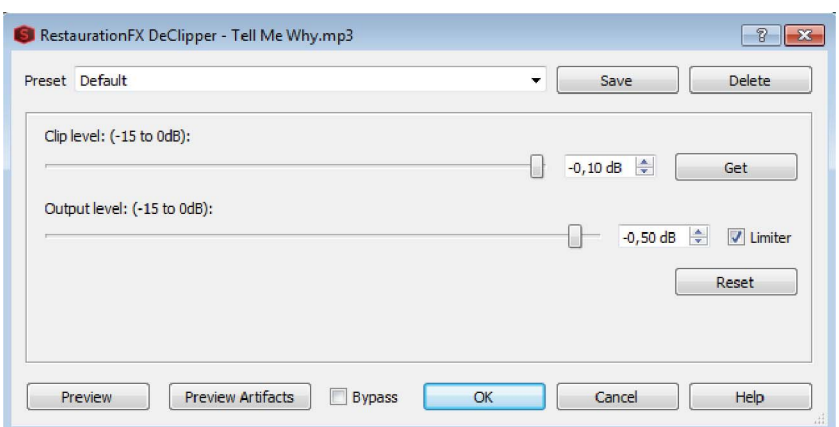
DeClipper

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.

Sound Forge Audio Studio 12 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.



Clip Level: Enter the level at which the algorithm registers the samples as distorted and corrects them if necessary here. This is important, as different sound cards show different clipping behavior.

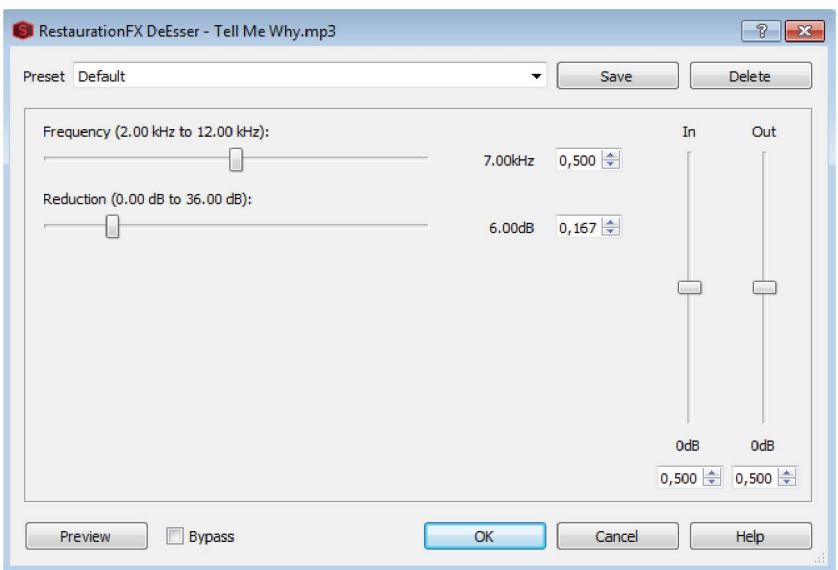
Get: Here you can determine the clip level automatically.

Output Level: The interpolated signal peaks create a change in the master volume that must be balanced with the output level fader to avoid further distortion. Watch the peak meter (if a peak meter is not open, you can open a new visualization window with Alt + F5).

To be safe you can activate the "**Limiter**" which reliably prevents clipping.

DeEsser

The DeEsser 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.



Frequency: 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.

Reduction: Regulates the filter attenuation in the signal route.

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

DeNoiser

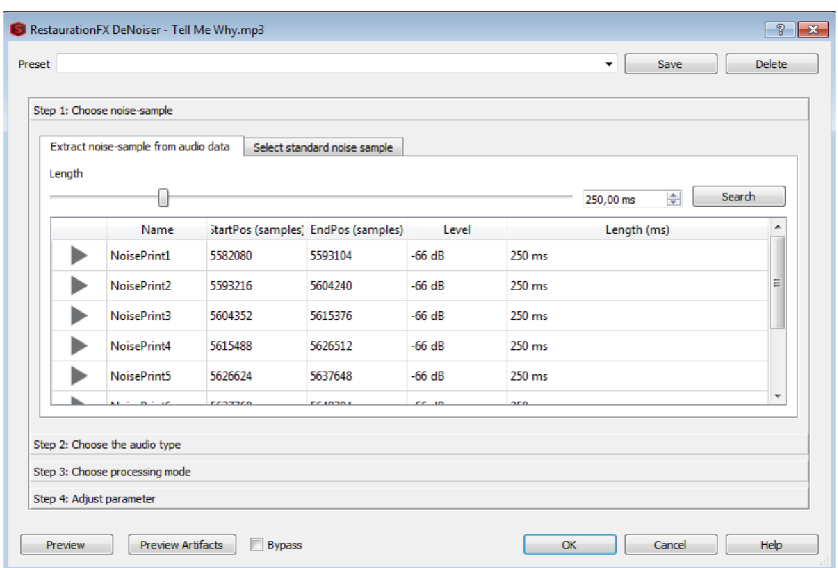
The DeNoiser removes persistent background noise like computer humming, hissing, noises from sound cards, disturbance from ground wires, interference from audio equipment with high-impedance outputs (e.g. turntables). Low frequency impact sound or rumbling can be removed effectively from LPs with the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

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

The DeNoiser can be closed at any time by clicking OK and the noise removal will be applied. For optimal results, however, it's best to follow the settings step by step. You can preview your editing results at any time by pressing the Preview button. You can also use the button "Preview Artifacts" for help with acoustic editing. When this is activated, you can hear the noise that you have removed, not the edited audio. You can then decide whether the noise that has been filtered out contains too much of the original material.

Step 1 - Choose noise sample

When the DeNoiser is opened, it will search for a sample of the noise in the audio material under the playback marker.

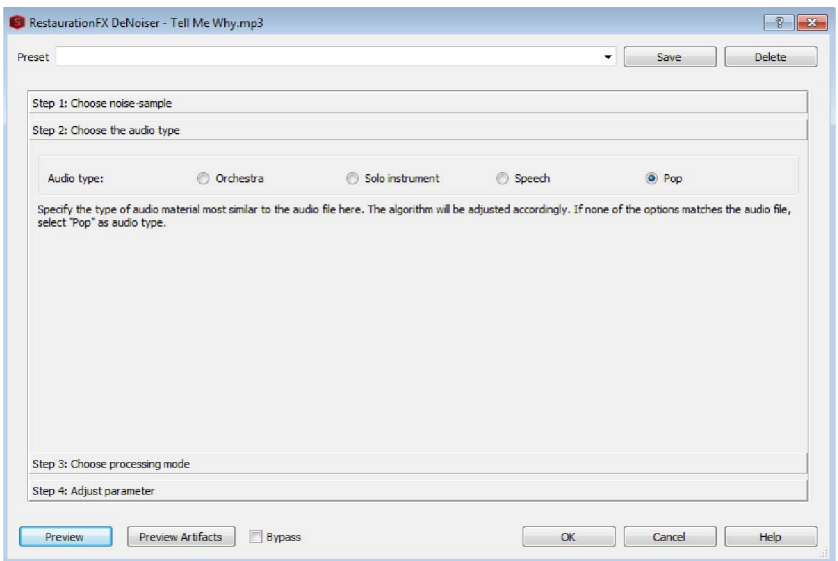


Use the Play button to preview the noise samples, select one of these and go to Step 2.

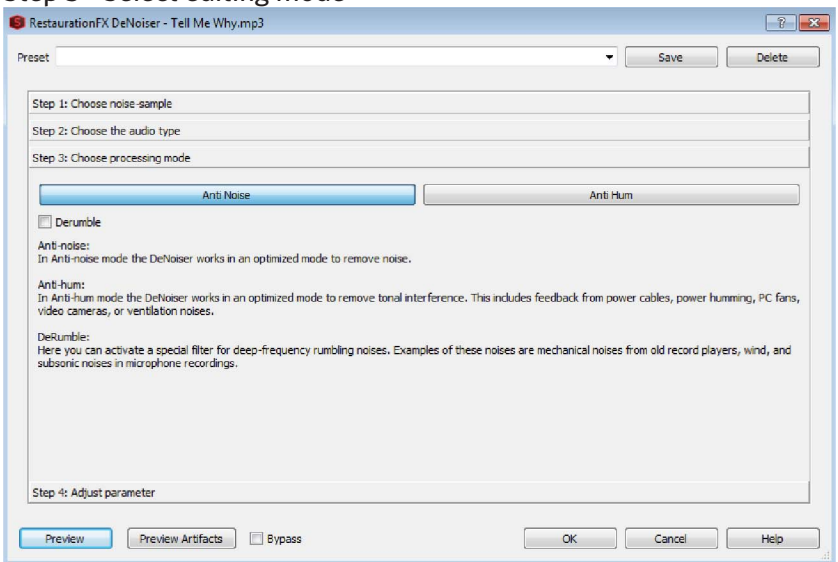
If you haven't found the right sample, you can choose one of the noise samples supplied in the "Standard Noise Samples" tab. You can also adjust the Noise sample manually by selecting a position and then choosing "Extras" > "Audio Restoration" > "Take Noise Sample".

Step 2 - Select audio material type

Set the type of edited audio material here, and the algorithm will be adjusted accordingly.



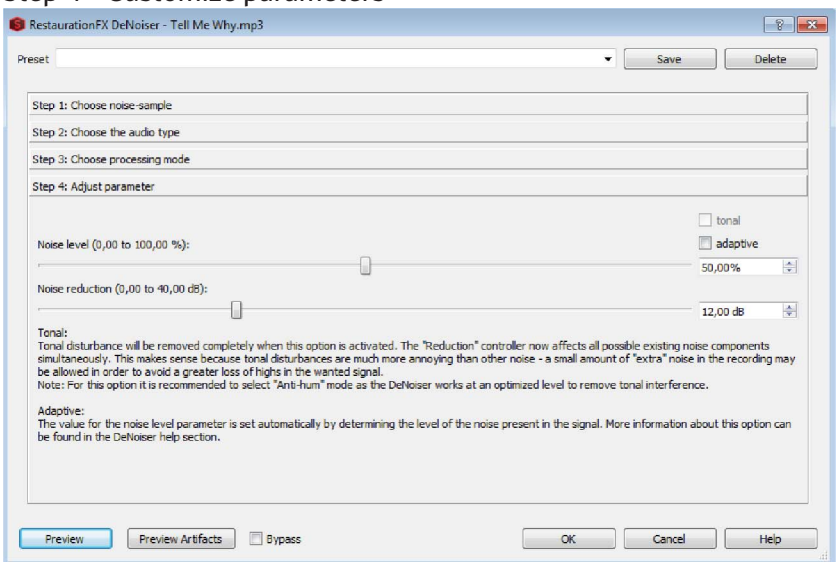
Step 3 - Select editing mode



In **Anti-noise** mode the DeNoiser works in an optimized mode to remove noise. In **Anti-hum** mode the DeNoiser works in an optimized mode to remove tonal interference. This includes feedback from power cables, mains hum, PC fans, video cameras, or ventilation noises.

DeRumble: Here you can activate a special filter for deep-frequency rumbling noises. Examples of these noises are mechanical noises from old record players, wind, and subsonic noises in microphone recordings.

Step 4 - Customize parameters



Tonal: Tonal disturbance will be removed completely when this option is activated. The "Reduction" controller now affects all possible existing noise components simultaneously. 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.

> Note: For this option it is recommended to select "Anti-hum" mode as the DeNoiser works at an optimized level to remove tonal interference.

Adaptive: The value for the noise level parameter is set automatically by determining the level of the hissing present in the signal. More information about this option can be found in the DeHisser section.

Noise level: The threshold of the noise reduction function should be set as precisely as possible. Values that are too low will result in insufficient noise dampening which causes artefacts like 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.

Noise reduction: This sets the balance between the original signal and the signal with the applied noise reduction. 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 buzzing, it's best to apply complete removal.

Options menu

Status Format

Here you can enter a measurement unit for how the time is displayed on the timeline, in the position fields under the waveform and in the time display window.

Loop Playback

Turns the loop playback on and off.

Shortcut key: Q

Scroll Playback

When activated, the visible section moves so the cursor remains visible. Learn more in the Playback settings (view page 154).

Shortcut: F6

Draw Gridlines

Turns gridlines on/off.

Shortcut key: F7

Gridlines

Here you can choose between various patterns of gridlines.

Aligning

Aligning makes the exact positioning of the play cursor and precise range selection easier. When "Aligning" is activated, the selection or play cursor "snaps" to a time position.

The "Snap to Grid" and "Markers/Slice edges" commands can be activated separately.

To grid

The grid aligns to the measurement unit selected. You can set this unit under "Options" > "Status format" or by right-clicking on the grid and marker bar. The time format settings correspond to the following grids:

Status format	Grid
Samples	1000 Samples
Hour/Minute/Second	1 s
Milliseconds	1 s
SMPTE (custom frame rate)	1 Frame
SMPTE/milliseconds	1 ms
Bars	1/4 bar (can be configured under Edit tempo)
CD MSF	1 CD frame
Feets/Frames	1 Frame

Snap to markers/slice borders

Snaps to all markers (marker, regions, CD indices) and slice edges (view page 52).

Shortcut:	Align In/Out	F8
	Snap to Grid	Ctrl + F8
	Snap to Markers/ Slice Borders	Shift + F8

Settings...

All Sound Forge Audio Studio 12 settings are contained in this dialog.

Audio settings

Driver system: A so-called "driver system" is used for the necessary communication between Sound Forge Audio Studio 12 and your sound card. In order to take full advantage of the program, we recommend that you use ASIO. "MAGIX Low Latency 2016" is included, a universal ASIO driver which can be applied by many sound cards which do not feature ASIO drivers.

Driver: Choose the sound card driver that you would like to work with. The drivers for all ASIO devices installed on the system will be listed here. Clicking the "Control Panel" button opens the settings dialog for the sound card driver. In the display field beside "ASIO buffer", you'll see the buffer size and bit rate set for the driver.

16/24-bit device/driver communication: If your sound card is able to play 24-bit audio, your project will be played at this higher resolution (internal 32-bit float calculation). This is applicable only to wave drivers.

Metronome

This is where you activate the metronome. The metronome can be set as an audio metronome or as a MIDI metronome.

Active During Playback: This option activates the metronome click during playback.

Active During Recording: This activates the metronome click during recording.

Precount For Recording (Measures): Here you can specify the number of beats that the metronome will count in before the recording starts. If you activate the "Precount Clicks Only" option, the metronome will stop before recording.

Audio Metronome: Under Beat you can specify the audio file you want to play for the first beat hit in the beat; under Beat Hit you can specify the audio file for the other beat hits. You can also adjust the volume of the metronome and specify which audio output to use.

MIDI Metronome: This metronome is played by a MIDI synthesizer; the Microsoft GS Wavetable Synth is set as the default. The MIDI channel and the note values for the beat and beat hit can be specified.

Playback

Spacebar and F12 Play/Pause instead of Play/Stop: Check the box if you want the playback marker to remain at the same position after stopping playback. If this option is not active, the playback marker will jump to the original position or back to the start of the range when playback is stopped.

The option "**Spacebar (Play/Stop) also functions when in background**" allows the spacebar to be used for "Play" and "Stop" in Sound Forge Audio Studio 12 while using a different software application.

An additional function can be used in the case of CPU overload, i.e. "**Escape key stops playback and recording (for overload situations)**".

Scroll Playback: The graphical display constantly moves even before the playback marker leaves the visible section. This provides a constant overview. You can switch to scroll mode by checking the box next to "**Active**" (you can also do this outside of the dialog by pressing F6).

"**Active for zoom < 1s**" causes autoscrolling to begin at very high zoom levels of less than a second.

You can choose between **Page** and **Soft** modes. With page scrolling, the section changes before the playback marker moves outside the section, while with soft scrolling, the playback marker always remains in the middle of your chosen section and the arrangement moves along beneath it.

Note: The autoscroll process can overload the processor in some cases which may lead to dropouts during playback. Deactivate Autoscroll Mode if this happens.

Scrubbing

Here can adjust the scrubbing mode settings for the preview tool. See "Edit" > "Tool" (view page 84).

Shuttle: This mode allows you to use the absolute position of the mouse in the window to control the speed. The mouse positions can be understood as follows:

Mouse pointer on the left border	=	double speed backwards
Mouse pointer in the middle	=	no movement
Mouse pointer on the right border	=	double speed forwards

Absolute: The distance between the playback marker and mouse position is used for tempo control. This means that the play cursor follows the movements of the mouse at max speed and automatically slows down when it gets close to the mouse.

One Speed: Playback follows the movements of the mouse at original speed. Holding the Shift key uses half original speed. The "Ctrl" key applies the speed set in the "Speed" field.

Two Speed: Here you can use two speeds for scrubbing. If the mouse pointer is moved a bit away from the cursor, playback happens at original speed. Playback slows down according to the value set under Scrubbing Speed when

the mouse pointer is close to the cursor (default is 0.25, i.e. 1/4 of the original speed).

Note: If a scrubbing speed greater than 1 is set, this speed will be used for the faster scrubbing. The slower scrubbing will happen at original speed.

Scrubbing Speed: Here you can specify a speed relative to the original speed for the second scrubbing tempo. The range of values stretches from 0.01 to 10.0, i.e. 1/100 of the original speed up to ten times the regular speed.

Default Device
If you are using a sound card with multiple outputs, or multiple sound cards (e.g. onboard + external), select the output you want to use for playback.

Program

Auto play on external launch: If Sound Forge Audio Studio 12 is launched with an audio file as an argument and this option is active, playback of this file will start automatically, allowing Sound Forge Audio Studio to be used as an audio player.

Automatically adjust sample rate when inserting: If you insert audio material from files with differing sample rates into another file, resampling happens automatically. If you do not want this to happen, deactivate this option.

Hide News: Current news about Sound Forge Audio Studio 12 is sometimes displayed when program is launched. If you do not want to see these messages, you can deactivate them.

Temporary Files: This preset path is directed to the standard temporary files folder. Please ensure that this folder is on a hard drive or partition with sufficient free storage space.

Reset global "Do not show this message again": Pressing this button reactivates all tips dialogs which you have deactivated over time.

Undo

Undo Active: This allows you to deactivate the undo feature entirely.

Undo for plug-in settings: You can also turn on the undo feature for plug-in settings.

Undo Steps: You can also set the number of undo steps that are saved. A value of 20 means that the last 20 changes to a project may be undone.

Keyboard / Menu

This dialog allows you to specify keyboard shortcuts for every menu function in Sound Forge Audio Studio 12. You might also find it useful to set hotkeys for the functions you use most often in order to enable quick access.

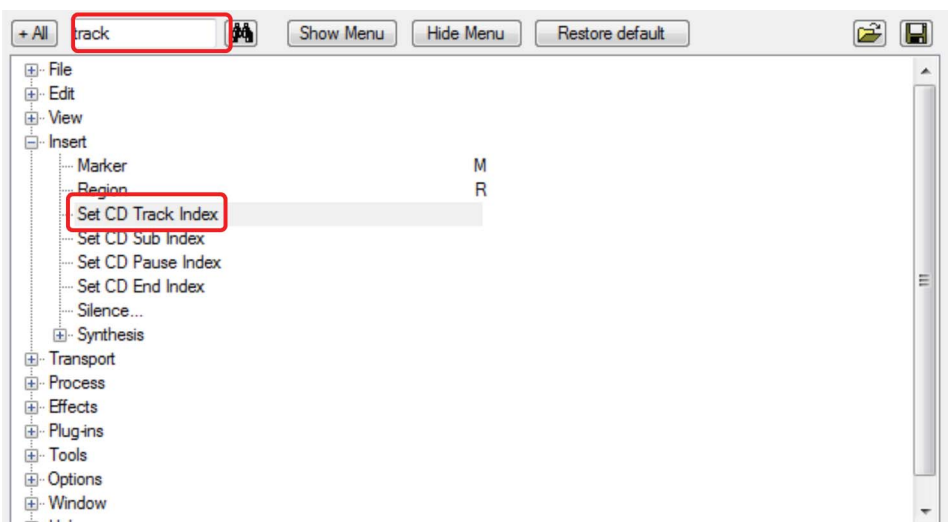
You can also hide menus that you don't really use.

When you close Sound Forge Audio Studio 12 the shortcut and menu settings will be saved automatically, so you can continue using them when you next open the program.

The main part of the dialog is the display of the complete Sound Forge Audio Studio 12 menu. Here you can select for which menu item you want to create a new shortcut or if the menu item should appear in the main menu of Sound Forge Audio Studio 12.

Displaying and Searching for Menu Items

The menu is displayed in tree form, and submenus can be opened by clicking on the "+" symbol. You can search for a specific menu item by entering a keyword in the search box in the upper menu bar and then clicking on the binoculars.

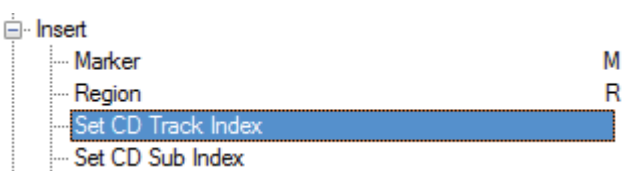


Show/hide menu item: Select a menu item which you would like to hide. "**Hide menu point**" removes the menu item from the menu. This menu item will appear grayed-out in the dialog's menu tree.

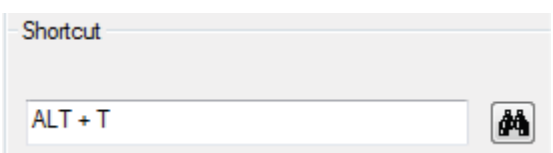
If you do hide a menu item, you won't be able to use a shortcut to select it. You can reactivate hidden menu items using "**Show menu**". "**Restore default**" resets the menu item to its default setting and makes the command visible again.

Creating shortcuts

Step 1: Click on the menu item you want to create a shortcut for.



Step 2: Now click on the entry box underneath the bar "Shortcut:..." and enter the desired key combination for the new keyboard shortcut.



Combinations of any key with "Shift", "Alt", or "Ctrl" can also be used.

If one of the shortcuts you choose is already assigned to another command, a dialog will appear which allows you to re-assign the shortcut or to choose a new one

Step 3: Activate the new keyboard shortcut by clicking on the "Assign" button. If one of the shortcuts you choose is already assigned to another command, a dialog will appear which allows you to re-assign the shortcut or to choose a new one

Deleting a shortcut
Click on the "Delete" button to remove the selected shortcut.

Export list: With this button you can call up and print a complete list of current shortcuts as a text file, Excel list or shortcut dialog.

Save/Load: Save your custom shortcut and menu settings by clicking on the "Save" button. You can load previously saved dialog settings by clicking on the "Load" button.

Special Keys/Mouse Wheel/Mouse

"**Special Keys**" redefines the keys for temporarily switching tools.

"Mouse" defines some special options for the keyboard and the mouse to achieve compatibility with older versions. These are:

- **Disable range zoom with double-click**
- **Zoom lasso allows vertical zoom without "Shift"**
- **Disable zoom with vertical mouse dragging on the timeline**
- **Knob characteristics same as faders**: If this option is active, knobs can be pulled up and down like faders by dragging with the mouse.

"**Mouse wheel**" allows you to define the mouse wheel behavior for zooming and scrolling in the virtual project. Define which modifier ("Alt", "Ctrl", "Shift") triggers which action in combination with the mouse wheel.

Colors

This is where you adjust all the colors used in the program. These color settings can be optionally loaded and saved as presets.

Restore

Restore last state: Restores the last status of the color settings before the dialog was opened.

Reset...: Here you can reset the color settings to those of the previous or original color state.

- Previous state: The previous color settings will be restored.
- Original state: Resets all colors to their default setting.

Dithering

No dithering, math. rounding of sample value: This mode converts the signals of 32-bit float via precise mathematical rounding without dithering. This rounding makes sure that surplus commas are not simply cut away, and it also prevents signal distortions.

Dithering with linear spread noise: This converts audio data at 32-bit float via dithering with noise featuring amplitude values that occur at regular intervals. The noise level may be set via the "Dithering depth in bits" option.

Dithering with triangular spread noise (standard dithering): In this mode, audio data at 32-bit float is converted via dithering with noise featuring amplitude values split into triangular intervals. Values in the medium range appear more frequently, and maximum or minimum values appear less frequently. This type of dithering typically produces more subtle results than linear dithering. The noise is not modulated through the signal, which results in a fading signal being enveloped by one constant noise signal.

Dithering depth in bits: This sets the level of the noise used in dithering. Input is in bits. Use this option to specify how many bits of the resulting 16-bits you want to be affected by dithering. In most cases, values between 0.5 and 2 will produce good results. Increase the value until you no longer hear any distortion effects. If you don't find any distortion effects, values below 0.5 should suffice. If you would like to add more noise to your signal, try entering values between 8 and 12.

VST

User VST Plug-in Folder: Set the path to your VST plug-in effects and VST instruments here. When you click the folder icon, a context menu will appear. This allows you to use the VST folder already set for your system as the plug-ins path or select any other VST folder. Sound Forge Audio Studio 12 then

scans the selected folder for VSTs. Not only are all the plug-ins imported, but they are also checked for usability within Sound Forge Audio Studio 12. This scan is only necessary once per folder, since available VST plug-ins are saved during this process.

If multiple VST plug-in folders are present on your computer, you can enter additional sources by scanning them with the option "(Re-) Scan selected VST folder". If individual plug-ins are not integrated as expected, then try using the option "(Re-) Scan selected VST folder including failed plug-ins" to check them. Sound Forge Audio Studio 12 only detects plug-ins in the specified folders. The corresponding data is saved in the file "VSTplug-ins.ini" in the folder "Program Files" > "MAGIX" > "Sound Forge Audio Studio 12".

The option "Scan user and system VST folders automatically for new plugins" runs an automatic scan for plug-ins each time the program is started. This can slow down the program start considerably, especially if you have a lot of plug-ins installed.

Toolbars

Edit Toolbar lets you add, move and remove toolbar icons. More information on this is available in the Toolbars section. **Reset Toolbars** sets the toolbars back to their factory settings. The two lower menu entries can be used to hide both toolbars and the status bar down below.

Window Menu

This menu contains commands for activating and arranging windows. All open file windows are listed in the bottom area of the menu.

New Window

Opens a new, empty window. This corresponds to the "New File..." command, but without the dialog for setting the sample rate, number of channels and bit depth. The previously selected values are used instead.

Shortcut key: Ctrl + Shift + N

Arrange/maximize

Switches between the maximized audio file window display and a display that arranges all windows next to and underneath each other.

Shortcut: Shift + F4

Activate Data Window/Activate Docking Window

These commands let you switch between the above section with the audio file windows and the docking window below.

Tip: Within these sections, you can switch between the windows with "Go to the next window" (Ctrl + Tab). This allows you to navigate between the different windows in Sound Forge Audio Studio using just your keyboard.

Shortcut:	Activate audio file window	Ctrl + P
	Activate Docking Window	Alt + P

Close docking window

Closes the current window in the docking area

Shortcut key: Ctrl + Alt + F4

Maximize/minimize docking window

Switches the height of the docking area between a maximized view and minimized view in which only the tabs in the docked window are visible. (This action is the equivalent of simply clicking on the tab).

Tip: You can adjust the height of the maximized view by dragging the border between the docking area and the file windows!

Shortcut key: Shift + F11

Go to Next Window

Switches to the next file window or docker area window, depending on which area is active. (see below)

Shortcut key: Ctrl + F6, Ctrl + Tab

Close All

Closes all file windows. If changes have been made, a query will first ask you whether you would like to save these first.

Help Menu

Contents and Index

This command displays the overview page of the help feature. You can jump to specific commands or read instructions on this page.

Shortcut key: F1

Tutorial videos...

This plays a tutorial video for Sound Forge Audio Studio 12. An Internet connection is required.

Deactivate Program

This menu item deactivates Sound Forge Audio Studio 12 with immediate effect. After deactivation, it is possible to immediately install and activate Sound Forge Audio Studio 12 on another computer

About Sound Forge Audio Studio 12

This displays copyright information and the version number of Sound Forge Audio Studio 12. At the very bottom, you can see the serial number required for support purposes.

Update online

This command lets you update Sound Forge Audio Studio 12 at any time. This feature checks online for available updates and installs them automatically.

Language:

Here you can select the language for the program interface. Sound Forge Audio Studio 12 restarts in the updated language.

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