MAGIX



SEQUOIA

PRODUCTION | BROADCAST | MASTERING

MANUAL

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What's New in Sequoia 16?

- **WaveColor:** This method for coloring waveforms replaces the old Comparisonics algorithm. But the fundamental principle is the same: The color stands for the average pitch, the saturation stands for the amount of noise (the less saturation there is, the less noise is contained in the sound).
- **Line representation in the spectroscope:** The spectroscope can now be represented as a smoothed or interpolated line. The familiar bar representation is available as well.
- **Improved ARA2 interface:** Improved switching between objects, improved pre-listen, more ARA plug-ins supported (e.g. SpectraLayers 6) Different ARA plugins can be used in the same project (e.g. Melodyne + SpectraLayers).
- **Tempo Track**: This is a new track type which is used exclusively for the tempo curve. It provides direct access to important tempo commands right in the track header.
- **Monitoring section: (view page 395) A dedicated monitoring section allows you to switch quickly between two monitoring paths, each with its own complete levels, output devices, effects and effects chains. An additional talkback signal path helps with communication between the recording room and mixing console.**
- New effects
  - **colorFX Suite** for sonic coloration with tape and tube amp simulation and bitcrusher.
  - **coreFX Suite** for modern all-round effects with a slick interface and with many handy presets.
  - **3D Reverb** algorithmic reverb for use in surround mixes (up to 9.1) that incorporates the position of the source in 3D space when generating the reverb.
- Track Panorama Settings:
  - Targets for stereo and surround panning are shown in one dialog.
  - A new output can be added with the plus button.
  - The panning law can be limited to the primary output.
  - When transferring settings to all tracks, busses can be excluded, the limitation to the primary output can be transferred to all tracks.
- MIDI
  - **MPE** (view page 354) recording, editing and playback with MPE controllers. MPE automatically assigns each note to different MIDI channels to allow individual controller data to be assigned to each note.
  - **Note Expressions:** MPE and other controller data can be assigned to VST3i Note Expressions if the VSTi supports VST3i Note Expressions. This enables controller data to be linked to individual notes, as with MPE.
  - **Controller Editor:** To allow for suitable editing of MPE/Note Expressions in the Controller Editor, the display of MIDI data from mixed channels has been

improved. Thanks to the increased resolution of the pitch bend, the controller range can be vertically zoomed by holding down Ctrl and scrolling with the mouse wheel.

- Clipstore Improvements:
  - Load options, e.g. load to timestamp, are now also available for virtual clips.
  - Tag handling: clips can be assigned to tags, there is a new column for tags and a tag filter function in the Clipstore toolbar.
  - Clips are loaded into selected subfolders.
  - An existing track and object automation is copied into the clip, and when pasting, there are options for using or merging the automation data.
  - The existence of track and object automation data can be displayed in a new column.
  - New column for the original object name.
- Better clipboard support: Copy and paste commands are available in the file manager and Clipstore, these also work in exchange with Windows Explorer with the usual shortcuts Ctrl+C/X/V
- Recording dialog: With the new option "Record standard output" (view page 836) in the recording dialog the standard output of the system can be recorded (e.g. Internet Browser).
- New object modes: For the two object modes " Link one track " / " Link all tracks " there are now 4, each one to the right / to the left.
- New SoX resampling algorithm for offline resampling: (rendering, CD burning), three modes for phase linearity during resampling are available.
- New algorithm for "Strip silence" with improved speed. For sample accurate operation, it can also be deactivated.
- Switching workspaces, mixer snapshots and mixer layouts via keyboard: There are new commands under "More..." in the View menu: "Select Workspace" to call up the Workspace menu, so that workspaces can be switched via keyboard, also: "Select mixer snapshot" and "Select mixer view snapshot".
- Automation:
  - Curve shapes for automation curves: Right-click an automation point of a track or object automation curve to set the shape of the curve from that point.
  - Keyboard shortcut for inserting and editing automation points with waveform.
  - Copied automation data of a parameter can be inserted on several tracks simultaneously.
  - Copy object automation to tracks.
  - New automation option Automation Preview for editing automated values without affecting the existing automation.
- Other improvements:
  - New toolbar buttons for Metronome and WaveColor.

- New system option for loading wave files with fades when auto-crossfade mode is activated.
- New project display option: Display the original clip name for objects
- New option "All markers" in export dialogs. It can be disabled, so all markers except CD tracks are ignored.
- Manual DeClicker: The shortcut Ctrl+Shift+D is now used for opening and closing, apply with Alt+D when the dialog is open.

Documentation And Help

In addition to the printed manual there are also other sources of information available that can assist you when you are working with Sequoia.

- PDF Documentation
- Help feature
- Online support area
- Online user forum

PDF Documentation

The informative PDF manuals included with Sequoia can be found in the program folder following installation. This folder also contains a clear, print-ready PDF document with the most important program keyboard shortcuts.

You can also access the main Sequoia documentation entitled "manual.pdf" by going to "Help" > "Manual".

Note: In order to read the PDF documents, you will need to install Adobe Acrobat Reader or another PDF viewer on your system.

Help features

In addition to this manual, we also recommend having a look at the help file via the "Help" menu; this includes more detailed information about the individual functions. You can use the "Help" command (keyboard shortcut "F1") to jump to the program interface overview page in the Help section and use the "Index" tab or "Search" feature to find specific topics. The "Favorites" tab can be used to save and display topics you often need to reference.

Sequoia also has a command search bar on the main program interface for finding menu commands and help topics (view page 75).

Forum

Visit the Pro Audio user forum oncommunity.samplitude.com/. As a registered user you will be supported by the professional Sequoia forum community and can get involved in all the conversations.

Contact

Support

Registered users get free technical support:

Online: www.magix.com/us/support/technical-support/pro-audio-support

Phone: +49 (0) 351 3164 9807, Mo - Fr from 10 am to 3 pm

Sales

If you have questions regarding licensing and upgrades/crossgrades please contact our Sales Department:

Tel. +49 (0) 5741-3455-25

Fax +49 (0) 5741 310 768

Email: sequoia@magix.net

System Requirements

Sequoia On Windows x64 (64-Bit)

If you have installed the 64-bit version of Windows on your system you can use Sequoia as a 64-bit or 32-bit version. The internal VST bridge makes it possible to use both 32-bit and 64-bit plug-ins.

Note: Because using the bridge can affect processor performance, it is recommended to use plug-ins that match the bit rate of the installed program version. If you do use the bridge, it is best to increase the ASIO buffer.

Recommended program versions:

32-bit	64-bit
Only 32-bit plug-ins	Only 64-bit plug-ins
Several 32-bit plug-ins and a few 64-bit plug-ins (e. g. VSTi)	Several 64-bit plug-ins and a few 32-bit plug-ins

The 64-Bit Version Should I Switch?

By switching to the 64-bit version you increase the available amount of RAM. On a 32-bit system the upper limit of addressable memory is 3.5 GB, for many programs it is only 2 GB. In contrast, on a 64-bit system the theoretical upper limit is a lot higher, namely 2^{64} = 16 Exabytes. Practical values are 16 GB for Windows 7/8 Home/Premium and 192 GB for Windows 7/8 Professional/Ultimate.

If you work with very large projects or memory-intensive VST instruments such as samples, changing to a 64-bit system is strongly recommended.

Before changing to a 64-bit system make sure that your computer has at least 4 GB of virtual memory and that 64-bit drivers are available for connected devices (sound cards, controllers etc).

General System Requirements

Supported Operating Systems

- Microsoft Windows 7 | 8 | 10 (32-bit and 64-bit)

Minimum System Requirements

- **Processor**: 2 GHz and higher
- **RAM**: 2 GB RAM (32-bit), 4 GB RAM (64-bit)
- **Graphics card**: Onboard, min. resolution 1024 x 768 pixels
- **Hard drive space**: 2 GB for minimal installation, 90 GB for Independence
- **Internet connection**: Required for registering and validating the program, as well as for some program functions. Program requires one-time registration.
- **Audio playback**: Sound card (ASIO-enabled sound card recommended)
- **DVD drive**
- **Optional**: CD/DVD ±R(W) recorder, MIDI interface

Hard disk

The maximum number of audio tracks depends on the rotation speed, access time, and data transfer rate of the hard disk. Using a modern SSD hard drive is recommended.

File formats

Import formats

Audio: WAV, Broadcast WAV, Sound Designer II, AAC, AIFF, DDP, FLAC MP3, MPEG, MUS, OGG, SD2, WMA

Video: DV-AVI, MPEG-1, MPEG-2, MPEG-4, H.264, HDV HD1, HDV HD2, MXF, MXV, DVCPRO1, AVC-Intra1, QuickTime, WMV(HD)

Image: JPEG, BMP

Additional: dira!Highlander, DIGISPOT, MID, GM, GS, XG, CD-A, DVD-A, CD/DVD data, AAF

Export formats

Audio: WAV, Broadcast WAV, AAC, AIFF, DDP, FLAC, MP3 , MPEG, MUS, OGG, WMA

Video: MPEG-2, DV-AVI, QuickTime MOV, MXV

Additional: dira!Highlander, DIGISPOT, MID, CD-A, DVD-A, CD/DVD data, AAF

Installation

1. Place the installation DVD into the DVD drive.

2. The installation program normally starts up automatically in Windows. If it does not, open Explorer and click the letter of the DVD drive. Double-click to start "start.exe".

   You can view the contents of the installation CD in the installation screen, as well as visit our website or install additional programs such as CodeMeter Runtime (view page 31).

3. Next choose the language in which you want to install Sequoia. In the language dialog window, you can choose between English, German, French, Italian and Spanish.

4. To begin the installation of Sequoia click on "Install program" > "Sequoia".

5. Follow the series of dialogs that appear by clicking "Continue" each time. If you choose the "User defined" installation style, you can also specify the target folder for the program folder and select additional components for installation. In the selection screen, you will see the total required memory for the installation.

6. After all files have been copied to the hard drive, confirm the end of the installation by clicking on "Finish".

After the initial installation, you can start the program via the Windows "Start" menu or the desktop link.

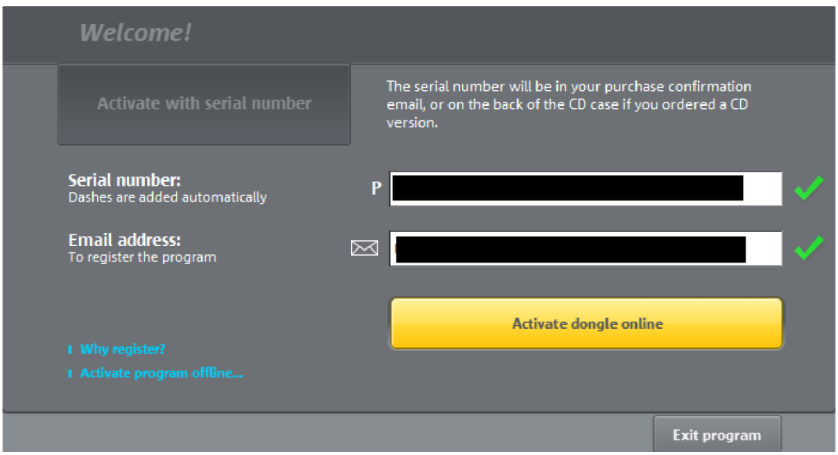
After installation, you can add or remove components by starting this installation program again and selecting "Custom" to select or deselect the respective components.

Code Meter Dongle Activation/Registration

Sequoia features hardware copy protection (dongle) which has to be registered when the program is started for the first time. The dongle enables usage of the program on different systems (but not simultaneously), without having to register each computer separately.

Automatic registration directly from within the program

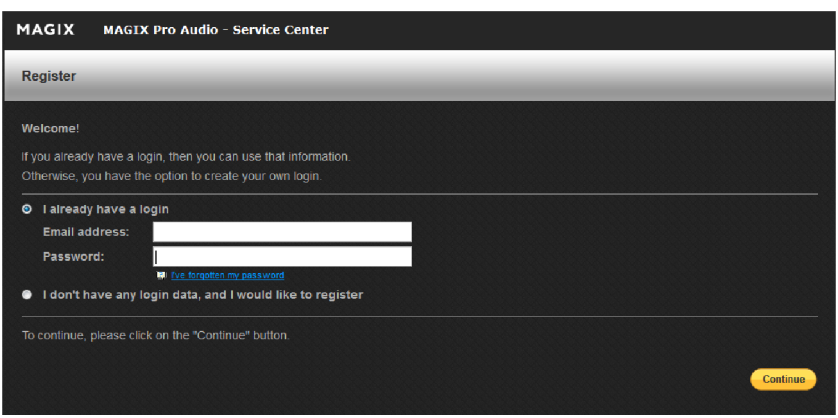
1. Insert your dongle and start the program. After starting up the program an activation dialog will appear.



2. Enter your serial number and your account email address in the activation dialog.

3. Click on "Register and activate dongle online" to proceed with activation.

4. Log in at www.magix.com in the support area. If you do not already have an account, you can register one here directly.



You can view your registered products and corresponding serials numbers in the Service Center after login. You can also update your personal data and get the latest downloads there.

Offline Activation of Sequoia

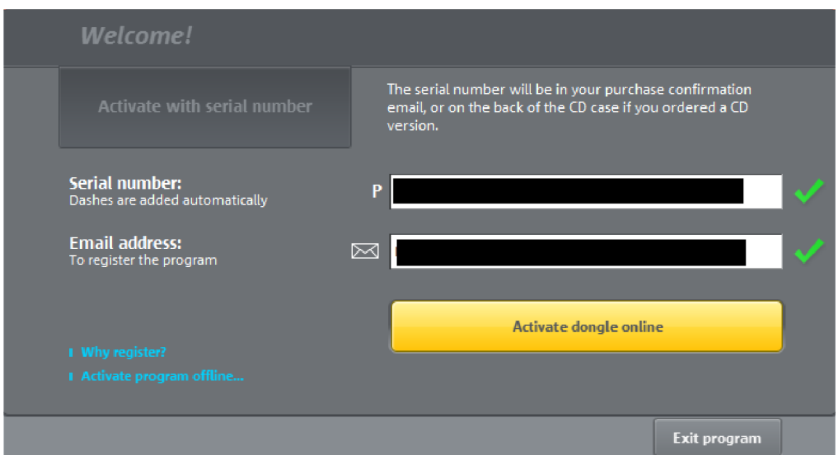
You can also activate your program on a different computer with an Internet connection if your studio PC does not have an Internet connection.

Activation can even be completed by fax, email or by filling out the form provided by the corresponding button, printing it, and sending it to the MAGIX address indicated.

Activation without direct internet access

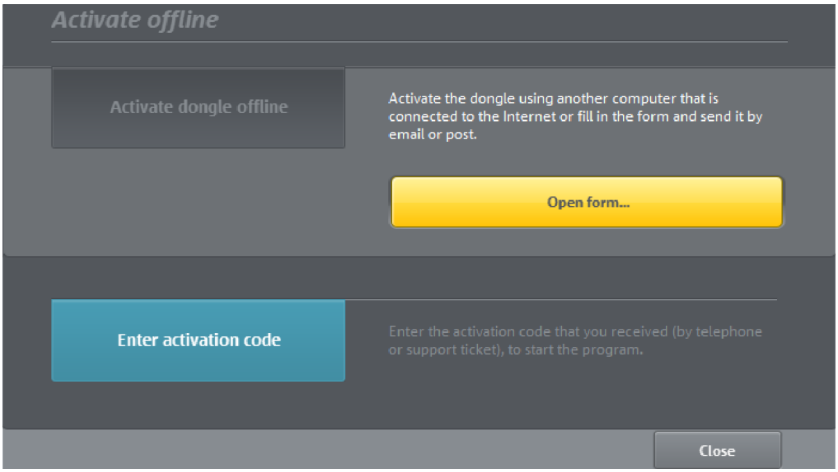
If you don't have direct Internet access, you can activate the program on another computer.

1. Insert your dongle and start the program. After starting up the program an activation dialog will appear.

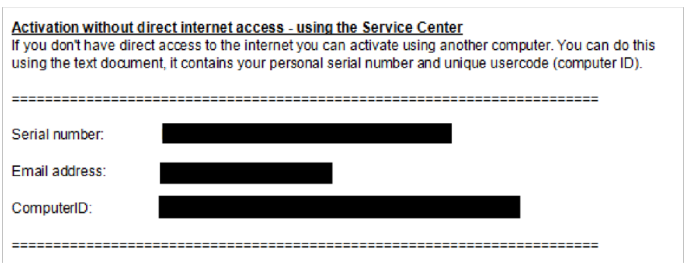


2. Enter your serial number and your account email address in the Start dialog.

3. Select "Activate program offline", click the "Activate dongle offline" in the following dialog and then "Open form".



You will receive a text document that contains your serial number and usercode (computer ID).


Activation without direct internet access - using the Service Center
If you don't have direct access to the internet you can activate using another computer. You can do this using the text document, it contains your personal serial number and unique usercode (computer ID).

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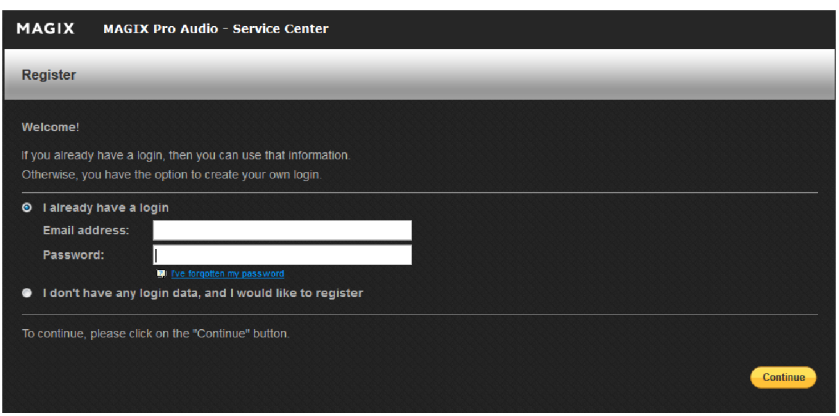
Serial number: [redacted]

Email address: [redacted]

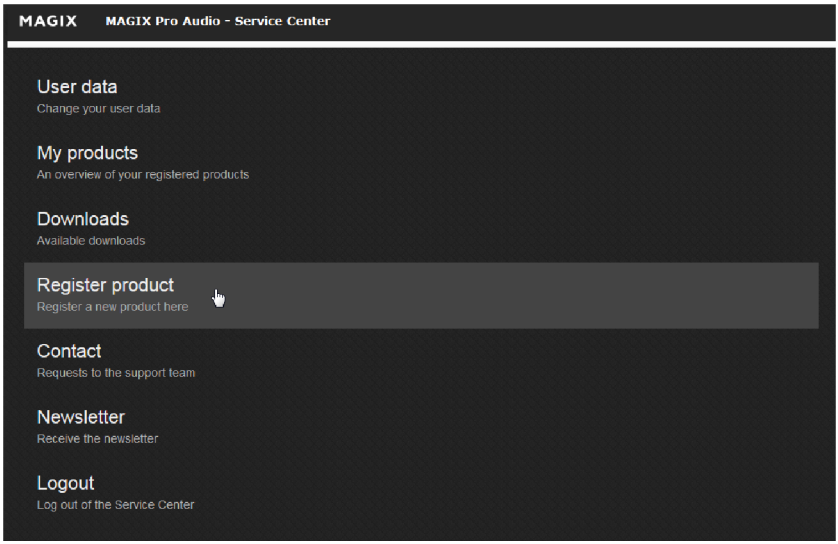
ComputerID: [redacted]

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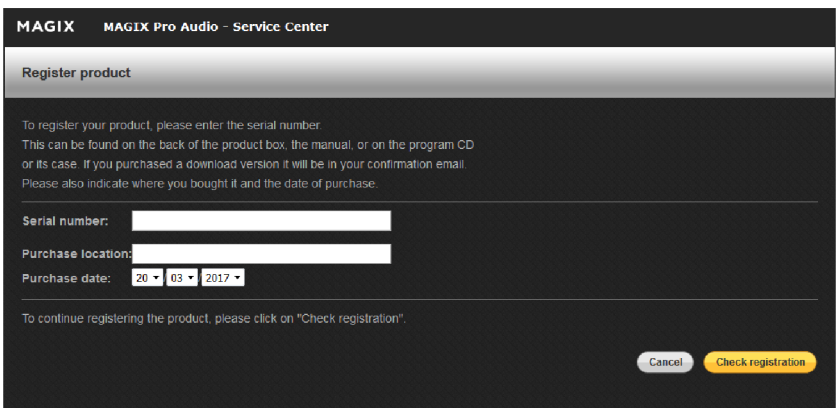
4. Go to a computer with Internet access and log in at www.magix-audio.com in the support area. If you do not already have an account, you can register one here directly.



5. After logging in select „Register product“.

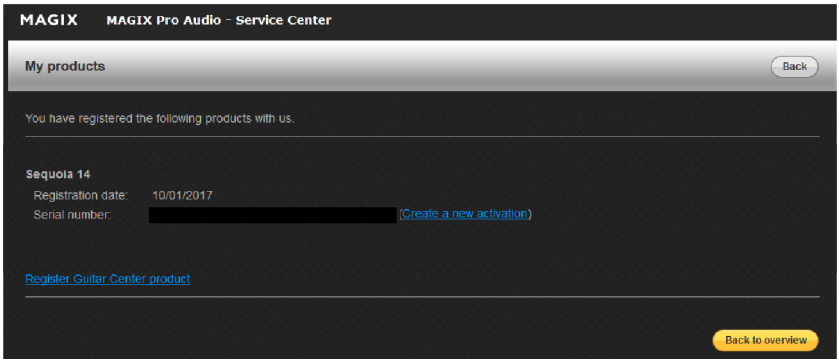


6. Enter your serial number and click on „Check registration“.



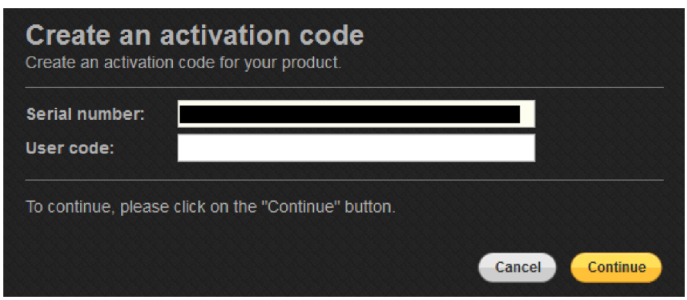
7. Now select the product to be registered and click on "Finish registration“.

8. Click in the start dialogue of the service center on „List of my products“.



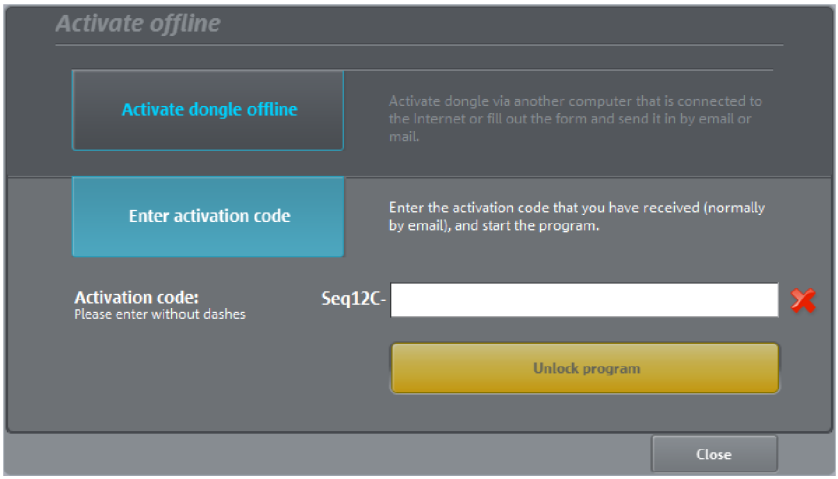
This will take you to an overview of your registered products.

9. Now click on the option „Create new activation“ behind the recently registered product and enter your usercode (ComputerID) from the form.



10. When you click on „Continue" you'll see an activation which is also sent by email.

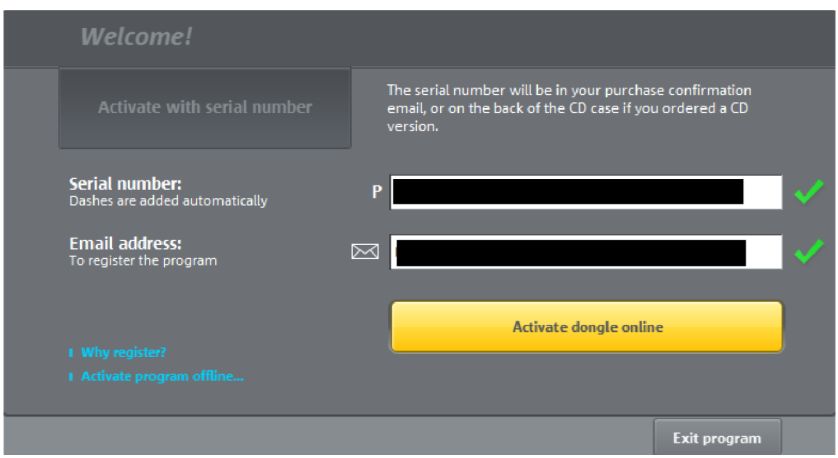
11. Go back to the activation dialog in the program and enter the activation code in the field „Enter activation code“ and finish the activation process by clicking on „Activate program“.



This completes the registration of Sequoia. You can now use the program. If you experience any problems during the registration process, please get in touch with our support (view page 20) team:

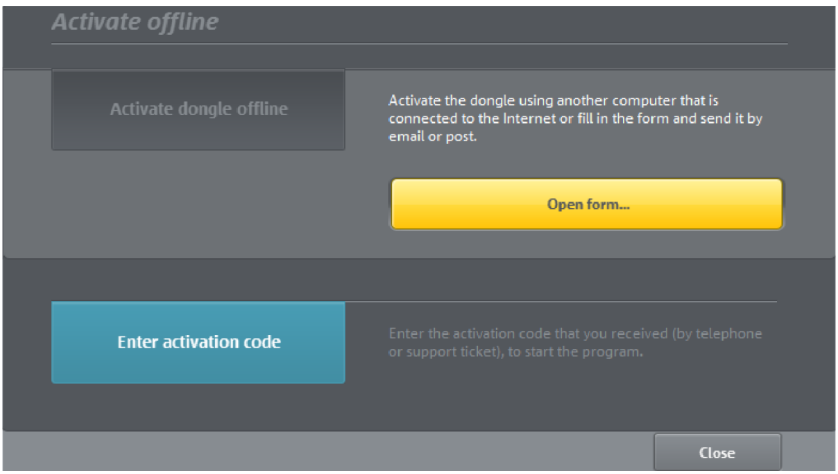
Activation by Email, Fax or Post

1. Insert your dongle and start the program. After starting up the program an activation dialog will appear.

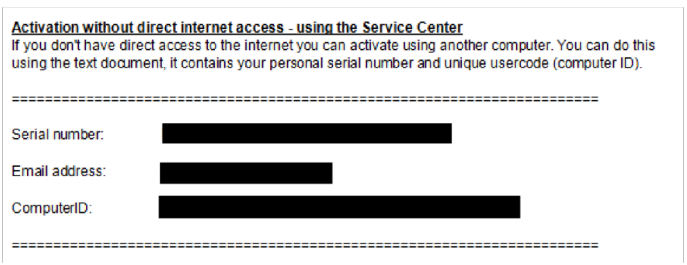


2. Enter your serial number and a valid email address.

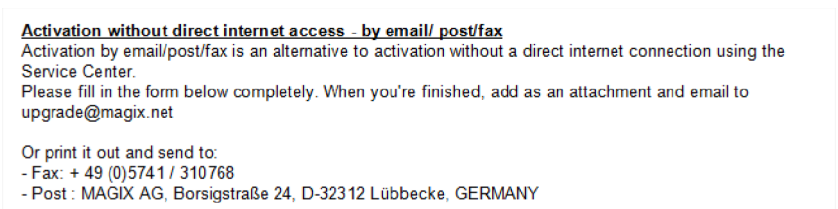
3. Select "Activate program offline", click "Activate dongle offline" in the following dialog and then "Open form".



In the open form, you will see the serial number and the ComputerID (=usercode).



4. Fill out this form and mail, fax or email it to the stated address.

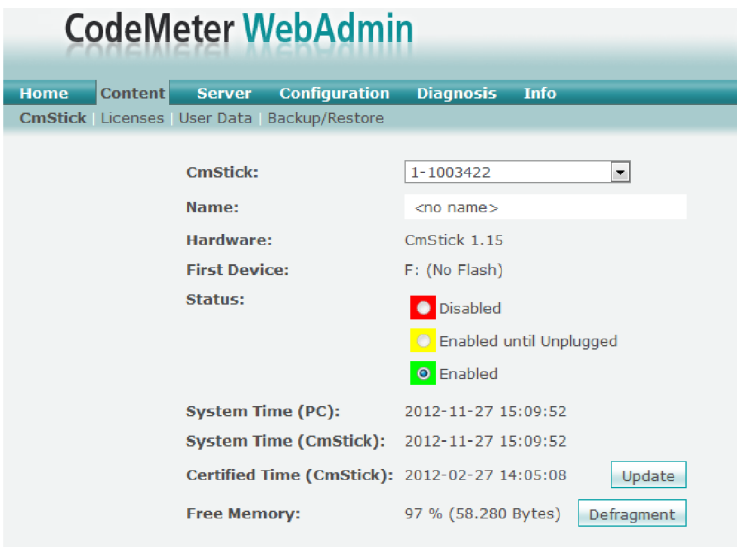


MAGIX will then send you the activation code.

CodeMeter Runtime And CodeMeter Control Center

The CodeMeter dongle manages program licenses and allows you to call up a program from various computer without requiring another activation. To do this, you need the "CodeMeter Runtime", which is also automatically installed with the program installation.

- You can find the latest version of "CodeMeter Runtime" in the Service Center at "Download/Tools" or in the download area at www.codemeter.de.
- Insert the CodeMeter into an available USB port on the server.
- Start the "CodeMeter Control Center" program via "Windows Start" > "Programs" > "CodeMeter" > "CodeMeter Control Center".
- Now click "WebAdmin" on the bottom left in the CodeMeter Control Center. Your Internet browser will open.



- In WebAdmin, go to the "Content" > "Licenses" tab. Here there is information about your dongle and license.

Network Installation

You can also run Sequoia on a network. Depending on the number of licenses you have purchased, Sequoia can be used on all of the workstations (clients) that are connected to the network server. The only requirement is that a registered dongle be inserted in the USB port of the server.

Configuration Of The Dongle/ Setting Up The Server

1. Install a current CodeMeter Runtime on a network server or a network server (Windows, Linux, Mac OS X).

2. Insert the dongle into an available USB port on the server.

3. Start the CodeMeter Control Center via "Programs" > "CodeMeter" > "CodeMeter Control Center".

4. In the CodeMeter Control Center, click "WebAdmin" on the bottom. Your Internet browser will open.

5. In WebAdmin, go to the "Settings" tab, check the "Start as server" option and press "Apply".

The server is now ready.

Note: The "Contents" > "Licenses" tab features a list of available license numbers.

Program Installation On Workstations

1. Install Sequoia on each of the client PCs using the program DVD.

2. Also install the current CodeMeter Runtime on the client PCs.

3. To register the dongle, insert it locally on a client PC or install Sequoia on the server and activate it there. If the dongle contains a multi-user license, then please select the offline activation and send the activation form to our Support (view page 20).

Note: Please make sure your firewall is set up properly. If you have any problems, deactivate your firewall as a test.

Tips for using the CodeMeter stick

CodeMeter Stick - Internet Access

Communication with the CodeMeter stick is performed using network components. For this reason, the TCP/IP protocol must be installed and access to the localhost (IP address 127.0.0.1) must be permitted. An actual Internet connection will not be established. However, access to the CodeMeter stick cannot be blocked by a firewall.

Note: Turn your firewall off if you are experiencing problems.

Restarting the CodeMeter Runtime Routines

If access to the CodeMeter stick is blocked during startup or while the program is running, restarting the CodeMeter Runtime may help.

To do this, start the CodeMeter Control Center and go to the "File/Exit CodeMeter". Next, start it at the same place with the command "Start CodeMeter".

This CodeMeter stick restart can be done while the program is running.

System Boot Problems on Some Systems

If the BIOS tries to boot from the CodeMeter stick, some systems may stall while the system is booting without displaying an error message.

If this happens, you can either remove the CodeMeter stick or deactivate the booting of USB devices in the system BIOS.

Note: If you experience recurring problems with your dongle, please contact our Support (view page 20).

Installing additional components

A range of additional components is integrated into Sequoia. When you start Sequoia for the first time, a dialog will appear with a list of these components. Click "OK" in the dialog to start downloading and installing the components.

- **Melodyne**: You register Melodyne when you use it in Sequoia for the first time. Enter the serial number provided during the registration process.
- **Birdline Platinum Pack** can be downloaded and installed from: http://skins.birdline.gr

There is also a short guide for installation on this page. The serial number for activating the package is included in the order confirmation for Sequoia.

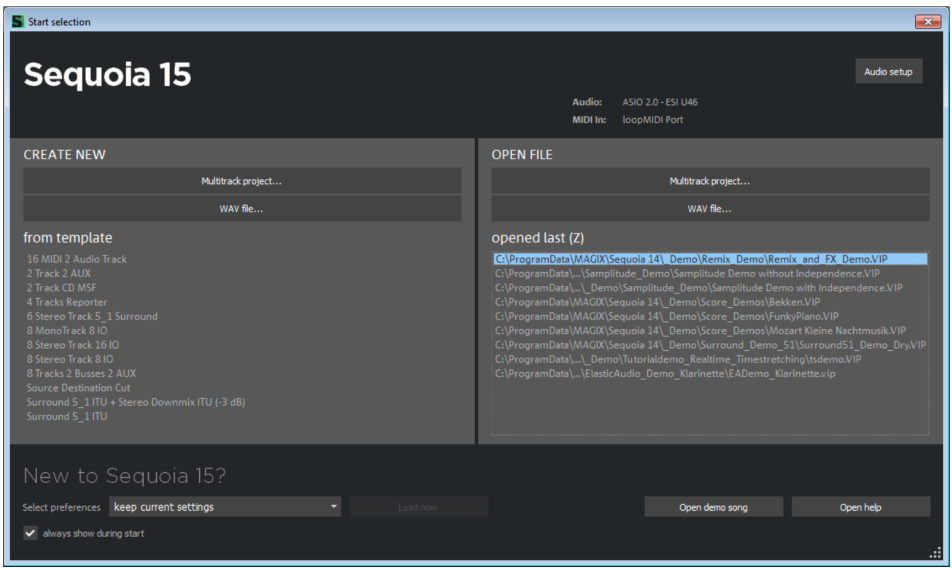
- **SpectraLayers Pro**: This must also must be installed separately. After installation, you can set up SpectraLayers Pro using the new "External tools" dialog (under "File" > "Program Preferences").

Getting around

In this workshop, we'll introduce new users to the basic functions in Sequoia.

Start dialog

After starting the program, the "Start" dialog will appear.



To the left of the dialog, you can create a new multi-track project or wave file. The templates underneath are for configuring the user interface in Sequoia for different tasks. Program settings will be configured differently too, depending on the task. All configurations can also be changed manually at a later stage in the project window.

To the right in the dialog, you can open a multi-track project created earlier or an existing wave file.

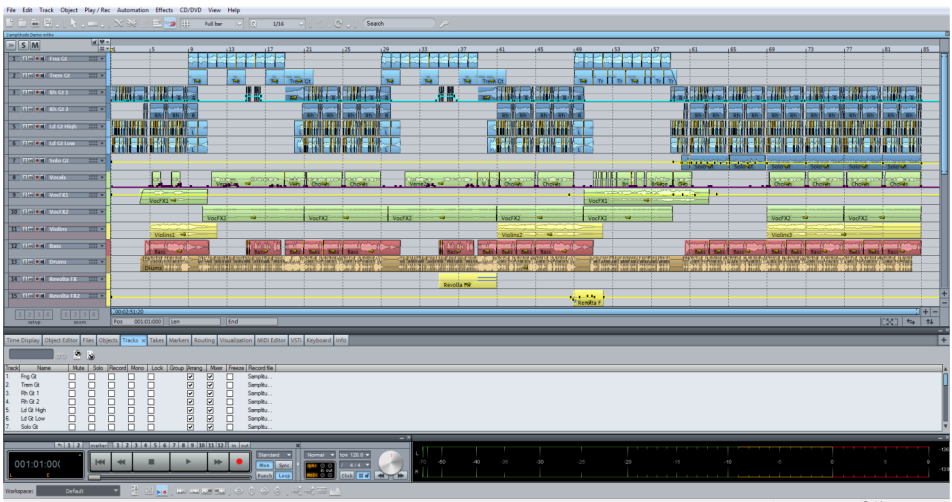
At the bottom of the dialog you can select preferences for the program settings. These settings are independent of projects and are the settings for general program functions. You can also open a demo song here.

User Interface

To get started, let's load an already produced demo song in order to explore the program a little.

- Click bottom right in the start dialog on "Open demo song".

The user interface opens with the demo song.



The user interface is split up into several areas. The large project window is in the middle. This includes a selection of tracks with objects in different colors.

The principle is very simple and familiar: Objects that are on top of each other in the project are played simultaneously and mixed together as they're played. Objects positioned one after the other will be played back in sequence. The project window displays the song structure in graphical form, with the beginning of the song on the left and the end of the song on the right.

The objects contain references to audio files, not actual audio material. Frequently objects represent only a certain section of an audio file, rather than an entire file. The audio itself remains unaffected on the hard drive. This enables you to work and experiment with audio the way you want in a process know as non-destructive editing, where the original recording is not affected or changed in any way.

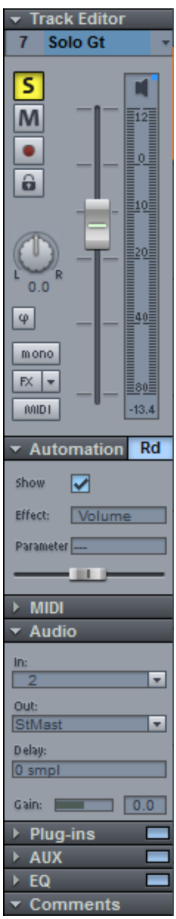
In a demo song, each track contains an instrument, as can be seen left on the track description. This is not always the case - all tracks can contains all kinds of sounds and objects.

Similar instruments are displayed in similar colors. For instance, all three brass section tracks below are in green colors, while above, the string tracks are in yellow and the bass and cello in orange red. You can change this color scheme if you wish.

In the illustration above, not all tracks in the demo song are visible. In the middle of the scroll bar on the right, you can see which tracks are in our demo song and which instruments are included.

Horizontal yellow lines are visible in each track in the demo project. These are volume curves. The curves ensure you can control the volume curve at any point. Panorama positioning and effects can also be continuously set using similar curves.

The track editor on the left is particularly important. Let's take a closer look at this.



The track editor settings only apply for the currently selected track. If you click in the other tracks, you'll see how the track selection changes each time. The settings for the track editor also change.

The track editor contains different categories which can be expanded or collapsed using the little arrows. In the image on the left, the categories for MIDI, plug-ins, AUX, EQ and Comments are still collapsed. You can expand each category to explore it.

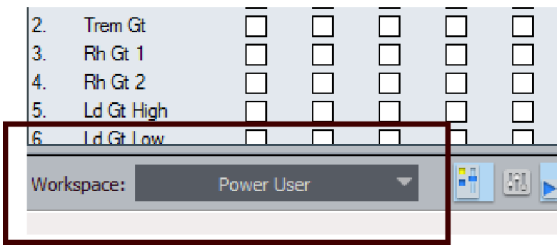
The track editor contains many functions that you can apply to the selected track. You can set volume and stereo panorama for the track, switch off or hide the volume curve, operate MIDI devices, set audio playback device for the track or use effect plug-ins for track effects.

You can also set the track to mute (M – "Mute") or solo (S – "Solo"). But we'll get to that in just a bit.

In addition to two toolbars on the top and bottom image border, the user interface displays the transport console with playback functions familiar from a tape machine: Start, Stop, Forward and Rewind as well as additional functions.

If the interface shown here appears cluttered, it's due to the fact that Sequoia is open in the "Power User" view. This view displays as many functions as possible to make working with the program as comprehensive as possible.

Below left you'll see the "Workspace" menu where you can switch the view.



Switch the view for specialized tasks such as mastering here. In these views, some buttons for functions are hidden or enlarged to make the interface simpler for your workflow. However, the functions are available at all times and you can switch them on whenever you need them.

Play

So far we've just had a look around - now let's have a listen.

- Press the spacebar on your keyboard.

This is the simplest playback method and one you are likely to be familiar with. Alternatively, you can use the play button in the transport console.



A thin line will move from left to right above the tracks during playback. This is the position line, which shows the currently playing position.

You should now be able to hear music. If you don't hear anything, try some troubleshooting. Are your speakers functioning and is your system volume set correctly? If all this is set up correctly, but the sound can't be heard or is poor quality, go to the end of this workshop to find out what might be the cause.

- To stop playback, press the spacebar a second time (or click the Stop button in the transport console).

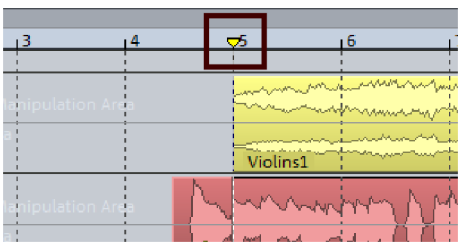
This stops playback and resets the position line to its previous location. In our case, the line has snapped right back to the start.

You can also keep the position line at the location where it currently is. Use the Pause button to do this.

- Start playback again and press the 0 key on the numeric keypad. Or, if you want to use the transport console, click again on the play arrow.

This stops playback but keeps the position line in the current position.

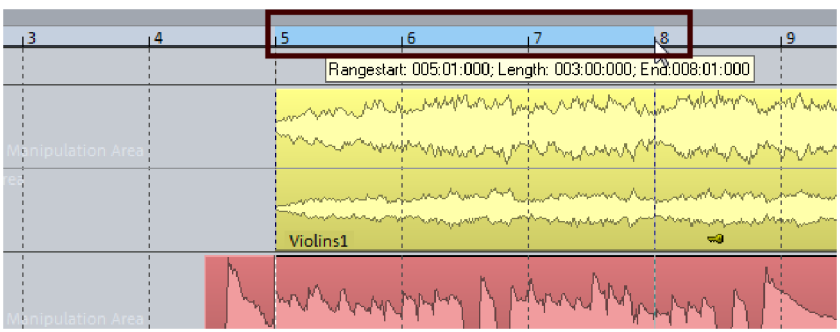
To jump to a specific location in a song, click above the first track in the grid list with the time information. This moves the playback marker.



If you start playback now, the track will play directly from this spot. You can also set the playback marker during playback. The marker will jump to the new position and continue playing from there.

To play a section as a loop, drag it to a range above the first track.

- Click above the first track and then drag & drop a loop range there.



An info tooltip to define the range area will appear while you are doing this. The range itself appears as a blue bar above the first track.

Ranges can be used for various editing tasks in Sequoia. In addition, there are lots of options for editing ranges and using them again at a later stage. Here, we want to use a range solely for the purposes of playing it as a loop.

- Activate the "Loop" function in the transport console.



- Then start playback.

You will see that the position line at the end of the range jumps back to the start and that playback continues from there without a pause.

You can use loops for tasks such as when you are editing a looped position and want to preview results right away.

- To switch off loop playback, deactivate the function in the transport console.

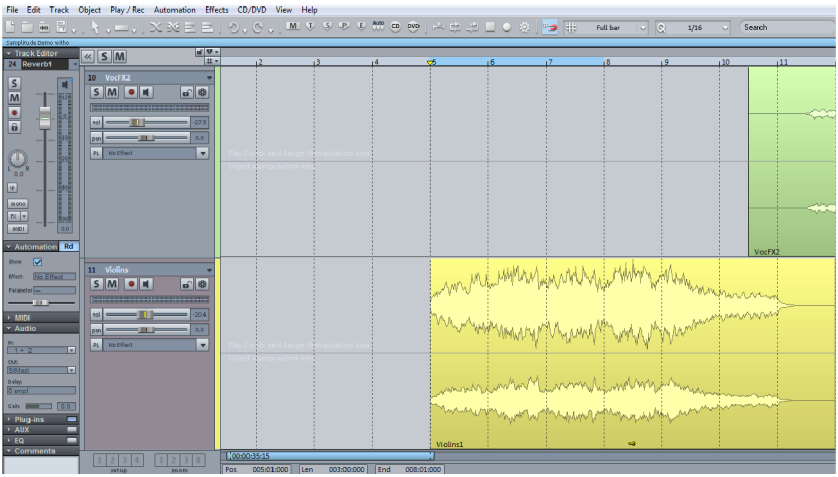
Zooming

Zooming is an important functions for making sections larger and editing details more precisely.

There are a range of additional zoom functions that will be explained in more detail later (view page 111). Using the mouse wheel is the most intuitive way to zoom for general tasks.

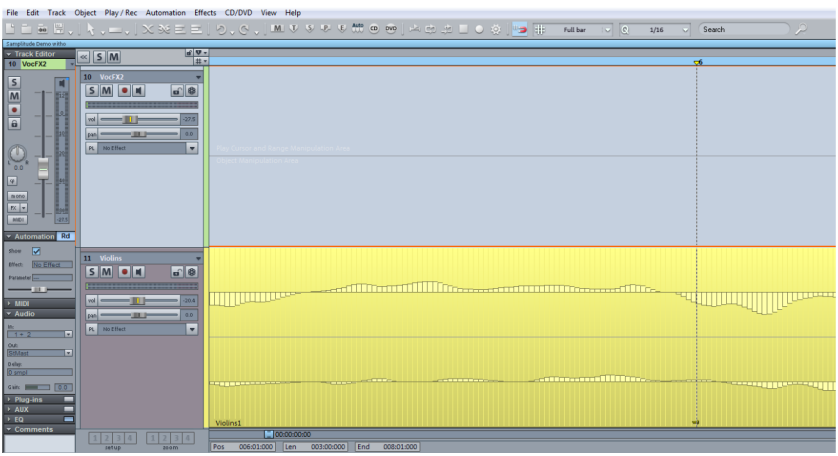
- Position the playback marker at the position where you want to zoom in.
- Hold the Ctrl key while using the mouse scroll wheel to zoom.

This lets you achieve a smooth vertical zoom. As a result, your tracks look wider, and fewer tracks are displayed simultaneously.



- Hold down the shift key along with the Ctrl key for horizontal zoom.

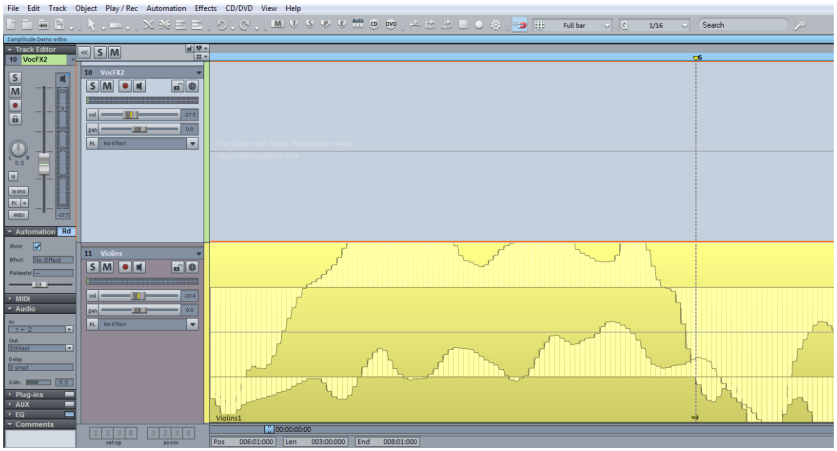
This means you can zoom in in horizontal view and that the time segment displayed is shorter. You can now make out more details in the waveform display.



In this illustration, we have zoomed out to the sample basis. The time segment pictured is so short that no time entries above the first track can be made out - things are at the millisecond level. The grid lines correspond to the individual (discrete) sampling values. These values are set by digitized audio material from the file.

A third option is to zoom in in a waveform display within the object. The differences between the peaks and troughs in a wave display will then appear larger. This makes it easier to see the zero crossover in a waveform, for instance.

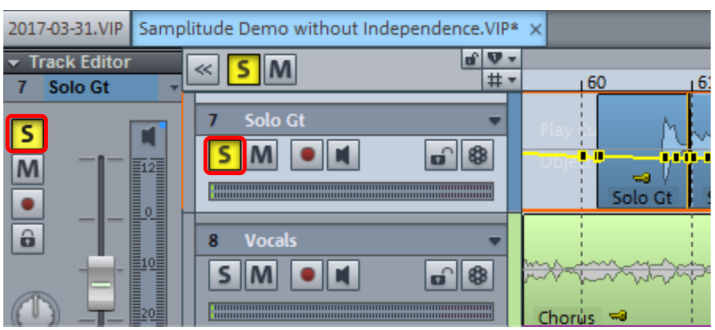
- To zoom in within an object, hold down the Alt key and use the mouse scroll wheel to zoom. The project section remains the same.



Object Editing

Editing selected objects is helpful for many tasks. To try this editing workflow out, select an object on any track.

Make the track solo so that you will only hear the sound of the object you've selected. To switch to solo mode, use either the S button in the track editor or the button in the track header.



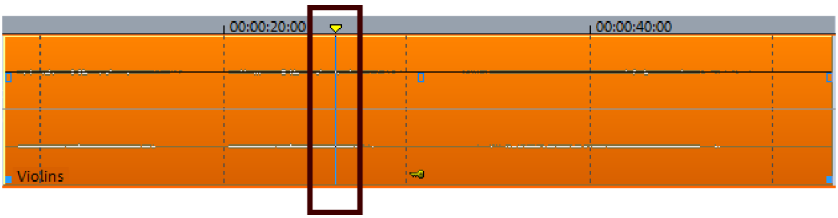
Place the playback marker before the object and start playback. You should now hear the object only.

The object can be edited in various ways.

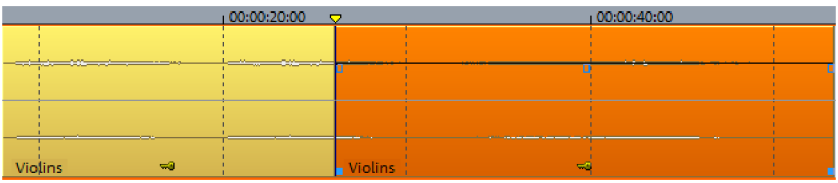
Object Editing

- Position the playback marker at the cut position.

- Click on the object you want to edit to select it.

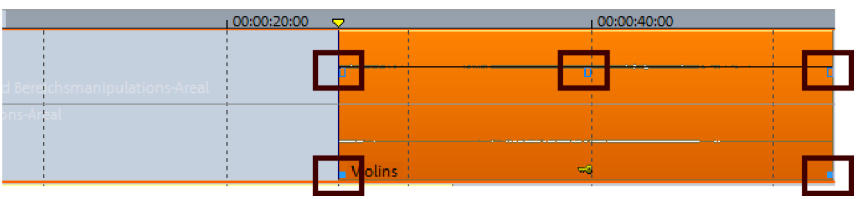


- Press the "T" key on the keyboard.



This creates two new independent objects. Each of these objects can be edited independently of one another. You can delete an object from a song.

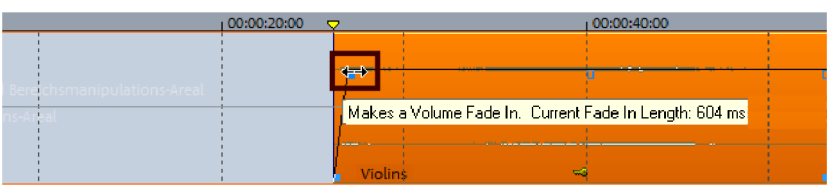
In the following illustration, we've deleted the left part of the first object.



Note: Simple object editing in the project is displayed here. Sequoia also offers many professional editing solutions, e.g. Source/Destination Cut (view page 400) or Multi-Synchronous Cut (view page 411, view page 422).

If you look closely, you will see a total of five small checkboxes on the selected object. The checkboxes are object handles which you can drag using the mouse.

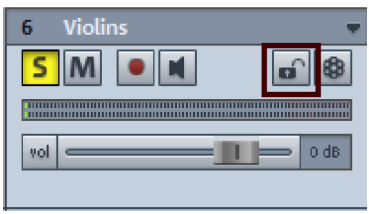
Object fades: Drag the checkbox in a bit top left. This creates a fast fade in. A tooltip will provide information about the exact progress.



Similarly, you can drag the handle in the background to the left in order to create a fade out.

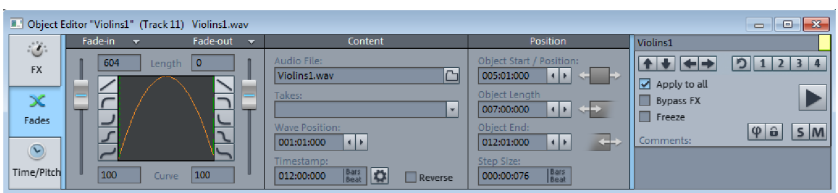
Object volume:You can use the handle in the center to adjust the object's volume. We have added the volume fader to the track editor above and volume curves to the tracks. In addition to the track-based options, you can add volume settings to all objects. This enables you to control the relative volume of each object.

Move Object/Set Objects: You can move objects any way you want, for instance horizontally within a track or vertically or diagonally between tracks. In our demo song, however, objects are fixed. To move them freely, you need to click the lock symbol in the track header.



You can then move objects in the track.

Object editor: Double-click on the object to open the object editor for more detailed editing.



To set object position and fades more precisely than with drag & drop, go to the "Fades" view. In the "Edit" menu, you can access a professional crossfade editor for adjusting fades and object transitions (crossfades) in detail (view page 425).

The object editor offers two additional views which you switch to on the left. "FX" gives you access to effects, volume and panorama settings for this object, while "Time/Pitch" lets you adjust pitch and time stretching for the object.

These are some of the key tasks that can be carried out with objects. You can find out more about object editing later in the book (view page 142).

Effects

Effects let you enhance your audio productions on a whole new level. A sophisticated use of effects adds depth to your material and enhances its sound. Sequoia includes a wide range of effects and effect plug-ins for all kinds of audio tasks, e.g. cleaning, sound optimization or distortion.

You can add audio effects to three different levels: Audio objects, tracks, and master. When played, effects are calculated in the audio objects first. Then the track effects are calculated, followed by the master effects.

First start with the object effects to distort individual objects while other objects in the same track remain the same. To apply object effects, select an object and open the "Effects" menu. We can set individual effects more precisely at a later stage.

You can select track effects in many locations in the program e.g. in the track editor in "Plug-ins" or in the mixer channel strips. These effects apply to all objects in a track and are integrated iwth the relevant object effect settings. Here you can set the necessary effects for the entire material for a track. These might include cleaning effects or an equalizer on an instrument track.

The mixer is where you set master effects at the end of your project that will apply to the entire sound. Let's take a look at the mixer.

Mixer

The mixer provides you with a separate user interface that's optimized for mixing. The goal here is to adjust loudness, panning (the left and right distribution of sound in the stereo field) and the sound of individual tracks in such a way so that you end up with a well-balanced overall sound.

The simplest way to open the mixer is using the keyboard shortcut "M", or you can also access it via the "View" menu.



The display of the mixer is designed to replicate a studio mixer. Each track in a project corresponds to a channel in the mixer. Each channel includes a volume fader. The pan control and the equalizer are located above the faders.

- Use the sliders to adjust the loudness for each individual track until you get a balanced sound and each instrument can be sufficiently heard.
- Use the pan control to distribute the tracks in the stereo field. Vocals and bass should always be kept in the center.

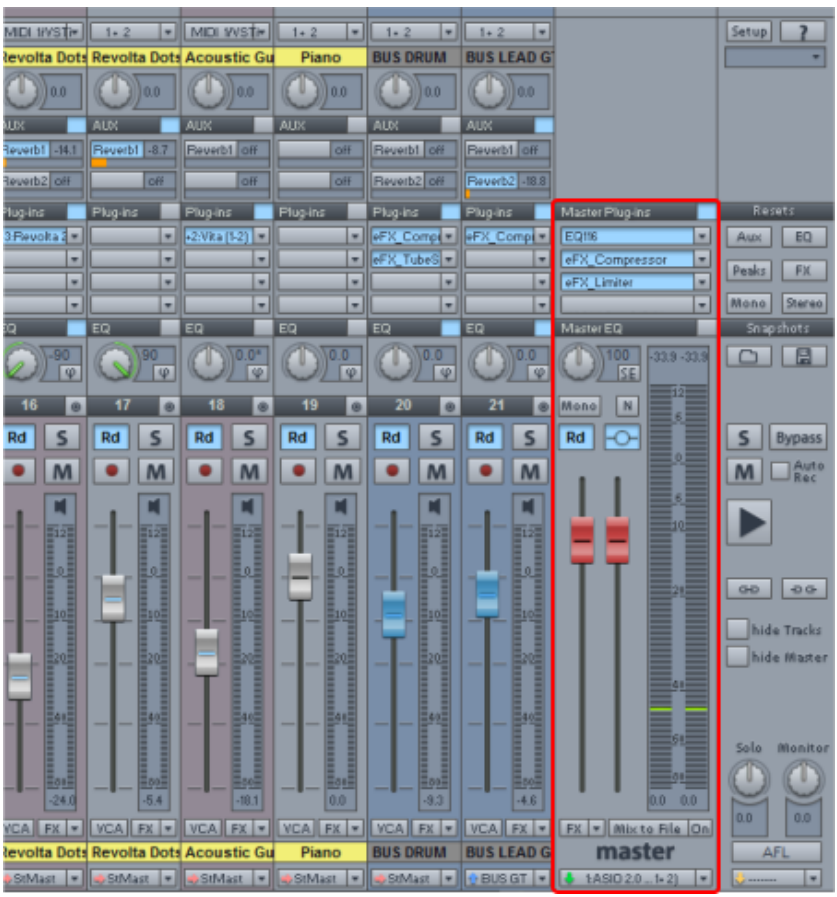
- Use the equalizer to trim frequency ranges (e.g. the highs on the bass guitar or the lows on the hi-hat). This creates space for other instruments in this frequency range. A right mouse click opens the equalizer dialog, where you can set effects more precisely.



You can load additional track effects in the "Plug-ins" range above the equalizer controllers. For instance, a compressor can be used to amplify the sound of the track or reduce individual instruments that feature sounds that are too loud.

Mastering

Mastering is the final step, the one in which we optimize the overall sound of the song. The mixer is also used for this task. On the right, you'll find the Mastering area.



It includes two volume faders for stereo output, a master equalizer for sound editing and section with master plug-ins. Four typical mastering plug-ins are selected in the demo song: a multiband stereo enhancer, an FFT filter, a compressor ("AM-Munition") and a room simulator for room sound.

If you click the small arrow on the plug-in slot, you will go to the selection menu with all effects that can be used as master effects.

As you can see, the mixer offers a powerful interface for production once you're finished with arrangement and object editing. We will come back (view page 363) to this topic in a chapter devoted to it.

Export

At the end of the production workflow, when everything sounds the way it should, audio editing with Sequoia is finished. It's now time to put the results to use. This process is called "export".

There a few different options to do this:

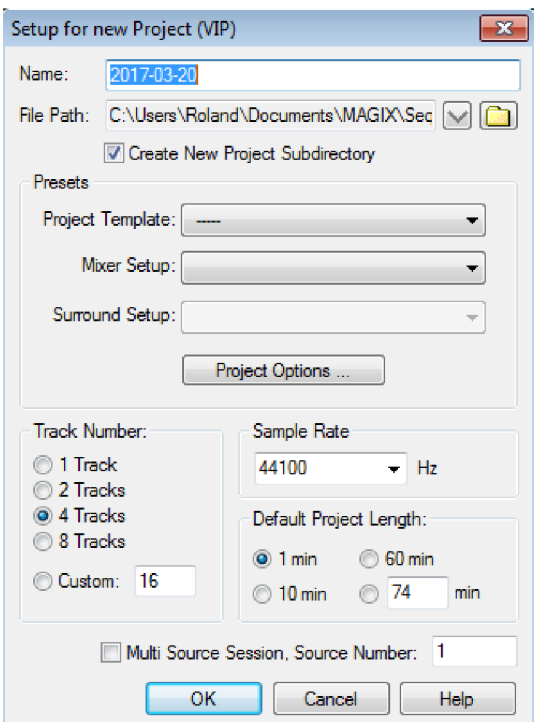
Burn CD or DVD: Burn an audio CD in the "CD/DVD" menu. This is the classic way to export your song.

Export file: You can save your production in a variety of file formats by going to "File" > "Export". The most important formats are wave and MP3.

That's the end of this short demonstration using the demo song.

My Projects

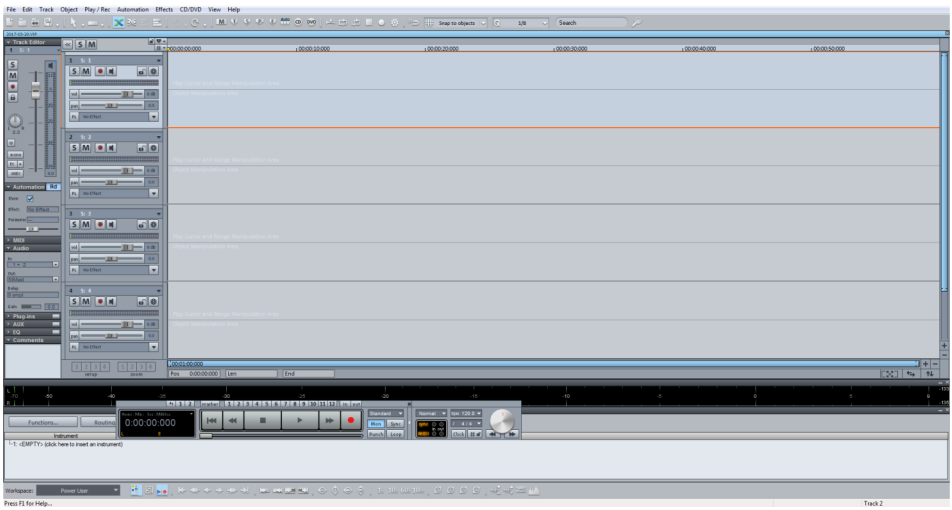
Add new projects either directly upon program start or via "File" > "New Virtual Project" while the program is running. Set presets for the project in the "Setup for new project" dialog.



We recommend you enter a name and file path that will make it easy to find the project later.

If you know how many tracks you're going to need, enter the number here. If you're not sure, you can add more tracks later (via the "Track menu").

Click on "OK" to open an empty project window.



There are two options for loading your own audio material to a project:

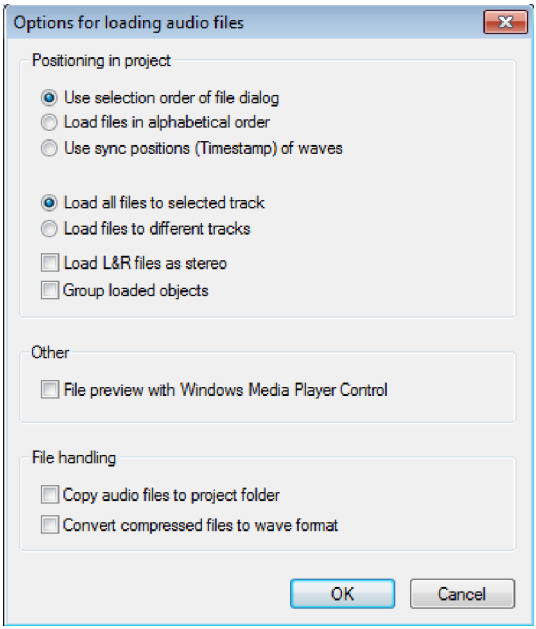
- Start an audio recording (we will explain how to do so later)
- or import one or multiple audio files.

To import a file, go to ""File" > "Import". You can import the following formats:

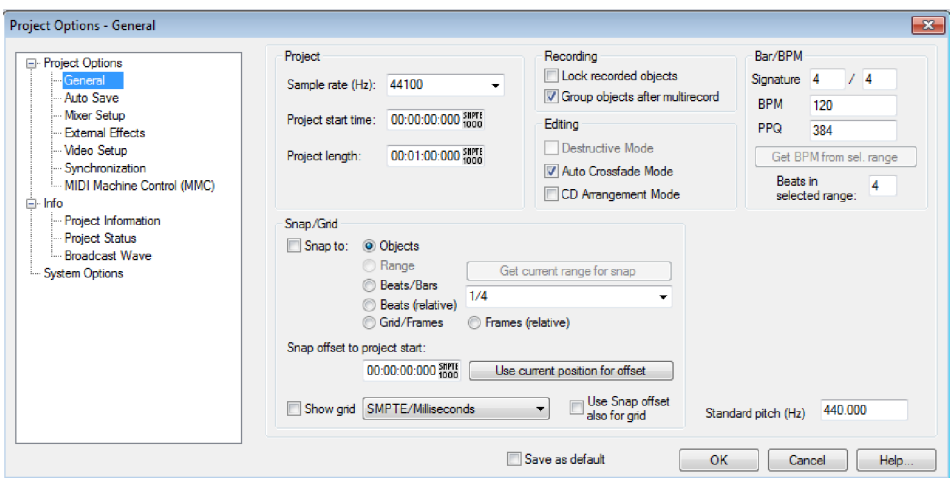
- Audio files in all supported formats (shortcut W)
- Sound Designer II files
- Raw data/dump
- MIDI files
- Video files
- Tracks from audio CDs or DVDs

Other formats can be found in the "File" menu.

When loading audio files, you can select several files in the selection dialog (with Ctrl or the shift key). You can then specify which objects you want to apply to the tracks (e.g. one after the other on a single track or on different tracks).

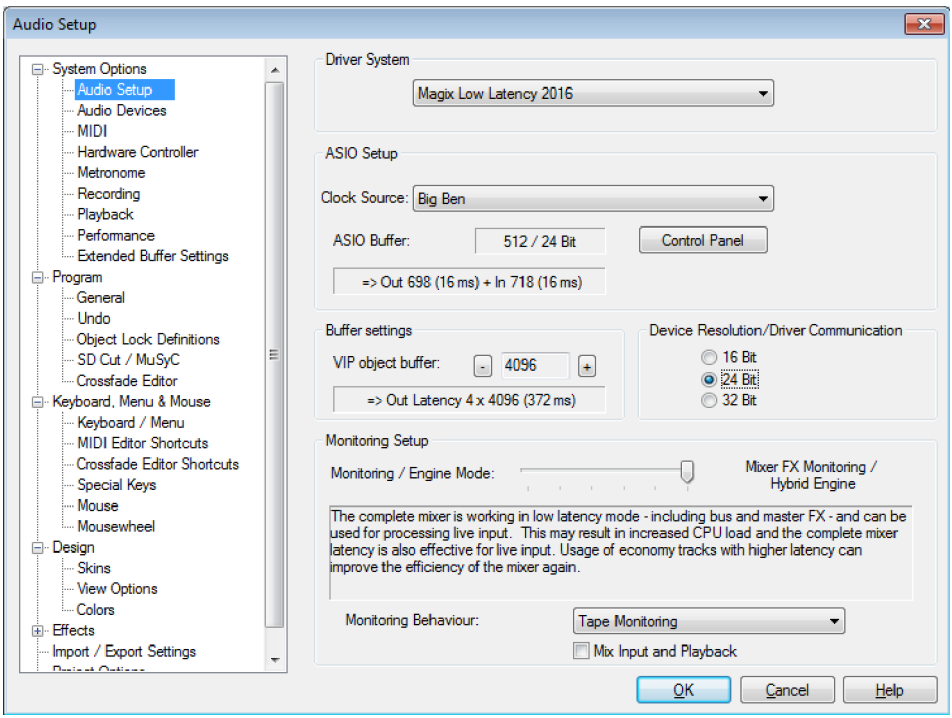


You can adjust settings for an individual project more precisely in the project options. Access the project options via "File" > "Project Properties".



You can switch on the grid and snap function, activate a beat grid or specify the speed of a track. Not all these options are necessary for any one project. However, it's good to get a general overview of your options.

We also recommend getting an overview of the system configuration. To do this, press the "Y" key.



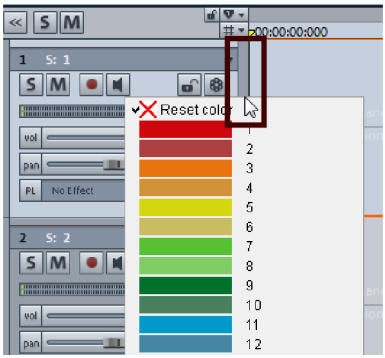
The system settings contains information and options about your driver system, available audio interfaces, buffer settings, monitoring options, plug-in paths, ReWire settings and lots more.

Tips for a smooth workflow

Finally, here are a few tips to keep your workflow streamlined:

Folder structure: It's a good idea to set up a folder structure on your hard drive for each new project. For commissioned work, for instance, you could give the top folder the name of your client, create a sub-folder for the actual project and additional sub-folders for source material, project files and partial projects.

Track names: Before you start working on the individual editing, give tracks descriptive names in the track editor, so they will be easy to identify. This way it's also easier to see what material is included in a track.



Coloration: Use different colors for different instrument groups (such as drums, guitars, wind instruments, etc.) to provide yourself with a better overview of your arrangement. Click on the area between the track header and track to assign the track a color.

Folder tracks: Many tracks include the option to group all tracks that belong together in a folder track (view page 132). This hides these tracks so that they don't take up unnecessary space in the project window.

Audio groups: Audio group (view page 383) (also called "audio subgroups or "submix buses") control the total level of all single channels assigned to them. Using audio groups also allows you to process the total signal by means of insert effects as well as automate it and route it to all available audio outputs.

Markers: Markers allow you to jump (view page 128) to specific points in the project. We recommend placing a marker at the beginning of each song part (e.g. intro, verse, refrain) in a song arrangement. This makes it easy to navigate through the project to these positions.

Comment function: Throughout the course of working on a production project, you will often have a lot of information and preliminary values to keep track of. And these can change quite a lot. It's more than normal to not be able to easily recall all the information about each individual track when you need it. This is where notes come in handy. You should try to get into the habit of keeping notes and workflow instructions within your project in the "Comments" section of the track editor or the info manager.

Create your own presets: You should put your carefully-assembled EQ, dynamic and channel strip presets in folders of their own that contain (view page 223) all the presets of a given category. Be sure to create a structure that provides you with a good overview of your presets and makes it easy for you to find and use the preset you're looking for. Naming the presets is important. Use detailed names that tell you what kind of preset it is and what it sounds like.

Track Bouncing: It's possible to "bounce" different intermediate stages of your workflow via this option in the "File" menu. Track bouncing is a good way to free up resources for your project.

Save project: Make sure you save your project regularly so you don't lose your work if something unexpected happens or your computer crashes.

Trouble Shooting

Peak meter not displaying peaks

Check to make sure the audio source (e.g. microphone, guitar) is connected to your sound card. Have you selected the right input? Is the controller for the input level on the sound card—or the gain controller on the mixing console—turned up? Most sound cards include their own mixing program where you can set the input sensitivity.

I can't hear myself (no monitoring)

In order for monitoring to work, you have to have an ASIO driver selected as the sound card driver. Sequoia includes the MAGIX Low Latency driver, which provides ASIO functionality for all audio devices.

The monitoring sound is hollow or strange, or I hear myself twice.

Most sound cards—including your computer's onboard sound—provide support for the Windows sound card mixer in addition to support for Sequoia monitoring. This mixer contains a signal path that leads directly from the input signal to the sound card output, and can also be used for monitoring.

If the monitoring sound is hollow, strange, or you hear an echo, both monitor paths are likely active and are interfering with each other. If this is the case, deactivate monitoring in the Windows sound card mixer.

1. Open the control panel, go to "Sound" and select the "Recording" tab or right-click on the speaker icon in the lower right hand corner of your screen and select "recording device" from the context menu.
2. Double-click on the device you want to use. If your audio source is connected and produces an audio signal, you should see a peak in the peak meter.
3. Go to the "Listen" tab and uncheck the box that says "Listen to this device". Then click on "Apply".

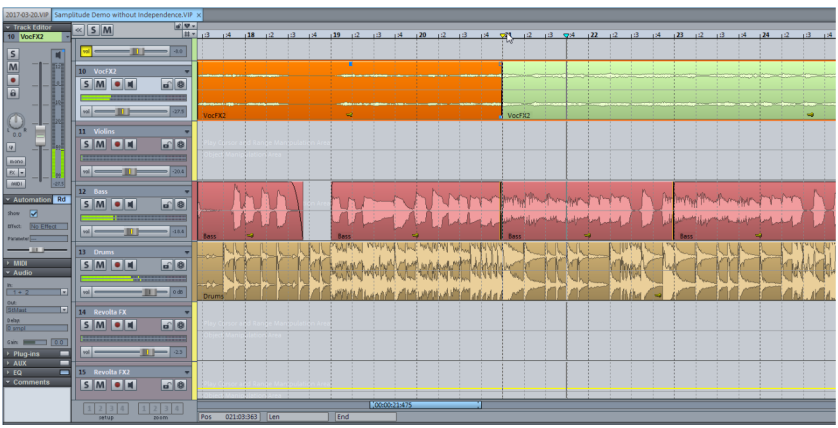
I don't hear any effects in the monitoring.

In order to hear effects, you must have at least "Track FX Monitoring" set as the monitoring mode (view page 596).

Sequoia Fundamentals - Basic Terms

VIP

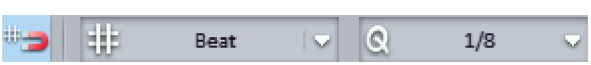
The virtual project (*.vip file) contains all the data from Sequoia. It does not contain audio data but only the names of the imported and recorded audio and MIDI files and the saved locations on the hard drive, the data for the contained objects and most of the effects processing. The object display in the project window of Sequoia is subsequently referred to as a VIP.



The file extension .vip is applied to all virtual projects in Sequoia.

Snap

The snap grid determines the step size for navigating and editing in a project. The basis for snapping can be beat intervals, objects, ranges, frames or the set quantization value. You can use the grid toolbar to adjust the snap settings and switch the grid on and off.



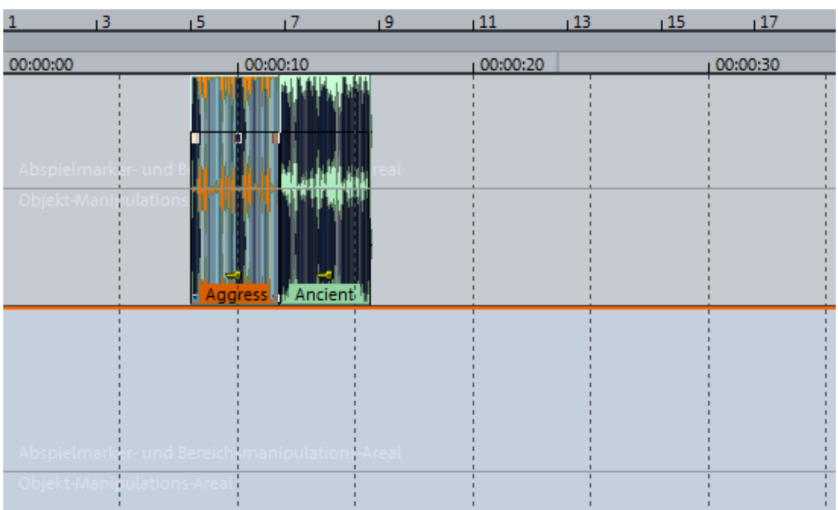
For example, the snap function is used to move objects and position them precisely on snap points in the grid.



Grid bar with the snap setting "Beat".

Grid

The grid consists of perpendicular guidelines that represent the time units appearing on the time scale for better orientation. You can select from various line types and units by right-clicking on the time scale.

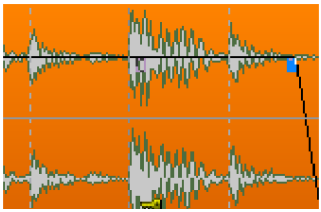


VIP with two grid bars - Grid setup based on the time unit of the lower grid toolbar.

Objects

Objects are units in the Sequoia project window that you can move, copy or cut within the track.

Audio objects contain references to source audio files that are visualized in waveform display.



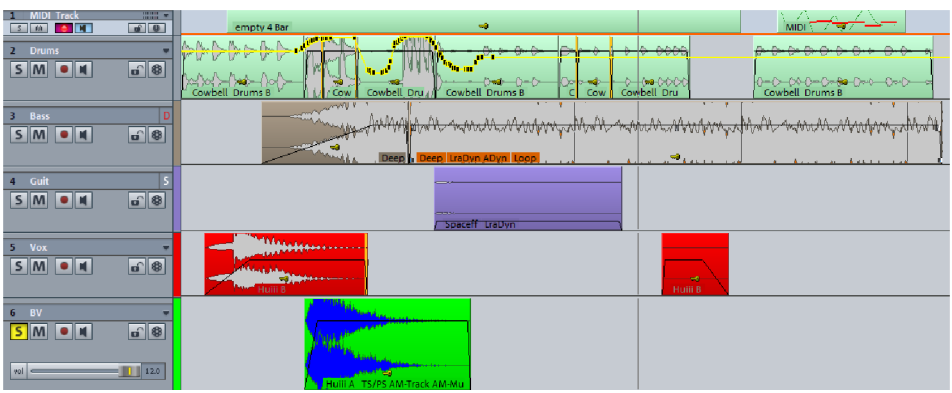
An audio object accesses audio files and calculates settings such as volume, panorama, object length, sound changes, effects, fades, pitch changes, time stretching, AUX send routing and plug-ins in realtime.

All of these settings are added to the original material afterwards - the original material remains unaffected on the hard drive (non-destructive editing).

MIDI objects contain playback instructions for MIDI devices or VST instruments in MIDI format. In contrast to audio objects, MIDI data is saved together with objects and is directly coupled to the corresponding objects.

Tracks

Multiple simultaneously played or recorded audio or MIDI objects are placed on top of each other on different tracks. Each track corresponds to a channel on the mixer (view page 363). You can use this channel to control volume, apply effects, mute or solo all of the objects in the track.



Section

"Section" refers to the visible part of a project in the project window.

There are many commands for moving (scrolling) the visible section and for customizing its size (zooming). The corresponding commands can be opened via the "View" menu, via the position bar, and via the shortcut keys.

Several Ranges

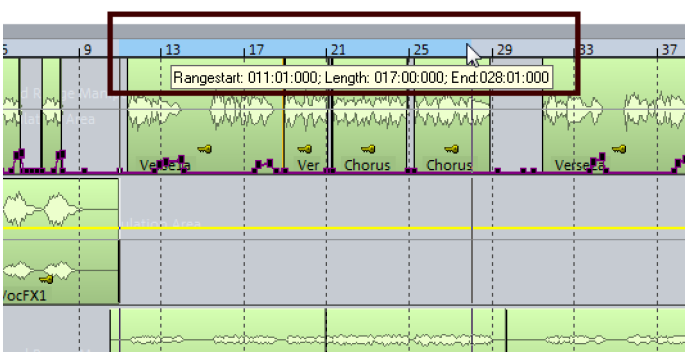
If you display your project in several ranges (menu "Edit" > "Range" . "Split range"), only one range can be active at a time. Activate a part by clicking it or clicking its controls. By clicking both double arrow buttons at the left lower border of a range, a corresponding context menu will appear that offers you different possibilities for moving the range horizontally or vertically.



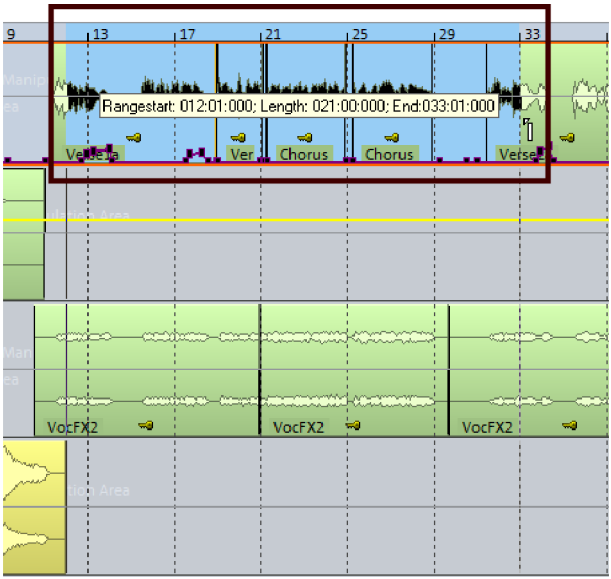
Ranges

Ranges are selected parts of an arrangement. Ranges are typically used for defining a playback range or for carrying out cut operations which are designed to cut or copy material within a given range and paste it somewhere else.

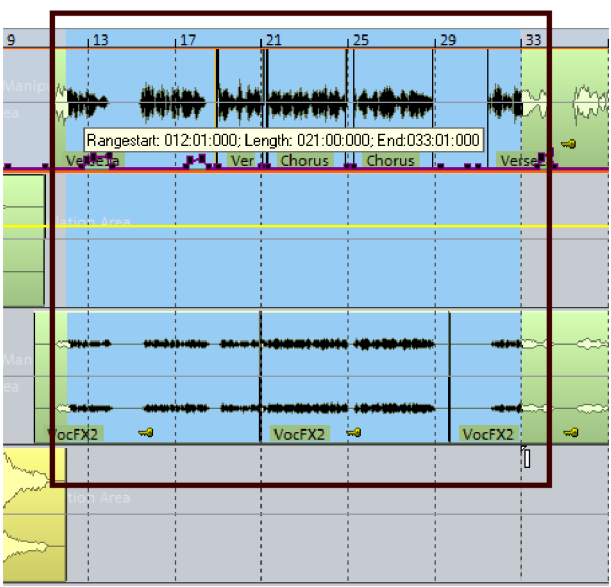
Playback ranges: Playback ranges can be selected above the first track in the grid bar via drag & drop.



Object ranges: A range can also be dragged out within an object. These ranges can then be used for editing or other tasks. The playback range in the grid bar will be dragged out too.



Object ranges can comprise several objects. To record an object lying below in a range, simply drag & drop the mouse pointer below.

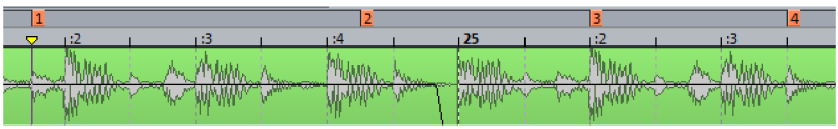


You can also use the grid feature when creating ranges. Ranges can also be saved for later use ("Edit" > "Range" > "Store Range/Get Range")

See Working with ranges (view page 122) for further details.

Markers

Use markers to select important sections in your arrangement.



When you have set a marker, you can jump directly to it later using menu command or by pressing a key.

Markers are shown in the marker bar above the grid bar. They can be set in a stopped state as well as during playback and recording.

Keyboard shortcut:

Set marker: Shift + number key or Shift + ' for the markers with the next highest number

Get marker: Number key

In addition to the standard markers, there are other markers for special functions: Audio markers, markers for CD burning functions, tempo markers, bar markers and grid position markers.

See the chapter on "Working with Markers" (view page 128) to find out more about markers.

Signal Flow

Record

During recording the record signal is routed through the ASIO buffer (ASIO only) and then recorded on the hard drive. Recording in Sequoia is designed in such a manner that it is actually not a real-time process. The input buffers coming from the sound card are collected in the buffer and passed on to the file system as larger blocks to be saved on the hard disk. Latencies are not important and the procedure is quite safe.

Playback

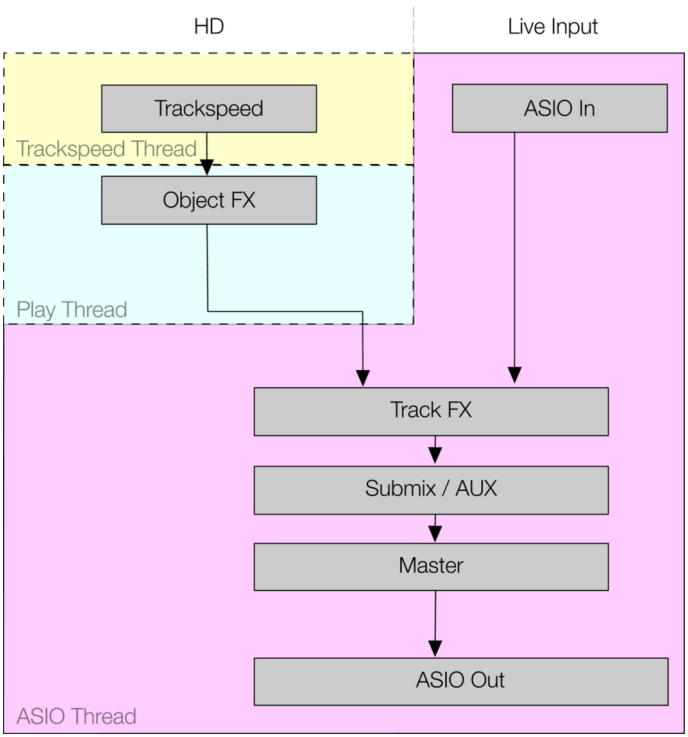
During playback, all files are read from the hard drive, and they pass through the track speed buffer and finally the effect sections. This not only ensures the allocation of

effects, but also setting of the volume and panorama as well as buses and outputs. The signals are processed in packages at the size of the VIP buffer and are finally reproduced by the sound card via the ASIO buffer (ASIO only).

Signal flow with monitoring

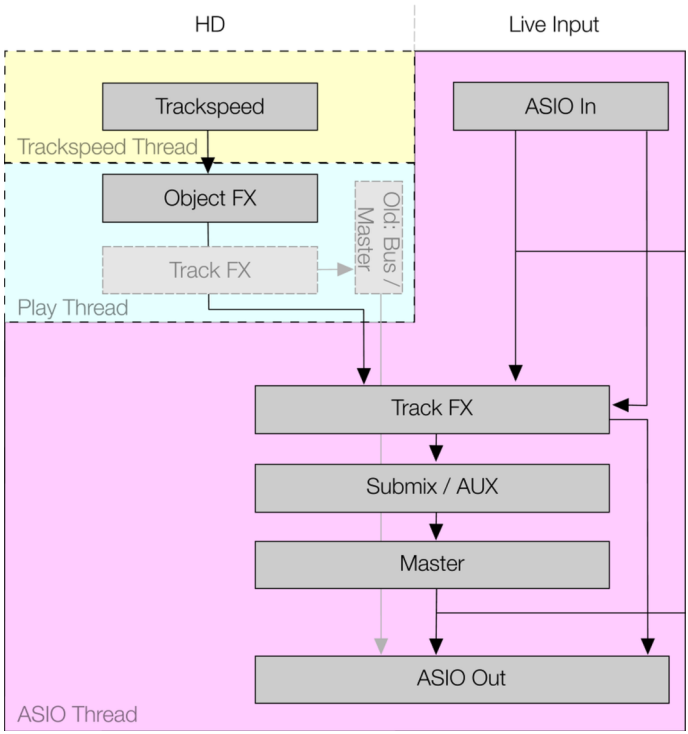
For all other models, three signal flows occur simultaneously: playback of the recorded tracks, recording of tracks to be recorded, and monitoring of the signals of the input.

Sequoia is designed to function like the sync head of an analog tape machine. Temporal offset of the individual signal currents is compensated automatically; the playback of the signal is drawn out by the latency of the effects and the buffer. This guarantees that the playback signal is not delayed vis-a-vis the recording and input signals.



Hardware monitoring

Audio monitoring via the sound card in use or via external mixing hardware. With MME drivers, this is the only monitoring type possible.



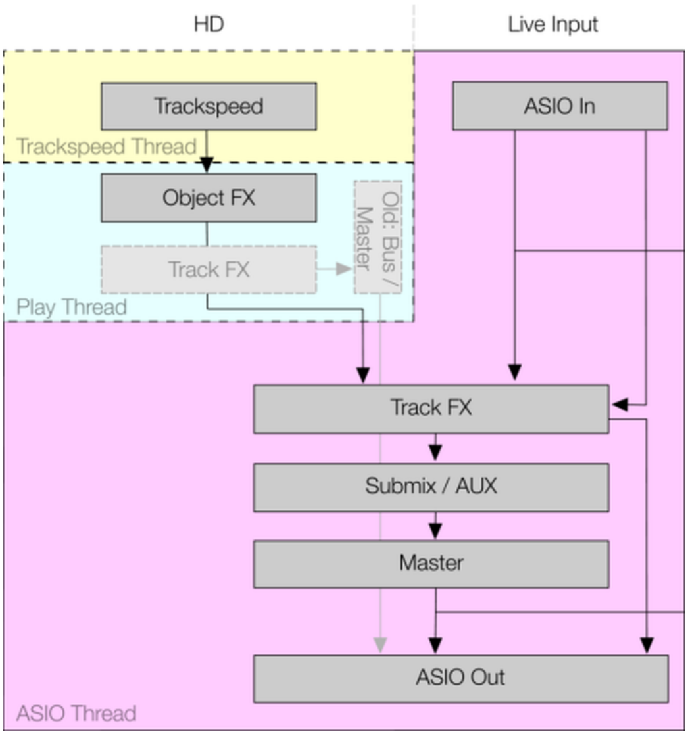
Software monitoring

Audio monitoring that includes the set recording track levels does not support effects. This monitoring option is only available when using ASIO drivers.

Whether track effects in the recording track are audible or not during software monitoring depends on whether the "Hybrid Engine" is activated/deactivated.

For more information about the Hybrid Engine, please also read the section "System options -> Monitoring settings (view page 596)".

ASIO software monitoring is essential when working with VST instruments. We recommend activating the "Hybrid Engine" for the lowest possible delay times.

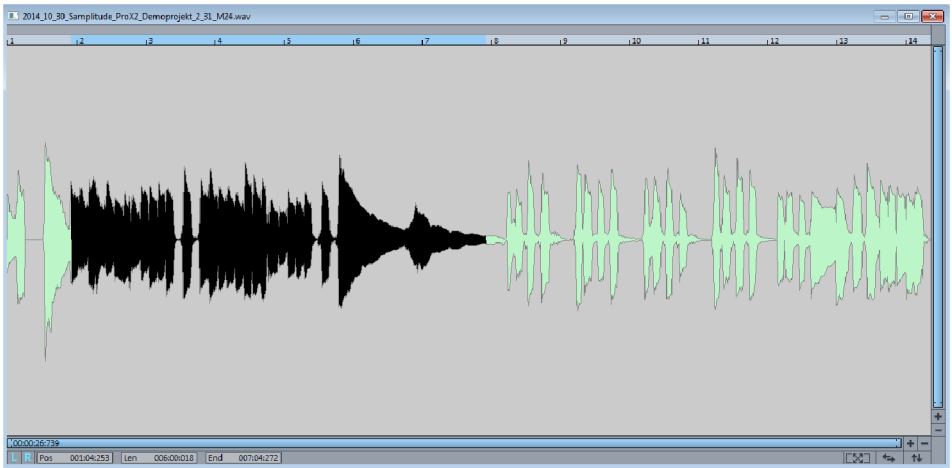


Audio file editing

Audio objects in the VIP reference corresponding audio files that are open in the background. They only reference these audio data and basically contain instructions for how these files will be processed in realtime. In Sequoia, most tasks can be carried out completely virtually in the VIP without touching the original audio material at all.

In certain cases, however, it's a good idea to make adjustments to the file material. In these cases, you are not only editing the reference objects within a virtual project, but also the audio files (.wav) on your hard drive. This type of audio editing is also called "destructive" editing.

To open an audio file from within a virtual project, position the mouse pointer at the lower half of the reference object while holding down the cmd and Shift keys and then double-click. Alternatively, you can also select the option "Wave Editing..." in the Object menu.



Audio file window: The audio file is opened in a separate window and displayed as a waveform. The function of the zoom, scroll, range and marker features does not change for the new window. 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.

Editing options: During destructive audio editing in Sequoia, you have nearly the same editing options available in the menu bar as you do when editing a VIP in real time (with the exception of realtime effects, which can only be used virtually in VIP, as well as some mouse modes such as universal, object, curve editing, object edit and pitch shift / time stretch modes).

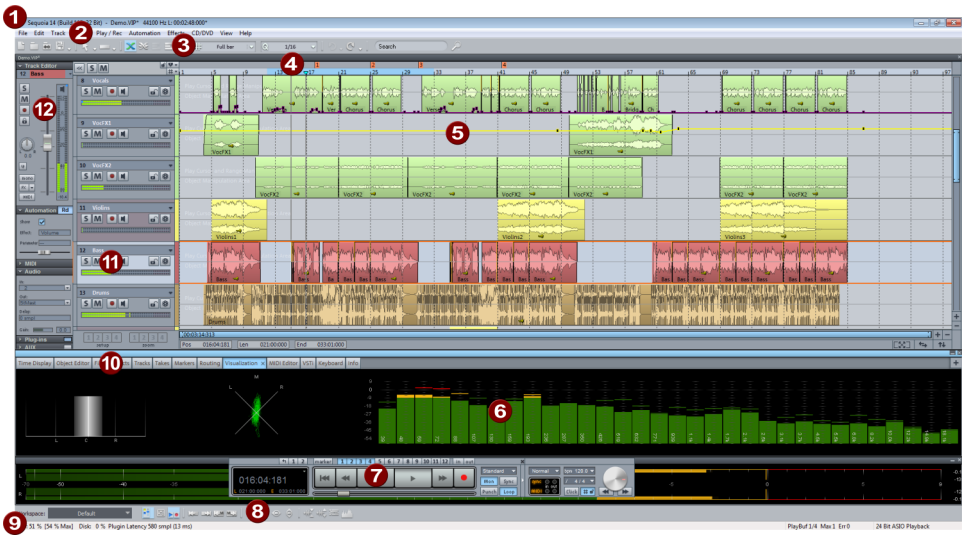
Mouse modes: The following mouse modes are available for audio editing: Range Mode, Wave Draw Mode (view page 156), Volume Draw Mode, Scrub Mode, Zoom Mode, Spectral Mode and Automation Draw Mode for realtime audio editing (view page 780).

File copies: When saving the files you've edited, you can create a copy of the original file ("File" > "Save Project Copy") so that you can access it whenever you need to by clicking "Undo". The "Object > Edit > Edit a copy of wave content..." command opens a copy of the audio file in the VIP.

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.

Screen Elements

Project Window



1. **Title bar**: The title bar is located at the top of the window. The program name can be seen here along with the name, sample rate and length of the current project.
2. **Menu bar**: The menus are located directly below the title bar. A keyboard shortcut can be assigned to each menu item.
3. **Top toolbars:** Both toolbars (view page 72) are made up of individual buttons for specific commands or basic functions. They are ordered above and below the arranger in freely configurable groups.
4. **Grid/marker bars**: The grid/marker bars are located above the first track in the project. In the upper half (if two grid bars are displayed between both grid bars), you'll find the marker bar, where you can position the markers and playback cursors. The grid bars display the project time in relation to the selected measurement unit. You can also define various ranges.
5. **Arranger**: Displays the project tracks together with the audio and MIDI objects. There are plenty of options for you to scroll (view page 116) and zoom (view page 111) through the visible section of the arranger.
6. **Visualization** (view page 1121)
7. **Transport console** (view page 65)

8. **Bottom toolbars:** These toolbars are for activating certain windows, for setting the play cursor and switching the view.
9. **Status display**: The status display appears at the bottom border of the VIP window. Here you'll find information about CPU load, latency, buffer, and current operations like loading, saving, effect calculation, etc. You can also open the status display by going to "View" > "Toolbars" > "Status Display".
10. **Manager/Docker** (view page 177)
11. **Track header:** The track header includes important control elements such as mute and solo function and recording next to the track names. At higher vertical zoom levels, some elements are faded out. More information about the individual controls of the track header can be found under "Track header" (view page 71).
12. **Track Editor**. The Track Editor is arranged far left next to the track headers. This lets you access all important settings for the selected track. More information about the individual controls can be found under "Track Editor" (view page 67).

Transport console

Shortcut: Ctrl + Shift + T



The transport console contains the most important commands for playback, recording, and positioning.

1. **Ranges:** Use the buttons "1" and "2" to save ranges. The arrow symbol can be used to open previously saved ranges.
2. **Marker:** Opens the Marker manager (view page 196).
3. **Marker 1-12:** The current playback position can be saved to one of the 12 marker buttons by clicking on it. If one position has been saved, the marker will appear colored. Another click on the same marker moves the playback position to the marker. You can delete a marker by right-clicking on it.
4. **Punch In/Out:** Sets the in/out point for a punch recording (view page 107). Additional Punch In/Out markers can be set by holding down the Alt key.

5 **Recording modes:** Here you can set up the appropriate recording mode. The following options are available:

Standard mode (playback while recording): This is the typical recording mode for multitrack productions. Here, the currently active tracks are recorded. The already existing tracks are played back in sync.

Recording without playback (read after write): If you have selected this option, playback of already existing audio is deactivated and the playback marker will remain at the recording start position. Playback can be started manually, e. g. if "read after write" is required. Set the playback marker at a different position and start playback by pressing the "Play" button. This will not interrupt the recording. This way, changes to previously recorded material can be made without affecting the recording process.

Punch marker mode: In this mode recording takes place only between punch start markers and punch end markers.

6 **MIDI record modes**: Select from the following MIDI record modes: normal, overdub, multi-overdub, and replace (view page 99).

7 **Tempo/Beat section:** You can change the playback speed and the tempo of the arrangement here. Adjust (view page 441) all objects to the current speed value by moving or stretching them.

8 **Scrub Control:** Using the Scrub Control wheel you can variably set playback speed. This is helpful when searching for specific audio passages. Right-clicking opens the scrubbing and Varispeed options (view page 118).

9 **Arrows**: The arrow buttons below can be used to start playback forwards and backwards at a slower speed. This is very practical for locating clicks, pops and other errors in the material.

10 **Grid**: You can activate the beat grid and have it displayed (view page 74) here.

11 **Click:** Use to switch on the metronome click. A right-click opens the metronome options window (view page 95) to configure the pre-counter and click volume.

12 **Loop**: Switch to Loop mode here (view page 105).

13 **Sync**: This button opens a dialog with the Synchronization options (view page 505).

14 **Mon:** Activates monitoring. All tracks with activated recording display the incoming signal on the peak meter. Right-clicking on the "Mon" button opens a selection of monitoring modes. You can find out more by reading "Monitoring settings" (view page 596).

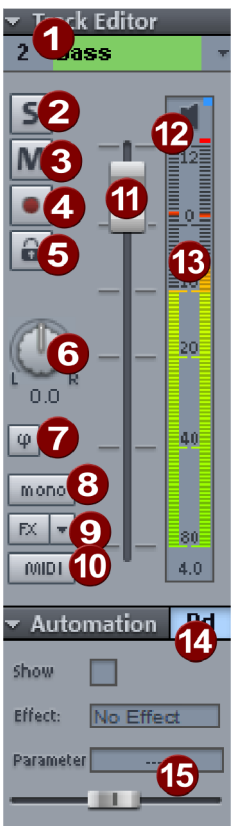
15 **Punch:** Switches Sequoia to punch marker mode (view page 107).

16 **Time display:** Displays the playback position. The unit of measurement can be selected by clicking the small triangle.

Track editor

The track editor is located on the left border of the arrangement window and enables quick access to all of the most important track parameters for the selected track. Record and monitoring status, volume, panorama, MIDI/audio in and outputs, plug-ins, AUX sends, and EQ settings are displayed in well-arranged sections and can also be edited in this view. All relevant settings for the corresponding track can also be viewed without opening the mixer or the track view in the arranger window.

You can open the track editor using the corresponding button in toolbar below or through "View" > "Track Editor" (Shortcut: Ctrl + Alt + Shift + E).



1. **Track number and track name:** Double-clicking on the track names allows them to be edited. Right-clicking the track name opens the "Track Options" dialog.
2. **The "S" button** mutes all tracks except the selected one (solo function). Detailed information can be found in "Global solo modes" (view page 382).
3. **The "M" button** mutes the active track (mute function).

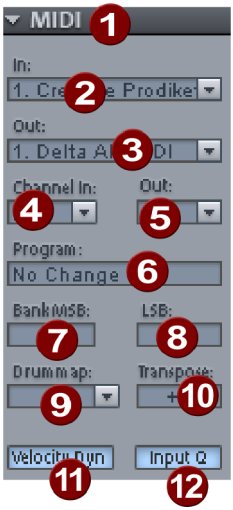
4. **The "Rec" button** activates the track for the recording.
5. **The "Lock" button** prevents objects in the track from being accidentally moved or deleted.
6. **The "Panorama" knob** controls the position of the channel within the mix. Right-clicking on this opens the Stereo panorama dialog/Stereo Editor (view page 392) where you can make adjustments to the stereo image, stereo width and the phasing.
7. **The "Phase reverse" button** reverses the signal phase 180 degrees. Right-clicking on this button also opens the Stereo panorama dialog/Stereo Editor (view page 392).
8. **The "Mono" button** switches the track to mono processing from the input up to the pan controller. In particular, all track effects preceding the pan controller operate in mono, which saves considerable CPU resources. However, the submix and AUX return buses always remain stereo.

   Here too, right-clicking on the button opens the Stereo panorama dialog/Stereo Editor (view page 392).

   If stereo objects are located in mono tracks, the mono share (L+R) is generally played. However, this can be changed via the "Mono side channel processing" preset.

   For standard routing, only the post DX/VST plug-ins and post AUX sends are arranged after the panorama controller and are therefore calculated as a stereo signal. The routing position of the pan controller can be freely adjusted in the "FX routing" dialog.
9. **The "FX" button** opens the dialog for specifying the effect sequences (view page 388) and adding VST plug-ins; right-clicking this button lets you access the track effects settings for the active track.
10. **The "MIDI" button** switches the track to MIDI recording and opens the MIDI section (view page 69) of the track editor.
11. **Volume input box** and **volume controller**.
12. The **loudspeaker symbol** switches on monitoring (view page 602), i.e. the playback of incoming signals when the track record button has been activated. "MIDI thru" will be active here for MIDI tracks.
13. **Level control display:** Both LED rows display the input and output signal for the track.
14. **The "Automation" button** activates automation (view page 457) for the track.
15. **Automation parameter selection field** and **automation control**: Select automation parameters and adjust the values with the corresponding controllers.

MIDI



1. **Arrow**: Opens and closes the respective dialog box.
2. **In**: Opens the MIDI input menu.
3. **Out**: Opens the MIDI output menu.
4. **Channel In**: Set the MIDI input channel here.
5. **Channel Out**: Set the MIDI output channel here.
6. **Program**: This slot is used for choosing the MIDI instrument. The first click activates this field and the second mouse click opens the program selection. When the menu is open, use the arrow and page up/down keys to make your selection.
7. **Bank MSB**: Here you can set the MSB (Most Significant Byte) for the "Bank Select" MIDI message that controls your external instrument.
8. **LSB**: Here you can set the LSB (Least Significant Byte) for the "Bank Select" MIDI message that controls your external instrument. The bank number is MSB value* 128 + LSB value.
9. **Drum map:** Here you can select a drum map (view page 325) for allocating MIDI notes to the device-specific sound. If a drum map is selected, the transpose function will not be available because the pitch is firmly assigned by the drum map.
10. **Transpose**: You can transpose the notes up or down here. This function works in realtime meaning that the MIDI files in the MIDI objects will not be changed.

11 **Velocity dyn**: This button activates velocity dynamics as a realtime track effect.

> Detailed information about this **MIDI effect**, which processes the MIDI velocity dynamics, can be found in "MIDI editors -> MIDI functions -> Velocity dynamics (view page 309)".

12 **Input Q:** In this case, the MIDI quantize settings (view page 313) are used to quantize every recording offline. The original position can be restored by going to "Object -> Quantize -> Reset Quantization".

Audio



1 **The "In" slot** determines the audio device, e. g. your sound card input.

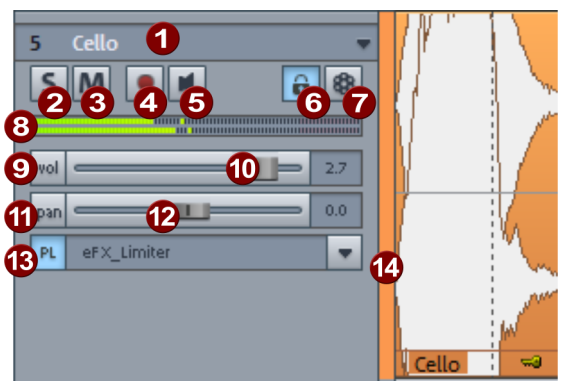
2 The **"Out" slot** determines the audio output device. For example, this might be a sound card output.

3 **Delay**: This sets the delay value for the track.

4 **Gain**: This regulates the amplification level of the input signal in dB.

5. **Plug-ins**: Access track effect inserts, MAGIX plug-ins and VST effects here. Clicking on the empty insert slot opens the plug-in browser (view page 222), where you can load a plug-in to the slot. Click on an occupied slot to activate/deactivate the plug-in. Right-clicking on the slot opens the interface of the plug-in. The arrow next to the respective insert slot opens a menu with various functions. You can ope the plug-in browser again to switch out the plug-in or remove it ("No effect"). The "Plug-ins" button activates or deactivates all effects in the channel. A visual indicator (*) is displayed for plug-ins that were previously active and are active the next time "Plug-ins" button is pressed.
6. **AUX**: Here you can specify the input of the AUX sends (view page 383) or right-click to switch to the output assignments (More information can be found in the section "Assign track to multiple outputs" on page 371) or sidechain sends (view page 387) display.
7. **EQ:** This contains the parametric EQ for the track. Right-click to open an advanced input window with six frequency bands.
8. **Comments**: Section where you can quickly add notes.

Track Header



The higher the vertical zoom level, the fewer options the track header will display. In order to see all options, you need to zoom (view page 111) into the display.

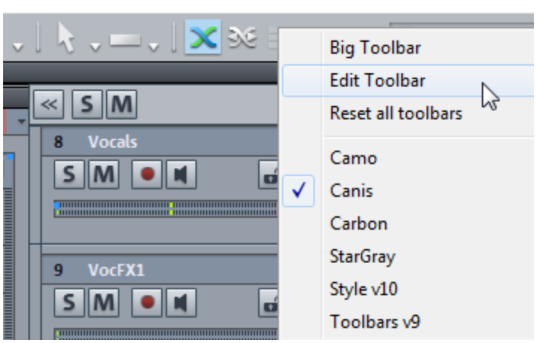
1. **Track number and name:** Clicking the track name or the track number selects the track. Repeatedly clicking the track number causes the track's display height to switch between flat and high. Double-clicking on the track names lets you edit them. Right-clicking the track name opens the "Track Options" dialog. Click the arrow to select additional track-relevant functions from the context menu.
2. **The "S" button** (view page 382) mutes all tracks except the selected one (solo).
3. **The "M" button** mutes the active track (mute).

4. **Record**: This button (view page 93) activates the track for recording.
5. **The loudspeaker symbol** activates monitoring (view page 596).
6. **The lock button** prevents the objects in the track from being accidentally moved or deleted.
7. **Revolver tracks**: Revolver tracks (view page 135) can be used to compile objects differently for each individual track.
8. **Peak Meter:** Both LED rows display the input and output signal for the track.
9. **Vol:** This button switches on the volume automation curve (view page 457). This way, level adjustments in the track can be controlled via an automation curve and drawn by moving the channel fader.
10. **Volume** controller
11. **Pan**: Switch on panorama automation with this button.
12. **Panorama** control (view page 392).
13. **PL:** The plug-in selector (view page 219) assigns insert effects or VST plug-ins to a track.
14. **Color selection**: If you click on the right edge of a track header, a selection menu will appear for specifying the color (view page 88) of the track and its objects.
15. **Automation tracks:** Use the small triangle to fade in or out the automation tracks.

Toolbars

Toolbars offer quick access to the most important functions in the program.

Right-clicking on a buttonenables you to swap out interfaces for the toolbars, increase the size of the buttons and customize (view page 91) the toolbars.

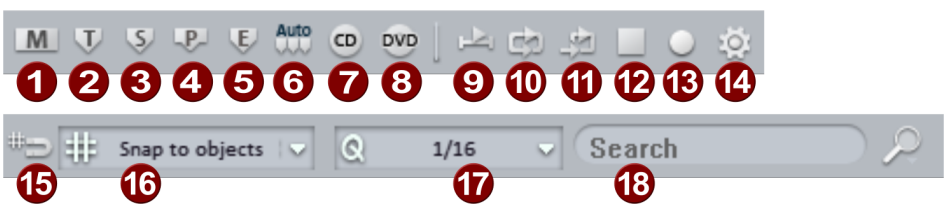


The following overview shows the presets in the "Power User" workspace:

Upper toolbar



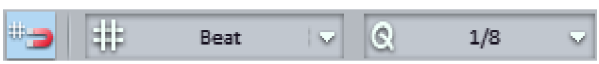
1. New Virtual Project
2. Load VIP
3. Load audio file
4. Export/Save
5. Mouse mode
6. Object mode (view page 89)
7. Auto Crossfade On/Off
8. Crossfade Editor (view page 425)
9. Group
10. Ungroup
11. Undo
12. Redo



1. Set Marker
2. Set CD title index
3. Set CD sub index
4. Set CD Pause Index
5. Set CD End index
6. Auto CD Track Indices
7. Create CD
8. Create DVD Audio

9. Play Once
10. Play Loop
11. Play into Loop
12. Stop
13. Record
14. Recording Options
15. Grid On/Off
16. Grid menu (see below)
17. Quantization menu (see below)
18. Search for menu commands and help topics (view page 75)

Grid and Quantization menu



You'll find the grid and quantization menu in the toolbar. The grid menu lets you change important grid settings without always having to open project options (view page 607). Click the magnet button to activate the grid. The button beside it opens the grid menu with the following grid settings:

Snap to objects: This option activates the object grid. This lets objects snap exactly to the edges of other objects.

Snap to range: This option activates the range grid and lets you use the current range as the basis for the grid.

Snap to beats/bars: This option activates a grid, which uses beat settings as its basis.

Snap to beats/bars (relative): Activates a grid with beats as the basis for snapping. The selected object will maintain a relative distance to the snap markings when moved.

Snap to frames: This option activates a frame-based grid.

Snap to frames (relative): This option also activates a frame-based grid. The selected object will maintain a relative distance to the snap markings when moved.

Here you must also specify the snap length (for beat/bar snapping only). The values range from "every 4 bars", "every 2 bars", "full bar", "beat" to "Snap to Quantization". The "Beat" setting means that signatures based on quarter notes serve as a quarter

snap unit and time signatures based on eighth notes serve as an eighth snap unit. The snap therefore follows the time signature and observes any time signature changes.

Warning: When the grid is active, the playback marker jumps according to the set grid value when forward and rewind buttons are used in the transport console. If you want to temporarily shut off the transport controls, press the "Alt" key simultaneously.

Activate grid: Here you can globally turn the grid on or off.

Grid and snap settings: This option brings you to the **Project options - General dialog** (view page 607), where you can change additional grid and snap settings.

The field marked with the "Q" is where you open the quantization menu. Here you can specify the quantization value for MIDI (view page 313) and audio quantization (view page 217). The "Snap to Quantization" option sets the snap interval accordingly. This links the audio and MIDI quantization (view page 606). Right-clicking on the field opens the MIDI quantization settings.

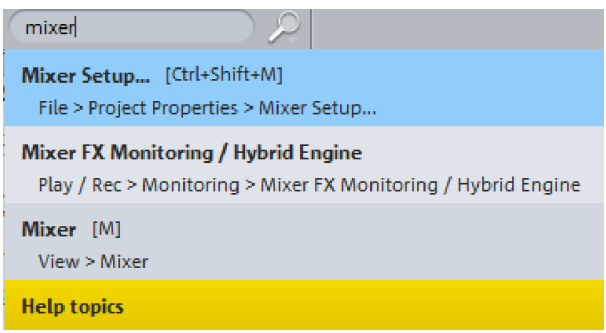
Note on "Input Q": In the MIDI section of the track editor beside the button for velocity dynamics you will find the "Input Q" button. If this function is active, recorded MIDI notes will be quantized immediately according to the current settings. The original position can be restored by going to "Object" > "Quantize" > "Reset Quantization".

Searching for menu commands and help topics

Sequoia provides a search field for finding menu commands and help topics.



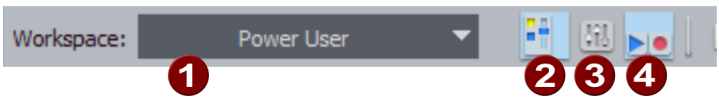
Enter a character combination into the search field which is relevant to a search term. Immediately up to five commands from the main menu will be displayed that contain the desired character combination.



All results that are listed can be directly selected. The commands in the upper section are listed immediately.

Shortcut: Ctrl + F

Lower toolbar



1. The Workspace (view page 109)
2. Track Editor
3. Open Mixer
4. Display transport console

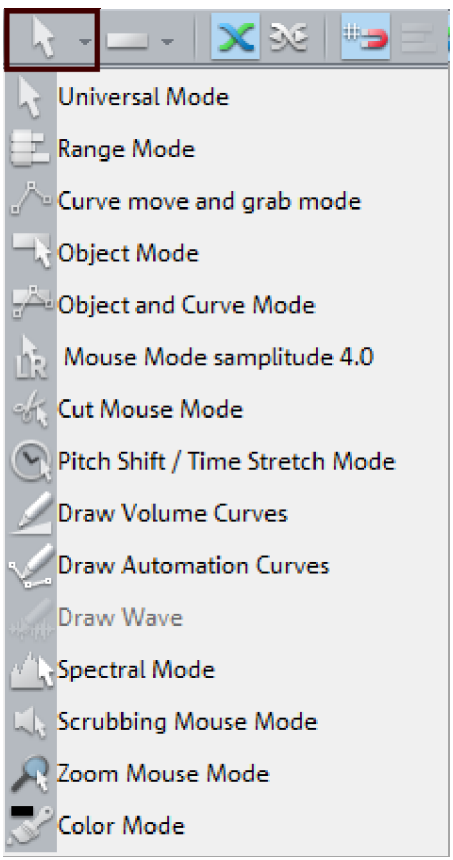


1. Section to start
2. One section left
3. Half section left
4. Half section right
5. One section right
6. Section to end
7. Playback marker to previous object edge
8. Playback marker to next object edge
9. Playback marker to previous marker
10. Playback marker to next marker
11. Show all
12. Show all vertical
13. Zoom to range
14. Section as zoom step



1. Zoom to predefined zoom levels 1s, 10s, 60s, 10min
2. Zoom to freely definable zoom levels 1..4. Shift+click on the icon saves current zoom value.
3. Zoom out of waveform
4. Zoom into waveform
5. Overview mode
6. Spectral Display
7. Waveform display

Mouse mode



The various mouse modes can also be accessed through "Edit" > "Mouse Mode". To open the selection menu for various mouse modes at a location in the project window, you can:

- Click and hold the right mouse button and then click the left mouse button
- Shift + click right mouse button
- Shift + click the context menu key (next to the right Ctrl+key).

Universal mode

This is the preset mouse mode for Sequoia. Different functions can be used depending on the relative position within a track. Right-clicking opens a context menu.

The vertical mouse position within the track serves to differentiate between object handling and area manipulation: In the upper half you can mark areas and set the playback marker. In the lower half, objects can be selected and moved.

Clicking makes the following mouse functions available to you:

Mouse functions for the upper half of the track:

- Click and drag without existing range or outside of existing range – Stretching a range
- Click and drag within existing range – Moving range borders
- Shift + click – Stretching a range between play cursor and click position
- Shift + click outside of existing range – Extending range to click position
- Shift + click + drag within existing range – Moving range
- Click: Set play cursor

Note. If you want to keep an existing range, move the play cursor by clicking above on the timeline, not in the track.

Mouse functions for the lower half of the track:

Selecting objects

- Selects individual objects by clicking
- Multiple objects can be selected by holding down the Ctrl key and clicking on them.
- All objects between two selected objects can be selected by holding down the Shift key and clicking on them
- Stretches a rectangle by dragging the mouse pointer to the right. All objects touched by the lasso are marked (see "Lasso function" (view page 79))

Moving objects

- Moves an object (or a group of objects) by dragging to the desired position

Holding down the Shift key means that when moving onto a different track, the horizontal position remains the same.

Holding down the K key means that the objects behind the object are also moved backward together with the object.

Duplicating objects

- Duplicates one or more objects by dragging while holding down the Ctrl key

Holding down the Shift key means that when duplicating the selected objects on another track, their position remains the same.

Lasso function

Object lasso: Stretches a rectangle by dragging the mouse pointer to the right. All objects touched by the lasso are marked and can be moved together.

Curve point lasso: Stretches a rectangle by dragging the mouse pointer to the left over an activated automation curve. All automation points touched by the lasso are marked and can be moved together.

Setting object volume and length with the object handles

You can adjust the volume and length of individual objects using the five object handles.

Middle handle (top): Changes the object volume. The exact value in dB is displayed in the tooltips.

Side handles (top): Set a fade-in or fade-out for the object. The fade curves can be edited in the object editor.

Lower handles: Sets the start and end position of the object.

Editing automation curves in universal mouse mode

For easy editing of automation curves, simply use universal mouse mode instead of switching to mouse mode:

Creating and deleting handle points: Double-clicking on the automation curve creates a new point; double-clicking it again deletes it. Selected points can also be deleted by pressing the "Del" key.

Selecting handle points: A handle point is selected by simply clicking on it. Select several points by holding down the Ctrl key when clicking. To select several handle points in sequence, click the first and last points while holding down the Shift key.

Moving a handle point: Handle points can be moved simply by dragging them with the mouse pointer.

- If you want to change the vertical positions of the curve points – that is, their values, not their positions – hold the **shift key** while moving them.
- Holding the **Alt key** in addition to this will only move the curve points horizontally. (So you'll be changing their positions and not their values).

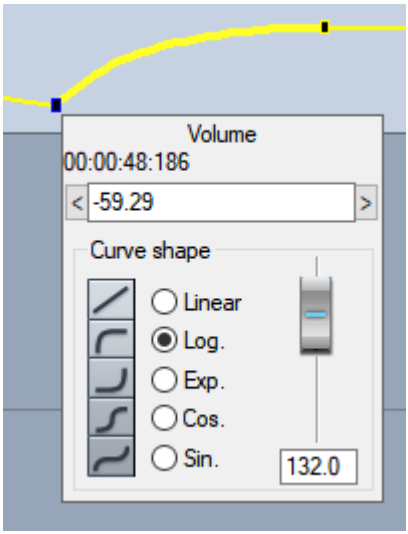
Moving curve segments: If you move the mouse pointer over a curve (the mouse pointer becomes a double arrow) and drag on the curve, both handles to the right and left of the mouse are selected and moved together.

Moving curves in a range: If a range has been selected and you drag on the curve, the entire curve and all its points will be moved inside the range edges. This creates two new curve points on the range edges.

Automation points lasso: Clicking and dragging next to the object activates the object lasso (view page 79) for selecting multiple automation points.

Right-click on a curve point: Numerical editing of the value and definition of the curve shape

A numerical entry field will appear where you can enter the values you want, or use the mouse wheel to change the value in 0.1 dB increments. Holding the shift key while doing this will adjust the value in increments of 0.01 dB. Holding down the "Ctrl" key, in turn, will change the value 1 dB at a time.



Close the entry field by pressing the Enter key. When the input field is open, you can jump to the next automation point using the Tab key.

Under Waveform you can define how the curve runs from the selected curve point to the next. The possible curve shapes correspond to the curves for objects fade in and out (view page 172). Using the fader you can adjust the precise settings of the curve form of the fade.

Range mode (safe mode)

Right mouse button: Displays context-sensitive popup menu.

Left mouse button: Selecting ranges and moving ranges ("Shift"). Objects or automation curves cannot be changed by accident (i.e. "Safe mode").

Create ranges

In range mode, ranges are sketched by clicking and dragging the mouse pointer while holding the mouse button.

The range mode is a "safe" mode, since objects or curves cannot be moved accidentally.

Setting playback markers

In range mode the start position of the playback marker is set by clicking in the arranger.

Zoom

Clicking in the arranger and holding down the Z key increases the horizontal zoom step while right-clicking and holding down the Z key reduces the horizontal zoom level.

Temporary switch to other mouse modes

Period key "." - Press this button to temporarily switch Sequoia to "Object" mode. You can select and move objects by holding down the key.

Hyphen key "-" - Press this button to switch Sequoia temporarily to "Curve" mode. You can edit volume or panorama curves by holding down the key.

These two special functions ensure you can reach basic editing functions quickly.

Curve mode

This mode is strictly for editing automation curves (view page 457). It can be opened temporarily from "Universal/Range" mode by pressing and holding down the hyphen ("-") key.

Set a new curve point on an existing automation curve by clicking. Curve points that have already been set can be moved easily using drag & drop.

If you want to simultaneously select several curve points for editing, simply span a range in the desired length. This is similar to using the object lasso, but for curve points. You can also select curve points vertically over multiple tracks.

Left mouse button:

Dragging to the right or left: Activates the curve point lasso function for selecting multiple automation points.

Clicking once on a curve point: Creates a new point; a double-click deletes it again. Another option for deleting curve points is to select them and press the "Del" key.

Clicking on a point: Selects a point.

Click on a point + "Ctrl": Allows selection of multiple points.

Click on a point + "Shift": Multiple selection of start and end points including all curve events in between.

Dragging a selected point: Moves the point.

Dragging a selected point + Ctrl: Moves multiple selected points.

Right mouse button: Context menu

Object Mode

This mode allows you to move objects and edit their start & end positions, the fade-in & fade-out phases, and the object volume.

Object mode is especially useful together with range mode, which can be accessed temporarily from within it. Press and hold the period key "." while using range mode. Once you let it go, Sequoia switches back to range mode.

Left mouse button:

Click: Selects objects.

Click + Shift: Selects two objects, including all objects in between the two objects.

Click + Ctrl: Selects multiple objects.

Dragging objects: Moves objects in the set grid steps.

Drag + Shift: Moves an object (or a group of objects) onto a different track; the horizontal position remains the same.

Drag + Ctrl: Duplicates one or several objects.

Drag + Shift + Ctrl: Duplicates one or more objects onto a different track without changing the time position.

Double-click on objects: The object editor opens.

Lasso function: Clicking next to an object while dragging to the right activates the object lasso to select multiple objects. Clicking next to the object and dragging to the left activates the curve point lasso for selecting multiple automation points.

Object handles:

Mid handles (top object edge): Changes the object volume. The exact value in dB is displayed in the tooltips.

Side handles (top object corners): Fade in or fade out of the object can be set here. The fade curves can be edited in the object editor.

Lower handles: Sets the start and end position of the object.

Object/Curve mode

This mode is a combination of Object mode and Curve mode. In this mode, you can move objects and edit curves. The playback marker and ranges are set in the bar ruler above the first track.

Left mouse button:

Click: Selects objects.

Click + Shift: Selects two objects, including all objects in between the two objects.

Click + Ctrl: Selects multiple objects.

Dragging objects: Moves objects in the set grid steps.

Drag + Shift: Moves an object (or a group of objects) onto a different track; the horizontal position remains the same.

Drag + Ctrl: Duplicates one or several objects.

Drag + Shift + Ctrl: Duplicates one or more objects onto a different track without changing the time position.

Double-click on objects: The object editor opens.

Lasso function: Clicking next to an object while dragging to the right activates the object lasso to select multiple objects. Clicking next to the object and dragging to the left activates the curve point lasso for selecting multiple automation points.

Object handles:

Mid handles (top object edge): Changes the object volume. The exact value in dB is displayed in the tooltips.

Side handles (top object corners): Fade in or fade out of the object can be set here. The fade curves can be edited in the object editor.

Lower handles: Sets the start and end position of the object.

Clicking on the curve: Creates a new point; a double-click deletes it. Another option for deleting curve points is to select them and press the "Del" key.

Clicking on a point: Selects a point.

Clicking on a point + Ctrl: Allows the selection of multiple points.

Dragging a point: Moves the selected point.

Dragging a point + Ctrl: Moves several selected points.

Right mouse button: Context menu

Left/Right Mode

Use this function to switch to the Left/Right Mode. The right mouse button then controls object functions and the left mouse button controls range manipulations.

Left mouse button:

Double-click on automation curves: Creates a new curve point; another double-click on the same curve point deletes it again.

Right mouse button:

Click: Selects objects.

Click+Shift: Selects multiple objects.

Dragging objects: Moves objects in the set grid steps.

Drag + Shift: Moves objects onto a different track, the horizontal position remains the same.

Duplicating objects: Dragging + Ctrl duplicates one or more objects.

Dragging + Shift + Ctrl: Duplicates one or more selected objects on a different track; the horizontal position remains the same.

Clicking on an automation point: An automation point is selected

Clicking on an automation point + Shift: Allows multiple selection of automation points

Dragging selected automation points: Moves selected automation points

Cut Mode

Left mouse button: Click on the object to separate it at the corresponding position. If the grid is switched on, the scissors tool which appears will move according to the grid settings.

Under "System Options > Keyboard, Menu & Mouse > Special Keys > Temporary key for cut mode (view page 688)", you can specify a keyboard shortcut to keep "Object Cut" mode active provided the shortcut key is held down.

Right mouse button: Context menu

Pitch shift/Timestretch mode

In this mode you can quickly and effectively carry out pitch and time adjustments on the object. The function of the object handles differs from that of the other mouse modes:

Mid handles – Pitch factor: The object pitch can be changed in the range by +/- 6 semitones. The process used can be selected in the object editor.

Lower right handle – time stretch factor: This handle controls the change in length of an object through time stretching. The same duration (starting length of the object in the audio file) can be extended or shortened by time stretching. Select the stretching mode used in the object editor.

This means that the object handles on the lower right and those in the object center can be used for directly setting the playback speed and pitch.

The time stretch mouse mode simplifies working with tempo markers in the timeline. Create BPM markers (tempo changes) by directly clicking the desired playback

position and holding down the Shift key. You can adapt the tempo "in one go" by moving the mouse vertically.

You can create grid position markers in "Time stretch" mouse mode by clicking the desired playback position and holding down the "Alt" key. You can move the grid with a horizontal mouse movement to manipulate the snap grid and adjust it to existing MIDI or audio objects.

Right mouse button: Context menu

Volume Draw Mode

This mode allows volume automation curves (view page 457) to be drawn with the left mouse button.

Automation draw mode

In order to draw an automation curve or a MIDI controller curve, select "Draw automation curve" mode.

Detailed information about automation is available in the chapter Automation (view page 457).

Left mouse button: Freehand draw function for volume curves.

Shift +
Left mouse button: Click and drag to draw an individual curve point

Right mouse button: Context menu

Wave draw mode

If you are in audio editing mode (view page 62), you can edit the waveform of a file using the pencil tool (view page 156). The waveform display shows a suitable zoom level and the mouse pointer becomes a pencil. Changing the waveform in the wave window is useful if you want to remove very short impulse disturbances manually.

Spectral Mode

With the "Spectral Mode" you can remove noise from an object. You can also edit the signal of the left and right channels individually too. Editing can be done directly in the arranger window.

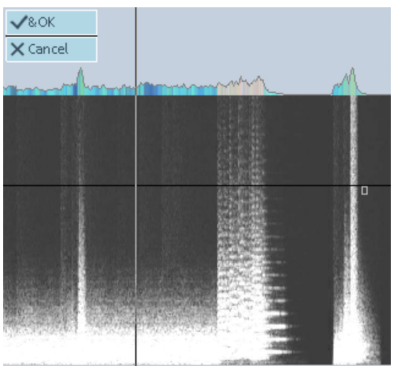
To work with "Spectral Mode" effectively activate the Spectral View in View Options.

The graphical representation of the music is displayed in a spectrogram. This displays the frequency proportions in a time curve. The volume of frequencies is visualized by a color code or by its brightness.

Audible distortion noises are usually louder than the wanted signal and limited to a certain frequency spectrum. They are highlighted with colors in the spectrogram. This allows you to easily select the interference with the mouse and remove it.

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.

After activating "Spectral Mode" highlight the unwanted noise by drawing a rectangle around it using the mouse. You can use the handle to make fine adjustments to the selected area.



To prevent an audible gap, removed components of the original frequency spectrum from the wanted signal that surrounds the distortion are recalculated into the recording. After making the correction, you can see the results in the wave/spectral display in the arranger window.

Left mouse button: Marks the selection box

Right mouse button: Context menu

This lets you edit the left and right channels of the signal separately by using the selection rectangle to select only the upper or lower part of the spectrogram.

Scrubbing Mode

This mouse mode enables you to monitor and control the playback speed (view page 118). In this case, the project can be played back in a forward or reverse scrubbing direction.

Left mouse button: Clicking the project activates monitoring with control of the playback speed.

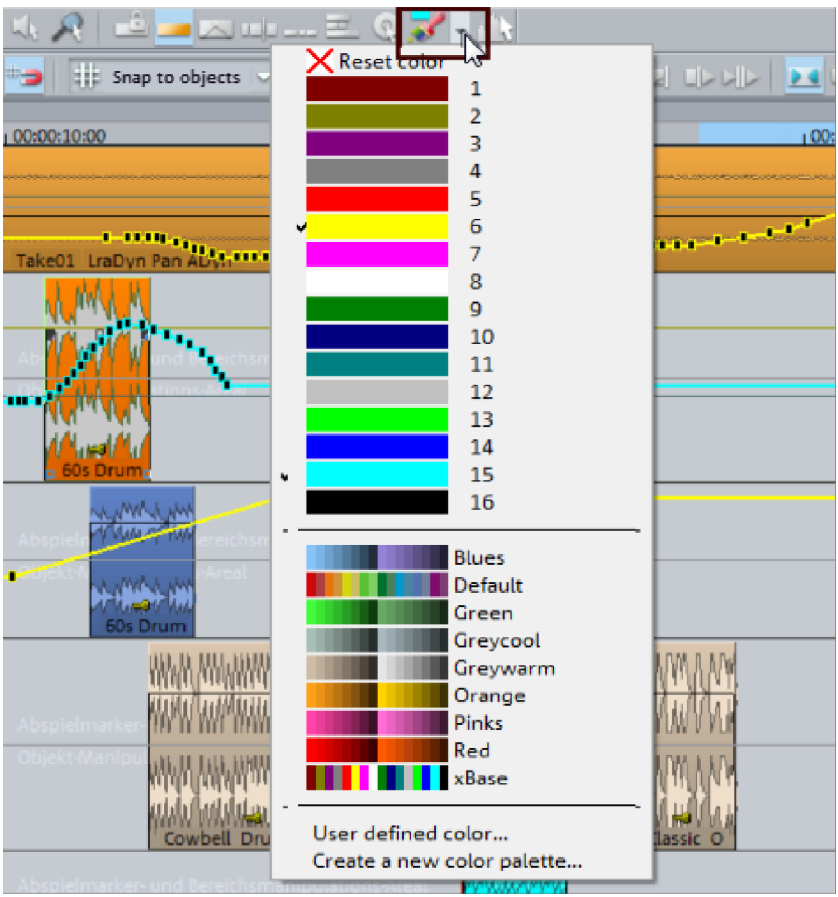
Right mouse button: Context menu

Zoom Mode

Use the right mouse button to zoom out of the project and the left button to zoom into it (view page 111).

Color mode

In Color Mode you can color objects or entire tracks. Click on the arrow next to the brush symbol first. Next, select the desired color. Now move the mouse over the objects and tracks in the arranger: the pointer turns into to a paint bucket. Click on the object or track box you would like to color.



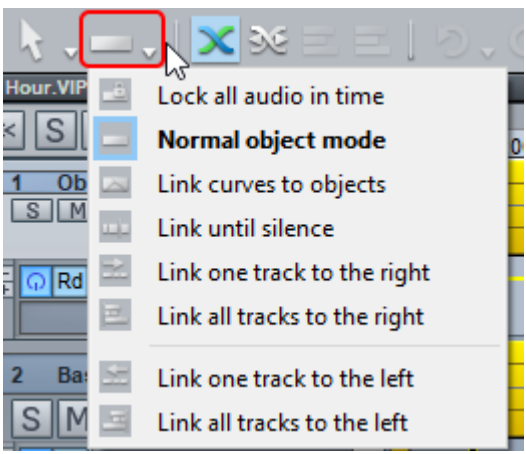
- You may also apply color to multiple objects across different tracks by dragging the lasso over the desired objects (the lasso appears when you drag the paint bucket). Release to color all selected objects.
- If you hold down the "Shift" key while doing this, you will color the wave form instead of the object.
- If you hold down the "Ctrl" key while doing this, you will color the object background instead of the object itself.
- If the preset color selection is not sufficient, you can also select your own color or even create your own personal color palette.
- You also have the option of accessing the color palette for each track by clicking the color selector on the right edge of the track header. This also allows a new color to be assigned to a track. If multiple tracks are selected (view page 130) beforehand, the color selection will apply to all of these.

Left mouse button: Changes the background color of the object

Right mouse button: Changes the wave form color of the object.

Object modes

The object mode determines the behavior of objects while moving.



The various object modes can also be accessed through "Edit" > "Object mode".

Normal object mode

In this mode you move objects one by one without affecting the position of the other objects.

Lock all objects

This mode locks all objects against moving and changing the start or end position, thus preventing such changes from happening accidentally. By simultaneously holding down the "Alt" key while clicking on an object, the lock can be temporarily removed.

Note: You can lock individual objects against moving by clicking the lock symbol on the object. It is still possible to change the start and end time then. In the System Options under Program > Object Lock Definitions (view page 708), you can specify against which changes exactly locked objects should be locked.

Link curves to objects

In this mode, all automation curve points of the track automation are moved together with the objects. If you delete the object, the automation data is also deleted.

Link until silence

If an object is selected and moved, all objects on this track adjacent to the right are moved together with the object. This way partial ranges of a project, if separated by pauses, remain unaffected from moves.

Link one track to the right

In this mode, all objects on the current track that are to the right of the clicked object are moved together with it.

Link all tracks to the right

In this mode, all objects in the project that are to the right of the clicked object are moved together with it.

Link one track to the left

In this mode, all objects on the current track that are to the left of the clicked object are moved together with it.

Link all tracks to the left

In this mode, all objects in the project that are to the left of the clicked object are moved together with it.

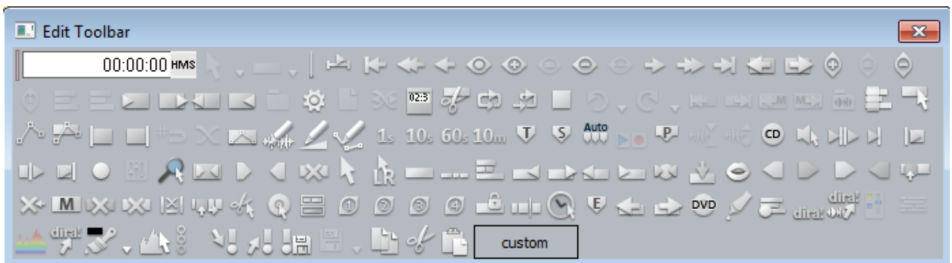
Note: If the object modes "Link until silence" and "Link one track to the left/right" are active, the markers in the top arranger track will be moved together with the objects. In object mode "Link all tracks to the left/right" the markers are always moved, regardless of which track the clicked object is located on.

More information about using the object modes is available in the chapter "Working with object modes (view page 142)".

Editing Toolbars

You can customize the toolbars to your requirements by adding icons, changing their positions or removing the ones you don't need. To do this, right-click to open the context menu 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.

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

Recording

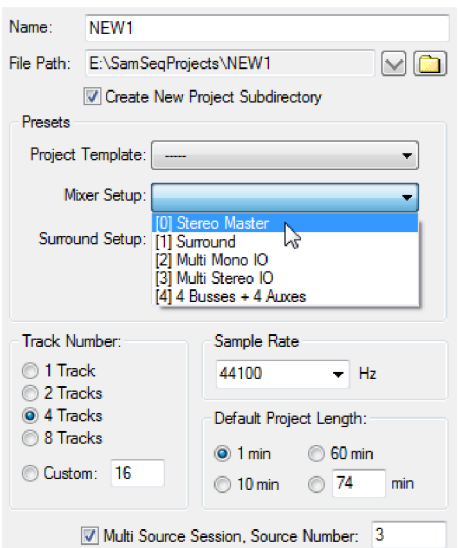
This chapter introduces all the functions for the recording process.

Audio Recording

This workshop will show you how to record using Sequoia. It will introduce you to all of the basic functions for the recording process.

Creating a VIP

First create a new project by clicking on the "New Multitrack Project (VIP)" button in the program start menu. If you already have Sequoia open, select "New Virtual Project (VIP)..." in the "File" menu



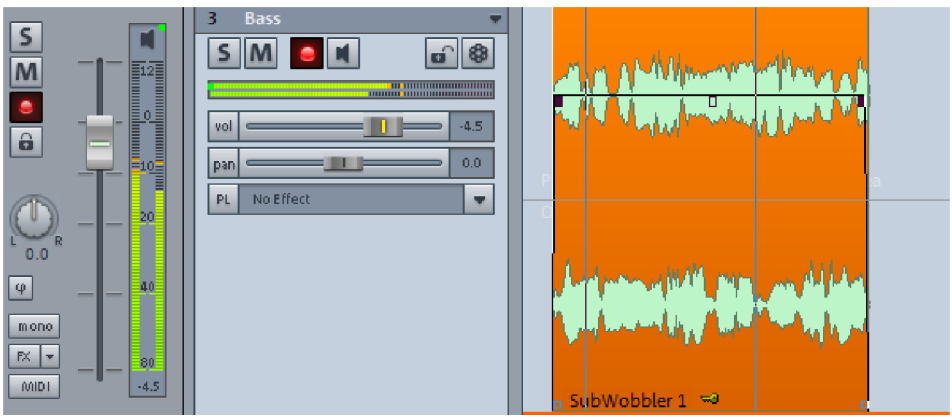
Now you can name your new project and select the file path for saving the VIP. You can also create a new folder where all files relating to this project can be saved. Select "[0] Stereo Master" as the mixer setup.

In the "Track Number" field enter the number of tracks you want to use. This doesn't mean that they are limited to this number because new tracks can be added at any time to the project. Adjust the "Sample Rate" to match the sample rate of your sound card and click on "OK".

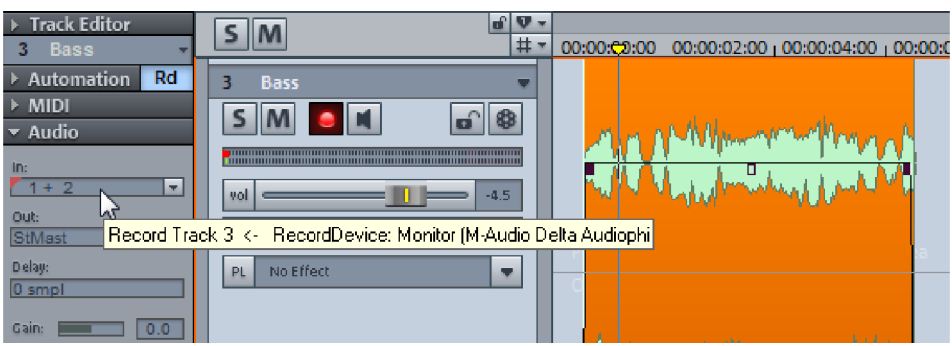
Note: If the preset project length is exceeded, the VIP adapts itself to the actual project length.

Preparing To Record

To record from your internal CD/DVD drive or from an external sound source using a cable/microphone connection, it is important that your sound card is connected to the sound source. You can check this by switching on the track that you want to record. Click the record button in the track header to do this; the button will glow red. The source's input level peak meter now appears in the bar.

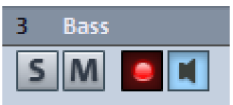


If you can't see an input level in the peak meter, check in the track editor ("View" > "Track Editor") to ensure that the sound device is connected to the same port listed under "Audio" -> "In". If this is not the case, set the track's input device to the port of the sound card's audio source. To do this, click on the field "Audio" > "In" and select the appropriate input.

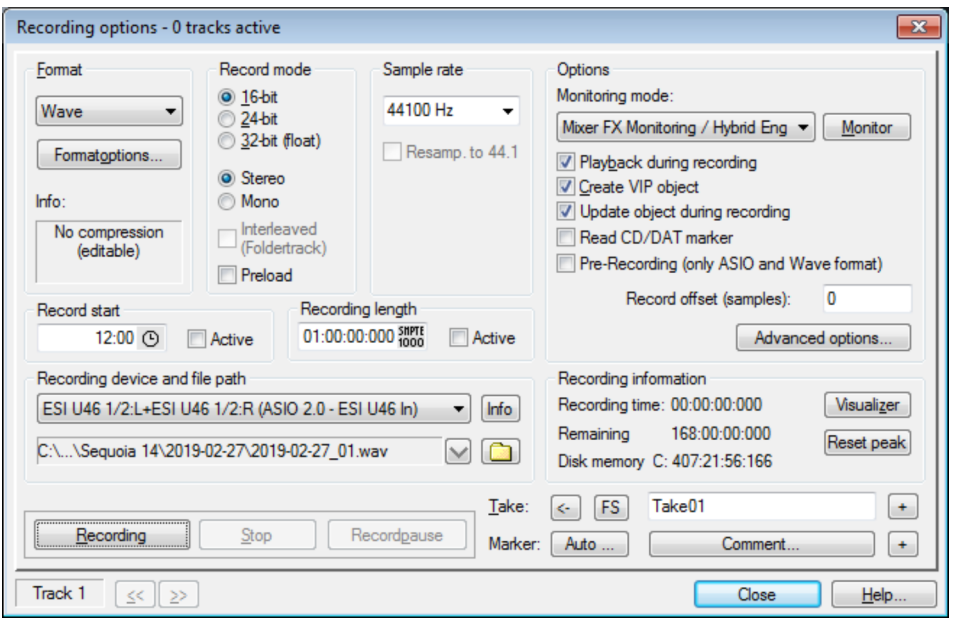


If you see the level displayed on the peak meter but still can't hear anything, make sure that the output device for the track corresponds to the output of your sound card.

If necessary, click on the loudspeaker symbol next to the level display on the track in order to view and hear the input signal.



Open the recording options by right-clicking on the record button in the transport console:



Select the format for the recording first. Detailed settings for the audio format are available in "Format Options".

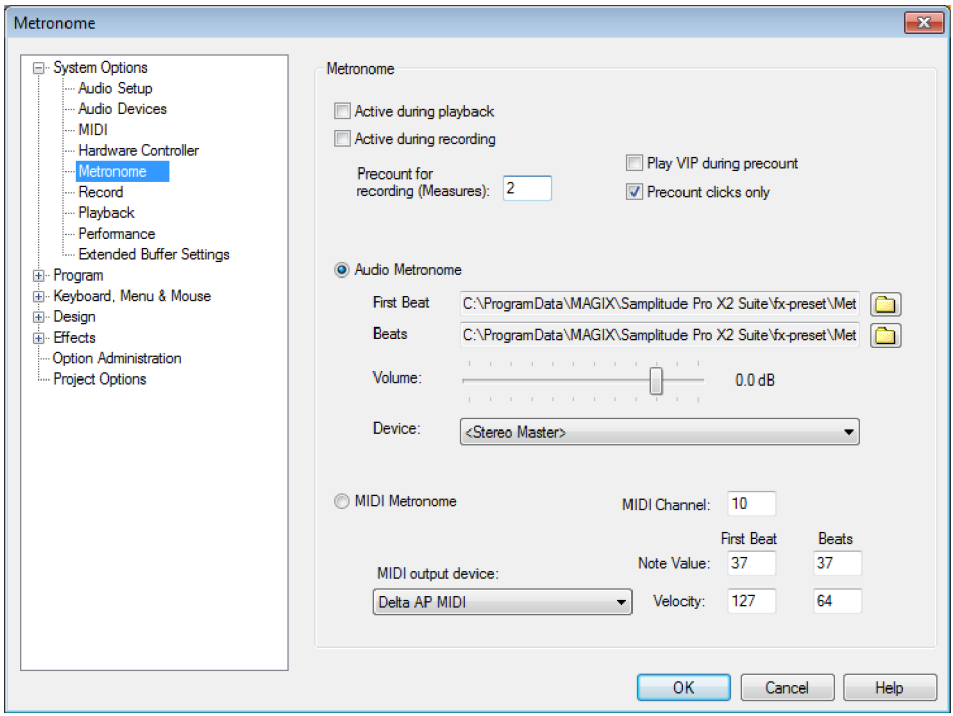
A record mode set to 16-bit at a sample rate of 44.1 kHz is equivalent to CD quality. A bit resolution of 24-bit or even 32-bit (float) ensures that the noise level remains as low as possible during subsequent editing of the audio. However, a higher bit resolution requires more disk space.

Now specify if the track should be recorded in mono or stereo. If you are recording from a sound source such as CD or DVD, it is best to use a stereo recording mode. For single instruments or vocals, it is better to use a mono recording mode.

In "Mono (Mix)" mode, the track input is switched to mono, the signal is recorded like a stereo recording (dual channels), and then mixed into a single channel.

Metronome Options

You can activate the metronome through "File" > "Program Preferences" > "Metronome Options...". 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. You can also make the VIP play back during the precount by activating the option "Play VIP During Precount". If you activate "Precount Clicks Only", the metronome will stop when the recording starts.

The "Punch Recording" button in the toolbar blinks during precounting and the "Record" button blinks in the transport console (view page 65). When recording begins, both buttons remain active until the end of the respective recording run.

Audio Metronome

First Beat/Beats: This setting allows you to specify individual samples ("Volume") for the metronome for the first beat of each measure ("First Beat") and the remaining beats of the measure ("Beats").

Both metronome sounds "Metronom1.wav" and "Metronom2.wav" are preset and are located in the folder "fx-preset" in the program directory.

Volume: This controller regulates the volume of the audio metronome sounds.

Device: Set the audio device for the metronome here. Stereo Master is preset but can be changed to another audio device output as the source of the metronome.

MIDI Metronome

MIDI Output Device: Here you can set the device that will produce the metronome click. This is normally the sound card.

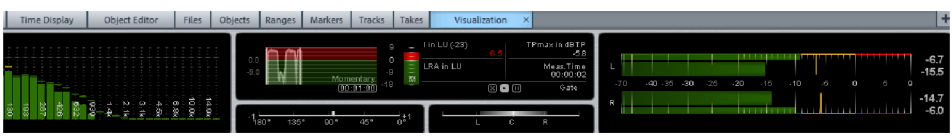
MIDI Channel: Here you can set the MIDI channel through which the MIDI commands are sent.

First Beat/Beats/Note Value/Velocity: Set various note values and the velocity for the first beat of each bar or the other beats of the bar here.

Note: The command "Edit" > "Tempo" > "Create Click Track" creates an audio track that contains all metronome clicks as objects.

Level Adjustment

1. Click "Visualization" in the docker. You should now see a signal.



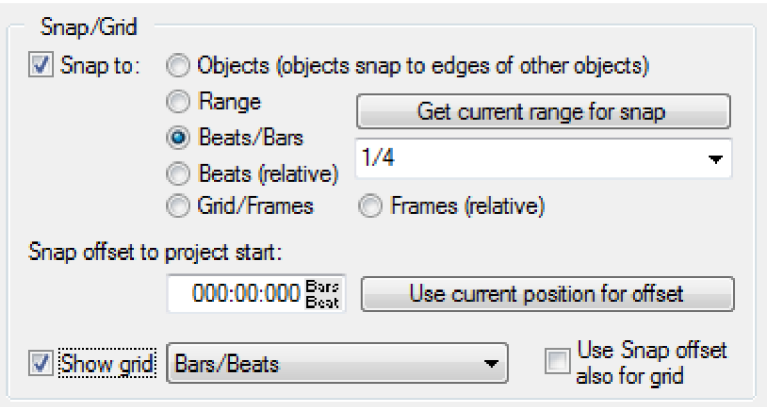
If there is no visual display of levels in the peak meter, click the "Mon" button on the transport console.

2. Adjust the input signal to your external amplifier or the sound card input so that the peak meter is as close as possible to 0 db but does not quite reach this value.

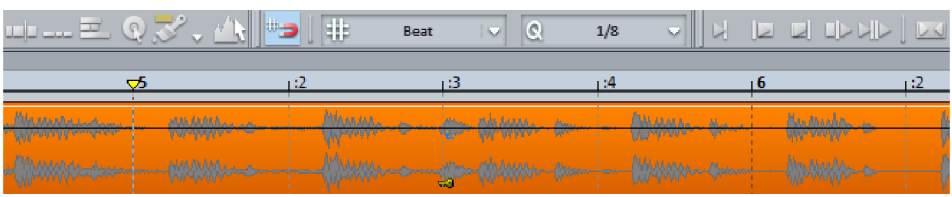
Snap And Grid Setup

The recording starts from the current position of the playback marker. For example, if you want to record starting at the fifth bar, place the playback marker at the beginning of the fifth bar.

1. To do this, open the project options under "View" > "Grid" > "Snap And Grid Setup" (shortcut: "I") and check the "Snap To", "Beats/Bars" and "Show Grid" boxes. In the selection menu, select "Bars/Beats" as the grid unit and confirm this by clicking "OK".

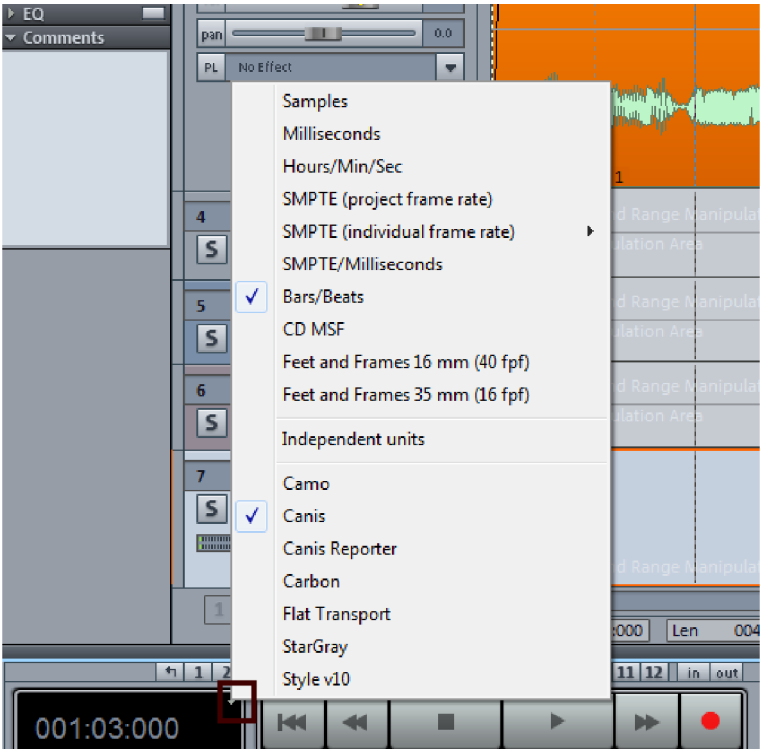


Now you can see the grid lines in the VIP.



These divide the project window, and the grid toolbar now indicates the bars.

2. Switch the display in the transport window (view page 65) to "Bars/Beats" using the start display area.



Since "Bars/Beats" is selected as the grid unit, you can easily use the left/right arrows on the keyboard to jump to the beginning of the fifth bar while monitoring the bar and beat position of the playback marker in the transport window.

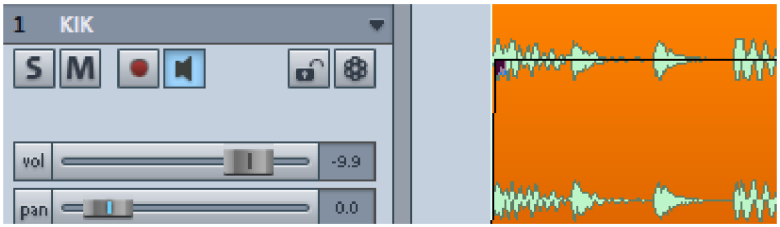
Starting To Record

Now if you click on the "Record" button, the recording will start.



When you are finished recording the audio you can click on "Stop". You will now be asked if you want to save your recording. If you are satisfied with the results, press the "Yes" button.

Your recording appears in the project window as a virtual object.



Now activate the next track by clicking on the record button in the second track.



As you can see from the level bars, the input signal is now selected for this track, which is ready for recording.

Note: You can bypass the dialog window and record straight into the VIP if the settings have already been made. Simply press the "R" key on the keyboard. If a track is activated, recording will begin immediately.

MIDI Recording

In Sequoia there is no separation of audio and MIDI tracks. Any track can contain both audio and MIDI objects. This means you can start working with audio and MIDI material within a project without needing to split tracks. You also have full control of VST instruments within a single track. When freezing the track, the MIDI data is converted into audio data. However, a track can only record from a certain device. This means that audio and MIDI files cannot be recorded simultaneously onto one track.

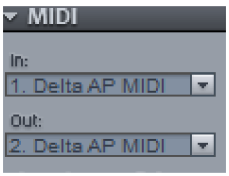
In a Sequoia, MIDI data is recorded similarly to audio data. You create an object for each recording, which overlaps existing objects.

Preparing A MIDI Recording

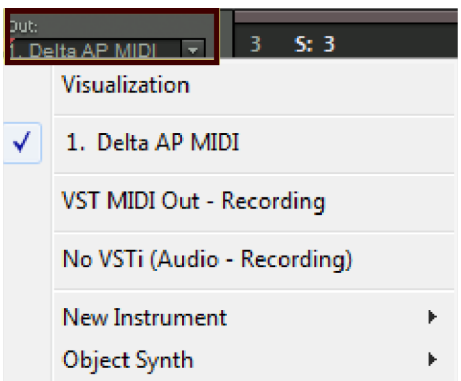
If the "MIDI" button is activated in the track editor, the MIDI section opens to prepare the track for MIDI recording.

Indicate to Sequoia which MIDI input device the program should use. Select the input device in the MIDI In slot (e.g. MIDI keyboard) to do this.

If you have multiple MIDI input devices connected to your system, you can also select <ALL> as the MIDI input device for a track.



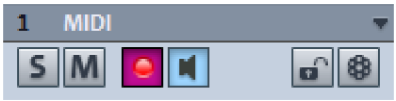
Select the MIDI port you want to use for the playback device (MIDI Output Device) through the MIDI Out slot (e.g. MIDI Out of your sound card or a VST instrument).



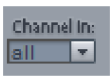
If you don't hear a sound when you press the keys on the input device, you probably need to activate the monitoring. Activate this with the loudspeaker button (MIDI thru) for the respective MIDI track.



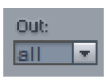
By right-clicking the "Mon" button in the transport console you can access a menu for adjusting the behavior of the "Record" and "Monitoring" buttons. If you activate the options "Automatic MIDI record switch on current track" and "Automatic MIDI monitoring (Thru)", the selected MIDI tracks will be immediately activated for the recording and monitoring will be switched on for the input signal. If you want to record multiple MIDI tracks simultaneously, deactivate the option "Automatic MIDI record switch on current track".



If you still don't hear anything when you play the keys, make sure that the MIDI out channel of the MIDI keyboard matches the "Channel In" slot in the track editor for the selected MIDI track.



Make sure that your MIDI synthesizer is sending to the same MIDI channel that you have selected in the "channel out" slot in the track editor. To be certain, set the Out channel to "All". Now you should here all of the synthesizers no matter what channel they are on.

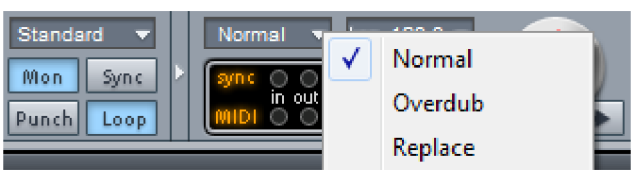


Note: Many drum machines send to MIDI channel 10, since this is preferably used as the percussion channel and is even specified as such according to the General MIDI (GM1) specifications.

MIDI Record Modes

The following MIDI record modes are available: Normal, Overdub, and Replace. These options determine how newly recorded MIDI data is added to the VIP.

You can set the MIDI Record Modes in the transport control.

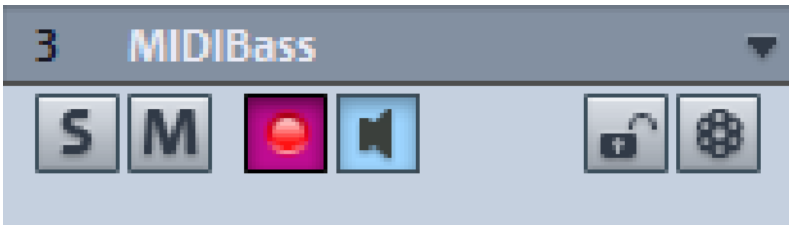


Normal: This recording mode is the equivalent of a normal audio recording. This means that a new MIDI object is created over the existing object for each recording. The old object remains, but it can be covered partially or completely by the most recently recorded one (both visually and acoustically). This way, you can record multiple takes of a passage and then compare them in the take manager (view page 201) later on.

Overdub: The data is recorded into an already existing object. The newly recorded MIDI data is mixed together with the data that is already present.

Replace: The data is recorded to an existing object and any MIDI data that is already present is overwritten.

After selecting the mode, make sure that the recording is activated for the track. The record button, which is set to standby, changes to violet to signal that this MIDI track should record MIDI data.



Now you can now start the MIDI recording using the shortcut key "R". After ending recording with the "Stop" button, you will be asked whether you want to keep or delete the recording. After confirming this with "OK", your newly recorded material will appear as a "MIDI take" in the VIP. In case you have recorded several takes in normal mode for the same selected range, you can select and play back the individual MIDI takes in the Take manager (view page 201).

Note: You can switch between the individual recording passes even faster by holding the "Ctrl" button down and right-clicking on the MIDI object. Now a menu will appear in which you can select the desired take and play it right away.

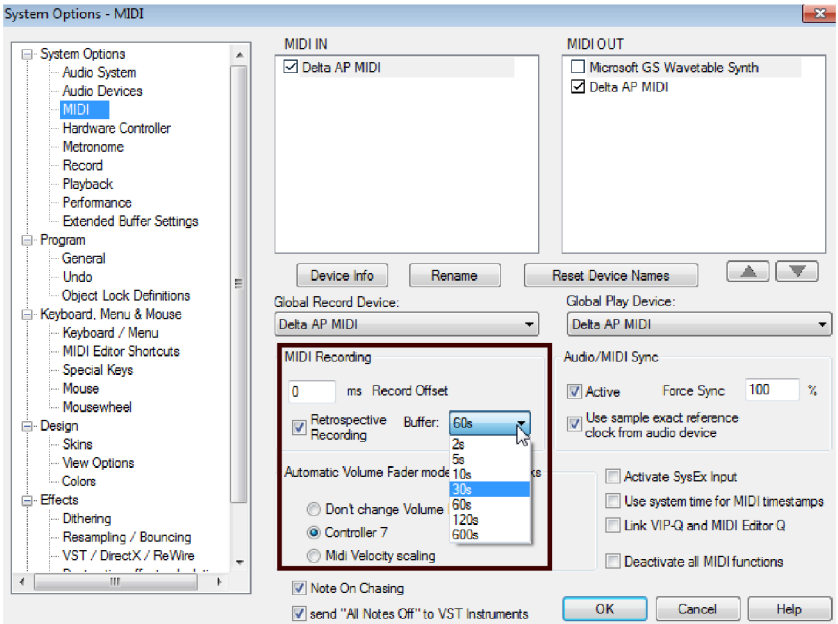
If you record within a previously created MIDI object (Object -> New MIDI object), you can follow the recording in the arranger track. The created events are displayed as blue bars. If individual MIDI events are muted after recording in the MIDI editor, these will appear in the MIDI object as gray bars.

Retroactive MIDI Recording

The "Retrospective MIDI Recording" feature is useful for saving MIDI improvisation, for instance if you're jamming to an already existing project with a MIDI/VST instrument. If you played something particularly inspired while you were jamming, you can retrieve it – even if you didn't recorded it earlier on.

To do this, proceed as follows:

- Press "Y" to open the systems options and go to the MIDI settings. Under "Recording", activate "Retrospective Recording" and enter the buffer for this function.



- Play the project and play along on your MIDI instrument until you've played the passage that you want to use.
- Stop the playback and select the "Retrospective Recording" function in the "Play/Rec" menu.

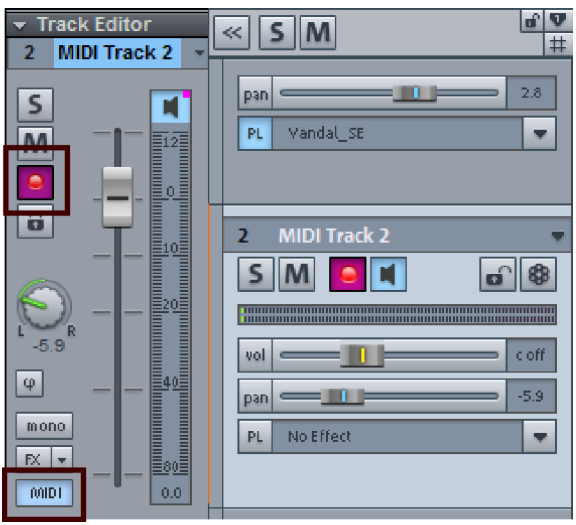
You will now be asked if you would like to insert the accompaniment you have just played at the point where you played or at the current playback marker position. Sequoia will then insert the part which was last played into the corresponding track. This preserves spontaneous moments of creativity and lets you integrate them into your project.

Tip: Recording MIDI notes using the "Retrospective Recording" feature also works in Stop Mode, i.e. without simultaneous playback of the arrangement.

VSTi Recording

VST instruments are recorded in the same way as MIDI hardware devices via MIDI recording. When recording software instruments, you need to open them in Sequoia before recording. Here we show you how to this using the example of a MAGIX plug-in.

- Simply select the track where you want to record and activate the MIDI button and the recording. Monitoring will be switched on automatically.



Note: If monitoring isn't automatically switched on, right-click on the "Mon" button and select the option "Automatic MIDI monitoring (Thru)".

- Select the plug-in you want to record in the plug-in slot of the track header (or in the track editor).

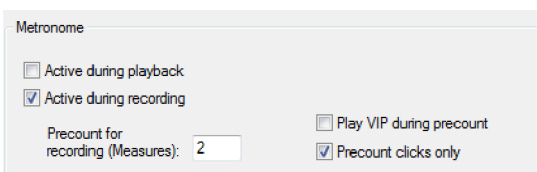
We'll use the MAGIX synth "Revolta" here.

- The graphic interface for the plug-in will open (in our case "Revolta"). Adjust the sound and test playback.



- Activate the metronome if necessary.

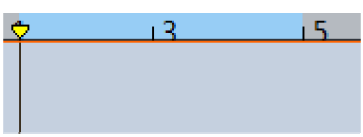
Right-clicking the "Click" button in the transport console opens the "Metronome Options" dialog, in this dialog select the "Active while Record" and "Precount click only" options. This way when we record we will only hear the metronome during the precount.



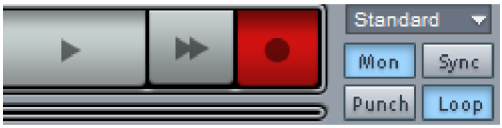
- Start recording by clicking on the record button in the transport console and play a melody using your keyboard.

Loop Recordings

1. If you want to record in a loop, first select the range in the arranger that you want to record.

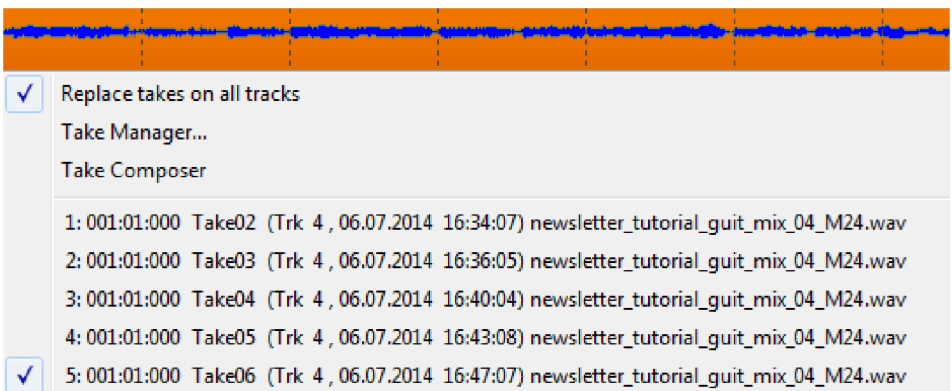


2. Click the "Loop" button on the transport console and start the recording.



The recording runs through this range repeatedly until it is stopped. Each run through creates a new take.

3. If you right-click the last recorded take while holding down the "Ctrl" key, you will see all of the takes that were made during the loop recording. To listen to one of the takes, simply select it in the displayed menu.



The Take manager (view page 201) is useful for organizing and editing recording runs. Use it together with the Take composer (view page 203) to produce the perfect take.

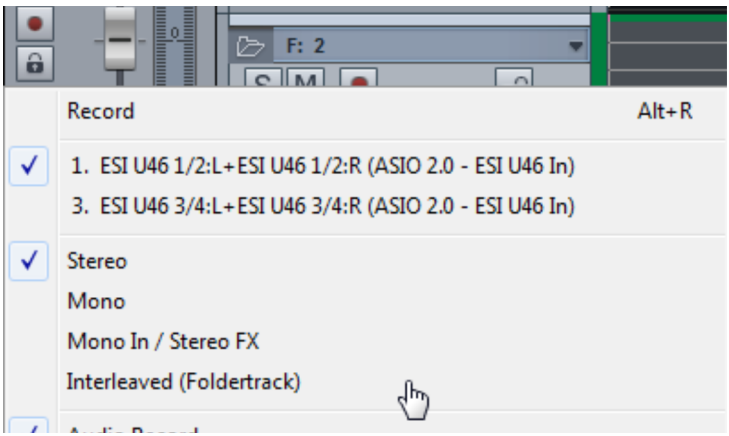
Interleaved Recording

You can conduct a multi-channel recording as interleaved recording. That means recording an arbitrary number of channels into one wave file.

Here's how you do it:

4. Create the desired number of tracks and do the appropriate input routing by assigning a different hardware input to each track. You can mix mono and stereo tracks at will.
5. Select all tracks by clicking on the track header of the first track and holding down the Shift key while clicking on the track header of the last track.
6. Now choose "Track menu" -> "Insert new tracks" -> "New track folder".
7. Now right-click on the record button in the track header of the folder track and choose "Interleaved (Foldertrack)" from the context menu.
8. Now activate recording in the folder track with a left click on the record button on the track header.

9. Click the record button in Sequoia to start the recording.



Notes:

- When loading an Interleaved Multi-Channel-Wave a Foldertrack will be created by default, optionally a surround master will be created. You can deactivate this behavior in "Options for loading audio files" (view page 616).
- Any number of audio tracks can be recorded into an Interleaved file. Loading the file into a folder track with simultaneous creation of a surround master works up to a maximum of 32 tracks.

Punch recording

In punch recording mode, only the range between the punch starter marker and the punch end marker is recorded. Punch recording can be carried out for all recording types described. You activate this mode in the transport console.



If you have not already set a punch marker, one will be set automatically during recording.

If the recording is started when punch markers have been set, the recording button will flash until the start marker has been reached and again when the end marker has been passed.

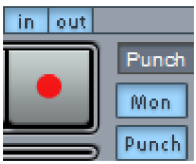
It will light continuously during the actual recording between the markers.

If you end the punch recording by clicking the record button again, playback will not be interrupted. If you press the "Stop" button, playback will stop.

Punch Recording With Markers

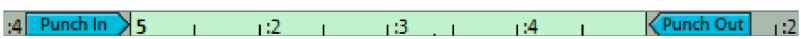
If you know the exact range that you want to record in, punch recording using markers is the best option.

1. To do this, select the area with the mouse where you want to use punch recording, and then activate the "In" (sets the punch-in marker) and "Out" (sets the punch-out marker) buttons in the transport console (view page 65).



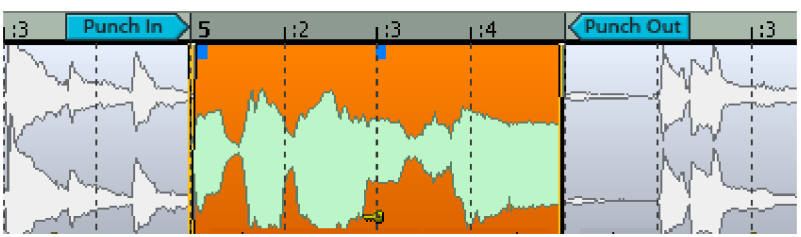
2. Now you can set the playback marker in position.

3. If "Punch" is activated, start the process with the "Record" button on the transport console. The actual recording will only take place within the punch range.



When the recording is started and the playback marker is not yet in the punch range, the "Record" button will blink. The button remains red during the punch recording.

Example: An error between beats 5 and 6 needs to be corrected. Playback starts before the punch-in position so that the introduction to the recording is made as easy as possible. During the punch process, the "Record" button flashes; the recording will start automatically from the "punch-in" marker (bar 5). The end of the object is already satisfactory so the recording will automatically end before it at the "punch out" marker (bar 6).



Working in the Project Window

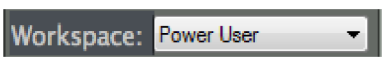
Here are some tips on how to work more effectively with Sequoia:

Workspaces

Workspaces are used to make menu items and toolbars more clearly arranged and to adapt the window layout to specific tasks.

For this you can hide certain menu items (menu "File" > "Program Preferences" > "Edit keyboard shortcuts and menu... (view page 688)") > "Hide Menu Item") and customize toolbars (right mouse click on the toolbar) so that they only contain the icons and menu items that are required for a specific task such as mastering, editing or recording. The arrangement, visibility and docking status (see below) of all windows in the Workspace are also saved.

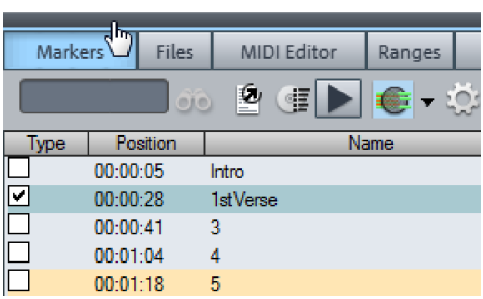
The drop-down menu for the Workspaces is located at the very left of the lower toolbar. Various work areas are already predefined, the "Power User" Workspace is the default. It contains all available menu commands and the toolbars in their default configuration and can be used as a starting point for your own customizations.



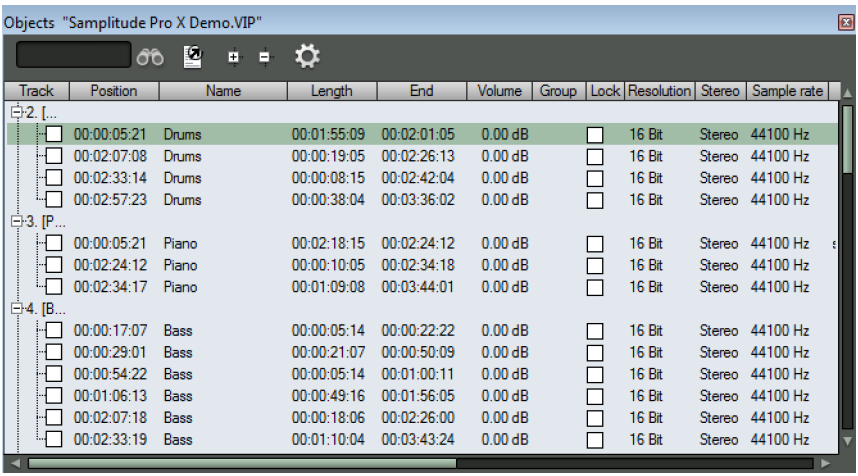
To customize a Workspace, right click on the drop-down menu and select "Edit Workspace". In further menus you can select which toolbars are displayed. There are also entries for customizing the menu commands and toolbars. Use "Save Workspace" to save your adjustments, preferably under a new name.

Docking

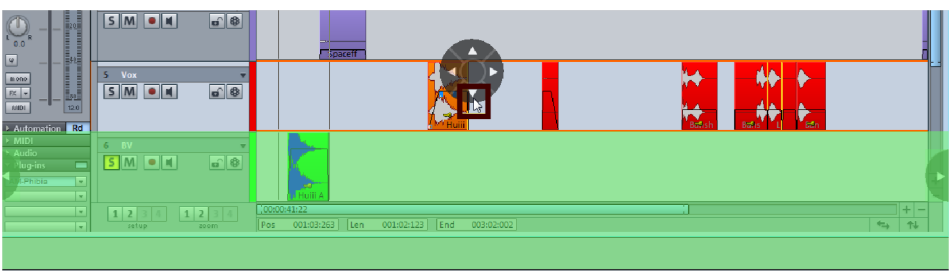
In Sequoia you can dock certain dialogs and windows to areas in the VIP interface or to the docker. The handles for docking and undocking (view page 177) are found at the top of the respective window and in a docked state are distinguished from the selected skin by a dark bar.



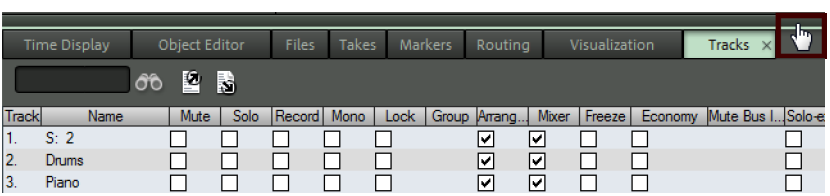
In an undocked state, the corresponding dialog appears with a conventional title bar.



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 pointer onto one of the arrow symbols, the area that docks when you release will be displayed as a transparent rectangle.



Conversely, an individual window can be undocked by clicking the title bar and dragging or by double-clicking the handle area.



You can place and arrange new windows inside the docker. To do this, simply move the respective window using the mouse pointer onto one of the arrow symbols in the docking bay.

The docking function is available for the following windows:

- All project windows
- All manager windows
- Visualization
- Time Display
- Object Editor
- MIDI editor
- Transport console
- Audio quantization

Zooming

Use the zoom functions to adjust the sections of a virtual project. The higher the zoom stage, the more precise the display will be.

Sequoia features the following zoom functions:

Zooming with the mouse wheel

Using the mouse wheel is the fastest way to zoom. To do this, hold down any key while using the mouse scroll wheel to zoom. You can zoom in by rolling the mouse wheel upwards. You can zoom out by rolling the mouse wheel downwards.

Shortcut keys for zooming with the mouse wheel

Ctrl + mouse wheel	Vertical Zoom
Shift + Ctrl + Mouse wheel	Horizontal zoom
Ctrl + Alt + mouse wheel	Vertical + horizontal zoom
Alt + mouse wheel	Zoom in/out for waveform display

Zooming with the keyboard

- The key combination "Ctrl + Arrow right" lets you vertically zoom out of the project; "Ctrl + Arrow left" vertically zooms in.
- Ctrl + Arrow up zoom into the waveform display, while Ctrl + Arrow down zooms out of the waveform display.

More shortcuts for zoom functions can be found in "View" > "Horizontal" or "View" > "Vertical".

Zoom using buttons

In the lower right-hand corner of the VIP window, the "**+**" and "**–**" buttons may be used for more precise adjustment of the horizontal and vertical zoom levels.

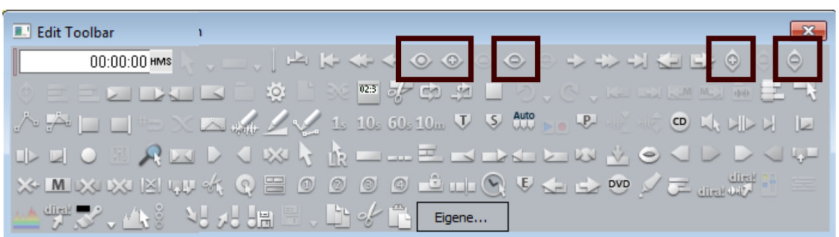


Buttons for important zoom levels are located in the bottom toolbar.



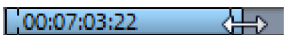
1. Entire project (vertical)
2. Entire project (horizontal)
3. Vertical zoom to range
4. Horizontal zoom to range
5. Zoom out of waveform display
6. Zoom in on waveform display

You can add additional zoom buttons to the context menu via the "Edit Toolbar" dialog. (view page 91)



Zooming with the Scrollbars

The scroll bars can also be used to change the zoom stages. When the mouse pointer moves over the left and right border of the horizontal scroll bar, it changes into a double arrow. This can be used to move the scroll bar borders, which also changes the length of the section displayed in the window. The length of the section will then always be displayed.



Similarly, you can use the vertical scroll bars to zoom into the track display.



Zooming with the Mouse - Vertical Zoom

An elegant method of zooming is to left-click on the time line, hold down the button, and drag the mouse vertically.

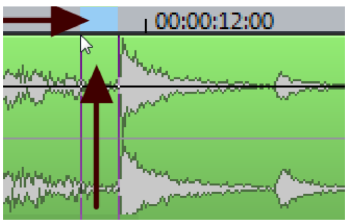
By dragging the mouse up, you can zoom out of the project, and you can zoom in with the reverse action. Simultaneously, you can change the range borders or the playback marker position with horizontal mouse movements, depending on whether you position the mouse at the beginning of the action on the grid toolbar or on the marker bar. 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. Next, move the mouse downwards while holding down the mouse button to zoom into the project.



3. Correct the mouse position by moving it horizontally and then zoom out of the project by moving the mouse upwards.



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

Zoom Mode

You can switch to Zoom Mode (view page 88) by clicking the magnifying glass icon (Zoom Mouse Mode). The mouse pointer will turn into a magnifying glass.



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.

The Zoom mouse mode can also be temporarily activated by holding the Z key.

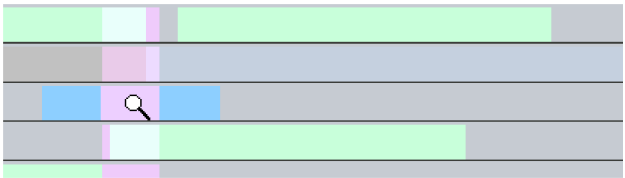
Zooming with the Overview Mode

Overview Mode offers you a view of all the objects in your project in a clear display underneath the arranger track. You can activate overview mode with this button

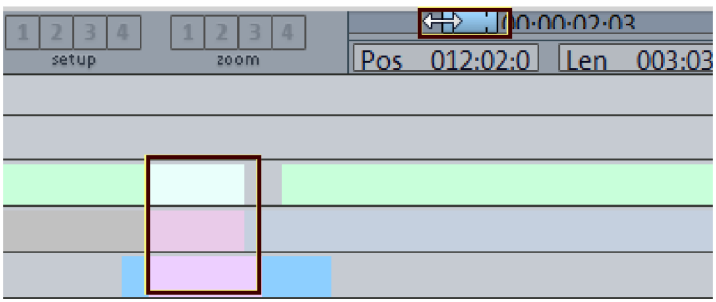


or with the corresponding command in the "View" menu. The mouse pointer turns into a magnifying glass in this screen area. The section chosen in overview mode determines the display size and the position of the section shown in the arranger. You

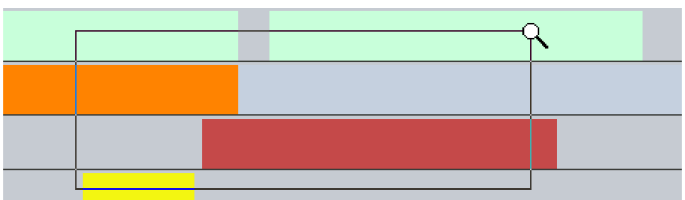
can adjust the section in the arranger by clicking on the desired position in the overview.



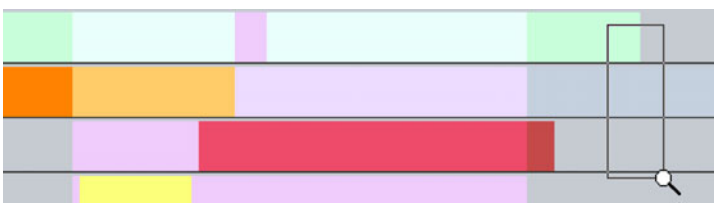
With the arranger scroll bar you can also control the selected section in the overview.



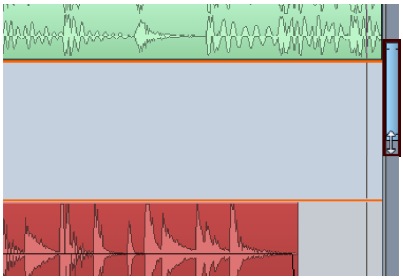
To select the desired section, drag out a lasso around the overview using the mouse.



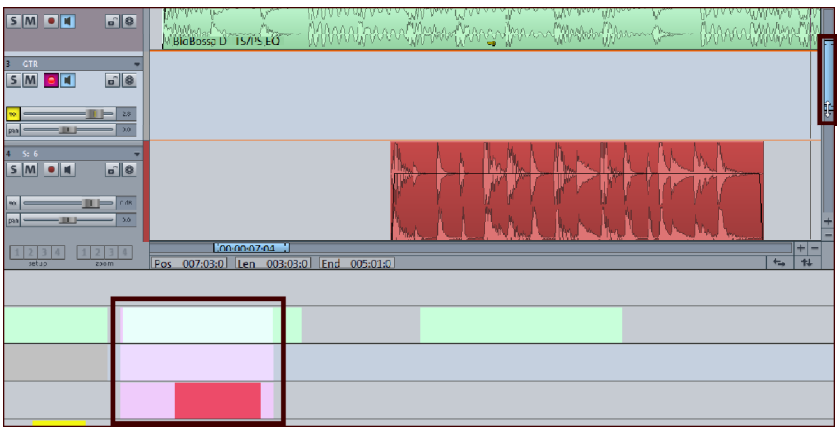
If another position in the overview is selected or a new lasso is drawn by left-clicking, the horizontal (i. e. temporal) position in your arrangement will change.



By extending or shortening the vertical scroll bar you can determine how many tracks will be shown in the arranger.



Overview mode also displays the tracks in the arranger that are shown at the current zoom level as a colored field.



Note: In single track virtual project (VIP) and WAV project a waveform will be displayed in Overview Mode.

Scrolling

Scrolling refers to when the visible section of an arrangement in the VIP window is moved.

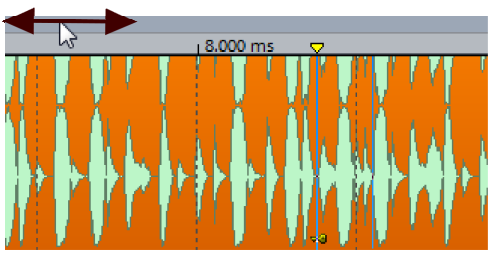
The visible section follows the playback marker during playback in what's known as Autoscroll Mode. This means that the section "jumps" to the next time section shortly before the playback marker moves off the right side of the screen (Page Mode). In "Soft Autoscroll" mode, the section moves constantly under one of the static play cursors in the middle of the section. You can switch between these scroll modes in the "View" menu.

The playback marker can also be moved using the arrow keys on your computer keyboard. Scroll with the arrow keys in Soft Auto Scroll Mode by holding down the ALT key.

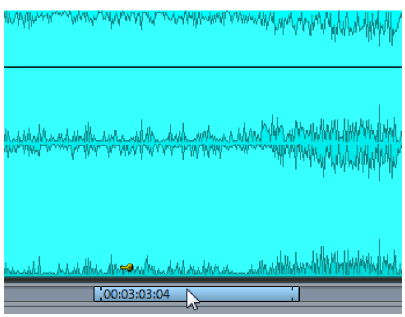
The Home and End keys move the playback marker and the visible section of the project either to the very beginning or the very end.

If you want to move the visible section without moving the playback marker, you have the following options:

Grid/marker bar: Moving the mouse pointer in the grid/marker bar while holding down the "Alt" key enables you to move the visible section conveniently from left to right.



With the scroll bar: By moving the scroll bar the contents of the window can be moved either to the left or to the right.



With the arrow buttons in bottom toolbar: The arrow buttons can be used to move the section to the previous or next object edge.



Tip: Other scroll buttons can be added (view page 91) in the "Edit Toolbar" dialog.

With keyboard: Under "View" > "Horizontal/Vertical" (view page 1149) you'll find a range of scrolling commands that can be carried out using your keyboard. Set up keyboard shortcuts for menu scrolling commands to be able to navigate through the arrangement quickly using keyboard commands.

Scrubbing



Scrubbing makes it easier for you to find a specific part of the music in the project simply by moving the mouse pointer. Playback occurs while moving the mouse forwards or backwards.

By varying the playback speed, it's possible to quickly locate an approximate position and then find the exact position at a reduced speed.

If you press the insert key ("Insert" or "0" on the number pad), Sequoia will switch to "Scrub Mode". Now move the playback marker with the mouse in order to hear the audio material beneath it. When you hold down the mouse button in "Scrub Mode", you can also hold down on the "Shift" or "Ctrl" keys to slow down the process slower to make it more precise. You can also use the mouse wheel for fine-tuned scrubbing.

The mouse mode list also contains a separate scrubbing mouse mode.

The following scrubbing modes can be selected via the playback options (keyboard shortcut: "P"):

Shuttle: The relative distance between the playback marker (positions bar) and the mouse position is used to control the speed. This means:

Scrub controller on the left edge	=	double speed backwards
Scrub controller in the center	=	no movement
Scrub controller on the right edge	=	double speed forwards

Absolute: This mode allows you to use the absolute position of the mouse in the window to control the speed.

Two speed: Two speeds are provided for scrubbing. The object plays back slower or faster depending on the distance between the scrub control fader and the mouse position, whereby slow scrubbing at a speed of 0.25, or 1/4 of the original speed, is preset, and fast scrubbing is set to 1.0, i.e. the original speed. Change the value for slow playback in the "Speed" field.

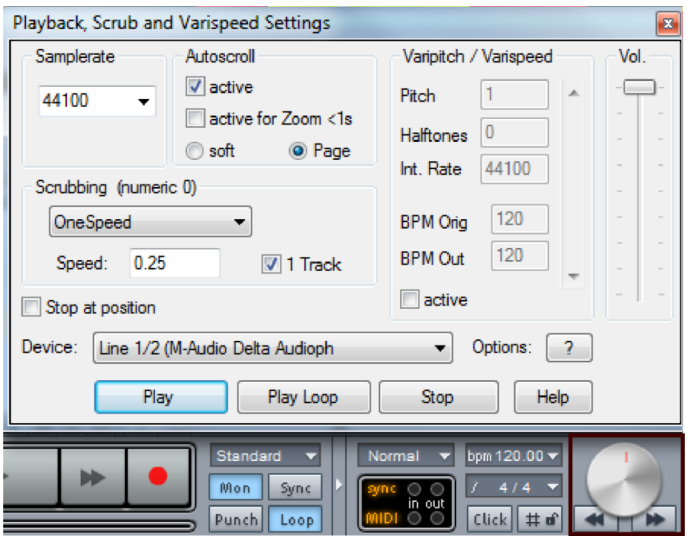
One speed: The preset scrubbing speed is 1.0, i.e. original speed. The "Shift" key halves this value. The "Ctrl" key uses the speed set in the "Scrubbing speed" field.

Scrubbing speed: This specifies the factor of the original speed applies to the scrubbing speed. 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.

The option "1 Track" enables you to limit scrubbing to the active track only.

Scrubbing in the transport control

Another way to scrub is to use the scrub control wheel in the transport controls. Right clicking on the scrub control wheel opens a dialog to set variable speed settings as described above.



Scrubbing tip: Scrolling is "softer" in case of smaller hard disk HD buffers (250-1000 samples). Test if your computer functions at this buffer size without any playback errors.

Advanced Ruler / Time Display

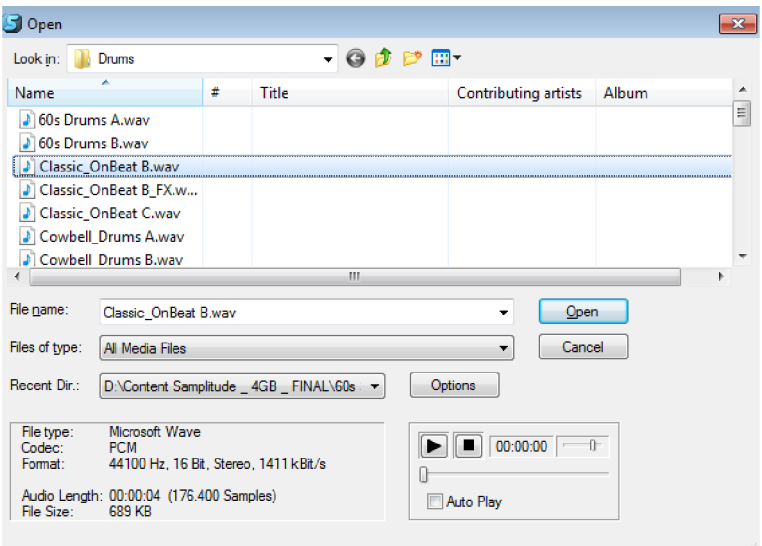
Use the grid bar's context menu to activate a second grid line in the arranger and set your own measurement unit (Show 2nd Grid Line). Both grid bar positions can be exchanged (Exchange Grids). If "Independent units" is selected in the transport control, (view page 65) the upper grid will not adapt to a chosen BPM grid (view page 448) like the lower grid does. This makes it possible to select an SMPTE format independent of the project frame rate, which provides an overview of two different SMPTE displays.

Load and import audio

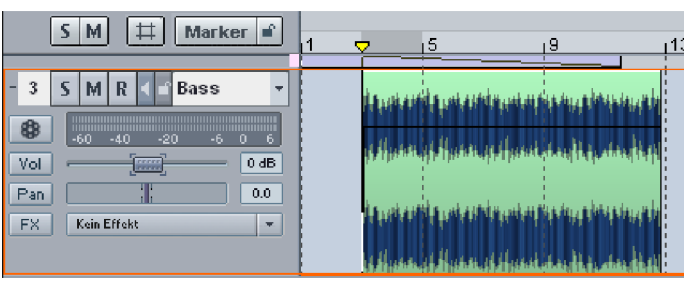
Loading Audio Files

1. Press "W" on your keyboard or select "File > Load/Import > Load Audio file" from the menu.

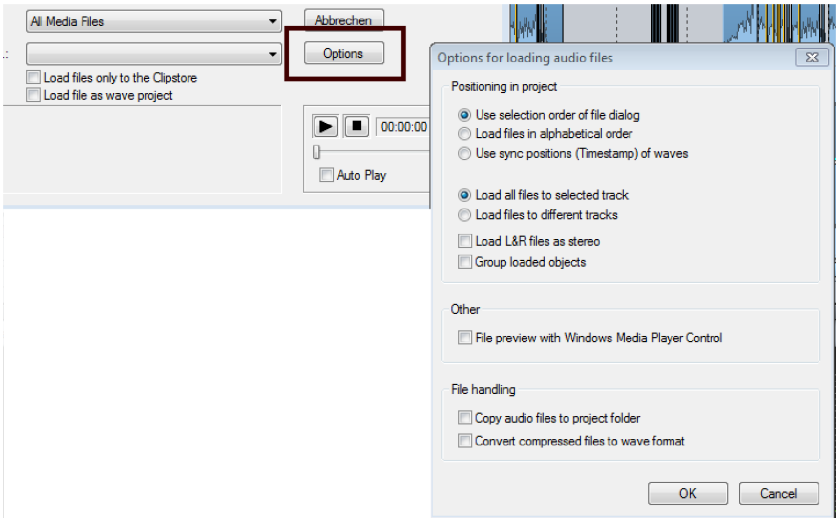
2. Choose an audio file from your computer. If you check the "Auto-Play" box, the selected file will be played straight away.



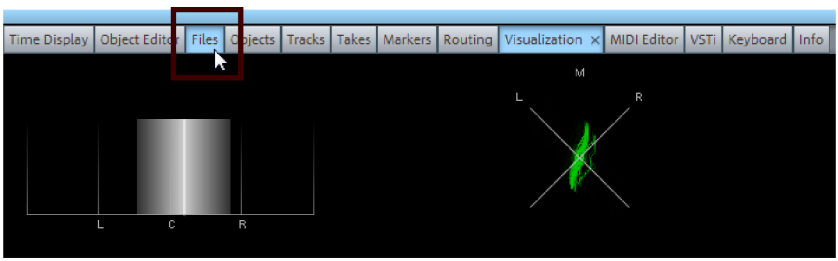
3. Using the "Open" button, the selected file in the activated track is inserted at the play cursor position.



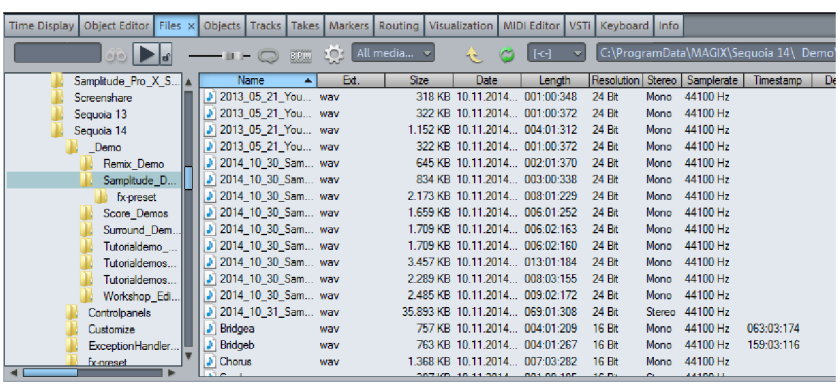
4. Don't forget that you can always use the "Shift" and "Ctrl" keys to select multiple files at once. Several options let you choose exactly how you want to order the files in the arrangement.



5. You can also load files using the file manager, which you access using the "Files" button in the docker.



There are extra features available when using this file manager, namely special search and sort functions. From the file manager, you can even drag and drop the selected audio files directly into the arrangement.



The difference between loading and importing audio files.

In contrast to loading a file via "File" > "Open" ("W" key), when you import an audio file ("File" > "Import"), it is converted to .wav format and then copied to your hard drive.

This is only necessary for file formats that cannot be opened directly using Sequoia, e.g. Windows Media Files and certain MPEG files. Increase the performance of your computer by importing compressed formats like MP3 or OGG Vorbis instead of loading them to bypass real-time decoding.

Integrating a Section of an Audio File as an Object

To integrate a section of an audio file into a virtual project (VIP) as an object, drag out the range that should be integrated into the VIP.

Version 1: Open the VIP and set the playback cursor to the position where you want the new object to be inserted. Select the track where necessary. Select "Object" > "More" > "New Object". The selected range is added as a new object to the VIP.

Version 2: Alternately, press "Return" to display the VIP and audio projects one under the other. Click the selected range of the range bar in your audio file and drag it up to the track you want in the VIP.

Integrating CD Tracks into a VIP as Objects

You can also import tracks from an audio CD into your project.

- Click anywhere in the upper half of the desired track at the position that will mark the beginning of the CD track. The playback marker now moves to the position at which the object will be inserted. The track is selected.
- If you go to "CD/DVD" > Import Audio CD Track(s)..." the CD track list will open. By clicking "Copy selected tracks...", you can import selected tracks into the system and load them into the project window.

Working with Ranges

With ranges you can make a selection within a project that stretches over as many tracks as you want. The selection area is not limited to object borders and can be extended well beyond them.

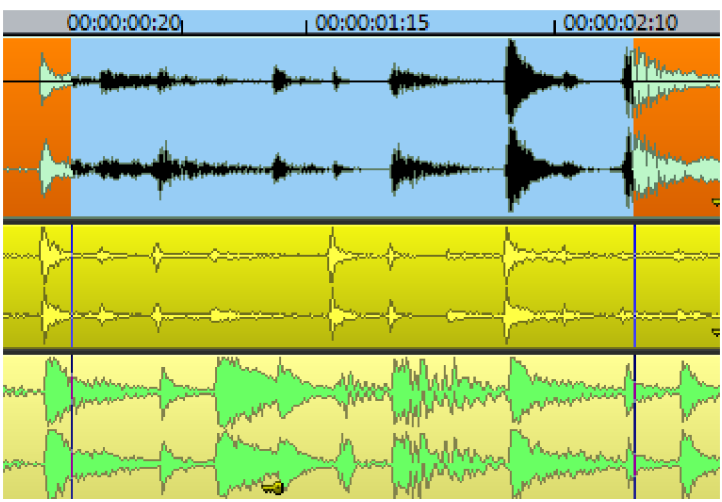
Selecting a Range

1. To select a range, set the mouse mode (view page 77) to "Range Mode" or move the mouse pointer to the top half of an object and press the left mouse button.

2. Move the mouse pointer within the object while holding down the mouse button. Now you can see an inverted rectangle between the starting point and the current mouse position.

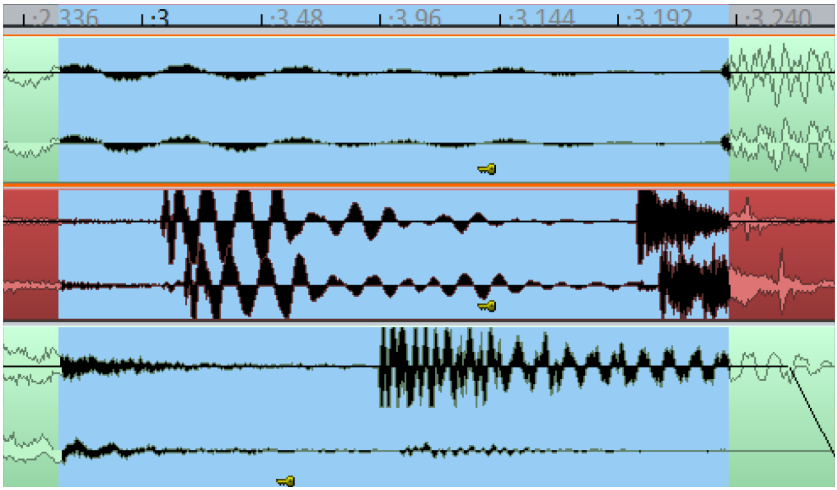
If you have activated the grid function, the range will automatically be adjusted to the set grid value.

3. Once you let go of the mouse button, the range will be selected.

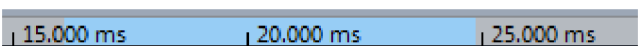


The playback marker always automatically remains at the beginning of the range, even if you can't currently see it.

4. To expand the range onto other tracks, click on the top half of the selected object again and drag the mouse down vertically while holding down the mouse button.



You can also select a range by dragging in the grid bar. This will then be indicated by a different color.



Double-clicking this area of the grid bar selects a range in the selected track; double-clicking again selects the range over all tracks, and double-clicking once more restores the simple grid bar selection.

It is important to note the following when doing this:

- Playback can be started before or within the loop.
- If you start playback behind a range, the loop button in the transport console will turn off.
- Range ends can be adjusted during playback as well.
- Click the range borders in the grid bar to position the playback marker at the playback borders either at the beginning or end of the range.
- The current playback range can be deleted by dragging to size 0.
- The current range can be deleted by double-clicking the grid bar outside of the range.
- The range remains in Loop Mode even if the playback marker is repositioned
- In Loop Mode the entire project is looped if you haven't marked a range.

Change Range Borders

You can change range borders by placing the mouse pointer on the edge of a range in the grid bar until it turns into a double arrow. Now you can reset the range borders by dragging in the horizontal direction.

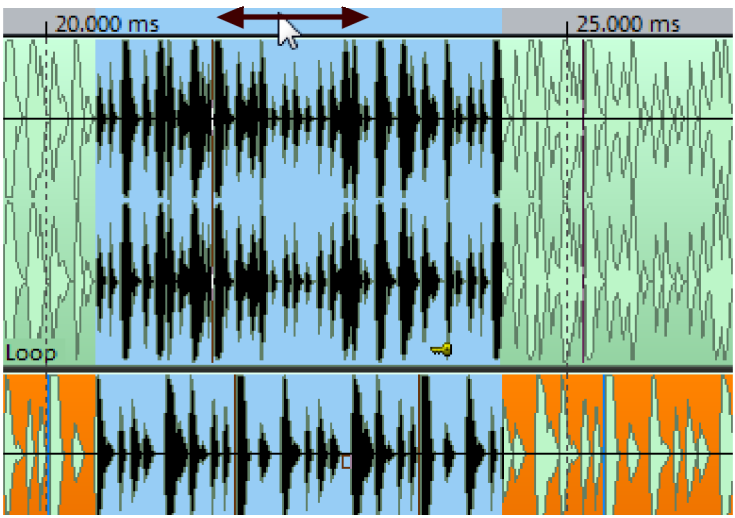


If the existing range stretches over several tracks and you only want to adjust one border (start, end, upper or lower border), click in the range and hold down the mouse button. Now drag the mouse pointer in the direction you want to change. As soon as the edge of the existing range has been crossed, the range edge will follow the movements of the mouse pointer. After you have redefined the range, you can release the mouse button.

You can also adjust the range start using the left and right arrow buttons. To adjust the range end you can use the arrow buttons while holding down the Shift key.

Horizontal Movement of a Range

Click with the left mouse button while holding down the Shift key inside an existing range on the grid bar. Hold the mouse button down and move the range in the horizontal direction.



Leaving a Range

To select a different range, click somewhere in the project other than the current range, and simply draw a new range.

Reactivate Range

To reactivate an earlier range, use key combination "Shift- + Back arrow" (or "Edit" > "Range" > "Reactivate last range"). By clicking on this command repeatedly you can restore the last five ranges. You can execute the same function by clicking on the button with the left arrow in the transport control.



Saving and opening ranges/Special range commands

Save selected ranges using the menu "Edit" > "Range" > "Store range“ or with Alt + function keys F2-F10.

Ranges can be recalled using "Edit" > "Range" > "Get Range" or with Ctrl + F2-F10.

Note: Alt + F4 is excluded since it is a Windows shortcut for closing windows. Similarly "Alt + F9" should not be used either, as it is used in Sequoia as a source destination edit command. However, you can redefine this keyboard shortcut any time you want via "File" > "Program Preferences" > "Edit Keyboard Shortcuts and Menu" (view page 688).

You can also save and rename ranges without any restrictions using Alt+F11.

All saved ranges are listed clearly in the Range manager (view page 200) and can be managed there.

Working with Ranges: Examples

Moving several neighboring objects together

1. Instead of selecting each object separately while holding down the "Ctrl" key, you can select a range that contains all objects.

2. Select the objects using the option "Object" > "Select Objects" > Select objects under playback marker/range" and move them in unison. You can do this faster and more efficiently by setting up a keyboard shortcut for this function.

Removing a section of a song from the project

If, for example, you want to remove a verse from a song, proceed as follows:

1. Select the verse by dragging on the grid bar to create a range and activate all of the contained tracks with a double-click.

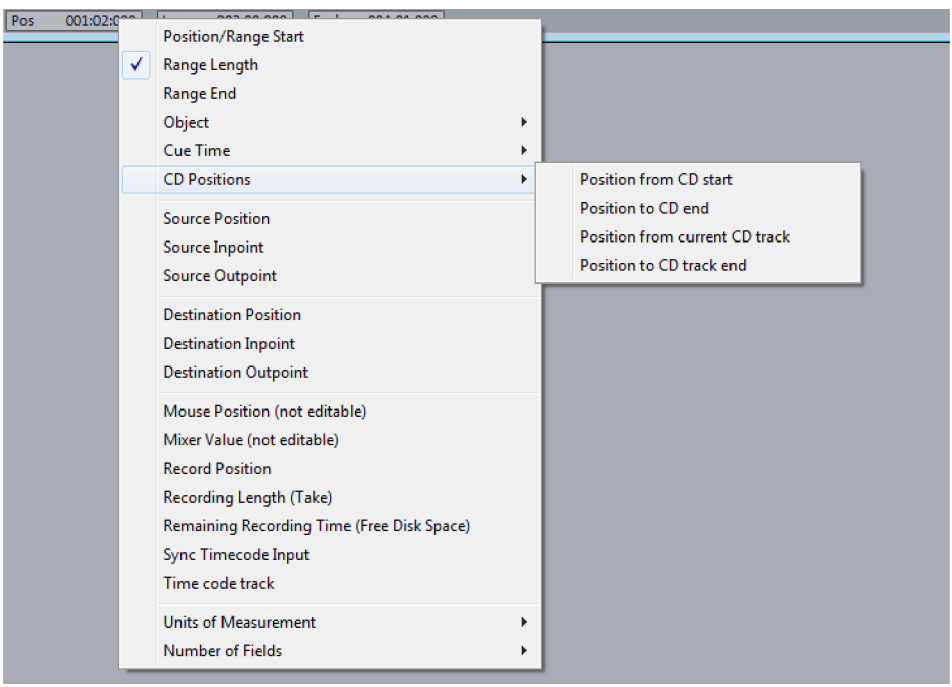
2. Now select "Edit" > "Delete" > "Ripple delete" to remove the verse.

Additional special functions for defining, changing and using ranges can be found in "Edit" > "Range".

All saved ranges are displayed In the Range Manager (view page 200) where they can be numerically edited and recalled. The Range Manager can be found in the docker or opened through "Edit" > "Range" > "Range Manager...".

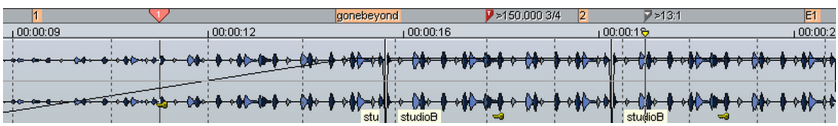
Under "Edit" > "More" you will find an additional dialog window called the Range Editor (view page 758) which can be used for editing the range start, range length and range end.

Other parameters such as "Pos" (Position/Range Start), "Len" (Range Length) and "End" (Range End) can be accessed by right-clicking. In the context menu that opens you can also display additional parameters in up to 5 fields, e.g. the current mouse position or the last edited mixer setting.



Working with Markers

Markers are specified positions in the project on the marker bar. Markers allow you to jump to specific points in the project. In a song arrangement, for example, it might be useful to place markers at the intro, verse and chorus, etc. This makes it easy to navigate through the project to these positions.



A project in Sequoia can contain any number of markers. The first ten number markers can be saved using the key combination "Shift + number key" on the corresponding playback marker position and directly opened again using the number keys.

If you want to rename a marker move the mouse pointer over its front edge.



This turns the mouse pointer into a double arrow. Now double-click. You can rename the marker in the dialog window that appears.

You can rename markers by going to "Play/Rec" > "Marker" > "Marker with Name...".



To delete a marker, select it by clicking its front edge, and then press the "Del" key on your keyboard. Markers can be moved by grabbing and dragging them - the mouse pointer changes into a double arrow (<->).

By using the object modes "Connect objects until pause" and "Connect objects on a track" together, the markers in the top arranger track can also be moved when objects are moved. Moving objects in the "Group objects" object mode simultaneously moves the marker independently of the selected track.

If you right-click inside the marker list or press the "Marker" button to the left of the marker bar, a context menu will appear that features the most important marker commands.

The Marker manager (view page 196) can also be accessed through "View > Manager > Marker manager". This is where you manage and edit the markers you set.

To stretch a range between any two markers, first click the first marker. Then click on the second marker while holding down the Shift key. This selects a new range.

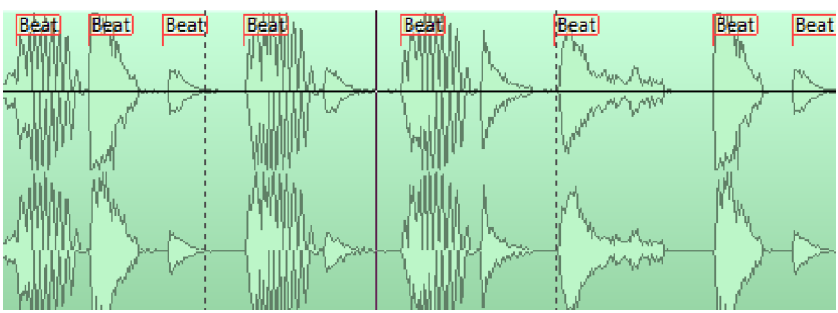
Use the function keys F2 and F3 or the shortcuts Alt + W/Alt + Q to quickly jump back and forth between the marker positions or select "Play/Rec > Move playback marker > Marker left/right" from the menu.

In addition to the standard markers, there are additional markers that can be used in Sequoia as well:

Markers in audio files are saved in the file (*.wav) as audio markers/beat markers and are available in this form in other applications as well. Audio markers/beat markers are linked to the audio material and are visible on the upper side of an audio object.

The purpose of the audio marker/beat marker is to mark positions within the audio material. This mark will therefore remain regardless of the positioning in the virtual project. Audio markers/beat markers can also be made visible in the view options ("Shift + Tab") in the "Objects" area by marking "Audio markers" with a check.

The audio markers/beat markers displayed in the virtual project object are identical to the markers in the corresponding audio file. If you want to place a new project marker in an audio file, as is done automatically when recording a new take, the audio markers/beat markers are visible at the same position in the audio material in all related objects.



Note: All time information for the audio marker/beat markers relates to time positions in the audio material and not to positions in the virtual project.

- **The triangular markers for CD burn functions**: CD track markers are displayed in red, CD subindex markers are green, and CD pause markers are blue.
- **Tempo markers** (view page 442) signify a tempo change at a specific position in the project. Double-clicking a tempo marker opens the "Tempo and Time Signature" dialog.
- **Bar markers** (view page 444) change the time signature starting at the marker position, e. g. 4/4 beat to 3/4.
- **Grid position markers** (view page 444) assign a specific musical position to a specific time position. This way, the beat grid and MIDI events can be easily synchronized with existing audio material.

Working with tracks

Select multiple tracks – group track controls

To select multiple tracks, choose the tracks by clicking the respective track numbers/track ranges while holding down the "Ctrl" key or the "Shift" key. The selected tracks are highlighted in the arranger, which indicates that they are grouped.

Alternatively, the select the desired tracks and then open the track context menu and select "Track properties > Group track controls".

If the status of a button or controller is changed now, then it will become apparent that the corresponding control elements of the other tracks in the group will also change. This applies to faders, panorama, EQ, AUX, mono, phase, solo, mute, record, MIDI/audio swapping, track color, track locking, Revolver track settings, bouncing settings, and for the selection of types of automation (track/object), plus the audio inputs/audio outputs. For the audio in and outputs, incremental functionality may also be selected, which distributes the in and outputs available in your system according to the selected tracks.

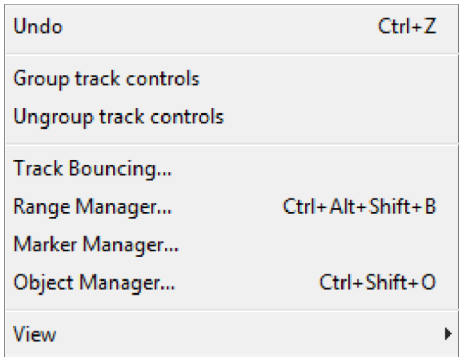
On the other hand, specific settings such as integrated plug-ins or the comments field remain as individual settings for the respective track and are not transferred to the group.

This group functionality also applies to the mixer. The multitrack selection function works additionally to previously added control groups (view page 377) of the mixer if the same control elements are used.

In order to remove a track from an existing multitrack selection, hold down the "Ctrl" key and click the track name.

In order to clear the group completely, click at any non-selected track. Alternatively, you may open the context menu in the track box and select "Track properties > Ungroup track controls to clear a group.

The context menu featuring commands to group or ungroup track controls can also be accessed by right-clicking in the track header area below the last track in the arranger.

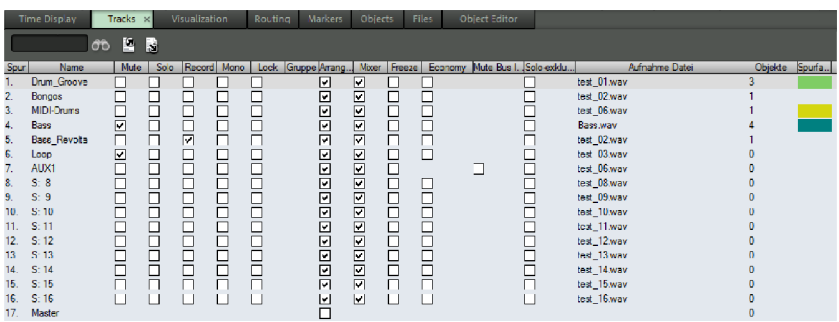


Note: Multi-channel selection in the mixer functions exactly like multitrack selection in the arranger, i.e. by clicking the track number/name while holding down the "Ctrl" key or "Shift". In order to clear a group of channels, click on a non-selected channel.

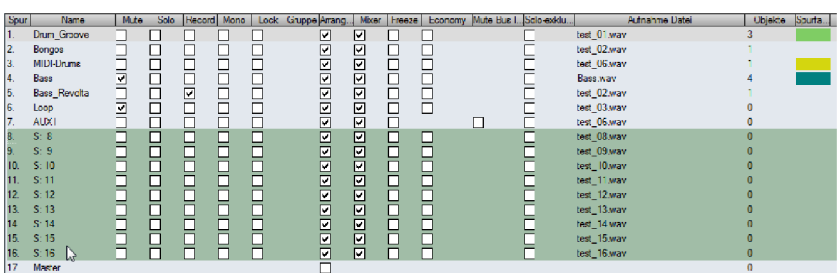
Hide tracks

If the project window contains tracks that are not being used, you can hide these tracks and mixer channels to create a less cluttered interface. The track manager can be used to do this.

1. Open the track manager ("View" > "Manager" > "Track Manager").



2. Use the "Shift + left-click" key combination to select the tracks you want to hide.



3. By clicking a track check box in the corresponding column "arrangement" and "mixer", you simultaneously remove all the ticks for the selected arrangement tracks and mixer channels.



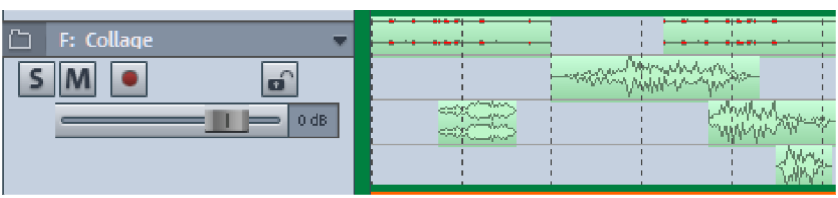
The tracks in the arrangement and mixer are now no longer visible.

You can reverse this process to make tracks and channels visible again.

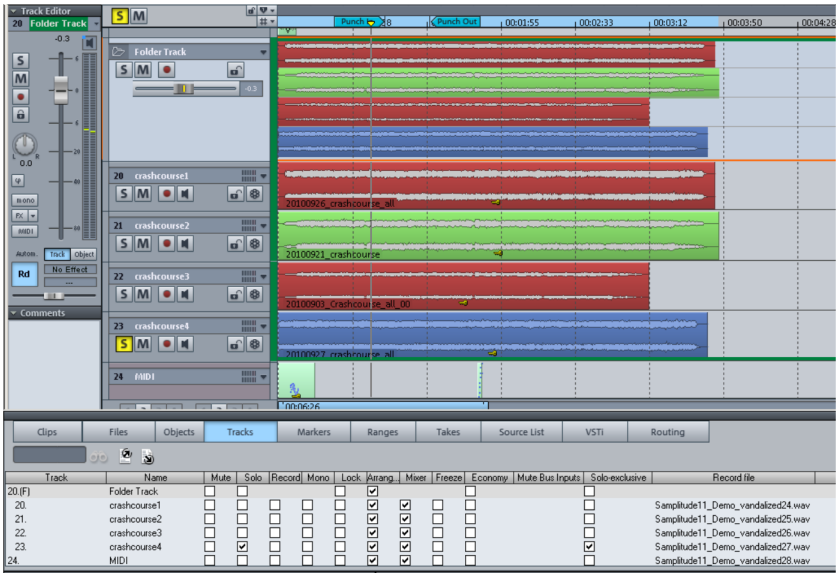
You can find out more about the track manager later (view page 194) in the manual.

Folder tracks

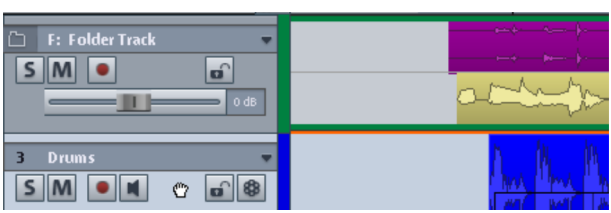
Folder tracks can help you keep a better overview of your arrangement by combining all tracks that belong together into one framework. The command "Track > Insert new Tracks > New Folder Track" adds a folder track in front of the selected track. If a range across several tracks is selected before creating the folder track, each of these tracks will be added to the new folder track.



The tracks in the folder track are displayed within a frame and the folder track number is appended with an "F" in the track manager.



If you want to delete a folder track, a dialog will ask to confirm if you really want to delete all of the tracks contained in the folder as well. You can drag & drop tracks into the folder track by clicking on the track name or an open area in the track header. The mouse pointer turns into a hand. Now you can drag & drop the track into the track header of the folder track.



Individual tracks can be removed from the expanded folder track again in the same way. You can also use the track menu to copy folder tracks.

The following functions are applied to all tracks found in the folder track:

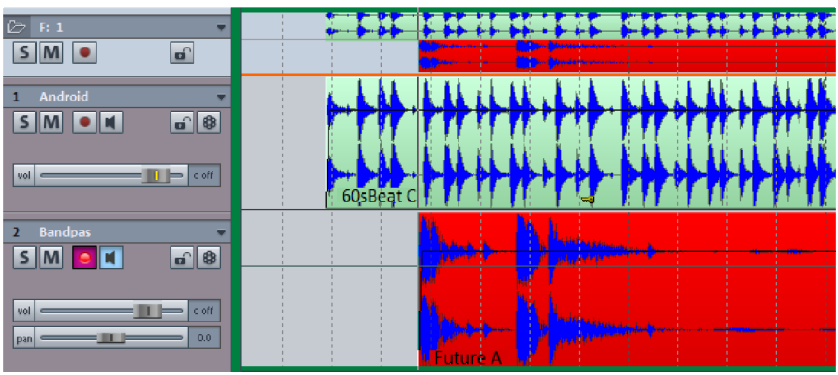
- Mute, Lock, and Solo

All tracks in the folder track are displayed when the folder track is open. Each track is given a border with the track color of the folder track. When minimized, the tracks found in the folder will be hidden in the arranger. They will still be visible in the mixer. Tracks of a folder track are marked with the border color of the folder track in the mixer. Normally all tracks can be seen in the folder track along with their objects in a minimized state. Range selection in the folder track is carried out across all tracks. You can use this function for range-based editing across all tracks in the folder track.

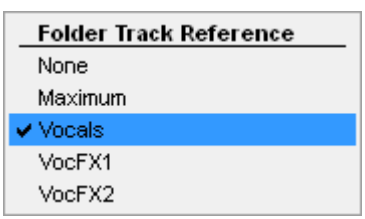
Reference Track in the Folder Track:

Right-clicking the track header of the folder track allows you to select a contained track as a reference track. The objects on this track are displayed in the folder and can be used for simultaneous object-based editing operations.

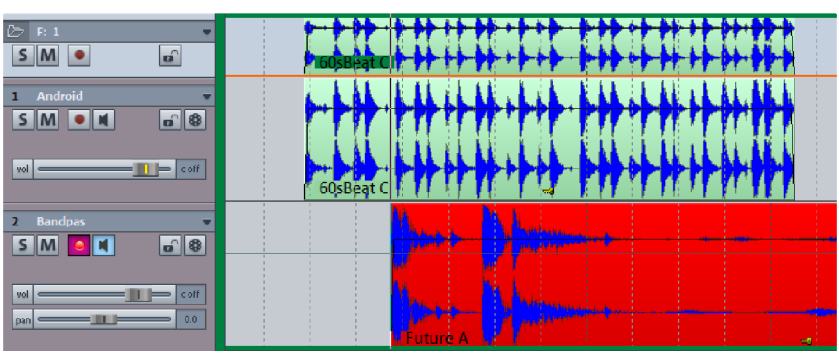
1. Create a folder track ("Track > Insert new Tracks > New Folder Track") and drag & drop several tracks into the folder.



2. Choose one of the tracks in the folder as a reference track (right-click on the folder track header).



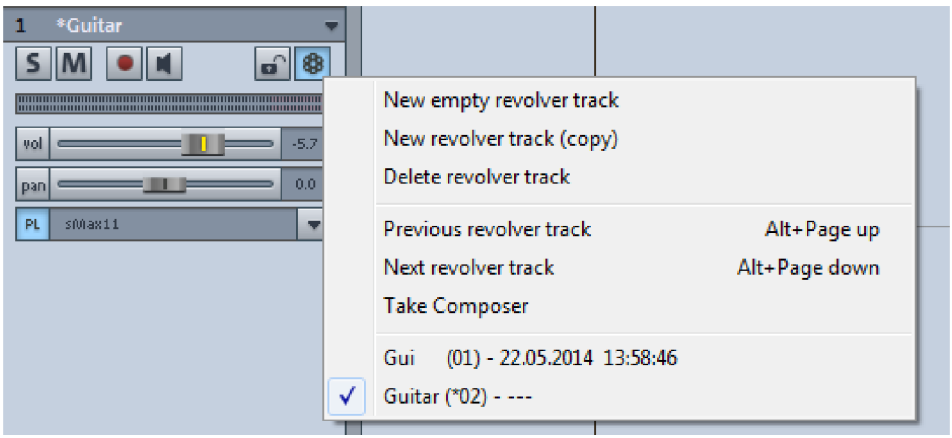
3. Instead of a symbolic display of all the contained tracks you will now see the object of the reference track in the folder track.



Revolver tracks

In Sequoia it is possible to work with revolver tracks. Revolver tracks can be used to compile objects differently for each individual track. The combinations created for the corresponding track can be accessed at any time in the revolver tracks menu.

Open the revolver tracks menu by holding down "Ctrl" and right-clicking the track name in the track header. There is a button for this beside the lock symbol in the track header.



Before rearranging the track objects, select the option "New revolver track (copy)". The track objects are copied as a new revolver track and a asterisk will appear before the track name. The original objects can now be edited or repositioned to produce a different version of the track.

The context menu can also be used to create a new, empty revolver track, to delete the current one, or to display the previous or next revolver track (keyboard shortcut: "Alt + Page up" or "Alt + Page down"). The command "Delete revolver track" deletes the current revolver track and displays the previous revolver track.

In the lower section of the dialog you can select which revolver track you would like to hear next from the list of existing revolver tracks. The individual revolver tracks are numbered and are listed by creation date and time.

To edit or view existing revolver tracks in the overview, open the Take Composer (view page 203) for a particular track. This editor enables revolver tracks and object takes to be edited, and the created revolver tracks are displayed under the individual takes.

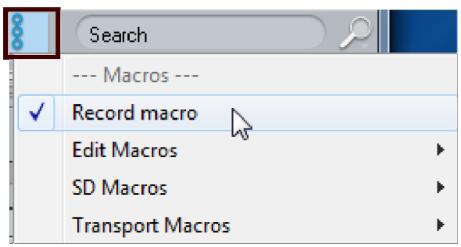
Macro functionality

The macro function gives you the option of creating and playing command orders. This is how you create a macro:

1. To record a macro, select "Record macros" by clicking on the corresponding icon in the toolbar,



the magnifying glass in the command search bar



or by going to "File" > "Program settings" > "Record macro".

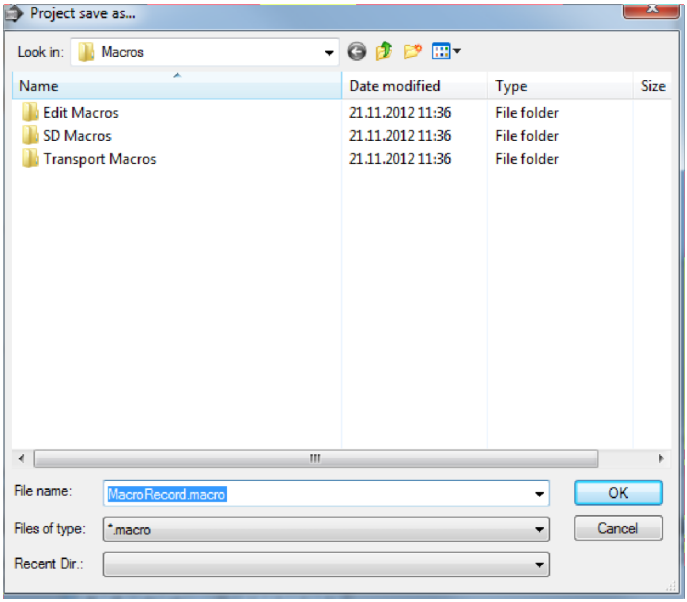
2. Now a tick will appear next to the command. This shows that the recording is activated.

3. Execute command orders that you would like to record as Macro.

Example: Place the playback marker over the selected object and press the keyboard shortcut "T" to split the object and then use the keyboard combination "Ctrl + D" to duplicate and move.

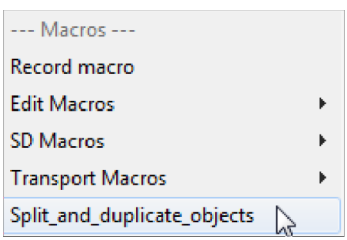
4. If you have executed your command orders, click on "Record macro" again. If the checkbox next to the command disappears again it means that the recording process is complete.

5. The save dialog will open automatically directly after the recording has ended. Give the created macro file a name then save it.



Note: Only simple command orders can be marked without settings dialogs in the described manner. For more complex macros we recommend editing the macro files in a text editor (see below). This also allows the use of other functions, e. g. crossfade editor functions. A complete list with short descriptions of the parameters can be found in the **file "MacroCommand.list**" in the program folder.

6. The created macro can now be opened using the command bar context menu.



Macros can also be called up in the command search bar by entering #run:

#run MacroName (in our example #run Split_and_duplicate_objects)

7. In the Macro menu of keyboard settings/menu settings you can also assign a keyboard shortcut (view page 688) to any macro.

Creating and editing macro files in a text editor

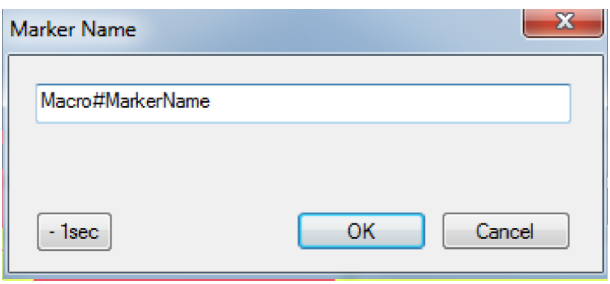
Macro files are simple text files. To create a macro file open the text editor and enter the desired command sequence from the „MacroCommand.list“ then save the file in the .macro format in the folder „Program Data“ > „Macros“.

To edit existing macro files, open them and insert additional command sequences.

Executing macros at project markers

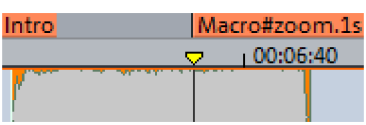
To execute macros independently from their playback position, you can use project markers.

To do this, set the playback marker at the desired position in the arrangement (view page 128) and go to "Playback > Marker > Marker with Name...", e. g.:



Use the macro markers like this: „Macro#MacroCommand“, e. g. „Macro#zoom.1s

The marker will appear in the timeline:



Each time the marker is reached, the corresponding macro will be executed; in this example, the zoom factor jumps to one second, meaning that the displayed screen view is exactly one second starting from the marker. You can also create named markers for all other macros listed in the MacroCommand.list in the program folder.

To execute macro files (e.g. "test.macro") in addition to single macro commands with a sequence of commands, name the marker in the following way: „Macro#run test.macro“.

Command search bar

In the command search bar, macro commands can be executed by first typing in # as in the following:

#tfind TrackName

#tfilter TrackName (Reset using #tfilter)

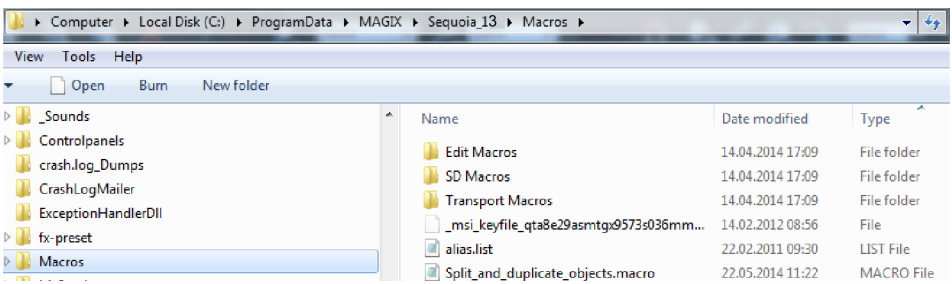
#mfind MarkerName

#ofind ObjectName

#rfind RangeName

#timeedit.startpos ms -1000

A complete list with short descriptions of the parameters can be found in the file "MacroCommand.list" in the program folder. **Program Data folder > Macros\alias.list**



In this file you can assign your own names to commands, e. g.

„zoom+“ and „zoomin“ for zooming in:

[Aliases]
zoom+=zoom.in
zoomin=zoom.in

or

„zoomout“ und „zoom-“ for zooming out:

[Aliases]
zoomout=zoom.out
zoom-=zoom.out

Note: These names cannot be used as macro commands in macro files.

Keyboard shortcuts and practical tips

To make working with Sequoia faster and more efficient, it's a good idea to familiarize yourself with the most important shortcuts. We have created the following list for this purpose and have added some practical tips for work in the project window.

- To start and stop playback, press the spacebar.
- To stop at the current position, press the 0 key on the numeric keypad (or click the play arrow in the transport console a second time).
- To set the playback marker, click on the desired position in the timeline above.
- To move the playback marker, use the left and right arrow keys.
- In order to jump to the most recent position, press the backspace key.
- To move the VIP under the play cursor, use the keyboard shortcut Alt + Arrow left / Alt + Arrow right.
- To create a range, use the mouse to draw in the timeline.
- To move a range, move it with the mouse while holding down the Shift key.
- To increase the size of a range, drag the end of the range using the mouse.
- To reactivate a deactivated range, use the shortcut keys "Shift + back space".
- To move an object, click on the lower half and drag it into its new position.
- To jump to the object edges with the playback cursor, select the shortcut keys Ctrl + Q / Ctrl + W.
- To create markers at the play cursor position, select "Shift + 1... 0" in the letter key block of your keyboard.
- To jump to a marker position with the play cursor, select the corresponding number "1... 0" in the letter key block of your keyboard.
- The keyboard shortcut "A" produces a range that covers everything.
- To set a range between two markers, double-click anywhere between the markers in the marker list.
- The keyboard shortcut "Shift + mouse movement" can be used to move the range.
- Drag the range border to increase the range size to the left or to the right, e. g. to adjust a loop.
- Use the "Tab" key to switch between two display modes in virtual projects. These modes can be edited using "Shift + Tab".
- The option "Objects" > "Lock Objects" can be used to protect objects from unintentional moving.
- If you press the right-mouse button in various areas of the VIP - e. g. above an object, above the Record and Play buttons, or above the scroll bars - a context menu with useful functions appears.
- Press "T" to split an object at the position of the playback marker.
- "Shift + mouse-click" can be used to save the current zoom level.
- The volume and panorama curves can be linked to the objects below by pressing the "Link curves with objects" button in the mouse mode bar. This allows objects and their corresponding curves to be moved together.

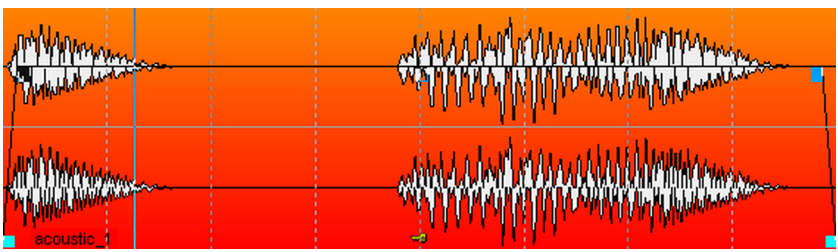
- Numerical values in dialog fields like the display of the playback position in the transport window can be changed by clicking the corresponding field and holding down the left mouse button while moving up or down. This turns the mouse pointer into a double arrow.

Methods for Working with Objects

Object selection

Selecting objects and clearing selection

Each object can be selected with the mouse. To do this, click the lower half of the object with the left mouse button. The selected object will be recognizable by a different background color and the five handles that appear on its corners and in the center of the object.



This selection can be cleared again by holding "Shift" and clicking the lower half of the object or by clicking beside the object.

Select multiple objects; clear selection of multiple objects; reverse selection

Objects can be selected in various ways. The selection can be expanded, reduced, or reversed at any time.

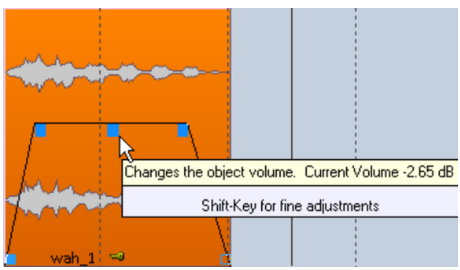
- Select multiple objects by holding down the "Ctrl" key at the same time.
- Left click on a free area in the lower part of a track and hold down the mouse button. Now drag the mouse to open the object lasso. All objects in the lasso are selected when the mouse button is released.
- If there is no free space available, access the object lasso via "Object > Select objects > Object lasso" (Keyboard shortcut: Ctrl + Alt + L). Now select individual objects by dragging the mouse.
- Hold down the "Shift" key and click to select all of the objects between the first and last selected objects.
- The complete object selection may be cleared by clicking anywhere in the lower part of a track. The complete selection will not be cleared if you click a currently selected object; clicking an object that is currently selected will remove that object from the complete selection..
- The current object selection may be reversed (invert). To do this, select "Object > Select objects > Switch selection".

Standard operations

- Duplicating one or more objects can be done using drag & drop while holding down the Ctrl key.
- Copying and inserting objects works with the usual keyboard shortcuts Ctrl + C and Ctrl + V.
- Using the keyboard shortcut Ctrl + Alt + V you can insert a copied object at the playback marker position while simultaneously moving subsequent object back on the timeline. ("Ripple").
- Object can be moved using drag & drop or a range of keyboard shortcuts ("Object" > "Move objects").
- By clicking the T key, a selected object is split into two objects at the play cursor. To undo object splitting at a later stage, move the objects together and select "Heal/Unsplit Object" under "Object" > "Edit".

Object Handles

If an object has been selected, five handles appear at the corners and at the center of the object.



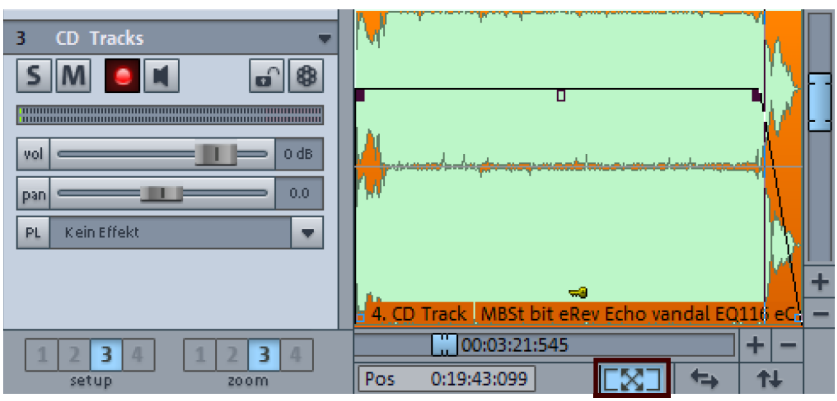
- The length handles are at the bottom left and right corners. Use these to adjust the length and starting time of an object within the audio file. The maximum length of an object depends on the audio file that the object is accessing.
- The fade handles (top left and top right) are used to fade an object in or out, and different curve shapes are possible. You can use the Object Editor to precisely adjust the curve shape.
- The volume handle is at the top center of the object. You can use this to adjust the object volume. The volume level is displayed above the tool tips.
- The object name is visible next to the left length handle on the right hand side and the key symbol to its right may be used for locking objects (view page 152).

Trimming objects

"Trimming" refers to trimming objects to a certain specific range. Make sure that the selected range does not exceed the object borders. Once a range is selected, right click the object and select "Trim objects" (shortcut: Ctrl + T). The selected object will now feature the selected length. You can also trim multiple activated objects in the same way, even if they don't have the same starting and end points.

Object zoom

If you use the "Zoom Into Selected Objects" button (also can be opened in "View" menu) the project section jumps to the selected object. If you click this button again, Sequoia will display the original section again.



Overlapping objects

Only one object can be played per track (channel) at any given time.

If an object is moved over a bordering object, it will cover the preceding object. The invisible part of the subsequent object can be made visible and audible again by moving the first object. To create a crossfade between two consecutive and partially overlapping objects, apply the "Crossfade editor" function in the "Edit" menu.

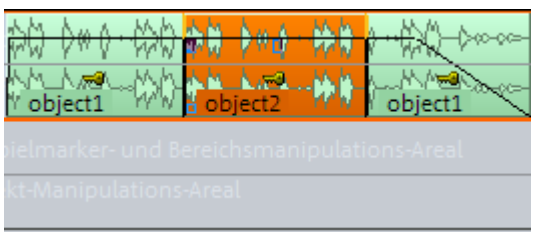
If "Auto Crossfade" is active, a crossfade will be created automatically as soon as the overlapping object is faded in.

Overlapping objects via "Ctrl + V"

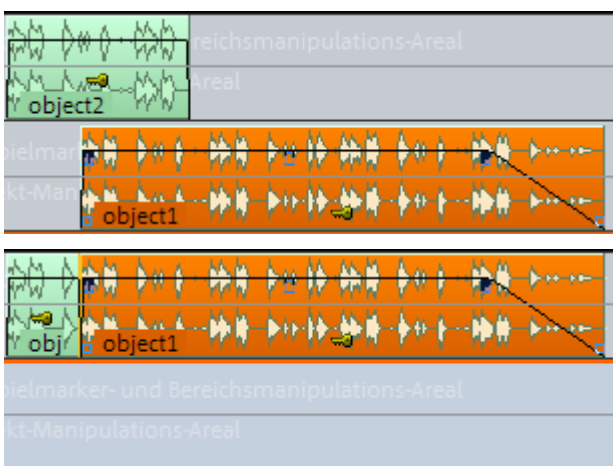
If you drag a shorter object completely into a longer one, the shorter object will replace the longer object from the start of the shorter object.



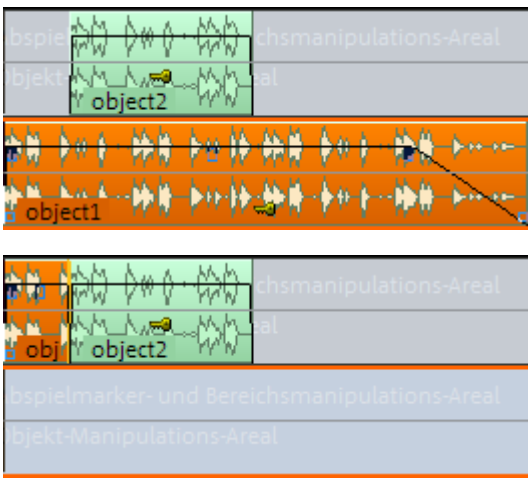
If you hold the keyboard shortcut "Ctrl + V" before releasing, then the shorter object will also replace the longer object across the entire length of the shorter object, but the end of the longer object that extends past will remain.



Now try it the other way around and drag a longer object over a shorter object; the shorter object will be replaced from the start position of the longer object if its start position lies behind that of the shorter object.



If the start position of the object being moved lies before that of the target object, then the target object will remain complete. Only the part of the object to be moved that is present before the start position of the target object will be added to the target track.



If you hold down the keyboard shortcut "Ctrl + V" before releasing, the object being moved replaces the target object across its entire length.



Group objects

Select all objects for the group.

Now either select the "Group" symbol in the toolbar or right-click on one of the selected objects and select "Object" > "Create Group" from the context menu. The selected objects now form a group.

To split all objects again, click on the "Ungroup" symbol in the program toolbar or right-click on one of the selected objects and select "Object" > "Ungroup" from the context menu.

"Object" > "Groups" > "Temporarily exclude object from group" command removes an object from an existing group according to when it was last clicked. By clicking on

an object again and repeating this function, the removed object is moved back into the group.

Shortcut: Shift + Ungroup button

"Object" > "Groups" > "Temporarily exclude all objects from group" temporarily ungroups all objects from the group. In this case, the "Ungroup" button will blink. If the function is accessed again or if the blinking buttons are pressed again, the groups will be restored and the button will stop blinking and return to inactive status.

Keyboard shortcut: Shortcut: "Shift + Alt + Ungroup"

Saving ignores the temporary condition, but saves the original grouping.

If group colors are activated in the view options ("Y" key > view options), a different color will be assigned to each group. This will ensure that you can easily distinguish the groups from each other. You can also activate the display of the group numbers in the VIP display.

Linking objects

This is how to link objects in a track:



Click the button "Link all tracks" in the mouse mode bar. All of the sequential objects on the timeline will now be linked with the selected object.



Click the button "Link one track" in the mouse mode bar. All of the sequential objects on the timeline will now be linked with the selected object.



If you now click "Normal object mode" in the mouse mode bar, all objects will return to their original modes.

Pressing the "K" key temporarily toggles the object mode. This activates the other selected mode for as long as the key remains pressed.

Gluing objects

If you want to combine two objects into one, select both objects and right click on one of them. In the context menu that opens, select "Glue objects". This creates a new file that is saved in the "FreezeData" folder in the project folder. If you uncheck "Freeze" in the object editor of the new object, it will be separated into its old components.

Note: In order to keep mono objects from changing into stereo objects during gluing, activate the option "System options -> Effects -> Resampling/bouncing -> Keep mono if possible".

Looping objects

Looping repeatedly plays an object or object range. You can change the length of objects by dragging out the length handles.

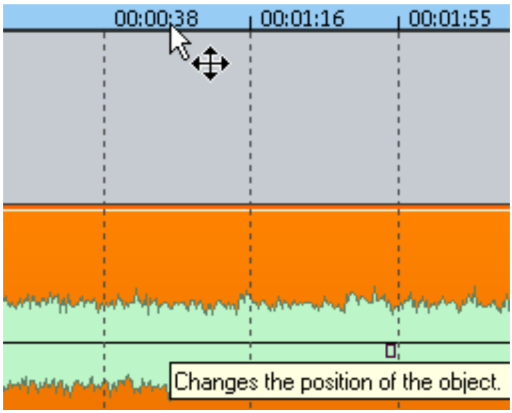
An object is normally played back between the object start and end. A loop object also starts from the beginning of the object, once the loop starting point has been reached only the set loop range will be repeated until the object end.

The easiest option to loop is to open the object's context menu and choose the option "Create looped object" (shortcut: Ctrl + L). The entire object is treated as a loop. If you now drag the length handle at the back to the right, you will see a vertical dash at the previous object end. For this marking onwards the object will be played back again, it is looped and can be stretched out to the right.

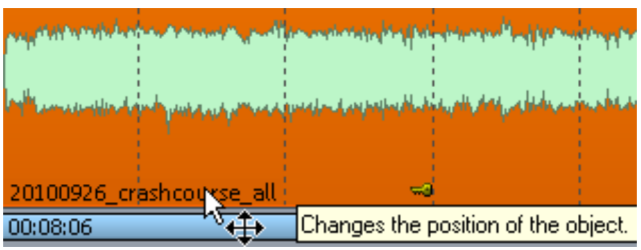
An object loop may be edited in far more detail by using the object editor (view page 174). In Loop Mode you can set and customize the loop start, loop length as well as the loop end within the object. Skilled looping omits time-consuming cutting.

Moving Objects

- A simple move: Left click an object in the lower half of the object and hold down the mouse button. The object will follow the mouse movement. Release the object at the desired position (drag & drop). You can also move an object to another track. In order to move an object to the left or right, use this keyboard shortcut: "Ctrl + Alt +Shift + Arrow left/right".
- To move multiple objects at once: Select the desired objects as described above. Click and drag one of the selected objects using the mouse to move all selected objects together.
- To preserve the time position while moving objects to different tracks: Select the objects as described above. Press the "Shift" key and move the objects to the desired track. The keyboard shortcut for moving a track up or downwards is "Ctrl + Alt + Shift + Arrow up/down".
- Moving objects upwards over the visible section: To move objects vertically upwards across the visible section, drag the objects across the section in the arranger. Move the mouse pointer across the top track in the section to the grid/marker bar; after a moment, the arrangement will scroll upwards so that the selected objects can be moved to the tracks above.



- To move selected objects downwards across the visible section in the arranger, hold the mouse pointer to the lower horizontal scroll bar; after a moment, the arrangement will scroll downwards so that the selected objects can be moved to the tracks below.



Moving Objects in Steps

- In "Object > Move Objects > Object Move Step1 (& Step2)" there are a number of commands for specifying individual keyboard shortcuts in the system options.
- To change the step widths 1 and 2, go to "Objects > Move Objects > Object/Fade Step Settings".
- Another option is to select the objects you want to move and open the Object Editor. In the Fades section click on the right or left arrow in the Position/Length dialog. The objects now move at the set step width.
- For step-by-step moving of an object you will also find the corresponding buttons on the range list. Go to "View" > Toolbars > Range bar" to make this visible. Buttons are also provided for moving the object's start or end, or the entire object.

Moving to a defined position

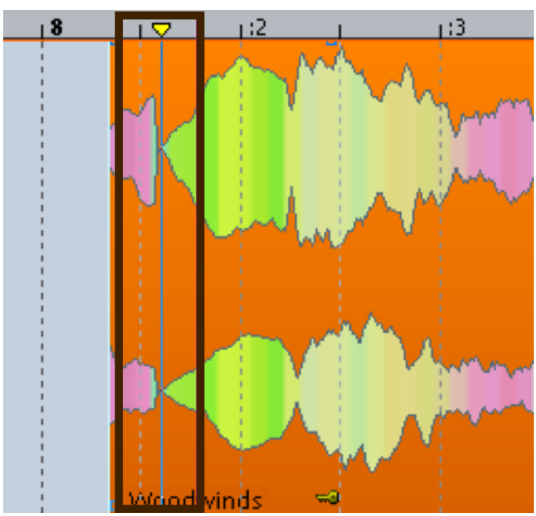
The following options are provided for moving an object to a specific position:

1. Right click on the object you want to move. In the context menu that opens, click "Objects/move crossfade/edit -> Move object...". A window opens to enter the value of the new starting position in samples, milliseconds, SMPTE time, or beats.
2. Open the object editor to enter the desired position under "Object start".
3. If you want an object to start at the position of the playback marker, right click on the object you want to move. In the context menu that opens select: "Object -> Edit/move crossfade/edit -> Object to playback marker position".

Move by snap point (Hotspot)

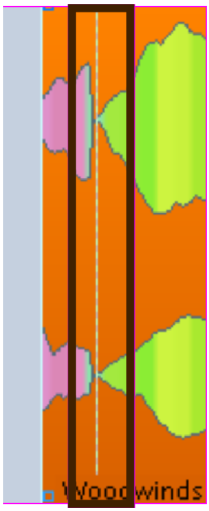
Sometimes the object beginning in your recording does not match the beat in the arrangement. This may happen when a recording contains an upbeat, a leading note or a snare pickup. Working with hotspots (snap points) is helpful for moving the selected object within the arrangement to fit the musical context.

1. Place the play cursor at the position where you want to create the hotspot for your selected object. This is often a position in the object at which a beat or start of a range should fall.

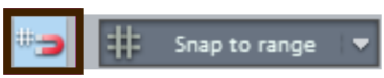


2. Create an object hotspot by selecting the object and entering the shortcut "Shift + H".

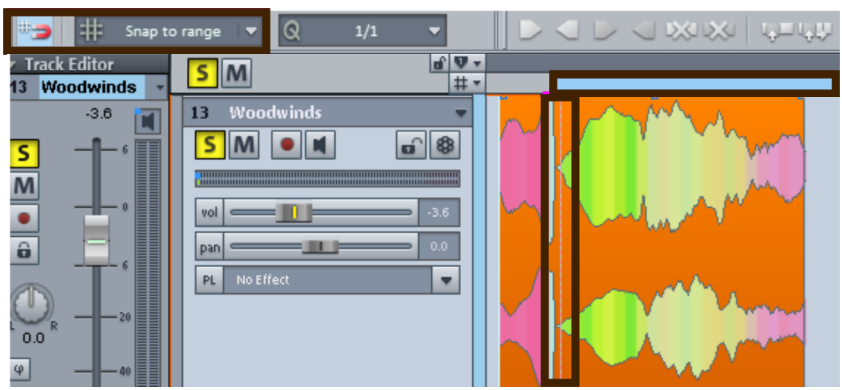
A vertical marking line should appear in the object. This line represents the newly created hotspot.



3. Activate the grid.

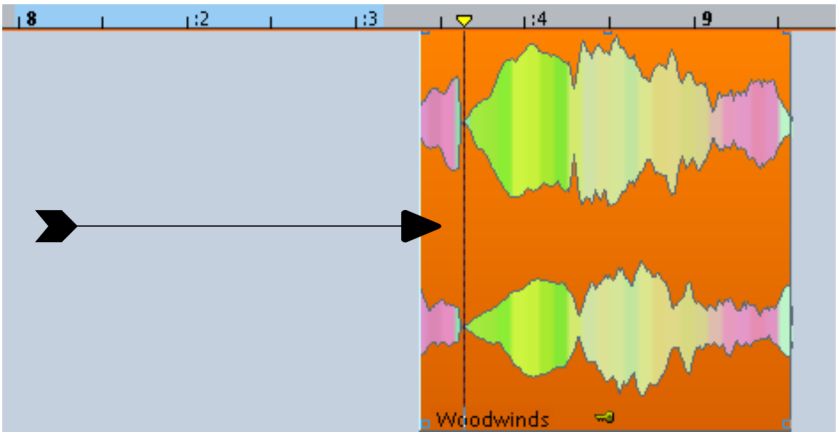


Move the object horizontally until it snaps in. The hotspot of the object is now placed at the respective grid selection. In the example below, this is the start of a range.



We have applied the grid range here. You can also use the object grid or beat grid to align the object at the hotspot.

4. Place the play cursor at a new position. You can make the selected object and its hotspot jump to the new position by using the function "Object hotspot to play cursor position" (shortcut: Ctrl + Alt + P).



5. To delete a hotspot, select the corresponding command in the object menu.

Locking Objects

To protect an object from being moved unintentionally, click on the key symbol on the object's lower border. This function corresponds with the "Lock" command in the object editor.

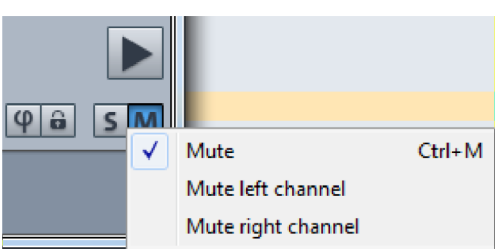
Clicking on the key symbol again unlocks the object.

Multiple selected objects can be locked with a mouse click. The menu option "Object > Edit > Lock Objects" applies to single and multiple objects. If you want to lock all objects on a track, click on the lock symbol in the track overview box or Track Editor.

Locking to prevent changes to fades, length, or deleting can be set in "Object > Edit > Lock Definitions...". You can temporarily disable the lock function by holding down the "Alt" key while clicking objects.

Mute objects

You can mute the activated objects using the "Ctrl + M" shortcut or with the mute button in the object editor.



You also have the option to mute either the left or right channel.

Note: If you've selected several objects and want to mute them collectively, make an object mute and use the "Apply to all" checkbox to apply this to the group.

Change waveform view

The waveform view can be adjusted in "View -> VIP display -> Define...".

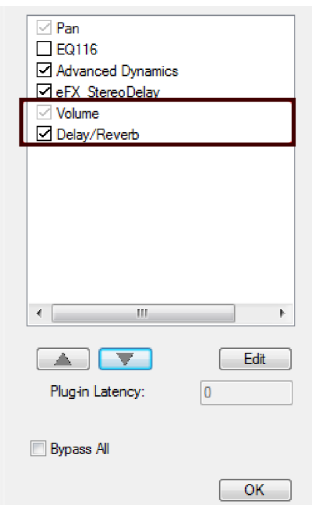
Detailed information about the display options is available via the menu reference under "VIP display (view page 695)".

Change the size of the waveform by holding the "Shift" key and rolling the mouse wheel.

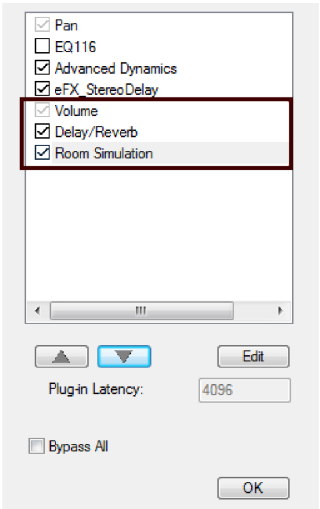
Zoom into a waveform at the position of the playback marker by pressing the "Ctrl" key and simultaneously scrolling with the mouse wheel.

Reverb in the object

Reverb times fade away if the Reverb effect in the routing is positioned after the "Volume".

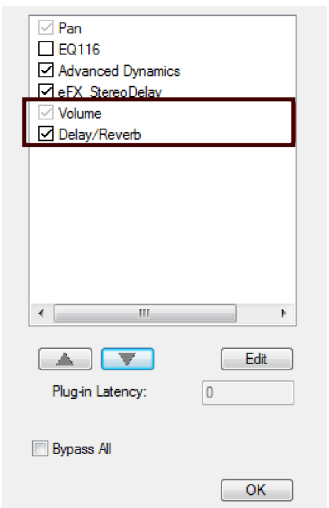


For reverb times with the Room Simulator, change the effects sequence so that "Volume" is placed before Room Simulator.



The reverb time depends on the option "Maximum reverb time for objects without fadeout (sec.)", which is located in the playback options (view page 830).

To ensure that reverb does not continue past the fade out, but rather that it breaks off, arrange the object's parameter "Volume" in the FX/routing dialog after the effect.



Replacing an audio file below the object

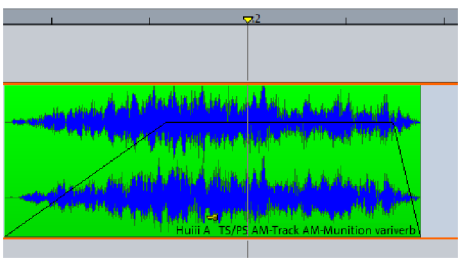


Open the object editor and switch to the "Fades" section or to "Max" view by right-clicking in an open area.

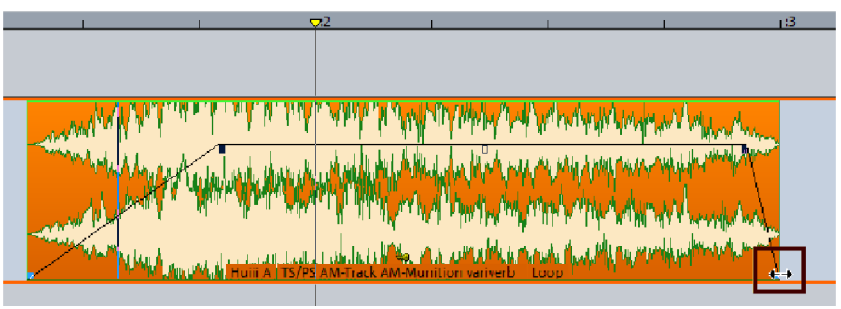
Now click the folder symbol in the object editor to the right of the name of audio file. You can select a new audio file from the dialog that follows.

Moving Audio Material Below the Object

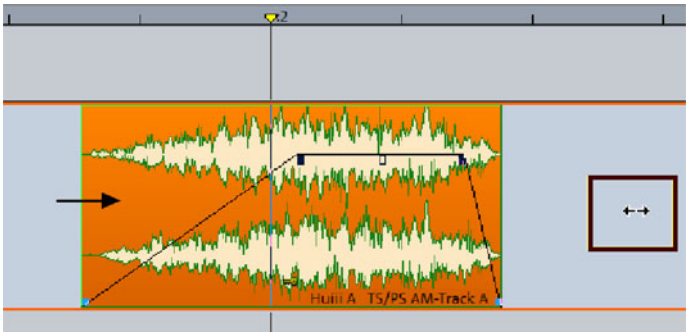
By pressing the right "Ctrl" button and clicking the lower part of the object, you can move the audio material below an object by clicking it. To do this, excess material must be available within the defined object borders, i. e. the audio file must be longer than the object that it is connected to.



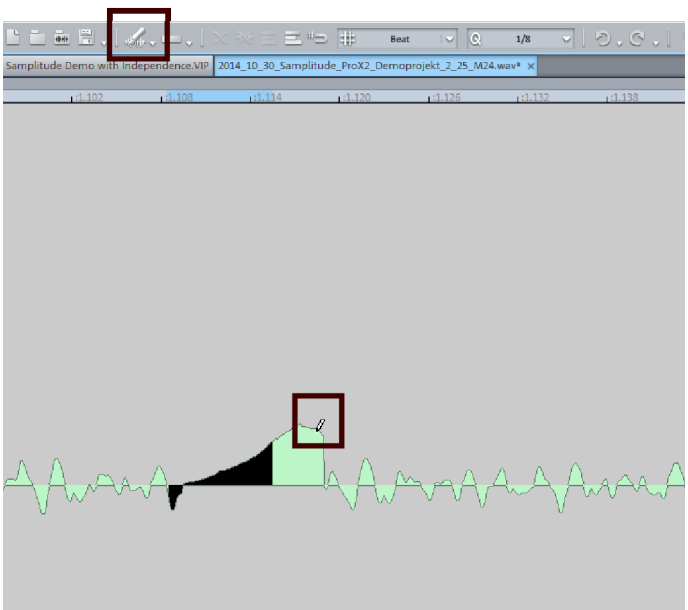
If the right "Ctrl" key is pressed and the length handles are dragged out on the left or right borders of the range, the adjacent object edge will be maintained while the other edge moves with the mouse.



If you hold down the right Ctrl and Shift keys and then drag the left or right length handle on the object to the start or end position, the edge will remain while the adjacent edge moves in sync with the mouse movement.



Drawing Waveforms with the Pencil Tool

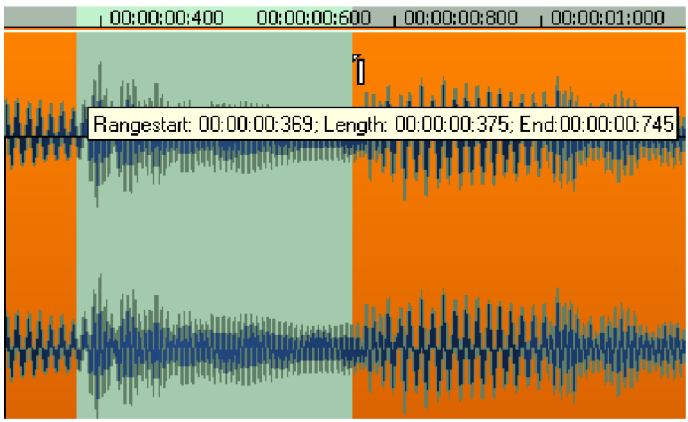


In "Wave editing (view page 62)" mode, use the pencil tool to draw or modify the waveform of a file. Waveform display shows the suitable zoom level, and the mouse pointer becomes a pencil. Changing the waveform in the wave window is useful if you want to edit very short impulse disturbances manually.

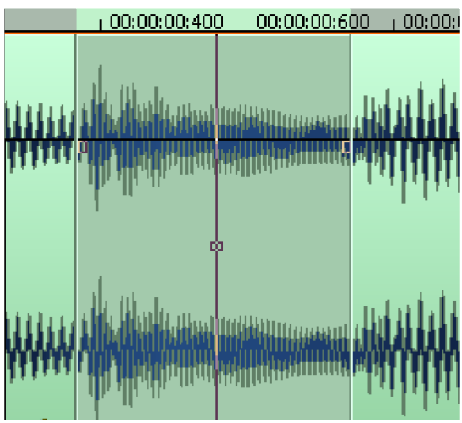
Pitch Shift-/Time Stretch-Mouse Mode: Object Handles

With the object handle you can use the Pitch Shift/Time Stretch mode to directly define the object length/playback speed and pitch. In our example we want to adjust the pitch and length of an individual bass sound.

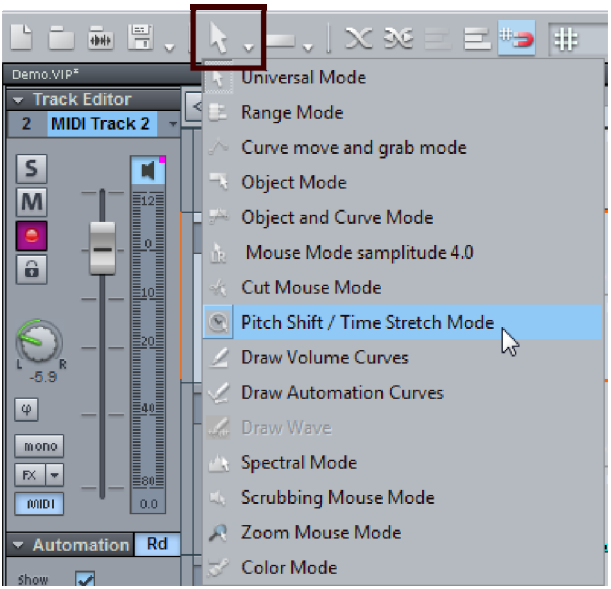
1. Zoom in to the place you want to edit. Now create a section above the passage where the sound you want to edit is located. Then drag the mouse horizontally in the upper half of the object.



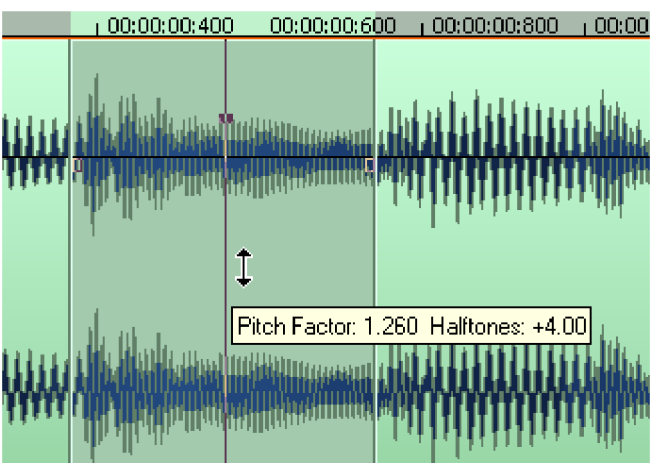
2. Using the Shortcut "T" cut the object at the range edges.



3. Click on the clock symbol (Pitch shift/Time Stretch mode) in the mouse mode menu.

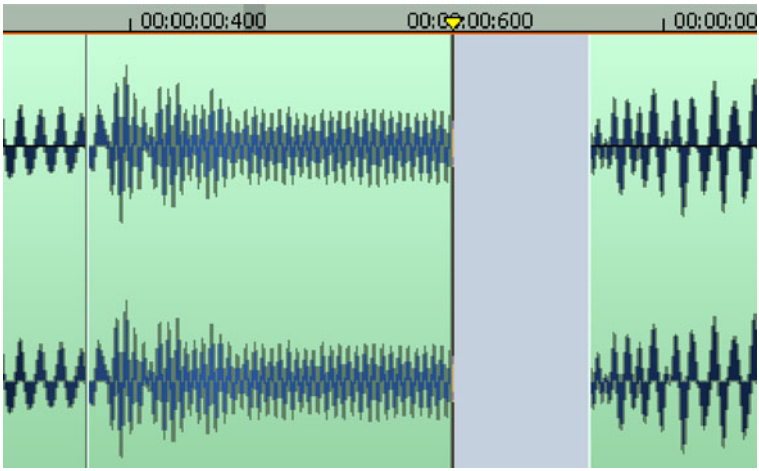


4. Next we want to adjust the pitch of the cut object. To do so, we simply click on the object. Now a handle will be visible in the middle of the object. If you hover the mouse over this handle for a moment it will change into a double arrow. Drag vertically to change the pitch of the object. Any changes made to the pitch will be displayed in positive or negative pitch factors, in half tones.

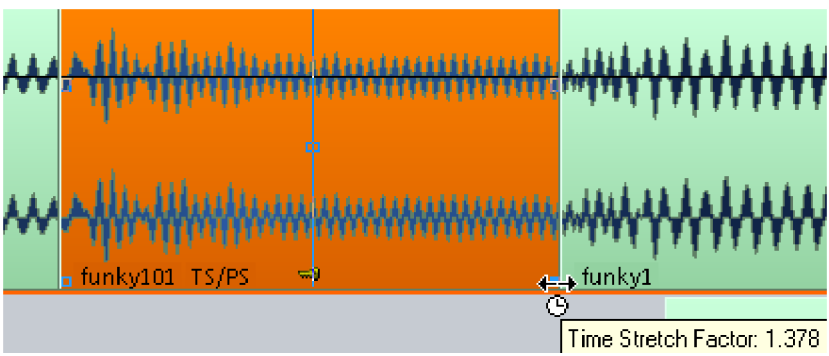


Holding the "Shift" key will allow to make more precise adjustments.

5. Since the bass tone dies away a bit early, we want to stretch the tone before the release phase To do so, we will set the play cursor to the beginning of the release phase, split it using the keyboard shortcut "T" and delete the separated object.



6. Close the gap by dragging at the bottom right-hand corner of the object. The timestretch factor is displayed. If you activated the object grid, the stretched object docks directly onto the subsequent object.



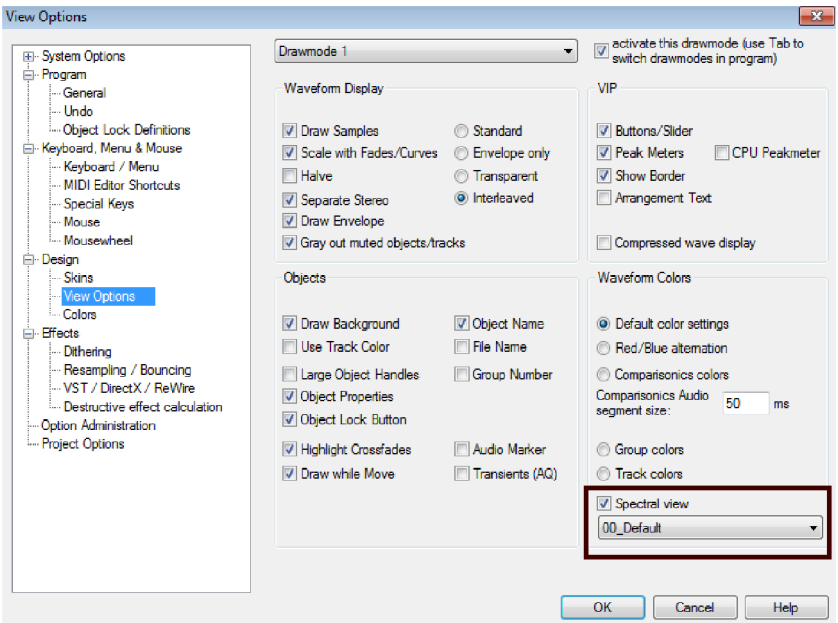
You have just stretched the bass tone without release phase, until the next tone begins.

Editing Objects in Spectral Mode

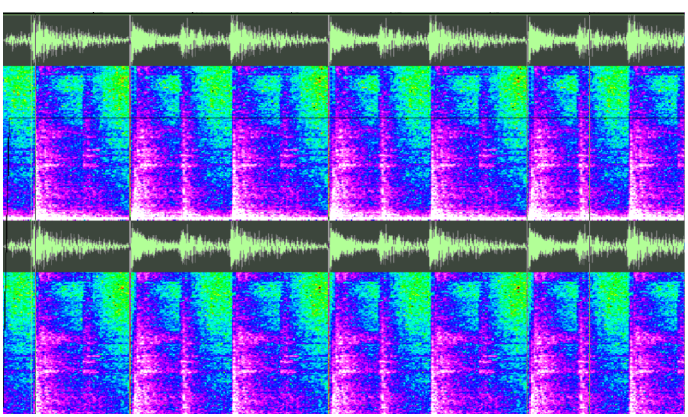
Spectral Mode can be used to remove noise interference from an object without audibly affecting the wanted signal.

Spectral Mode is useful for removing all kinds of short, random sounds in music such as popping, coughing, or even short drop-outs. Use the DeHisser or the DeNoiser to permanently remove noises like humming, etc.

1. First activate spectral display from the display options: System options (shortcut "Y") > "Design" > "Display options"

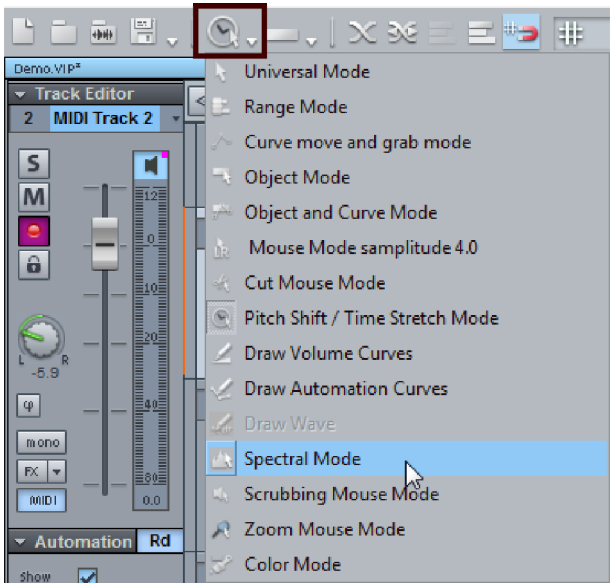


Now you can see the audio signal in the spectral display.

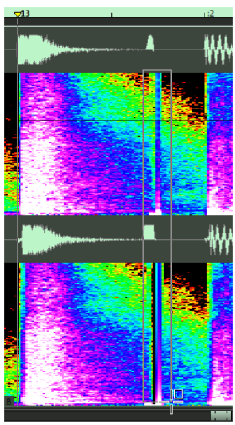


The spectrogram displays the frequency proportions in a time curve. The volume of frequencies is visualized by a color code. A continuous sound is displayed by a pattern consisting of horizontal lines, which correspond to the sound components or overtones of the sound. A distortion with an impulse quality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

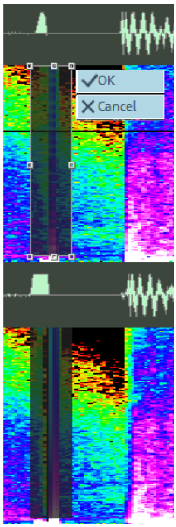
2. Switch to the "Spectral Mode" mouse mode.



3. In the following example, we want to remove a percussion cracking noise from the loop. We need to zoom in on the object a little and highlight the interference by drawing a rectangle around it using the mouse.

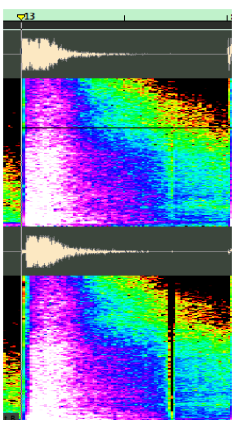


4. Use the handle to adjust the selected area.



You can listen to the results while editing.

5. Click "OK". Now you should be able to see the result in the spectral display in the arranger window: The colored signal in the spectrogram was removed.



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.

Tip: This also enables you to edit the left and right channels of the signal separately by selecting only the upper or lower part of the spectrogram from the highlighted rectangle.

External editing

You can send each object to an external editing program. After editing, you easily import your file back into Sequoia.

You can configure this data excchange together with the corresponding context menu option with the help of the the dialogue "External Tools (view page 710)" which is located in the menu "File" > Program Settings" > "External Tools".

Editing objects in SpectraLayers Pro

SpectraLayers Pro is an external audio editing program which is part of the Sequoia program package. It has to be installed seperately. You can export every objekt from Sequoia to SpectraLayers Pro and re-import it after editing back to >Programmname>.

SpectraLayers Pro offers audio editing in a graphical way. Edit with a pencil, the noise spray or magic stick and other tools inside Spectralview and treat your audio like you already know from picture-editing software.

In SpectraLayers Pro your audio is displayed in layers instead of tracks. 2D- and 3D-views allow you to take a close-up look inside your material.

With SpectraLayers Pro you can unmask overlapping frequencies and get rid of the them in no time. Single Instruments gain presence and the whole mix become more clearer, without moving faders.For audio restauration and forensics, SpectraLayers Pro is your weapon of choice and companion for fast and prime results. With tools like frequency tracking, noise printing and noise reduction, SpectraLayers Pro makes the most out of your archive or live recordings and your master will appear in maximal clarity. Useful precision-tools like the overtone selector, lasso selection, dodge, clone stamp or eraser give you plenty of options to manipulate your audio.

For many Sound Designers SpectraLayers Pro has an additional value. Filtered noise prints, artefacts and other side products unveil to be the beginning of a new path to make abstract sounds.

Offline Editing of Objects

By editing an object offline (e.g. to add complicated effects to the audio material) severe loads on the CPU can be reduced during playback in the virtual project (VIP).

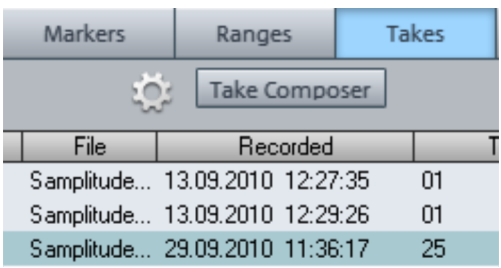
To edit an object offline, right-click the object to access its context menu and select "Wave Editing". This opens the audio file as a "wave project". The wave can now be edited as desired. Sequoia now functions as an audio editor (view page 62).

If an audio file that is used in multiple VIP objects is edited in offline mode using the option "Object > Wave Editing", all of the associated objects in the VIP will access this audio file. For example, if reverb is added to an audio file in offline mode, all objects that reference this audio file will now feature reverb.

However, if you only want to edit specific objects offline, select the option „Object“ > „Edit“ > „Edit a copy of wave content“. Sequoia will create a copy of the audio material and add this to the project folder. The selected objects will now refer to the copy that is created. Offline effects can now be applied to the selected objects without influencing other objects.

The button "Advanced options (view page 858)" in the corresponding effects dialog provides access to options for saving the copy.

The Take Manager (view page 201) provides an overview of the versions of destructively edited files and lets you select them for the respective object.



As a safety measure the option "Create copy" is activated by default for all destructive effects. This creates a temporary copy of the unedited audio file. Activate this option in the offline effects dialog to be able to use the undo function in the VIP later - this allows you to undo offline editing.

Note: Temporary files for the "Undo" function are only created with offline effects calculation if the "Undo" function for wave projects is also activated (keyboard shortcut: "Y" – Program > Undo) and "Create copy" is activated in the corresponding effect dialog. Deactivate the option "Create copy" if you are sure that the original audio file will not be used for other objects. This also applies to objects in other VIPs.

Object Editor

The foundation for object-oriented working is the object editor which can function as a tool for a single object or as a "realtime channel strip" for multiple objects.

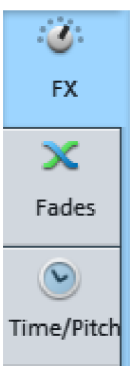
Note: There is an object editor for MIDI objects as well. Detailed information can be found in the chapter "MIDI in Sequoia > MIDI object editor (view page 295)"



The object editor is divided into three sections – "FX", "Fades", and "Time/Pitch". The object editor can always be left open to edit objects - the view automatically adjusts to the selected object.

To open the object editor, select an object and press "Ctrl + O" or double-click on the object. The object editor opens in the Docking bay (view page 177).

Object Editor: Basic Functions



The selection options for the three different dialog windows of the object editor are displayed on the left.

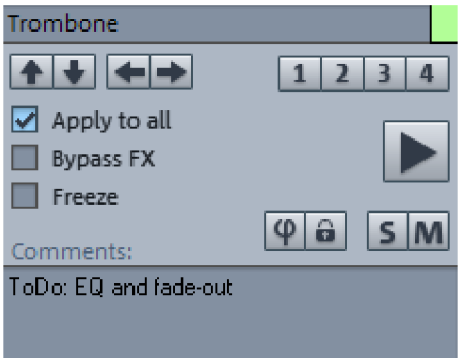
FX: Here you can set the gain, AUX, plug-ins, EQ, panorama and automation.

Fades: Here you can set the fade-in, fade-out, wave position and object position.

Time/Pitch: Here you can set time stretch / pitch shift algorithms and loops.

Note: By right-clicking in an open area you can display and select various graphic interfaces which are also known as "skins". The skins that have the "Editor Max" in the name display all of the parameters in a single overview.

The right section is the same in all three dialog windows:



The object name is displayed in the upper text field and can also be edited. In the square field beside this you can select the color for the object waveform.

Note: Please make sure that you have selected the option "Default color settings" in the "Waveform Colors" section of the display options (keyboard shortcut: Shift + Tab).

Using the arrow keys you can "jump" up or down to the object of a neighboring track. With the horizontal arrow keys you can jump to the previous or next object on a track. If multiple objects are selected, these buttons will not be active. Clicking the Undo button (spinning arrow) returns the selected object to its original settings.

1-4: You can save the various object editor settings in the 4 snapshots. This can be done by pressing "Shift" and left-clicking on the corresponding button. If you haven't yet set a snapshot for the respective button, a simple left-click is enough.

Clicking on the respective number keys will switch between snapshots. This means that you can quickly compare several object editor settings.

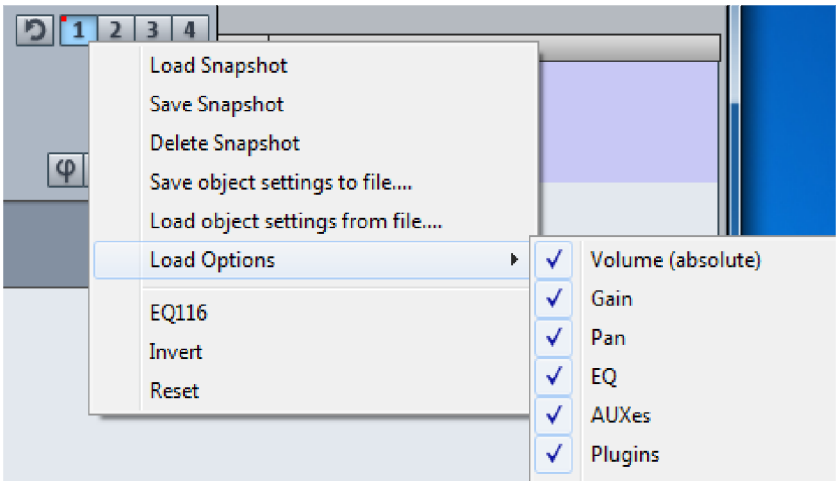
The active snapshot is marked by a bullet point.

To apply object settings from one object to another, first save the object settings from the original object as a snapshot, then switch to the next object and open the snapshot for the currently selected object by simply clicking on it.

Right-clicking on a snapshot number opens a context menu where snapshots can be saved, loaded and deleted.

In the context menu you can also save the object effect settings as a file or load them from a different file. When loading a file you can also specify exactly which settings

from the snapshot you want to apply to the activated object. Here you have a choice between volume, gain, panorama, EQ, AUX and plug-in settings.



To all: If the "To all" option is active all selected objects are updated when the object editor is opened. No matter in which of the three object editor windows, all recently applied settings that have been made in a selected object, will be applied to all other selected objects.

Note:

Volume changes are transferred relatively in an object, i. e. they are increased or decreased relative to the volume of other selected objects.

The EQ settings are always completely transferred, in contrast to previous versions where only the edited bands were transferred.

Bypass FX: This deactivates all object effects, EQ/panorama settings, and AUX sends.

Freeze: If you click on the Freeze function for an object, this will be calculated as a new audio file with all the associated effects. The new freeze object takes the place of the original object. This way you can bypass the realtime processing of effects and reduce the load on the CPU.

Play/Stop: This button corresponds to the normal playback function (spacebar).

Phi: The Phi button causes a 180 degree phase rotation. By right-clicking on the Phi button you can invert the left and right channels individually.

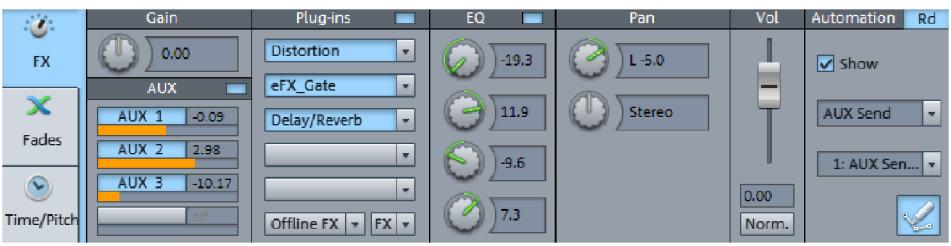
Lock: The "Lock" button has the same function as the key button in an object. The object is then protected against any inadvertent horizontal movement. Additional locks against vertical moving, volume adjustment, fades, length adjustments, ripples or deletion can be set in the system options (keyboard shortcut: "Y") under "Program" > "Object Lock Definitions (view page 708)". Locking can be temporarily cancelled by pressing the "Alt" key.

Solo: During playback this button moves the playback marker to the start of the selected object and plays that only. Track-dependent busses and AUX paths are also included in the playback.

Mute: This button mutes the selected object. By right-clicking you can also mute the left and right channels individually.

Comments: You can type comments relevant to the selected object in this field.

Object effects



In addition to the numerous effects available for individual tracks and the master area of the mixer, audio objects can also feature effects. All settings are retained when moving or copying objects.

By crossfading between objects of different settings, you can also fade between the different effect settings. CPU-intensive effects can therefore be used more efficiently, since they are only processed as required (unlike a track effect set up as an AUX send effect).

"Effects" or the context menu also makes it possible to access the object effects.

Manipulation using the level and pan curves or the volume controllers is only calculated after object level effects.

Gain and AUX sends

Gain: Set the gain for the object here.

AUX Sends: Any object can send to all available AUX busses. You will see, however, initially only to AUX1 AUX 4. You can adjust the corresponding slider by dragging in the respective AUX Send field.

By right click on the blue button in the header the advanced AUX send dialog will open, which provides full access to all AUX send paths.

By left clicking on the blue button in the track header you can bypass all AUX sends (Bypass function).

Plug-ins and EQ

The plug-in section provides you with quick access to installed effect plug-ins on the object level.

Clicking on the empty insert slot opens the plug-in browser (view page 222), where you can load a plug-in to the slot. Click on an occupied slot to activate/deactivate the plug-in. Right-clicking on the slot opens the interface of the plug-in. The arrow next to the respective insert slot opens a menu with various functions. You can ope the plug-in browser again to switch out the plug-in or remove it ("No effect"). The "Plug-ins" button activates or deactivates all effects in the channel. A visual indicator (*) is displayed for plug-ins that were previously active and are active the next time "Plug-ins" button is pressed.

The "FX" button opens the effect routing selection dialog (view page 388) for modifications to the plug-in series. To load additional internal effects, you can use the FX/Routing dialog. If you click on the arrow button next to the "FX" button, you can save, load or reset the object effect settings.

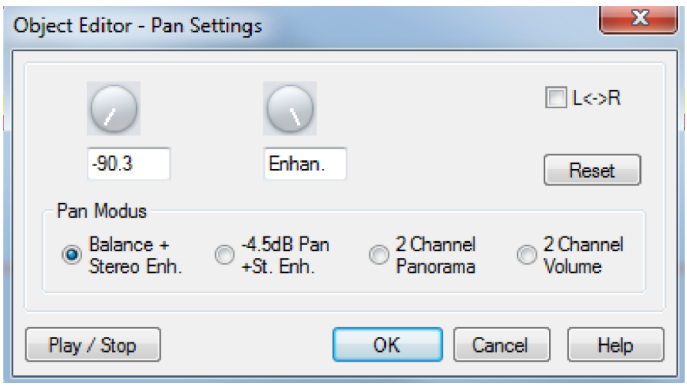
The "Offline-FX" button opens a selection dialog for using offline effects (view page 219). Select the effect in the corresponding effect dialog and confirm using the "OK" button. The effect will now be included in the object offline. It is also possible to edit either the left or right channels individually.

To remove a plug-in or an effect, select "No effect" from the corresponding plug-in slot in the menu.

Under "EQ" you can set four frequency bands of the parametric EQ directly via the knobs. The small square button in the header switches the equalizer on and off. By right-clicking on the knob/button you can access the dialog window for the fully parametric 6-band equalizer. The selected EQ values can be cut, copied, inserted and deleted by right-clicking on the corresponding fields.

Pan

Here you can edit the panorama as well as the stereo width of the active object. Right-clicking on the knob opens the "Pan Settings" dialog.



Use L <-> R to switch between the left and right channels.

Pan: With the knob on the left you can specify the panorama in the stereo image.

Stereo: The knob on the right can be used to set the stereo base width.

Balance + Stereo Enhancer: The pan knob sets the stereo balance. The mid position corresponds to a level of 0dB on both channels. When the panorama is moved to the left, the level of the right channel is reduced and vice versa.

The stereo button sets the base width of the left attack = 0% = mono via mid = 100% = stereo to right attack = 200% = Enhan. (extreme base widening).

-4.5 dB Pan + **St. Enh.:** The panorama fader controls the panorama position in such a way that when the levels of both channels are set to the middle it reduces by -4.5 dB. In the outer settings one of the channels is faded out, the other is raised from -4.5 dB to 0 dB through the control range. This compensates for any perceived drop in volume in the mid position. Use this mode to place mono objects in the stereo panorama.

2 Channel Panorama: In this mode the two faders turn into two panorama faders for both channels, just as would be expected on a mixer with mono channels.

2 Channel Volume: In this mode you can control the level of the two object stereo tracks separately just as you would on a mixer with mono channels.

Reset: This button resets the panorama mode and the knob.

Note: For a three-times more powerful frequency-selective setting of the base width you can use the multiband stereo enhancer (view page 963)as an effect insert in the plug-in section.

Vol

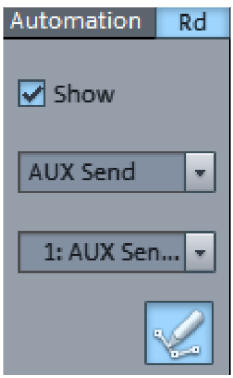
Vol.: The volume can be controlled using this fader.

Double-clicking on the fader sets it to 0 dB.

Norm.: The Norm. button normalizes the maximum level of the object to 0 dB.

Automation

You can change object automation settings here.



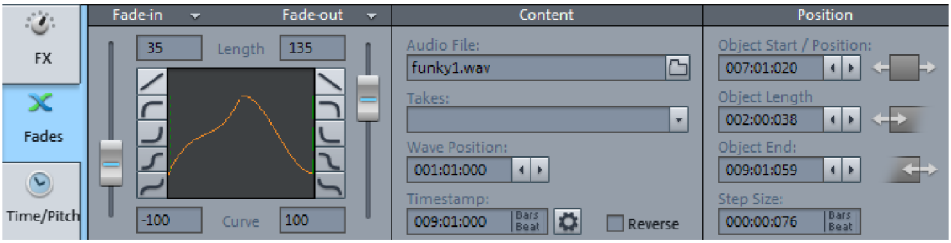
You can make the automation curves of the selected parameter visible in objects with the "View" option.

In both of the lower fields you can select the desired automation parameters.

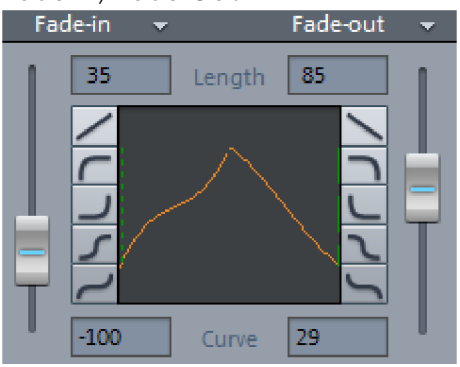
Activate the Automation drawing mode button and draw the object automation curve in the corresponding object.

Position/Fades

Use this dialog to set object-related time positions (start, length, end) and fade parameter for the selected object. In the entry fields "Timestamp" and "Step size" the measurement units can be changed by simply clicking. The following units are available: samples, milliseconds, hours/mins/secs, SMPTE (24, 25, 29.97 non drop, 29,97 drop, 30), SMPTE/milliseconds, CD-MSF, feet and frames 16 mm (40fpf) and feet and frames 35 mm (16fpf), noise reduction (meter).



Fade In/Fade Out



Fade in and fade out can be edited in this dialog.

Using the faders you can adjust the precise settings of the curve form of the fade.

If you wish to produce the original fade, click on the arrow button and select the "Reset" option. The same context menu also features presets for "linear", "exp." (exponential), "log." (logarithmic), "cos." (cosine) and "sin." (Sine) curves. You can also load additional custom curve forms here. Double-click the fader to switch the fade between "Linear" and the last setting.

Additional options for fade-ins are available via the arrow button:

Enable crossfade (left): Use this option to extend the fade-in object to the left over the original object border, depending on the values set in "Fade length". The fade-out of the previous object is shortened accordingly; both fade-in objects will remain linked.

Asymmetrical crossfade (left): This option adjusts the fade-in independent of the fade-out to the left past the object border.

There are also "Fade offset" option presets which determine which portion of the fade should lie within or outside the borders of the original object.

Fade within: The fade is located completely within the object

Fade symmetric: The fade is symmetric to object borders

Fade outside: The fade is completely outside the object's original borders

"Fade inside" (corresponds to 0% "Fade offset"), "Fade symmetrical" (corresponds to 50% "Fade offset") and "Fade outside" (corresponds to 100% "Fade offset").

The object will be stretched accordingly. The original object edge that now acts as the "axis" of the fade will now be depicted as a dotted line.

If the fade offset is above 0%, make sure that there is enough audio material in the audio file so that the object can always be faded in or out. For example, if the object starts at the exact beginning of the curve display, you will no longer be able to crossfade as soon as the "Fade offset" rises above 0%.

You can also use the "Get global crossfade" function. Here the crossfade values of the standard settings are acquired for automatic crossfades. Use "Set global crossfade" to set the current crossfade values as standard values for automatic crossfades.

The resulting curve form for fade-in and fade-out is shown together with the "Fade offset" in the graphic.

Note: For detailed crossfade editing in Sequoia you can also use the Crossfade Editor (view page 425) (Shortcut: Ctrl + F).

Content

Audio file: Here the referenced audio file can be exchanged, copied and renamed directly in the object editor. Open the file exchange dialog by clicking the folder symbol beside the name of the audio file. If the audio file is used by multiple objects, you can indicate whether it should be exchanged with the currently selected object or with all referenced objects.

Takes: Select the take you want here.

Wave position: The arrow buttons do not affect the position of the object or its length, but instead move the wave display in the object left or right.

Timestamp: This value assigns a unique time to the respective object.

If you click on the "Gear" button the Broadcast Wave Manager will open (view page 663). This feature lets you define and save metadata for the underlying audio files.

The "Reverse" function can be used to play selected objects backwards. This process does not change the audio files.

Note: Please note that there are some limitations when using additional effects on an object that is played in reverse. This applies to audio editing mode or the use of time stretching effects and Elastic Audio.

Position

The starting time of the object on the timeline, object length, and object end can be set here by entering a number or using the arrow buttons.

Using the arrow buttons beside object start / position the object can be moved forward to the start of the project or towards the end of the project.

The arrow buttons beside object length move the start of the object, they work just like the front lower object handle.

The arrow buttons beside object end shorten or lengthen the object, they work just like the back lower handle.

The step size shows the position/length adjustment made upon clicking on the arrow button. The unit of measurement to the right beside the input value can be changed in the context menu (right-mouse click to open).

The lower part of the menu features corresponding presets for each selected measurement, e.g. for the unit of measurement "Bar/beats", the following step sizes are used: 1/64, 1/32, 1/16, 1/8, 1/4, 1/2, 1Beat, 2Beats, 1Bar, 2Bars, 4Bars.

Loop

In Loop mode (view page 148), Loop Start, Loop Length and Loop End are active whereby the unit of measurement to the right beside the "Step size" input value can be changed with a mouse-click. The loop with the specified step size can be corrected using the arrow buttons:

Loop start: The upper arrow buttons change the loop starting point without changing the loop length, that is, the loop end is moved as well.

Loop length: The middle arrow buttons change the starting point as well as the loop length, the loop end remains unchanged.

Loop end: The lower arrow buttons move the loop end, the loop length changes.

All settings can be seen in the selected object in the form of vertical dashes. Once an object has been selected the whole object turns into a loop. If, however, you selected a range, only this will turn into a loop.

Pitch shifting / Time stretching

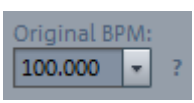


Pitchshifting and timestretching can be executed simultaneously and independently of one another (but not in "Resampling" mode).

Pitch shifting: You can shift pitch either in half tones (mouse wheel + Ctrl), in cents (mouse wheel + Shift) or enter it as a pitch factor of the original value.

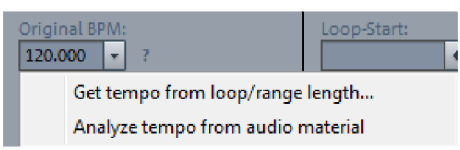
Time stretching: Speed correction can be made either as new object length/stretch length or as stretch factor of the original value, or you can indicate the desired tempo as BPM (Beats Per Minute).

The original tempo is required to calculate the used stretch factor of the BPM entry.



A question mark next to the entry field indicates that the displayed value is a default value and may not correct.

You can either enter the original tempo manually or let the program calculate it. To do this, click on the arrow next to the value.



Get tempo from loop/range length: A range section above the object is used to calculate tempo. Select the option and enter how many quarter notes should be included in the selection.

Analyze tempo of audio material: Melodyne will calculate tempo. Melodyne must be installed for this. You can learn more in Melodyne integration (view page 234).

Analyze and round off tempo of audio material: The calculated tempo will be rounded off to a whole BPM value.

Transfer object tempo to project tempo: The calculated object tempo is used as project tempo.

Mode: Four algorithms may be used for realtime time stretching and pitch shifting: élastique Pro, élastique Efficient, Resample and Monophonic Voice.

> You can learn more about individual algorithms in the menu reference under "Effects > Time / Pitch > Resampling / Time Stretching / Pitch Shifting (view page 912)".

"Edit" opens a clear display of the already named parameters and some others. If during algorithm selection you move through the individual modes with the mouse, a detailed explanation window will appear for each mode.

The "Elastic Audio (view page 914)" button opens the selected object in the Elastic Audio editor.

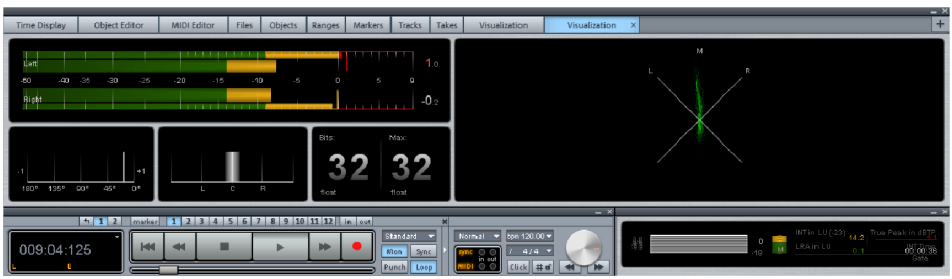
Use Elastic Audio (Pitch Automation): Changes the pitch using Elastic Audio (view page 914).

Apply musical tempo adjustment: Adjusts the playback tempo for the object to the tempo map of the arrangement.

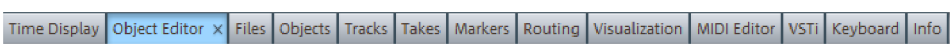
Manager/Docker

Menu: "View" > "Manager / Docker"

The docker is where you can customize the display windows you need for your workflow. In addition to the manager windows (view page 178), now the visualization (view page 216), time display (view page 214), transport console (view page 65), object editor (view page 165), MIDI editor (view page 299), Monitoring section and audio quantization (view page 217) are opened as a tab window in the docker.



The docker can be minimized by clicking on the "Minimize" symbol or on an already selected tab in the docker. If you click a tab again, the docker will be maximized again.



Using the "+" key, you can open a new tab in the docker.



This displays a list of the tab windows integrated in the docker. Tabs that have already been integrated appear with a (+) symbol.

To rearrange the docker tabs simply drag & drop them to a different position in the tab bar.

If you want to remove a tab from the docker, simply double-click on the tab or drag & drop it. The dialog window will now appear in its own window.

You can return the window to the docker by double-clicking again on the individual window or by dragging the window onto the tab bar.

Using the "Ctrl + F4" shortcut, you can remove the currently selected tab without opening it as a separate window. The "Ctrl + Tab" shortcut activates the next docking tab. "Ctrl + B" shortcut activates the docker window.

Manager

Depending on the functional area, individual managers bundle often-used administrative and control functions in a manager window of their own.

The following manager windows are available:

Manager	Key shortcut
ClipStore	Ctrl + Shift + Alt + C
File Browser	Ctrl + Shift + B
Object manager	Ctrl + Shift + O
Track manager	Ctrl+ Shift +S
Marker manager	Ctrl + Shift + Alt + M
Range Manager	Ctrl + Shift + Alt + B
Take manager	Ctrl + Shift + Alt + T
Source List Manager	Ctrl + Shift + Alt + S
VSTi manager	Ctrl + Shift + I
Routing Manager	Ctrl + Shift + Alt + R
Soundpool Manager	
Info Manager	

In order to open a manager, select the manager you need in "View" > "Manager". The selected manager will open either as a separate window or in the Docker (view page 177), if it is activated.

Clip Manager ("Clipstore")

Sequoia contains a Clip Manager in which all audio files of all loaded projects are displayed together with "clips" from these projects in a common list for cross-project use.

A clip is a selection of single or multiple objects or a range selection from a VIP that can be stored in the Clipstore for further use in the same or other projects.

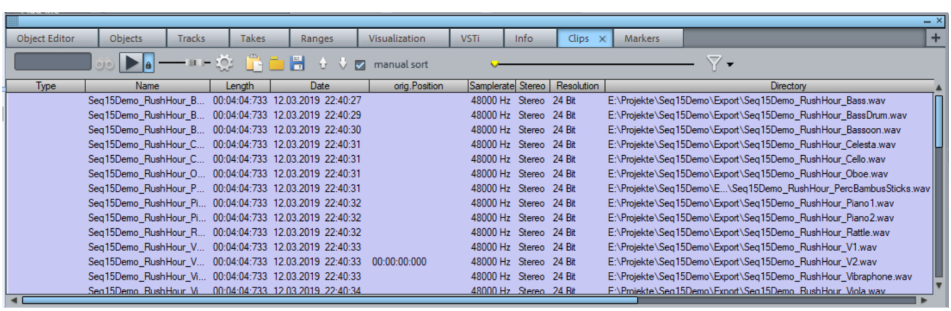
You can imagine this as a comprehensive functional extension of the clipboard. After you have copied parts of a project to the clipboard (or in Sequoiabriefly "Clip") with Ctrl+C, you can now (but only now!) paste them into the same or another open project. A next copy operation overwrites the clipboard and the old clip is no longer available. In the Clipstore you can keep such copied project components and save them together with the project. Even project independent Clip Lists can be saved and loaded as Clipstore presets.

Note: Technically, the clips are small stand-alone VIP files. They are stored in the ClipStore subfolder next to the project from which they originate. As a VIP, a clip contains not only the reference to the audio files but also all other object-related settings (effects, fades, timestretching, etc.) as well as volume and panorama automation.

In order to show the Clip Manager, select "View" > "Manager" > "Clipstore". The manager opens either as a separate window or in the docker (view page 177).

Menu: View > Manager > Clipstore

Shortcuts: Ctrl + Shift + Alt + C



Clip List Display

The Clip List initially contains all audio files and clips of all currently opened projects.

A star in the "Type" column in front of the name indicates that this audio file is loaded in the current project. The label "Clip" in the "Type" column indicates that it is a clip created manually by a copy action.

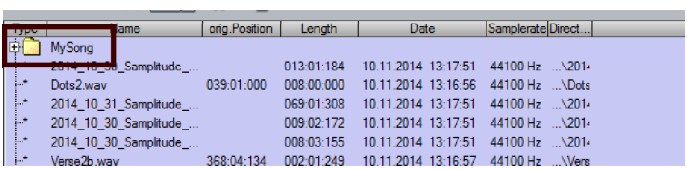
Entries in the Clipstore with a blue background belong to the currently active VIP. This means they are either audio files referenced by the active VIP or clips manually created from this VIP. Entries without a blue background belong to other, simultaneously opened projects or are wave projects opened independently from the VIPs.

All Clipstore elements (blue) assigned to the active VIP are saved with the VIP and are available in the Clipstore the next time the VIP is opened. To assign a clip in the list that comes from another project to the active project, right-click on the entry and select "Assign selected clips to current VIP".

Note: A clip can only be assigned to one VIP at a time. If you assign it to another VIP, the assignment to the original VIP is removed.

To view the file location of each item in the Clip List, select "Show Clip in File Browser" from the item's context menu. It switches to the Sequoia file manager and displays the file.

You also create folders in the Clipstore to keep everything clearly organized. To do so, right-click in the manager and select "Add new folder...". Specify a name for the folder and confirm with "OK".



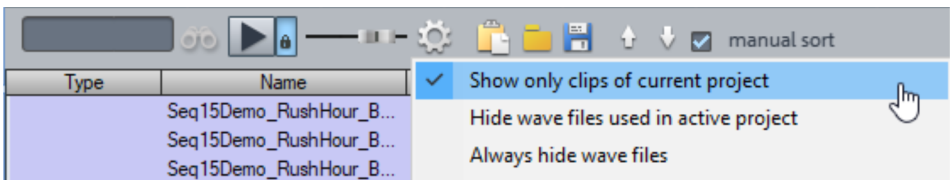
All clip list entries can be moved into the folder with the mouse. With the small +/- symbol in front of a folder you can open and close it. Folders in the Clipstore are saved and loaded together with your project.

Filtering the Clip List

To find files or clips in the Clip List faster, it can be filtered in various ways.

Clipstore

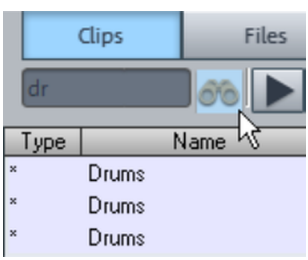
In the upper part of the "Options" menu in the Manager toolbar, there are several options to hide certain types of entries in the Clip List:



- **Show only clips of the current project:** All entries with white background, i.e. files from other VIPs, are hidden.
- **Hide wave files used in the active project:** The audio files used in the active project (those marked with *) are hidden. This means that only unused audio files in the active VIP and files from other loaded projects are displayed.
- **Always hide wave files:** Audio files are generally hidden and only clips are displayed.

Search feature

To search the list for entries, enter their names fully or in part into the ClipStore toolbar query field and click the magnifying glass symbol. The Clip List then only displays clips or audio files that contain the search term in the file or clip name.



To find the file associated with a clip on the hard disk, right-click on the clip and select the entry "Show clip in file browser" in the context menu. If necessary, the file browser view is opened in the path of the file and the file is selected.

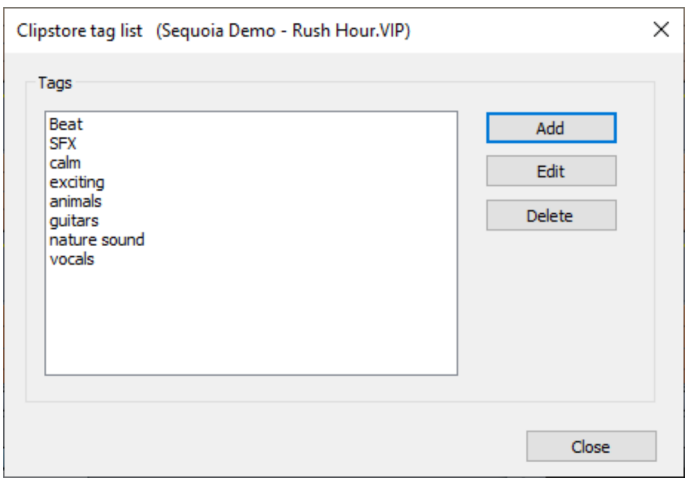
Tags

Clipstore entries can be provided with "tags" (keywords), according to which the clips and audio files can be arranged in categories. The list can then be filtered by these tags.

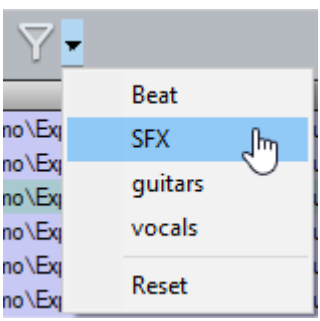
To add a Tag, double-click in the Tags column in the list and enter the name. You can also enter multiple tags. Since tags can also contain spaces, use a semicolon (;) as separator.

You can now assign the tag to any other list item by selecting it and choosing the corresponding tag from the submenu in the context menu item "Add tag". Any number of tags can be assigned to a clip.

With the entry "Edit tag list...". in the context menu of the clip list you can create, edit and remove tags in a separate dialog.



The list can also be sorted by tags by clicking on the Tags column header.



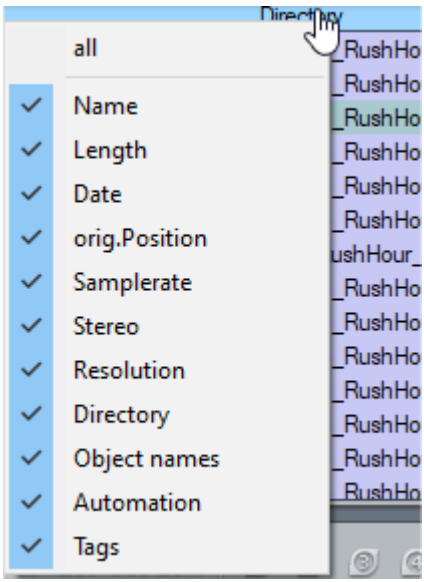
By clicking on the Filter icon in the Clip Manager toolbar, you can filter the list by a specific tag.

To set the tag, click on the small triangle and select the tag from the menu.

Sorting the Clip List

You can move selected entries up or down in the list using the arrow keys. You can also manually sort clips in the list. Click the check box in the Clipstore toolbar for "Manual sorting". Now you can drag entries to the desired position with the mouse.

By clicking an attribute in the list header you can sort the entries in ascending or descending order according to this attribute. Any previous manual sorting will be discarded.



Right-click on the list header to select which attributes should be displayed. Besides name, length, date and information about the sample format, there are some special attributes:

- **Original Position:** This is the time position of the sample start of the first object of a clip in the project from which it was copied.
- **Automation:** Indicates whether a clip contains track or object automation.
- **Tags:** The tags assigned to an entry in the list. (See above)

Play clips

 A selected entry can be played back using the playback symbol on the ClipStore toolbar. If the "Auto play" button (lock symbol) is activated, all selected list entries will be immediately played back. A complete list can be previewed simply by using the keyboard's arrow keys. You can stop playback by clicking on the playback button again.



With the position slider you can jump to a certain position in longer files.

To minimize the time until playback start, select all files listed in the Clipstore and click "Preload all clips" in the options menu. Clip list entries that aren't yet part of a project will also be opened in the background.

Hint: Playback is not routed through the mixer or internal effects. The audio device selected for playback in the playback options ("P") under "Global playback device" will be used.

Adding audio files to the Clipstore

Instead of loading audio files directly into your project, you can first collect them in the Clipstore. To load audio files into the Clipstore, you can:

- Drag & drop files from the Windows Explorer or Sequoia File Manager on the Clipstore tab or directly into the clip list.
- select the files in the Windows Explorer or the Sequoia File manager, copy them with Ctrl+C and right-click in the clip list and select "Paste file from clipboard".
- In the "Load Audio File" dialog from the " File" > "Import" menu, select the option "Load files only to the Clipstore". If the check box is activated, all files selected in this dialog will only appear in the ClipStore list, and not in the project. Even if you want to open an entry using drag & drop from a database connected to Sequoia such as DigaSystem, the selected file will only appear in the Clip list, not the current project.

Creating Clips

To create clips in the clip list, you can:

- Select single or multiple objects and drag and drop them from the project window into the Clipstore or
- Select individual or multiple objects or drag a range over one or more tracks in the project and use the menu commands or keyboard shortcuts "Edit Menu > Copy > Copy to ClipStore" (Keyboard shortcut: Ctrl + Shift + C) and "Edit Menu > Cut > Cut to ClipStore" (Keyboard shortcut: Ctrl + Shift + X).

Enable the "Insert new clips at list start" option in the Clipstore Options so that new clips are always sorted at the top of the Clip List. The clips are saved together with the project and are displayed again in the clipstore when the project is reopened.

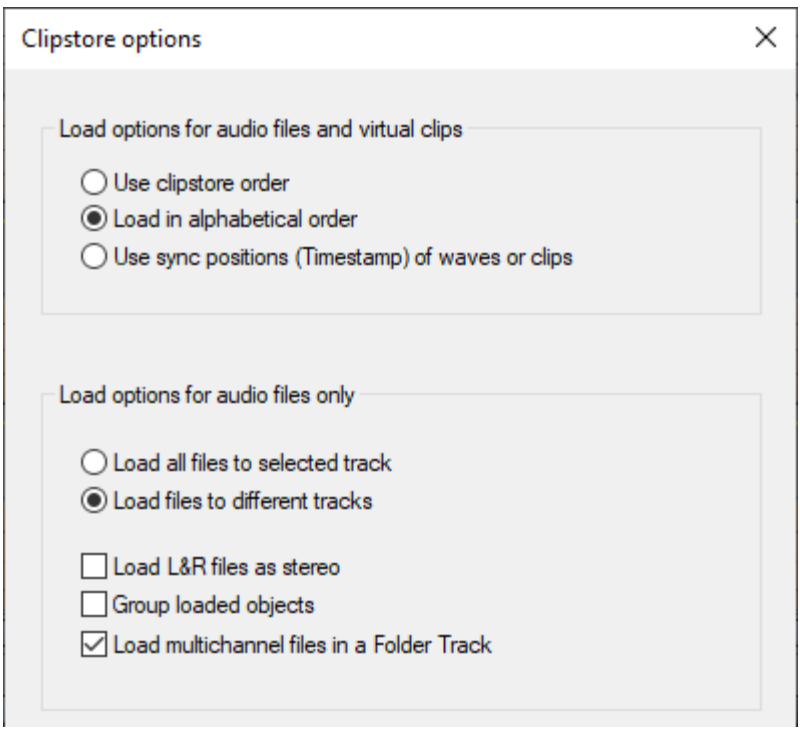
Insert audio files and clips into projects

To insert clip list elements into a project, you have several options:

- Select items and drag and drop them to the desired position.
- To insert clips at the playback marker position, select them in the Clipstore and choose "Edit menu" > "Paste" > "Paste From Clipstore" (keyboard shortcut: Ctrl + Shift + V), click the Paste button in the Clipstore toolbar or press Enter (only when the clip manager is selected).
- Right-clicking on a clip in the Clipstore > "Load clip to the project" or "Insert clip at cursor position".

Tip: When inserting multiple elements, you should sort the entries in the Clip List beforehand, because by default the list order determines the order of insertion.

More options for positioning in the project can be found in the "Load Options" in the Settings menu (gear icon on the Clipstore toolbar).



Use Clipstore order: Audio files and clips are arranged in the order of the clip list.

Load in alphabetical order: The selected files are sorted alphabetically in the VIP. For clips, the clip name is relevant.

Use sync positions (timestamp) of waves or clips: Broadcast wave files containing timestamps are positioned precisely at this position in the VIP. Clips are loaded at the same time position from which they were copied.

Load all files to selected track: The selected files are loaded successively in one of the selected tracks.

Load files to different tracks: The files are now sorted in vertical order from the selected track to the next one. If necessary, an additional track is added.

Load L&R files as stereo: If you select this option, Sequoia loads all L & R files as stereo.

Group loaded objects: All loaded files are grouped. They can be ungrouped at any time.

Load multichannel files into a folder track: This option is activated by default. When loading multi-channel files (i.e. files with more than two channels), it is assumed that these are Surround files in Interleaved format. A folder (view page 132) track with the required number of tracks is created and routed to the surround master (that is also created if necessary). The tracks in the folder track get the surround panning according to the determined surround format. If this option is not active, the individual channels are loaded onto the track and the tracks below it.

Save and load the Clip List as Clipstore Preset

 You can save a clip list by clicking the Save button in the manager toolbar. It is available across projects, so that it can be loaded into Sequoia at any time and the content can be used for other projects. To do this, click on the "Load" button in the Clipstore header.

The Sequoia "fx preset" folder is used as default when saving and loading. You can also choose a different storage location if you want. A ClipStore preset with the "*.clp" extension will be created. This is a text file that can be viewed and edited using a conventional text editor.

When loading Clipstore presets, the current Clipstore content is retained and the loaded Clipstore preset is added to the list. The list entries of the Clipstore preset are also assigned to the current VIP and are available in the Clipstore the next time the VIP is opened after saving the VIP.

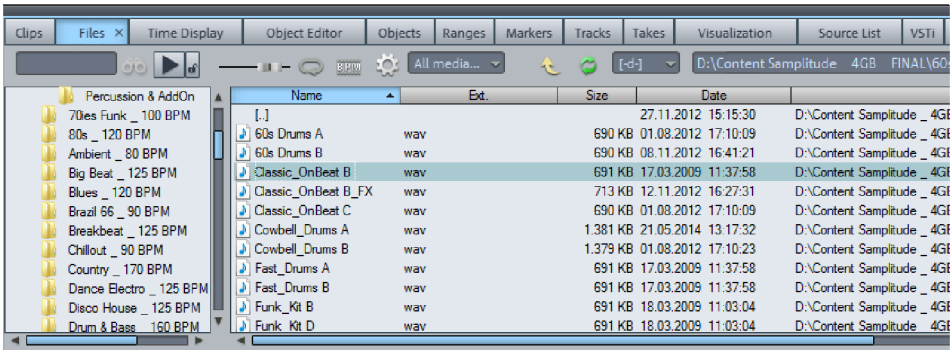
File Manager

Sequoia contains a file manager for previewing and loading files directly into projects. The browser can be used to create folders for favorites and provides access to all recently opened folders.

To display it go to "View" > "Manager" and select "File Manager". The manager will open either in its own window or in the Docker (view page 177).

Menu: View -> Manager -> File Manager

Keyboard Shortcut: Ctrl + Shift + B



Navigation/favorites

The file manager includes a directory tree that may be switched on/off; this behaves just like Windows Explorer. Directories and files are displayed with directory and file symbols in the form of a list.

The following file information is displayed:

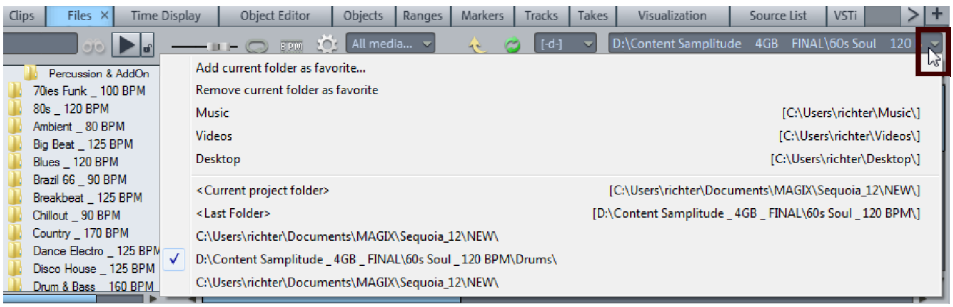
- Name
- Extension
- Size
- Date
- Length
- Resolution in bits
- Stereo/mono
- Sample rate
- Timestamp
- Description/title
- Originator/artist
- Path
- BPM: Specified by the loop length

Navigation is done via mouse or keyboard. A drive selection menu located in the toolbar lets you quickly switch between all data storage devices connected to the workstation.

A display filter located in the toolbar limits the display to certain media types, e.g. .WAV, MIDI or project files, only. "All media files" is displayed by default.

Just like other manager windows, the "Search" option searches in the current window for directories or files. Enter a search item into the box and press the search button. Any located directories or files are highlighted. You can find and highlight several entries that meet the search criteria by entering one or several first letters.

To the right in the file manager, the current directory may be added or deleted as a favorite to your favorites list. Click a favorite to open it.



Underneath the favorites list, a selection menu is located featuring a list of paths used last. You can switch into the current project folder here.

Preview audio files

A selected audio file can be previewed with the playback button in the file manager. Any highlighted audio file is played immediately if the “Autoplay" box is checked. A complete list can be previewed simply by using the cursor keys. Clicking the play button again stops continuous playback. The fader symbol can be used to control the volume.



Playback does not proceed via the mixer or internal effects, but rather via the global playback device selected in the playback parameters (keyboard shortcut: "P"). If no playback occurs, check to ensure that the routing settings for the device are correct.



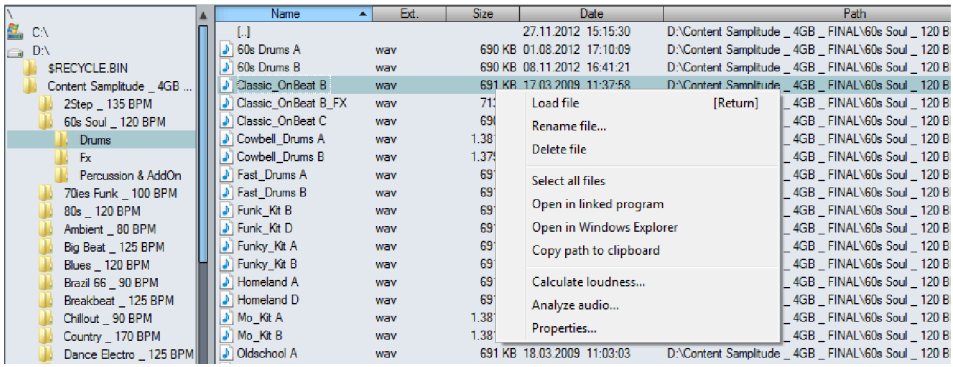
BPM sync: This option enables the selected loop to be previewed in the VIP's tempo. The original BPM is specified in this case by the length of the file. When playback is running, loops are positioned at the beginning of the beat by double clicking and then added into the arrangement as loop objects.



Looped preview enables the selected file to be previewed in a loop.

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.

Load Files into Projects from the File Manager



Media files in the file manager can be added to an open project by drag & drop or by double-clicking on the desired playback marker position. If the file manager list is in the foreground, pressing “Enter” produces the same result. Right-clicking the desired file opens a context menu. Here you can load, rename and delete files and folders.

Note: Deleted files are always sent to the Windows Recycle Bin. The keyboard shortcut "Shift + Del" plus a confirmation deletes the file permanently without sending it to the Trash.

This also provides access to the functions "Select all files", "Open in linked program", "Open in Windows Explorer", and "Copy path to clipboard". This allows you to calculate loudness values (view page 750), analyze audio and display the properties of the respective files. During the audio analysis Sequoia calculates loudness values for all selected files and displays the result in a report window.

Insert file into new track: The keyboard shortcut "Alt + Enter" creates a new track for the file. The track that is created is named according to the file.

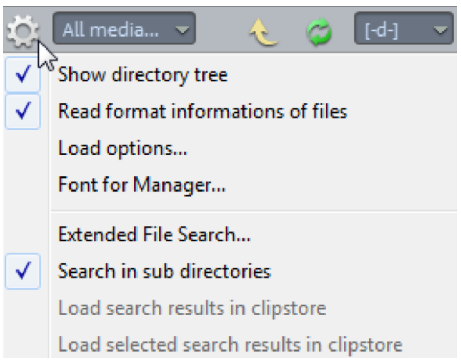
Open file in wave project view: Drag an audio file from the list and drop it into an empty space or into the title bar of the program window.

If several files are selected from the list, all files will be inserted at the current playback marker position as new objects with the loading options ("Options") taken into account. This way you can specify whether all files should be inserted into the current project one after the other, one underneath the other, alphabetically, or according to their timestamp positions. The load option settings are also valid for files loaded using the "Load" dialog.

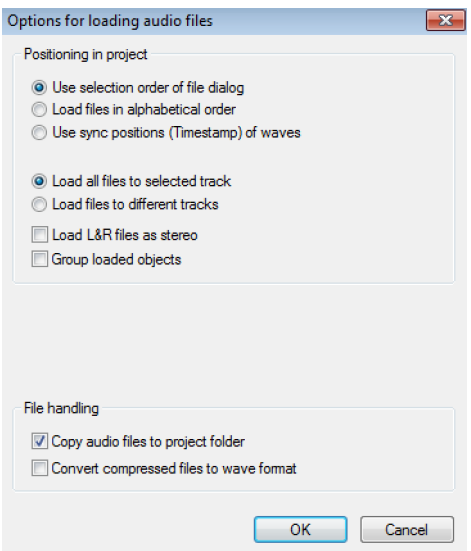
File manager options



Additional options can be accessed by clicking the small gear icon:



Loading Options



Use selection order of file dialog: If this option is activated, Sequoia remembers the sequence in which the files were selected and then sorts them accordingly.

Load files in alphabetical order: If this option is activated, Sequoia sorts the selected files alphabetically in the VIP.

Use sync positions (Timestamp) of waves: Broadcast wave files containing timestamps are positioned precisely at this position in the VIP.

Load all files to selected track: The selected files are loaded successively in one of the selected tracks.

Load files to different tracks: The files are now sorted in vertical order from the selected track to the next one. If necessary, an additional track is added.

Load L & R files as stereo: If you choose this option, Sequoia will load all L & R files as stereo files.

Group loaded objects: All loaded files are grouped. They can be ungrouped at any time.

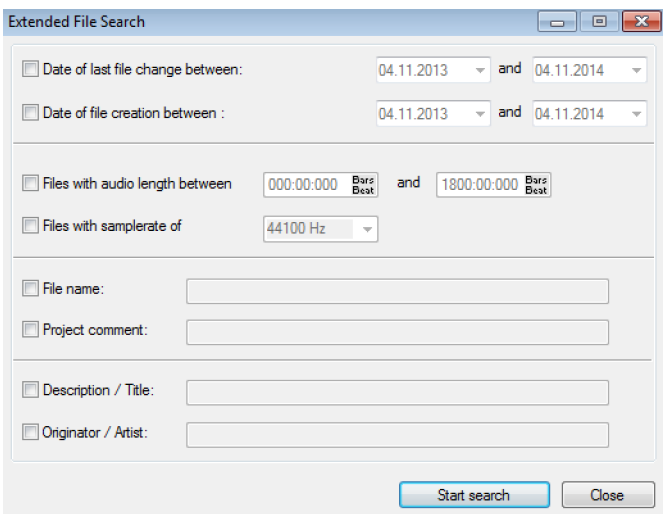
File Handling

Copy audio files to project folder: Copies the file to the corresponding project folder.

Convert compressed files to wave format: Compressed audio formats such as MP3 can also be loaded and played directly in Sequoia. This requires increased CPU calculation. Activate this option to convert files like this into an uncompressed wave format.

Extended File Search

Additional criteria are available for advanced 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, Originator / Artist:

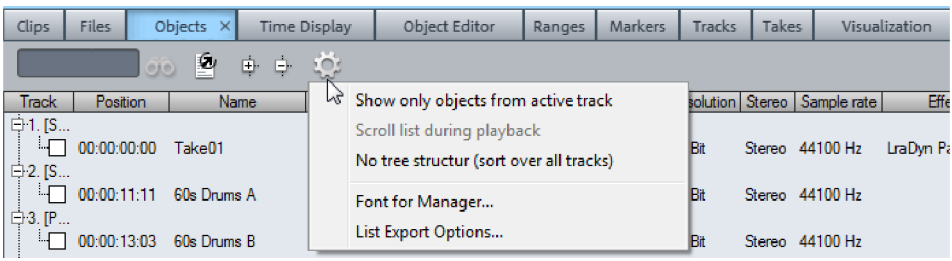


Object Manager

The object manager lists all objects contained in the current project and makes it possible to edit selected object parameters. To display the object manager go to "View" > "Manager" and select "Object Manager". The manager will open either in its own window or in the Docker (view page 177).

Menu: View -> Manager -> Object Manager

Keyboard Shortcut: Ctrl + Shift + O



You can export object manager information as a text file. Click "Export text" in the toolbar to do this. The Windows text editor opens with an excerpt from the object manager list. The following information will be saved:

- Project name and project path
- Track and object name
- Start position in the project
- Path of source file

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

Object View and Selection

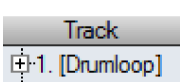
Objects are displayed in a tree structure according to tracks. Tracks are only listed if they also actually include objects. Further instances of the listed objects can be moved into the arranger using drag & drop. If you hold down the "Ctrl" key while doing this, a copy of the object will be created.



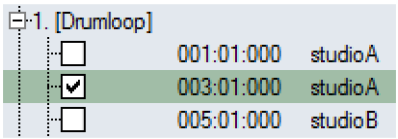
You also have the option of displaying objects in the current tracks only (toolbar: "Options").



You can use the +/- keys on the toolbar to open and close the track object display for the complete window.

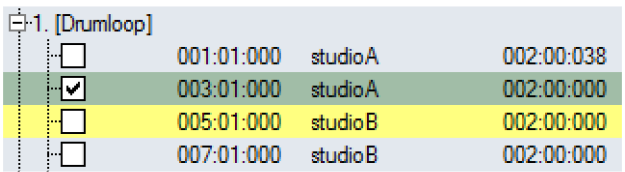


Click on the “+” symbol left of the track number to expand the view of a single track.



Select objects: The check box in front of the object name can be used to select one or multiple objects. This is also visible immediately in the project window.

If an object that is part of a group is selected, the other objects of the selection will also be selected in the project window and object manager. Grouped objects are also recognizable by the number of the object group in the "Group" column.



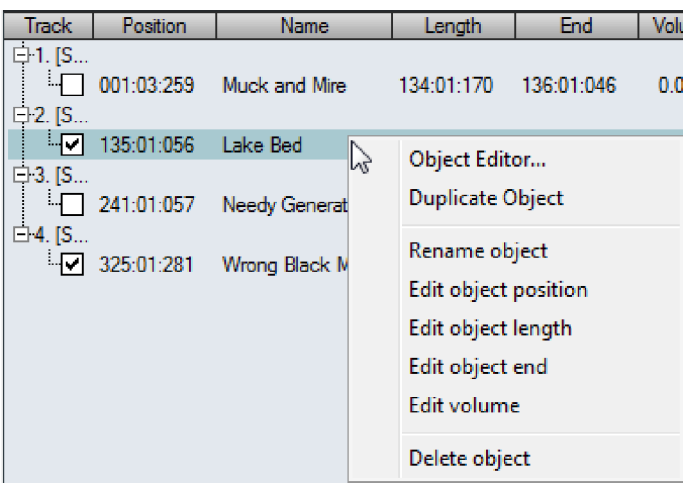
Objects that are currently playing are highlighted with color in the object manager.

Search for objects: As in other manager windows, the Object Manager also provides a search option that you can use to search for objects in the current window. Enter a search term into the box and press “Enter”. Found objects are highlighted, but are not selected in the project window.

Delete, Rename and Edit Objects

You can delete objects directly in the object manager by selecting one or several objects simultaneously and pressing the “Del” key. Alternatively, you can delete an object in its context menu. You can rename objects by double-clicking on the object names and entering a new name. You can also use the context menu.

To edit an object directly in the object manager, right-click the object entry you want to edit and start the object editor using the context menu that appears. In the same context menu you can also duplicate objects.



Edit object parameters

You can edit the following parameters in the object manager:

- Start position
- Name
- Length
- End
- Volume
- Group
- Lock

To edit a parameter, double click the respective entry and enter a different value. Numeric values can be changed by dragging with the mouse, and "Ctrl" and "Shift" allow larger or smaller changes. "Tab" moves to the next editable value. "Playback marker up/down" navigates vertically within a column, provided this is a text column.

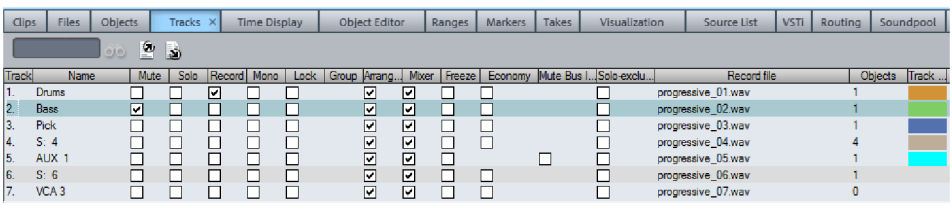
Track Manager

All the tracks in the current project are displayed in the track manager, which allows direct access to the "Solo", "Mute", and "Record" functions, as well as hiding tracks in the project window and mixer.

To display the track manager go to "View" > "Manager" and select "Track Manager". The manager will open either in its own window or in the Docker (view page 177).

Menu: View -> Manager -> Track Manager

Keyboard Shortcut: Ctrl + Shift + S



Track View and Selection

The track manager contains all of the tracks in the current project. To select a track, double-click on the track number. The selected track will be highlighted in the project window. If the selected track is located outside the visible area, the project window will automatically scroll to display it.

Search for tracks: As in other manager windows, the track manager also provides a search option that allows you to search for tracks in the current window. Enter a search term into the box and press “Enter”. Found tracks are highlighted.

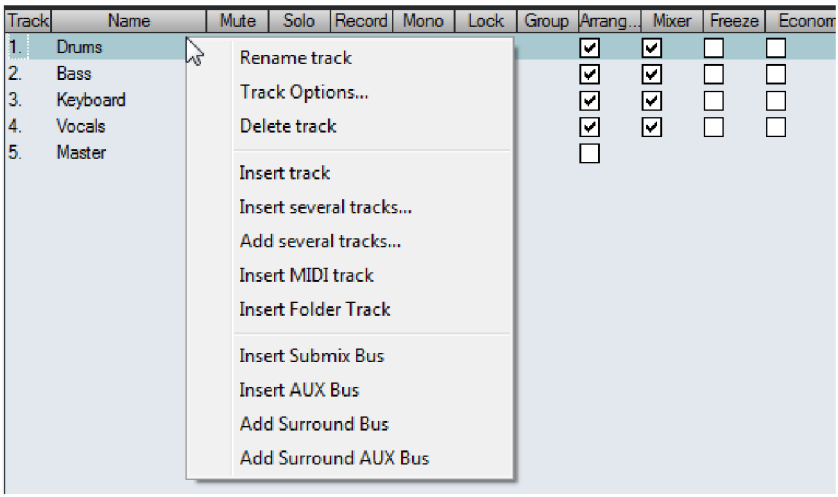
Sorting, Renaming, Deleting and Inserting Tracks

To sort tracks in the track manager, simply drag & drop them vertically to the desired position. You can delete tracks in the manager by selecting one or several of these tracks and pressing the "Del" key. Alternatively, you can also use the context menu to delete a track.

You can rename tracks by double-clicking on the track names and entering a new name. You can also use the context menu for this. Pressing "Tab" moves the cursor to the next editable value. You can use the up and down arrows to navigate vertically within a text column.

New tracks can be added directly in the track manager. To do this, right-click on a track entry and select “Insert track” from the context menu. You can also add or insert

folder tracks, submix buses, AUX buses, or surround buses. The "Track Options" are also made available by right-clicking.



Hiding and Displaying Tracks

In addition to direct access to the functions „Solo“, „Mute“ and „Record“, you can also hide tracks. To remove a track from the project window or mixer, click on the box in the “Arranger” or “Mixer” column to remove the check. Hidden tracks will still be played back and grouped objects on these tracks will still be edited.

To display a hidden track, reactivate the check box.

Write a Text File

Click the "Export text" button and Sequoia will create a text file with a listing of the tracks used in your project. The button beside it can be clicked to import a track list from a text file.

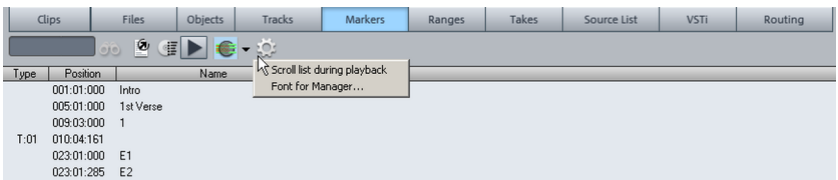
Marker Manager

The marker manager lists all of the markers contained in the current project and makes it possible to jump directly from the list to them or to play them.

To display the marker manager go to "View" > "Manager" and select "Marker Manager". The manager will open either in its own window or in the Docker (view page 177).

Menu: View -> Manager -> Marker Manager

Keyboard Shortcut: Ctrl + Shift + Alt + M



The following marker types can be displayed in the marker manager:

- Markers with numbers or names
- CD track markers
- CD subindex markers
- CD pause markers
- Source markers
- In/out point markers
- Tempo markers
- Beat markers
- AQ marker (audio quantization marker/transient marker)

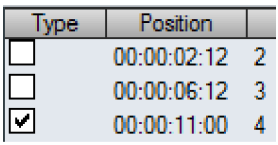
Marker manager information can be exported as a text file. To do this click on the "Export text" button in the toolbar. The Windows text editor opens with an excerpt from the marker manager list. The following information will be saved:

- 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).

Playing from and jumping to markers

To position the playback marker on a set marker, click on the type column of the marker in the marker manager.



Play back from a marker: Select the marker and click on the "Play marker in a loop" button.



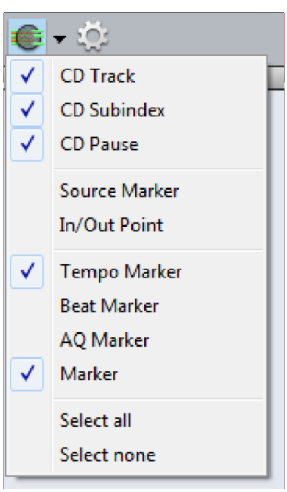
A range of two bars surrounding the selected marker is created and loop playback is started.

Marker View and Selection

Marker/CD Index Manager: This dialog lists all of the set markers and CD track indices in the project.

Detailed information about the Marker/CD Index Manager can be found in "CD/DVD" > "CD Track/Index Manager" (view page 1110).

Marker filter: An optional marker filter lets you limit the view to selected marker types. In the filter section click on all of the marker types you want to display. All marker types that are not selected will be hidden.

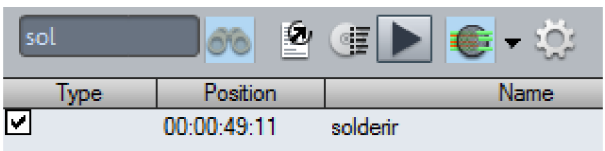


Export text: If you click on the „Export Text“ button, Sequoia will create a text file with a list of all the markers in the project.

Search for markers: As in other manager windows, the marker manager also provides a search option which allows you to search for markers in the current window. Enter a search term into the box and click on the binoculars button.



Found markers will be displayed.



Right-clicking on the marker list opens a context menu where you can specify the CD track, CD subindex, CD pause, CD end or position markers. In the context menu you can also insert new markers, rename markers, adjust marker positions and delete markers.

Set range between selected markers: In the context menu you can set a range between the selected markers.

Deleting markers, changing a marker's name, position, and type

You can delete markers directly in the manager by selecting one or several of these markers and pressing the "Del" key.

You can rename markers by double-clicking on the marker names and entering a new name, or you can use the context menu.

To change a marker position double-click on "Position" before entering a new position.

The "Tab" button switches to the next editable value. "Arrow up/down" navigates vertically within a column.

Tip: You can change the type in the context menu for the respective marker any time, e.g. if you want to change a position marker into a CD track marker.

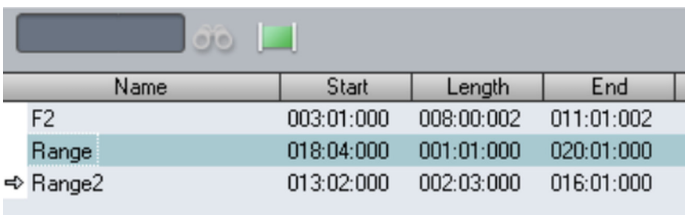
Range Manager

The range manager displays all of the ranges in the project and makes it possible to select them directly from a list.

To display it go to "View" > "Manager" and select "Range Manager". The manager will open either in its own window or in the Docker (view page 177).

Menu: View -> Manager -> Range Manager

Keyboard Shortcut: Ctrl + Shift + Alt + B



Define and Search for Ranges

Defining Ranges: To save a range in the manager, you first have to define a range in the project window (view page 122). Next, click "Define new range" in the toolbar of the range manager. Ranges that are saved using the program's “Save Range” feature (Alt+F2, Alt+F3, etc.) appear in the list appended with F2, F3, etc.

Search for Ranges: As in other manager windows, the range manager also features a search option that lets you search for ranges in the current window. Enter a search term into the box and click on the binoculars button or press Enter. Found Ranges are highlighted.

Edit Range Parameters

You can edit the following parameters in the range manager:

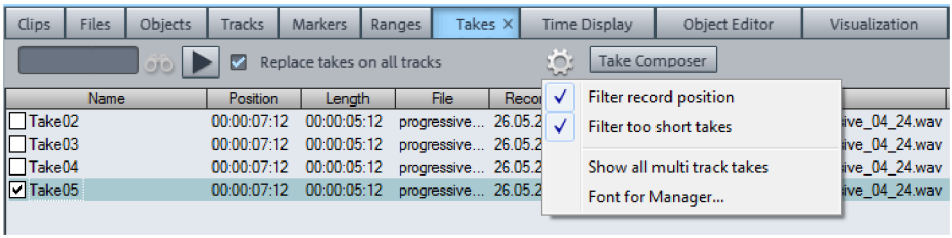
- Name
- Start
- Length
- End

To edit a parameter, double-click the value and enter a different value. Numeric values can be changed by clicking and dragging; "Ctrl + Shift" enables larger or smaller value changes. Pressing "Tab" moves the cursor to the next editable value. You can use the up and down arrows to navigate vertically within a name text column.

Take manager

Menu: Tools - > Manager -> Take manager

Keyboard shortcut: Ctrl + Shift + Alt + T



The take manager offers you a convenient option for selecting and organizing recording takes, loop recordings (view page 105), and destructive object edits (view page 163).

When graphical comping with the take composer (view page 203), you can edit together a new take from available take sections.

Note: A new take is also added after effects calculation and this is displayed immediately in the take manager if the option "Create copy" is selected in the destructive effects dialog.

Take manager – basic working principle

In addition to the audio and MIDI data, Sequoia also saves additional information like the time position and the track where the data was recorded in each recorded file. This data is stored as take information in the wave project and the MIDI object. For multitrack recordings, the takes also contain information like which other tracks were involved in the recording.

This enables each recorded passage to be assigned to a section, marking it as a take of a certain recording.

First, record as many takes of a certain passage as required. If a new take in an arranger track is recorded over an old one at the same position, no existing data will be overwritten. The new recordings will be attached to the end or saved to a new file.

If an object from a recording is selected and the take manager is opened, takes from the same track and original position are searched for in the whole audio material of the project, and the results are shown as takes.

Note: The take manager only accepts objects that were created by recording, but not those created by wave or CD import.

Take Manager – Application Examples

- To select the best takes after some punch loop recording sequences
- To find the best takes from several recording sequences of a production by means of predefined bar positions.
- To display a clear overview of all available takes in a specific SMPTE period.
- To edit the best parts of existing takes of recorded vocals into a "perfect take"

The basis for working with the take manager is always a selected object, e. g. the most recently created object after a punch-in recording. The current object take can be identified by the checkmark preceding the names in the take list. Now all the takes suited to the selected object are shown in the list. In the basic settings, this refers to all the takes from the same track and original position.

Take manager – options

Filter recording position: Only takes that overlap the same original time positions are shown, e.g. the selected object.

Filter out takes that are too short: Only takes that are at least as long as the currently selected take are shown.

Show all tracks: Takes of all tracks will be displayed.

Manager font: Use the dialog to specify the font, format, and size for the take manager

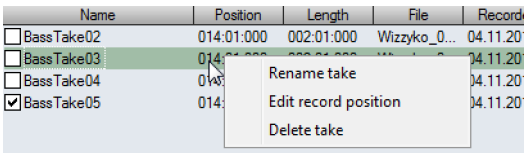
Select Takes

To select a take for the object, place a check mark before the desired take, or press Enter after you have selected the corresponding take in the take manager.

Select a take directly in the arranger window by applying "Ctrl + right click" to the object.

All of the takes shown in the take manager and the currently selected take are arranged one below the other in the take composer (view page 203). Takes can be compared directly in the take composer and a combination of the various takes can be edited together.

Rename take: Right-click on the take to open the context menu and rename it or edit the recording position in the take manager. Multiple selection is also possible.



Edit recording position: This option enables you to make changes in the position field of the take manager.

Delete takes: Right-click on the take to open the context menu and delete it. Multiple selection is also possible. Once you have deleted all of the takes except for one, you can decide if you would also like to delete the referenced file. Note that the deleted audio file will not be usable in other projects.

Note: If the audio file contains several takes but is not referenced through an object, it can be completely deleted from the hard drive without having to delete each take. To do this, use the "Remove unused samples (view page 655)" function in the "Clean up" section of the File menu.

For a better overview, easy-to-use take-naming options are available in the recording window (view page 835) even during recording.

Take management for multitrack recordings

For multitrack recordings, each recording creates associated classified takes on each track. All simultaneously recorded objects are grouped and also given the same take name.

Switch takes on all tracks: This allows all takes of a multitrack recording to be exchanged in one step.

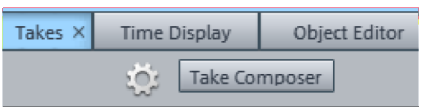
Note: Since the take manager display refers to the last clicked object, the reference track may be switched at any time.

Take Composer

The Take Composer is a function of the Take Manager that displays a selected track's individual object recording runs (takes) and the created Revolver Tracks (view page 135) below each other. Produced Revolver Tracks are located below the takes.

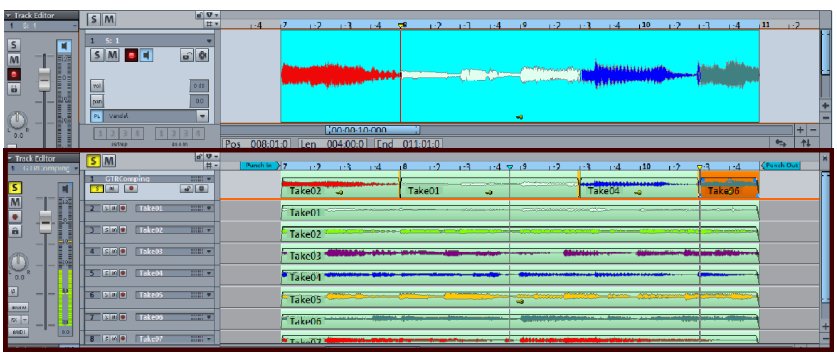
Use the Take Composer to add the best parts of individual recordings of an object to create the perfect take, or edit the Revolver Tracks for the corresponding track.

Open the Take Composer by clicking the corresponding button in the Take Manager.



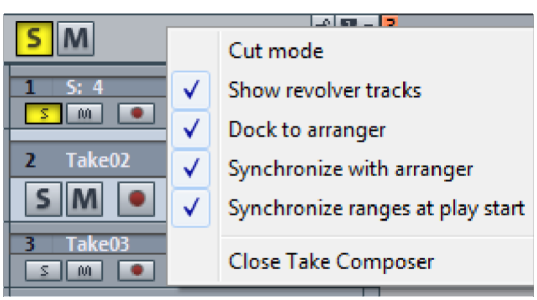
Note: Please make sure to select a take object before opening the Take Composer.

By default the Take Composer opens as an extra window below the arranger.



"Punch-in" and "Punch-out" markers will be displayed for the last selected object in the arranger.

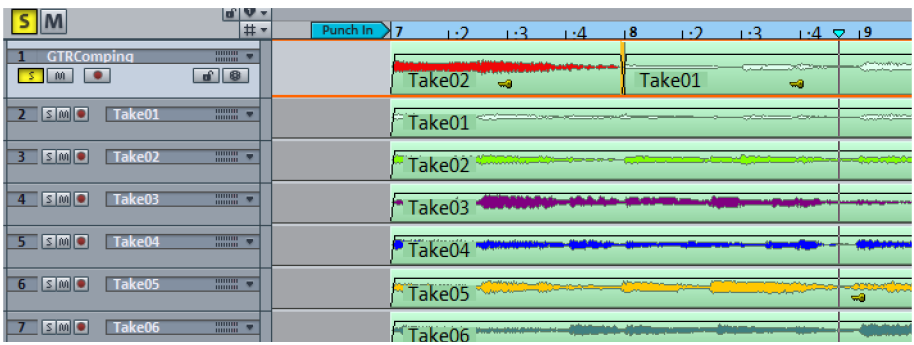
In the Take Composer use the space bar to initiate playback in sync with the arranger.



In the Take Composer's marker menu you can specify whether you want to display revolver tracks (view page 135), whether the Take Composer should open in its own window or should be docked to the arranger, whether the Take Composer should run in sync with the arranger and whether the last selected range in the Take Composer should be synchronized with the arranger before playback.

Here you can also switch to cut mode to quickly execute "comping" (view page 206) (see below) or close the Take Composer.

The current take selection for the respective track is displayed in the first track of the Take Composer. During "comping", you can edit the best passages of the individual takes together in this "Comping Track" by using the editing operations described below.



The individual recording takes are arranged in the Take Composer 2 onwards under each other.

Monitoring with the arrangement

To listen to the recorded passages individually, switch the corresponding track to "Solo" and start playback. The solo mode "solo exclusive" is used automatically. Start the playback with the spacebar, the take is played together with the arrangement.

Monitoring without the arrangement

To play a take without the arrangement, press Ctrl + Spacebar to start playback.

Note: If you select an object in the Take Composer, you can listen to only that object with "Ctrl + Space", ignoring the "Solo" settings for the take tracks.

If multiple takes should be played back simultaneously in the Take Composer, click the corresponding track header and press the shortcut "Alt + S". This switches the track to (non-exclusive) solo mode.

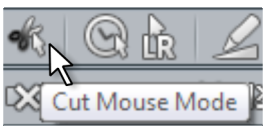
Comping

"Comping" refers to the process of combining several takes to create a single one.

Comping in Cut Mode

For this process, the take composer offers a specially modified scissors tool for quickly cutting/copying take sections into the first take composer track.

Select the scissors tool by selecting "Cut Mouse Mode". The mouse pointer becomes a scissors icon.



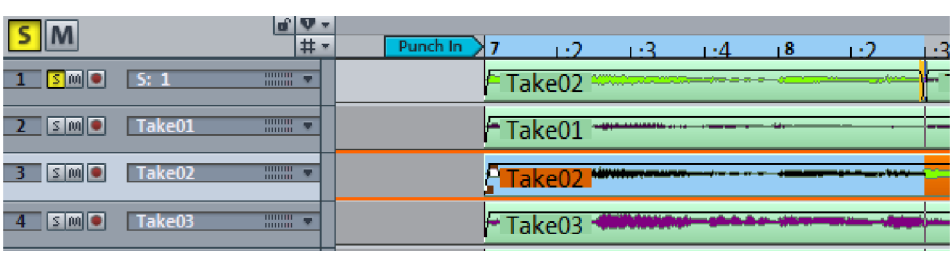
Now click on the desired cut point in the respective take. The selected take is inserted from the current position up to the end of the first track in the take composer. Auto crossfade (view page 735) mode is activated as a default during the process. Cut the suitable sections from the respective take in this way along the timeline into the top track.

If you only want to transfer specific ranges of a take into the comping track, drag out a limited range with the scissors tool by holding down the mouse button. When the mouse button is released, the section indicated in the respective take will be copied to the first track.. The keyboard combination "Shift + click" replaces an object in the comping track with the contents of the take composer track over which you are holding the scissors tool.

To play a take from the cursor position in "Cut" mode, use the keyboard shortcut "Ctrl + mouse click".

Comping in Universal Mode or in Range Mode

Another comping option is to select the desired take range in a track. In "Universal Mode" or "Range Mode", drag out a range in a take composer track. Use the keyboard shortcut "Shift + C" to transfer the selected object's section to the first track. Auto crossfade mode (view page 735) is also activated by default during this process.



If however you don't mark/select a take range, the shortcut "Shift + C" will insert the selected take from the current position up to the end of the first track in the take composer.

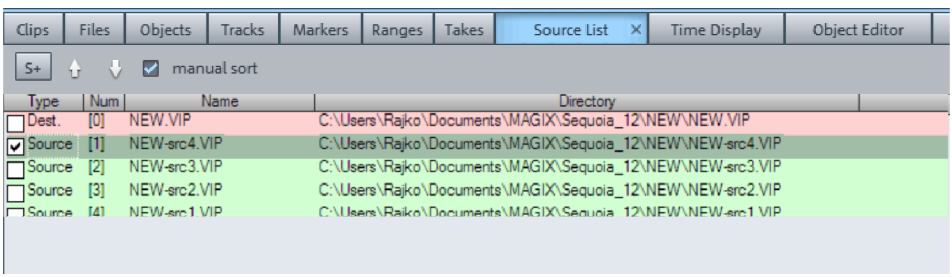
This enables you to select the best parts of the available takes and add them to the top track in the take composer. These parts may be combined to create the perfect take. When the take composer project is closed, changes may be confirmed, which transfers the take manager's comping track into the original project.

Source List Manager

The Source List Manager shows all source and destination projects contained in the current session and makes it possible to directly activate projects and place them in the foreground. In order to show the source list, select "View" > "Manager". The manager opens either as a separate window or in the docker (view page 177).

Menu: View > Manager > Source list manager

Keyboard shortcuts: Ctrl + Shift + Alt + S



To enter a project into the source list, click the "S" key ("Add active project to the list") in the source list toolbar. The source list will show the entry only when at least one project with a destination track is found in the current session.

Activate individual projects in a session by clicking the checkbox in front of the project type. Source list entries can be sorted according to increasing or decreasing type, creation number, name and project file. To sort projects manually using drag & drop, activate the "Manual sort" check box first.

Tip: You can remove projects from the source list and close them simultaneously by selecting the project in the list and pressing "Del". A confirmation request prevents deleting a project that has not yet been saved. More information about this is available in "Source Destination Cut -> Source Destination Cut with several projects" (view page 407).

VST Instruments Manager

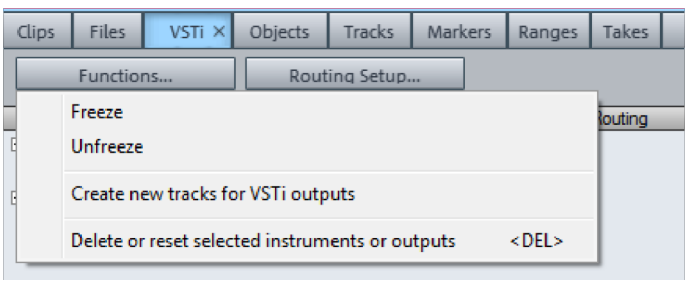
The VSTi manager enables easy display, insertion and deletion of VST, MAGIX, and Rewire instruments in the current project.

Menu: View > Manager > VST Instruments Manager

Keyboard shortcuts: Ctrl + Shift + I

Right-clicking on the selected instrument opens the plug-in dialog of the selected VSTi. If the plus sign beside the instrument display is clicked, the outputs to the assigned tracks will be shown. By right-clicking the "Output Routing" column you can select a track that will only receive the output of the respective individual outputs. The assigned output then appears in the plug-in slot of the target track.

The general instrument outputs are routed in the VSTi manager in the "Routing Setup". You can specify how the outputs will be routed in the dialog that appears. You can either route all outputs to the current track or create new mono or stereo tracks for each output. Pressing the "Del" key removes a selected instrument completely from the project. If only individual output signals are selected, routing to a track can be canceled by pressing the "Del" key.



Detailed information about instrument routing settings (view page 279) and about managing single outputs is provided in the chapter "Routing VST instruments with the VSTi Manager (view page 282)".

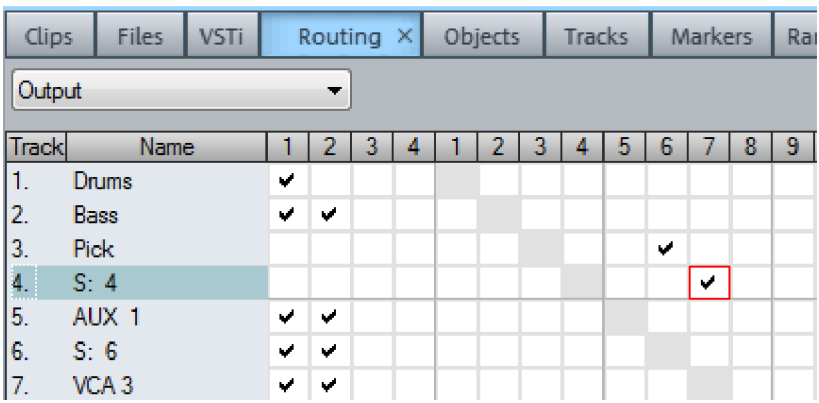
Through "Functions" you can access the "Freeze" and "Unfreeze" commands of the integrated VST instruments. Here you can "Create new tracks for single VST outputs" as well as "Reset or delete selected instruments or outputs".

Routing Manager

The routing manager provides you with a clear matrix display of the inputs, outputs and AUX sends and VCA groups of all tracks.

Menu View > Manager > VST Instruments Manager

Keyboard shortcut: Ctrl + Alt + Shift + R



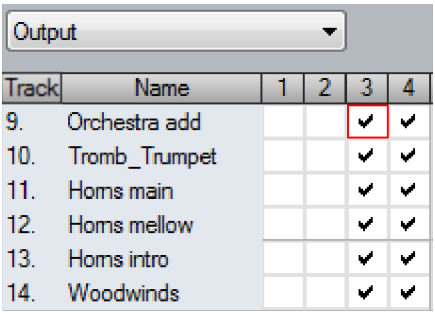
By simply clicking in the matrix fields you can assign all available inputs and outputs to your project's tracks or lift existing routings.

By right-clicking you can define the selected output as a "Direct out" (red fader display), a "Pre-fader out" (yellow fader display) or "Post-fader out" (orange fader display) or open the stereo panorama dialog to set the panorama (view page 392) for the respective outputs.

This is how you can assign multiple sequential tracks to the same inputs/output (vertical):

1. Select the input/output of the first assigned track by clicking on the corresponding Matrix field (e.g. "track 9", "outputs 3+4").

2. Assign the same input/output using Shift + mouse click to the Matrix field of the last track to be assigned, e.g. "track 14", "outputs 3+4").

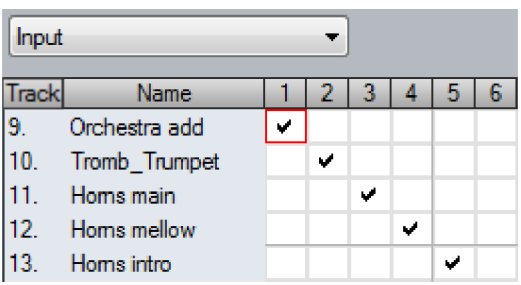


In vertical routing all selected and intermediate tracks are assigned from top to bottom to the same input or output.

This is how you can assign multiple sequential tracks to the next row of inputs/outputs (diagonally):

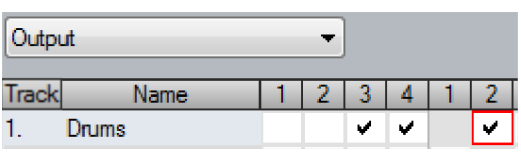
1. Select the input/output of the first track to be assigned by clicking the corresponding matrix field (e.g. "track 9", "input 1").

2. Following the diagonals in the matrix, assign the corresponding input/output using Shift + mouse click to the matrix field of the last track to be assigned (e.g. "track 13", "input 5").

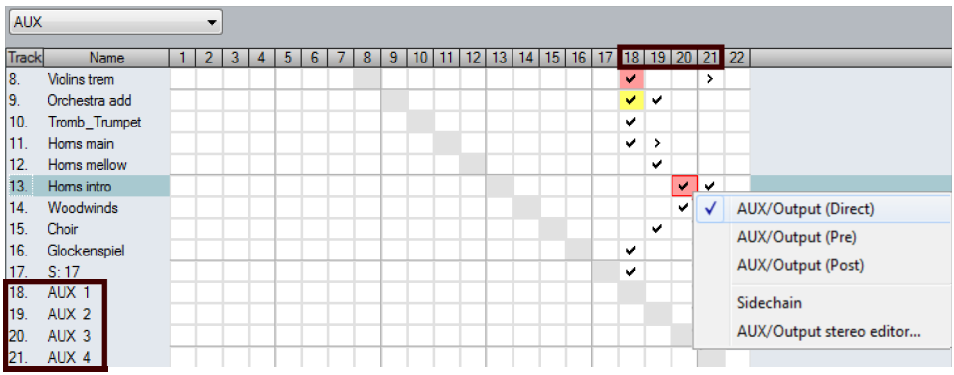


In diagonal routing, all selected and intermediate tracks will be assigned to corresponding inputs and outputs in sequential numbering.

In the output section, you can repurpose any track as a submix bus. Clicking on the corresponding matrix field behind the available output transforms the track into a submix bus.

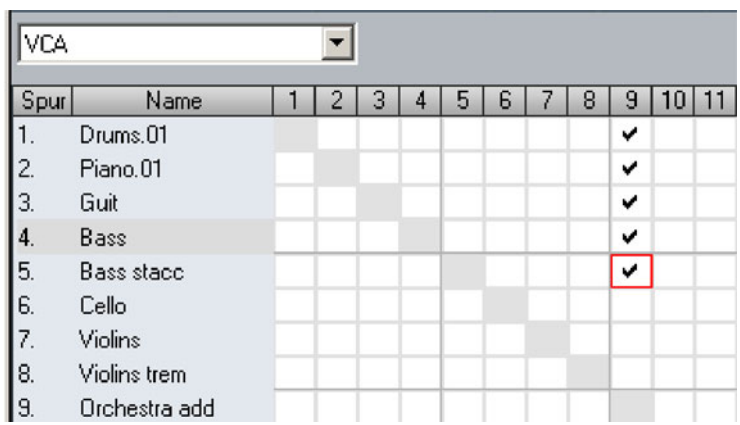


You can also route Aux busses to the individual tracks (view page 384). By right-clicking you can define the selected AUX bus as a "direct out" (red field), "pre-fader send" (yellow field) or "sidechain send" (view page 387) (">" symbol), or open the stereo panorama dialog to set the panorama (view page 392) for the respective AUX bus.



VCA Group Assignment in the Routing Manager

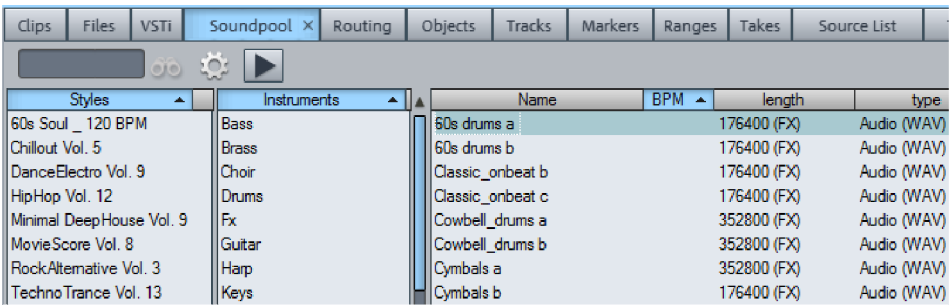
Here you can select each track as VCA Group (view page 372) and assign it as many tracks as you like.



In the example above, you define track 9 to VCA group. By clicking on the "VCA" button in the respective mixer channel, you can see that the status of the fader has changed: "Fader is VCA master". Tracks 1 to 5 are controlled by the VCA group in track 9.

Soundpool manager

In the Soundpool Manager you can integrate you Soundpool loop content in a clear form.



In the input field next to the magnifying glass enter a combination of letters which should be in the loop you're looking for. Clicking on the magnifying glass symbol starts a search for the corresponding loop name.

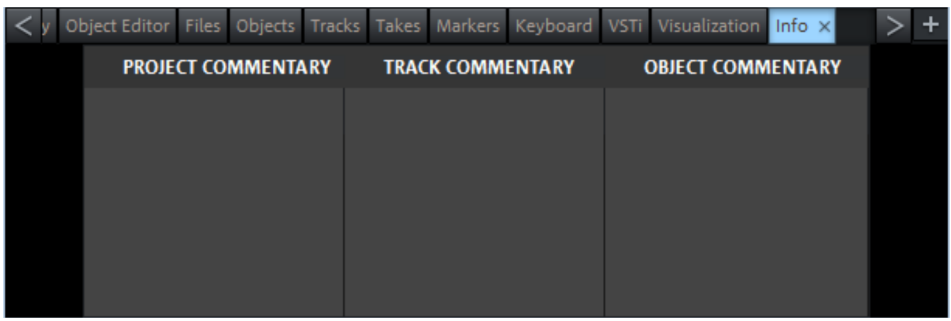
You can access a selection menu by clicking on the gear symbol. Here you can:

- Add styles: Search the folder system for the desires Soundpool style
- automatically play the selected loops
- Hide instruments without available loops
- reset the Soundpool. The list of styles shown in the Soundpool will be deleted.
- Clean Soundpool: With this function only the styles available on the hard drive will be displayed in the Soundpool. Soundpools on storage devices as well as styles deleted or moved from the hard drive will be removed from the Soundpool view.

With the play arrow symbol you can start and stop the selected loop.

Info manager

The info manager displays notes for individual objects, tracks or the entire project on a clearly-arranged interface.



You can write notes concerning the currently selected project, track or object in the three fields.

Keyboard

The keyboard enables software synthesizers to be played and recorded directly via the on-screen keyboard.

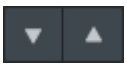


If a track synthesizer is not present, then a new track will be added and a new synthesizer plug-in will be added when the view is switched to “Keyboard” (Vita with Sound Acoustic Bar Piano).

The keyboard always controls the synthesizer in the track for which MIDI recording has been activated.

You can click the keyboard with the mouse to play the instrument. The closer to the bottom edge you click on the “virtual keys,” the louder the sound will play. Of course, you can't seriously play music by clicking with the mouse (this function is more suitable for testing out sounds quickly). That's why you can also play the keyboard with the keys on your computer's own keyboard.

Note: This works only after you first click once on the keyboard using your mouse. Otherwise, pressing the computer keys will function as keyboard shortcuts for the different functions in Sequoia. If the computer keys control the program's keyboard, then the piano keys will display the corresponding keyboard characters.



Use the vertical arrow buttons to move the octave range that the keyboard can be played in.



You can use the horizontal arrow keys to select the next or previous sound of the synthesizer or they can be selected directly in the list field on the side.

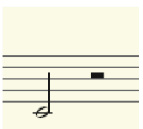


This button opens the editor window for the synthesizer for fine tuning the sound.

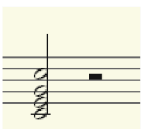
Arpeggiator



The arpeggiator is a special function that can be used to create full chords or broken chords (arpeggios, i.e. the notes of a chord played in quick succession) by pressing a single key.



C note



C major chord



C major arpeggio with 1/16 notes

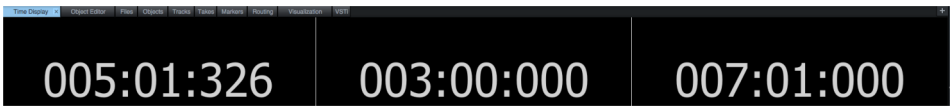


1. This button activates the arpeggiator.
2. This switch determines whether the played note will generate no chord, a major chord (Dur) or a minor chord (Moll).
3. This switch determines the type of arpeggio. When set to the far right, a normal chord is played. The other positions are up, down or up and down. The figures are repeated as long as the note is played.
4. The tempo of the arpeggio is set here and can range from 1/4 notes (slow) up to 1/32 notes (very fast).

Time display

Shortcut: Ctrl + Shift + Z

This function shows a zoomable time display. This makes it possible to maintain a good overview of the project even at longer distances. The time display is opened in the docker with three display fields, one for position, one for range length and one for the range end.



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..."). In the next section you will find a list of all the possible display fields.

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 colors settings to each field.

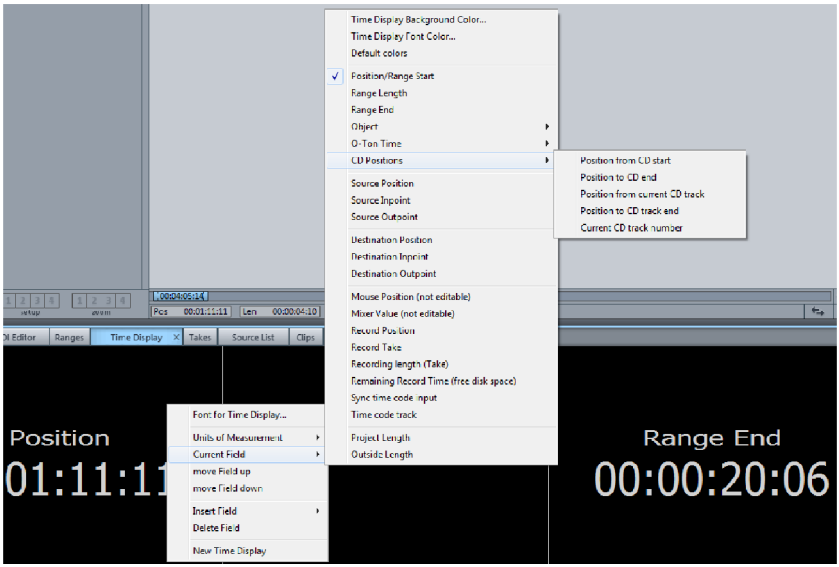
Move field forward / Move field back This function can be used to change the order of the fields.

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.

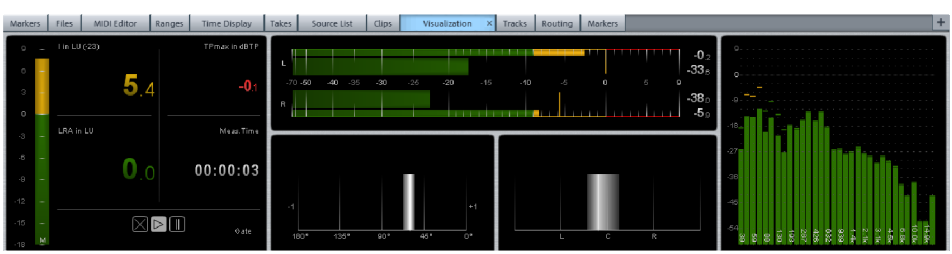


Visualization

The visualization displays the audio material in graphic form. You can integrate the visualization into the Docker (view page 109) or open it in its own window.

Menu: View > Visualization

Keyboard Shortcut: Ctrl + Alt + Shift + V



Normally the visualization is divided into several display windows. By right-clicking on each window, you can choose between the following display options:

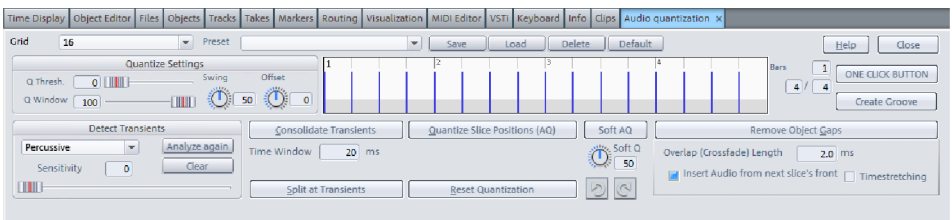
- Peak Meter (peak meter display)
- Loudness Meter
- Surround Meter
- Phase oscilloscope (vectorscope)
- Correlation Meter
- Direction meter
- Spectroscope

- Spectrogram
- Bit meter
- Oscilloscope
- tuner

Find out more information about these options and settings can in the menu reference in the manual PDF (View menu > Visualization (view page 1121)).

Audio quantization

Audio quantization is a special function for adjusting and snapping a recording to a grid.



These commands reflect the typical working procedure when "adapting" a multitrack drum recording to the VIP grid and are applied to the object selection. The "One click button" automatically executes the "Determine transients", "Consolidate transients", "Cut at transients", "Quantize object positions (AQ)", and "Remove Object Gaps" commands sequentially.

During determination of transients, you can set the sensitivity of the detection with the "Sensitivity" controller. The higher the value, the more beat markers are created.

By using "Recalculate" and "Delete", you can recalculate or delete an already created audio material analysis.

If you would like to connect the slices you have created to crossfades, then enter the length of the fade into the respective field "Overlap (crossfade) length".

Several gaps between slices can be closed either by object time stretching or applying audio material from the slice to the right of the gap.

For a detailed description of the audio quantization wizard, see the menu reference > "Object" menu in the manual PDF.

Example of use:

1. Enable display for audio markers/transients in the view options (System options ("Y" key) > "Design" -> "View Options" -> "Objects" -> "Audio markers/Transients").
2. Load a VIP with a multitrack drum recording where the drummer has played to the click, so overall the VIP grid already matches the drummer's tempo.
3. Select a range for audio quantization. We recommend about 8 to 16 bars to guarantee a quick calculation. The analysis of transients for an entire song can take a long time. Trim the objects to the selected range size.
4. Select all drum objects, open the audio quantization dialog, and move the "Sensitivity" slider. Beat markers will now appear in the objects at the positions of the transients. Set the slider so that the main hits correspond to the beat markers. You can use the F2 and F3 keys to move the playback marker to the previous or next marker.
5. To cut objects for all tracks at the beginning of transients, specify the beat markers with the "Consolidate transients" command. Sequoia adjusts the beat markers of all tracks within a period defined in the "Time window" parameter to the beat marker position farthest to the left.
6. To separate all quantization objects, use the "Split at transients" command.
7. Use "Quantize object positions" to adjust the individual positions of the slice objects. You also have the option of changing the quantization settings here.
8. In some cases the quantization will result in gaps between the slice objects. These can be filled with the option "Remove Object Gaps". The relevant objects will be stretched.

Effects

Overview

Sequoia contains the following effects and effects plug-ins:

- **Amplitude**: Normalize, normalize (quick access), loudness adaptation, fade in/out, set zero
- **Dynamics:** Advanced Dynamics, Multiband Dynamics, sMax11, eFX_Compressor, eFX_Gate, eFX_Limiter, AM-Munition, AM-Track, AM-Phibia, AM-Pulse
- **Frequency/Filters**: EQ116, FFT Filter/Spectral Analysis, Brilliance Enhancer, Filtox
- **Delay/Reverb:** Delay, Room Simulator, efx_Reverb, efx_StereoDelay, Ecox, Variverb Pro.
- **Time/Pitch**: Melodyne, Resampling/Time stretching, Elastic Audio
- **Distortion**: Vandal, BitMachine, eFX_TubeStage
- **Restoration**: DeClicker/DeCrackler, DeClipper, DeHisser, DeNoiser, get noise sample, eFX_DeEsser, remove DC offset (offline), manual DeClicker, spectral cleaning (offline), spectral editor - Algorithmix reNOVAtor (offline)
- **Stereo/Phase**: Switch Channels, Multiband Stereo Enhancer, Invert Phase (both channels, left channel, right channel), eFX_TremoloPan
- **Modulation/Special**: Vocoder, eFX_ChorusFlanger, eFX_Phaser, eFX_VocalStrip, Corvex, reverse
- **Sample manipulation**: Adjust sample rate (offline), reverse, form loop (offline)
- **essential FX Suite**: eFX_ChorusFlanger, eFX_Phaser, eFX_Reverb, eFX_StereoDelay, eFX_Compressor, eFX_Gate, eFX_Limiter, eFX_DeEsser, eFX_VocalStrip, eFX_TubeStage, eFX_TremoloPan
- **Vintage Effects Suite**: Corvex, Ecox, Filtox
- **Analog Modelling Suite**: AM-Munition, AM-Track, AM-Phibia, AM-Pulse
- **VariVerb Pro**
- **Vandal**
- **Cleaning & Restoration Suite:** DeClicker/DeCrackler, DeClipper, DeHisser, DeNoiser, Brilliance Enhancer and Spectral Cleaning

Track and Master levels also include:

- **Independence FX** (over 40 high-quality filters and effects)

Effects – Organization and Workflow

Internal real time effects

During effects editing, effects are calculated during playback or export in real time. The effects settings can be changed at any time. Real-time effects can be applied to different positions in a project:

- **Object effects** (access via the object editor or the "Effects" menu)
- **Track effects** (accessible via "Track" > "More" > "Track effects", the track editor, track header plug-in selection field or the insert section of the mixer)
- **Master Effects** (can be accessed via the Master section of the Mixer)

The signal passes through the real-time effects in the following order:

1. Object effects
2. Track effects
3. Master effects

Internal offline effects

The object effects can also be added as offline effects. To do this, activate "Process effects offline" in the "Effects" menu.

In this editing process, effects are added on a one-time basis to the original wave file or a new wave file. In contrast to real-time effects, no additional CPU resources are required during playback of the effect.

You can switch between non-destructive and destructive offline audio editing in the effects dialogs ("Create copy" option):

- **Non-destructive audio editing:** The effect for the selected area will be written into a temporary file. When saving the WAV file, the original and the changed areas will be recombined again.
- **Destructive audio editing:** The effect for the selected area will be written directly into the original file. The "Undo" function is available for reversing operations.
- **Application on a VIP object:** The effect will be calculated for the selected area of the object. Depending on the settings for the application of destructive effects, calculation takes place directly in the original material, at the end of the original material or in a new file.

To see precise settings for effects editing, go to "Destructive effects editing" in the system settings ("Y" key).

Effects plug-ins

Along with internal effects, additional effects can also be applied as plug-ins. Sequoia supports plug-ins in the following formats:

- **MAGIX plug-ins**: A selection of internally integrated VST effects that come with the program. Includes the Analogue Modelling Suite (only for Samplitude Pro X5 Suite and Sequoia), the Vintage Effects Suite, AM-Munition, essentialFX (eFX), Vandal, and VariVerb II.
- **VST effects-**: This includes all external VST effects

These effects can be accessed in the track editor, object editor, track header or via the "Insert" section of the mixer, the routing dialog and the master section.

Effect slots

Effects are loaded to their location via the effect slots.



Effect slots in the object editor, track editor, mixer and track header

Clicking on the empty insert slot opens the plug-in browser (view page 222), where you can load a plug-in to the slot. Click on an occupied slot to activate/deactivate the plug-in. Right-clicking on the slot opens the interface of the plug-in. The arrow next to the respective insert slot opens a menu with various functions. You can ope the plug-in browser again to switch out the plug-in or remove it ("No effect"). The "Plug-ins" button activates or deactivates all effects in the channel. A visual indicator (*) is displayed for plug-ins that were previously active and are active the next time "Plug-ins" button is pressed.

Plug-in browser

The plug-in browser or FX browser allows all real-time effects and plug-ins to be loaded to the slot at object, track and master levels. It offers centralized access to all available plug-ins and instruments.



1. **Full text search:** After opening the FX browser, the input field is selected for the full text search. Simply start typing and enter the name or part of the name of the plug-in you're looking for. The search function filters the plug-in list below and selects the first search result. You can use the enter key to load the selected plug-in to the track slot.

2. **Plug-in list:** All plug-ins in the list are listed in a tree structure according to setting under **4.** Using the arrow keys you can move through the list to select a plug-in. Ctrl + Arrow left collapses the entire tree and Ctrl + Arrow right opens up the entire tree.

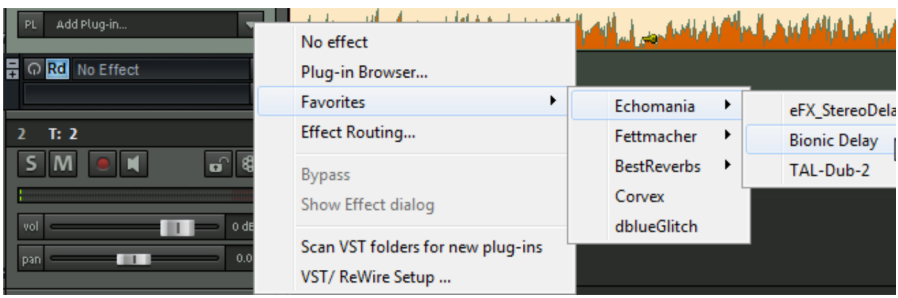
3. **Instruments/Effects:** The display can switch between instruments and effects. Instruments are only used at track level and can therefore only appear there.

4. **Path/Manufacturer/Category:** The tree structure can be sorted according to storage location, plug-in manufacturer or plug-in category.

5 **Last used:** This view lists the previously used plug-ins.

6 **Favorites:** You can mark your favorite plug-ins with a star. If the "Favorites" button is activated above, only your favorite plug-ins will display in the list.

Double-click on "New folder" to create folders and double-click on the name of the folder to rename it. Drag & drop to sort your favorites into folders – you can also move a folder within another folder. Favorites are accessible via the plug-in slot menu and the folders are displayed there as sub-menus.



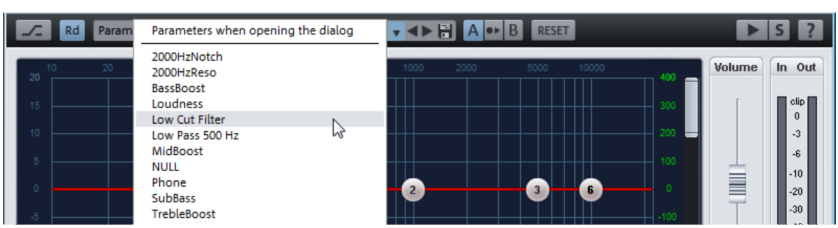
7 Here you can see whether a plug-in is VST2 or VST3, and whether it's a 32-bit or 64-bit version.

8 **Remove/Replace:** These buttons are only active if you open the FX browser by clicking an arrow button on an occupied slot. With "Remove" you can remove a loaded plug-in, with "Replace" the plug-in is replaced by the plug-in selected in the list.

9 **Add:** Loads the selected plug-in and closes the plug-in browser.

Saving Effect Parameters (Preset Mechanism)

Most of the effects include preset selection boxes. Presets appear in the selection box when they are moved to or saved in the folder "Program Files" > "MAGIX" > "Sequoia" >fx-preset".



If a preset is not saved in the "fx-preset" folder, you can open it using the "Load" function. However, when opened this way the presets will not appear in the selection box. In this case the path to effect preset has to be set manually.

Effect Dialog Bar

In several effect dialogs you have direct access to the automation button which also provides access to the automation menu. This is made possible by a newly created header in the respective dialogs.



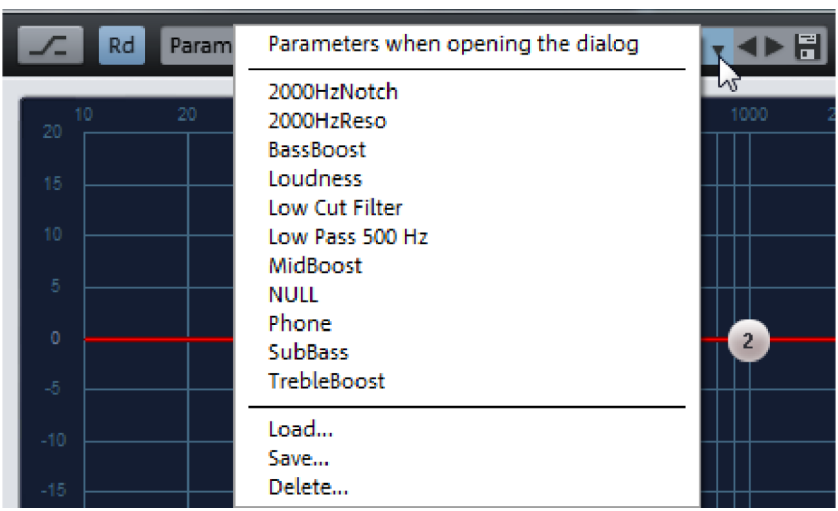
In the presets input field you can load, save or delete settings.

By selecting "Parameters when dialog is opened", you will undo all changes made in the dialog since it was opened.



By closing the dialog, you will apply all current settings.

Clicking on the downward pointing arrow, you will see all presets at a glance. Below the preset list you can open dialogs for loading additional presets and for deleting selected presets. You can navigate through presets using the two arrows pointing in horizontal directions. Current settings are saved to disk.



Here is an overview of the other control elements:



Bypass: The effect is removed from the signal route. This enables the unedited signal to be easily compared to the result of the current effect settings.



Mode: By clicking on this button you can toggle between the selected automation write/read mode. By right-clicking you can get to the Automation context menu.



A/B comparison:This control element is only displayed when it is supported by an effect. You can copy each selected setting using the arrow symbol to a different location. This way you can experiment with successful settings without losing them.



Reset: This control element is only displayed when it is supported by the respective effect. All parameters are reset to default settings.



Play button.



Solo button. This control element is only displayed when it is supported by the respective effect. The Solo mode will be activated for the corresponding effect.



Help: You can get extra information in the help section.

Track effect settings

By right clicking on the track/channel number in the Track Editor, Header or Mixer a Context menu will open. In the "Track effects" category you can copy, reset, save or load current/discarded track effect settings. There are also a wide range of presets available, which offer you possible effect combinations and instruments for various applications.

The track effect settings are .TRK files. Save your personal track effect settings in the program directory in "FX presets" -> "Track FX". You can also create subfolders here. We've already included a selection of useful presets, e. g. for "Mid-side processing". The track effect settings of a VST instrument can also be saved (including parameters and all subsequent track effects), and transferred to other tracks.

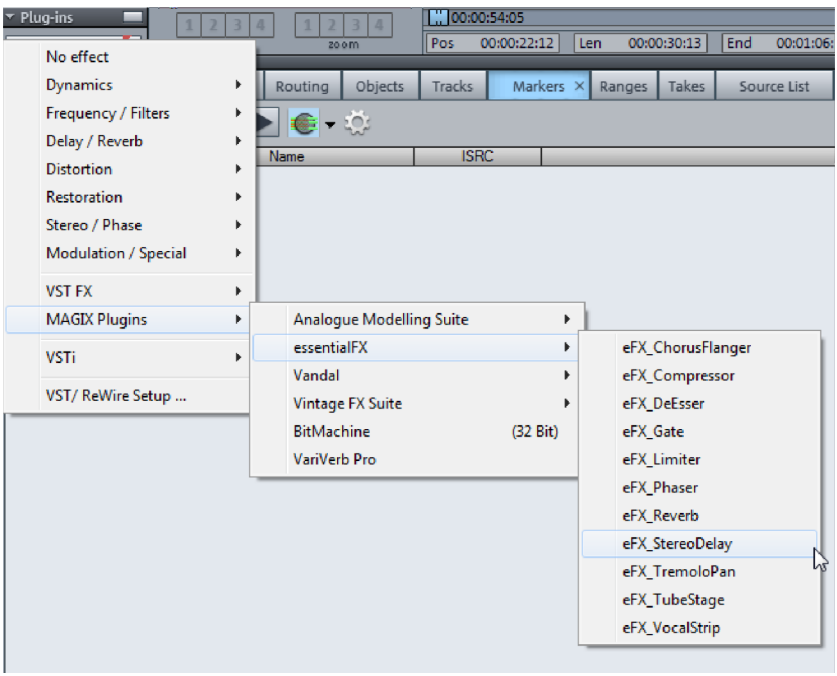
Effect Routing

The sequence of all realtime effects can be adjusted individually for each track. This enables individual sequences to be set for the effects.

More information about this can be found in the chapter „Mixer“ > „Effect Routing/Plug-ins Dialog (view page 388)“.

MAGIX Plug-ins

MAGIX plug-ins are **effects** that are accessible via the plug-in selection field in the object editor, track editor or track header, the mixer, or via the "Effects" menu.



The following will give you a brief overview of the MAGIX plug-ins.

essentialFX

The essentialFX Plug-in Suite offers high-quality audio tools for a wide range of tasks. Each plug-in has a number of important qualities for producing quick and accurate results:

- Low-resource DSP functions despite top quality algorithms
- Low space requirements for interfaces
- Easy operation thanks to reduction of function elements featuring only the most important parameters
- Consistent operation of all interfaces
- Simple, straightforward preset management with the tried-and-true file/folder basis; simple upgrades for users
- Color coding: Each plug-in has a color code in the top part of the window for quick identification; modulation effects = blue, dynamic plug-ins = orange

The following essentialFX are available: eFX_Reverb (view page 982), eFX_StereoDelay (view page 984), eFX_ChorusFlanger (view page 979), eFX_Phaser (view page 980), eFX_Compressor (view page 985), eFX_Limiter (view page 989), eFX_Gate (view page 988), eFX_DeEsser (view page 993), eFX_VocalStrip (view page 995), eFX_TubeStage (view page 991), eFX_TremoloPan (view page 994).

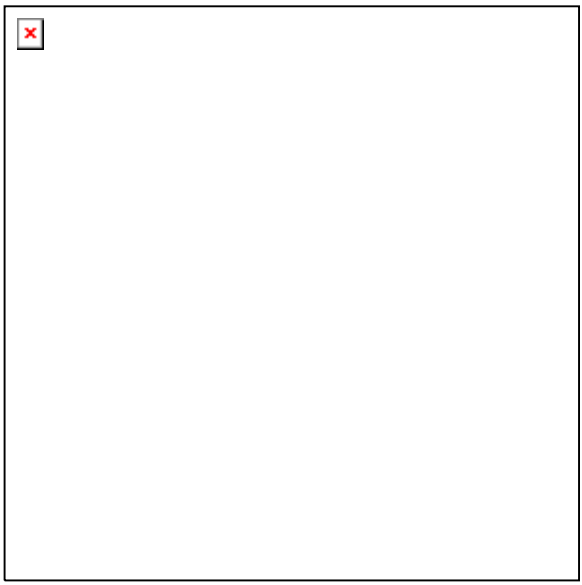


Vintage Effects Suite



This suite enhances the repertoire of Sequoia with a chorus/flanger, delay and filter plug-in. CORVEX, ECOX and FILTOX are each based in the same basic handling principle: as required, a modulator (LFO) controls almost all knobs accessible via the "frontplate". VariVerb is a classic reverb unit which performs classic and modern algorithm-based reverb generation without accessing impulse responses.

coreFX Suite

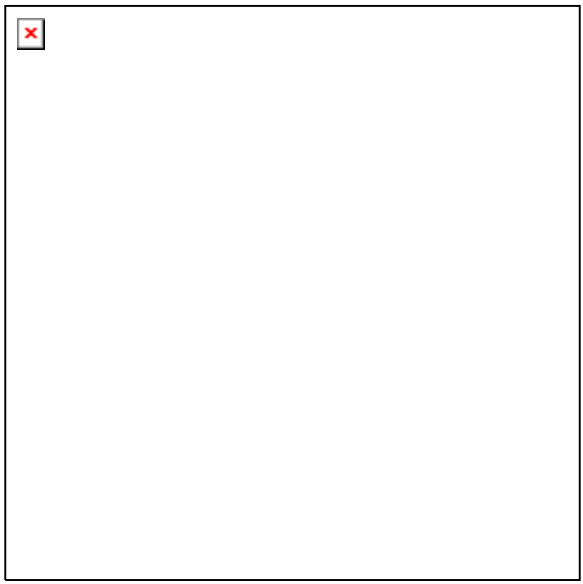


The coreFX are a collection of several effects plug-ins with a clear user interface that ensures a uniform operation. There are 9 core FX that can be categorized in two groups: delay-based and dynamic effects, plus the Volume Former. Inside these groups the effects differ just from the range of the parameter values which results in different audible effects. The Volume Former is a special case, it cannot be assigned to either of these two groups.

Tip: Each of the plug-ins contains many hands-on presets for daily use. Just try out the presets to get an idea of how the plug-ins work!

For a detailed description, click on the name of the plug-in in the top left corner of the plug-in window and open the help document.

colorFX Suite



Digital recording and playback technology has been brought to perfection over the last few decades, so it's no longer a big deal to record music that is free of noise or distortion and faithfully reproduces every little detail of an instrument.

But in the course of developing towards this, we have learned to live with the shortcomings of the old recording techniques. Their special timbre corresponds to our listening habits and is indispensable for many types of music: the analog distortions of tube amplifiers, the warmth and noise of tape recordings, or the dirty, punchy sound of early 8- or 12-bit digital technology.

The effects of colorFX Suite give you back this very special sound coloration: **Tube Distortion** simulates the behavior of tube amps for authentic guitar sound and analog warmth by adding harmonic overtones and distortion. **Tape Machine** authentically simulates the sound of the magnetic tape recording, with all the shortcomings such as speed fluctuations, tape wear or noise that make up the nostalgic charm of old recordings, and adds extra pressure through tape saturation. **Bitcrusher** enables authentic lo-fi sounds by reducing bitrate and sample rate. Brutal digital distortions for the very hard industrial sound are just as possible as hardly audible digital noise.

An automatic level control for the input and output level ensures that the effects can always be optimally used. An additional modulation unit, which can modulate levels, filters and other parameters depending on the input level, the sidechain input or an LFO synchronized to the project tempo, makes it possible to use it as a dynamic or creative effect.

For a detailed description, click on the name of the plug-in in the top left corner of the plug-in window and open the help document.

3D Reverb

3D Reverb is a unique and powerful algorithmic reverb effect. It produces the sound of a wide range of acoustic spaces, from small rooms to large concert halls. It works equally well in stereo and 3D multi-channel applications with up to 10 audio channels.

When used on a surround AUX bus or surround master, the input signals are routed into the virtual room according to their 3D panning, which supports the spatial impression of surround sound mixes.

The calculation of early reflections is based on a geometric spatial model that is automatically adjusted to the audio channel configuration to obtain a reflection pattern that exactly matches the positioning of the incoming audio signals. This allows the existing 3D panning in a project to be used directly to create a reverberation that matches the placement of the source signal to produce a convincing 3D sound.

For more information on the 3D reverb please read the corresponding chapter (view page **Fehler! Textmarke nicht definiert.**) in the menu reference "Effects menu".

Analogue Modelling Suite



AM-munition is an extremely versatile, dynamic tool for editing groups or signal sums, especially in the domain of mastering. It has separate units like compression, filtering, side-chain, limiter and clipper. All modules and parameters are optimized to perform the fundamental function without any compromises: Effective enrichment of

the program material without causing bothersome artifacts, a high reachable volume and an 'analogue' behavior with an individual sound signature.

AM-phibia is a tube amplifier/channel strip. It combines an optical compressor with a pre and post filter unit. The filter presets offer appropriate settings for various types of input signal. When interacting with the compressor section, am-phibia may be used as a vocals pre-amp, tube guitar amp, or simply for creating a warm sound.

AM-pulse is a "transient modeler", a creative tool for editing envelope and sustain processes on percussive or dynamic signals.

AM-track is a combination of an analog compressor and a tape simulator in a single device. This is used primarily for so-called "tracking", i.e. editing individual channel strips or subgroup signals.

VariVerb Pro

VariVerb (short for: VARIable reVERB) is a high-quality reverb plug-in that runs on an algorithmic basis. It provides a series of rooms, halls, reverb plates, spring reverbs and so-called non-linear effects, which can be edited comprehensively and easily.



Vandal

The main concept of the Vandal guitar amp (view page 1047) consists of virtual switching that models three different preamps and two power amps. These can be accessed in the functional design of a unique frontal display.



BitMachine

There are some situations where a more imperfect, lo-fi sound would suit a drum loop or synthesizer sound perfectly. Remember that the first hardware samplers from the '80s that usually only ran at 8 or 12-bit rates and at low sample rates.

With the BitMachine, changing the sound with such an "antique" device is no problem. You can use the BitMachine to bring back to life the times when minimalist and scratchy sound chips in home computers were commonplace.



The BitMachine opens up a gateway to "acoustic time travel" where you can encounter bit and sample rate reduction and downstream filters based on analog

models. The effect also has a modulation section with which you can control individual parameters using an oscillator (LFO) or the input signal.

We have designed a range of "typical" presets to demonstrate the time travel abilities of the BitMachine. These can be opened at the top right of the interface.

Independence FX (only as track and master plug-in)

Independence is an all-round sampler that's available as a MAGIX synth (view page 243). It includes its own effects section which can be used separately for track or master effects.

Select these effects in the mixer, track header or the "Independence FX" option under "MAGIX plug-ins".



Independence offers over 40 high-quality filters and effects.

- You can add unlimited Insert filters and effects.
- You can also change the order of the Inserts easily using drag & drop.

- Any Insert that contains a "sync" button is fully synchronizable to your sequencer software (host) or to the Sequoia internal tempo and you can also choose the mode of synchronization.
- In addition to the the main "insert fx bank" menu that enables you to load, save and copy/paste already prepared presets of multiple inserts, each individual insert filter or effect contains its own "preset" menu which includes all these options.
- You can minimize the editing display of each insert with a click on the open/close triangle in the upper left corner.
- Regardless of where you save a preset of your custom insert settings, it will always be available in the preset menus of both the Mixer and the Flexible Modifiers areas.

Detailed information about "Independence FX" can be found in the PDF document.

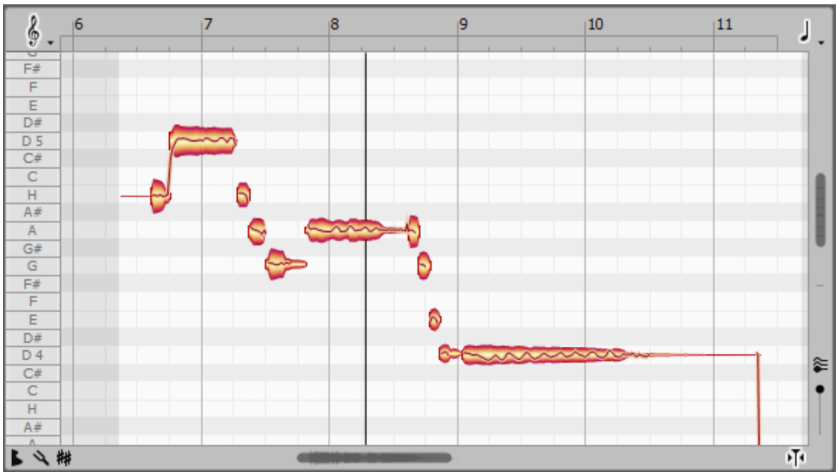
Melodyne integration

Melodyne is a software that lets you make extremely detailed edits of pitch, timing and spectrum of audio material, as well as change the melody of individual instruments within a complete mixes. It was possible to use Melodyne in older versions of Sequoia as a VST plug-in, and starting from Sequoia X5, a special program interface from Melodyne (ARA = Audio Random Access) had been implemented, letting Melodyne be seamlessly integrated (starting from version 4) into Sequoia interface.

Melodyne essential, the basic version of Melodyne, is included with Sequoia. You can find more information about Melodyne capabilities and its various editions on the manufacturer's website from Celemony.

Edit audio objects with Melodyne

To edit one or multiple audio object with Melodyne, select them, right click and select "Edit audio file in Melodyne" from the context menu. Melodyne will be loaded for these objects as an object effect, the objects will be analyzed and the Melodyne screen will be opened. There you will see the last select object as a typical Melodyne "blob".



The blob view is a mix of piano roll and waveform display. You can copy, move, lengthen, shorten and play back all notes here. The scope of possible edits depends on the Melodyne version and spans pitch and timing changes up to detailed intonation control in polyphone material and spectral editing of multiple tracks.

Tip: To be able to use Melodyne to its full capacity, we recommend you watch the video tutorials on the Celemony website http://helpcenter.celemony.com/hc-2/!

If you select another object, the display in the Melodyne window will adjust accordingly.

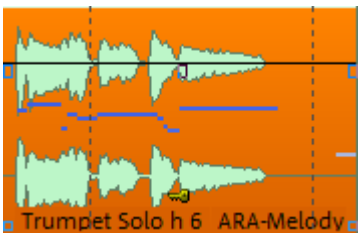


You can let the "blobs" representing the other objects appear for reference purposes by activating this symbol to the left in the Melodyne window.

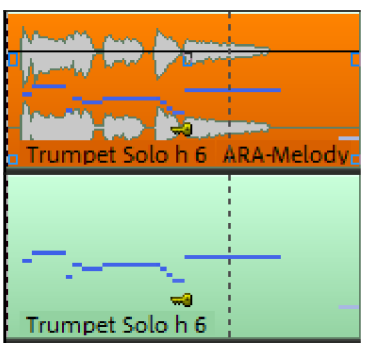
If the objects originate from different tracks, these will be analogously created in Melodyne (only Melodyne studio), with the track names taken from Samplitude.

Audio to MIDI

If an audio object has been analyzed or edited using Melodyne, you will also see the pitch on the audio object as a piano roll, similar to how MIDI objects in VIP are represented.



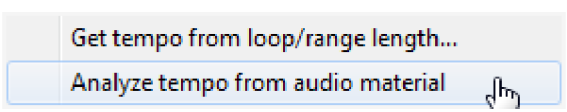
Selecting "Audio to MIDI..." from the object context menu or the "Object" menu, you can create a new MIDI object.



The MIDI object will be added to the track directly below the audio object. If there are already objects at this time position, on this track the MIDI object is inserted to a newly created track instead.

Determine object tempo

If Melodyne is installed, it can be used to determine object tempo. You can learn more in the Object Editor section, Pitchshifting/Timestretching (view page 175)



Notes

- The VST 3 version of Melodyne is required to work with Sequoia via ARA.
- Melodyne integration via ARA works only if you set Melodyne as an object effect, which is easiest set in the object context menu. You can also load Melodyne from the object editor or the Effects menu.

- If Melodyne is loaded in an audio object, right-clicking on the object will open the Melodyne Editor, not the Object Editor. Object Editor can be accessed via the context menu, the Object menu or the Ctrl + O keys.
- Melodyne can't be used simultaneously with Sequoia tempo and pitch functions (Elastic Audio, Timestretching/Pitchshifting, musical tempo adjustment).
- If an audio object is split, Melodyne edits will be applied to both objects, just like if this object is duplicated with the command or using Ctrl + mouse drag. Melodyne edits will not be transferred if you copy an object to the clipboard and then paste it!
- Object editor functions reverse and loop aren't allowed.

External hardware effect integration

(for "Mixer FX Monitoring/Hybrid Engine" only)

To integrate external effects devices into your setup, you will require a multi-channel audio interface. Reserve a channel pair for every external effect device you want to integrate.

The integration of external effects or synthesizers can be done in the dialog "File > Project Properties > External Effects". Here you can set up the inputs and outputs for 32 external devices, create new "Effect send" and "Effect return" tracks, as well as specify effects latencies so that they can be taken into account during latency compensation.

The order of inputs and outputs for the program is saved globally. The setup of external effects integration is project-specific.

Note: Return tracks have to be recorded first in order to be available for track bouncing and CD burning. For this reason, external effects are integrated through the program's own tracks and are not available in the tracks themselves as plug-in inserts.

You can open the "External Effects" dialog in the project options or by using the keyboard shortcut "I".

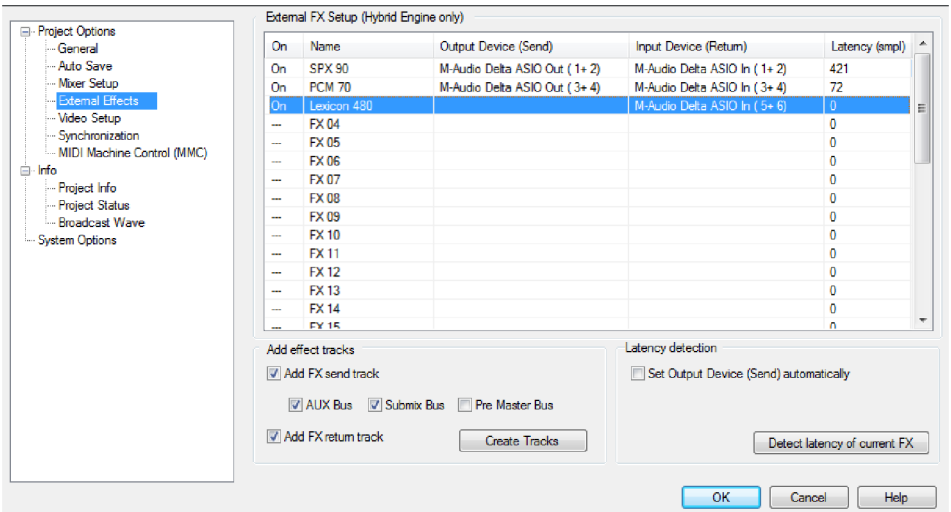
Routing Effects

1. First replace the descriptions FX01, FX02 etc. in the name column with the name of the effect that will be inserted.

2. Clicking on the next column reveals a selection of available output devices that can be used as effect sends for the effect device. Select an output.

3. In the „Input Device“ column select a free input as the effect return channel.

4. Now you can activate the external effect device by clicking in the first column and connect it to the set physical audio inputs and outputs.



In all areas of the program where input and output channels are set up, you can also integrate external effect devices, e.g. in the mixer, in the track editor or by right-clicking on "Mute" or "Record". The names of the assigned external effects are displayed behind the device name.

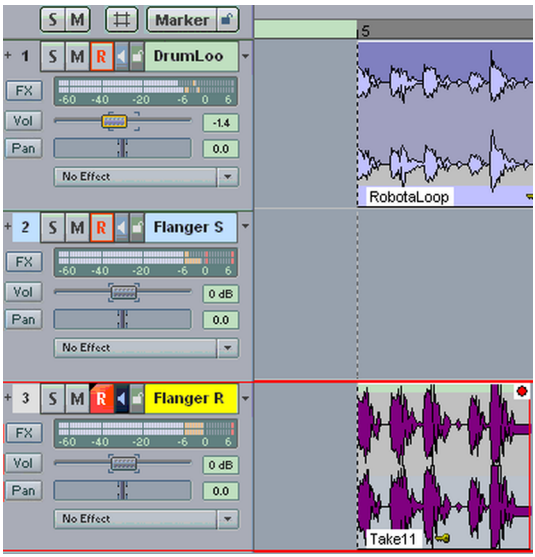
Creating effects send and effects return tracks

The lower part of the "External effects" dialog lets you create effect tracks for your external effect device.

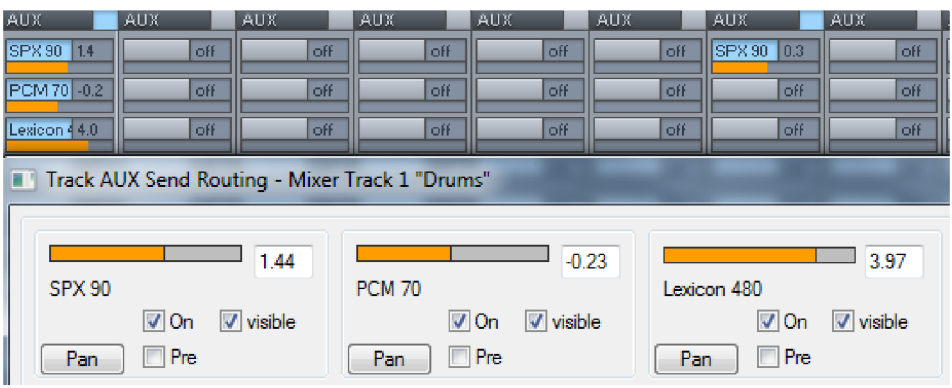
Effect as AUX

1. Under "Add effect tracks", select the options "Add FX Send track" and "AUX bus", then tick the box beside "Add FX return track".

2. Click the "Create tracks" button. You'll now see two newly created tracks or channels in the arranger and in the mixer that control the send and return signals of the external effect device.

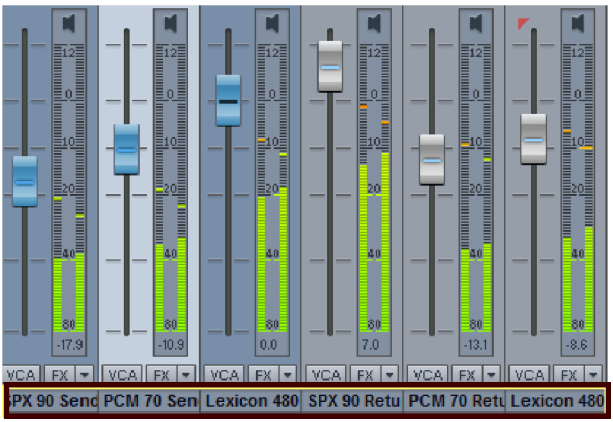


3. You can now set up the send level of every integrated effect device in the AUX slot of every mixer channel



The volume fader of the send channel works as master send level.

4. Position the effect signal in the mix using the fader of the effects return channel.



Effect as Insert

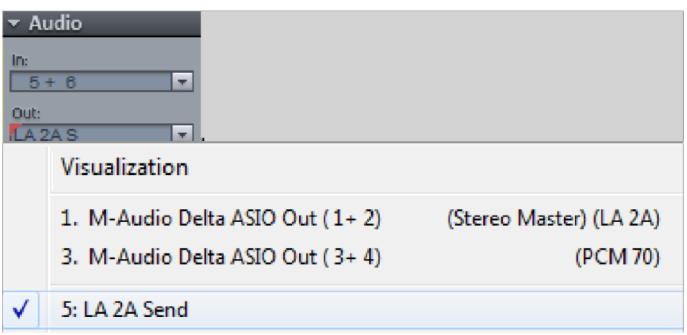
To add an insert effect:

Step1: Select "Add effects track", then the options "Add FX send track" and "Submix bus", then set a check next to "Add FX return track".

Step2: Click "Add tracks".

Two newly created tracks will now appear in the arranger and in the mixer to control the send and/or return signal of the external insert effect device.

3. For the track that will receive the insert effect, select the Send (Submix) track to the external effect device as the output device.



4. With the fader of the effect return channel you can set the return level of the effect signal.

Note: Make sure input monitoring is switched on for the effect return track. This can be done by clicking on the track monitoring button (loudspeaker symbol). The loudspeaker symbol is highlighted and you will hear the effect return signal.

The routing for each channel can be set up in the track settings dialog:

1. Right-click on the track name.

For the effect send channel, set the output for your sound card that is connected to the external effect input in the "Playback" field.

2. For the effect return channel, set the output for your sound card that is connected to the external effect output in the "Record" field. For the channel that is intended to work with the external device, set the effect send track in the "Playback" field [e. g. Submix: LA 2A Send (track 16)].

Use the double arrow buttons beside the track name field in the track options dialog to switch the track options to those of the neighboring track.

It is also possible to use an effect as an AUX as well as a submix bus:

To do this, activate both options before creating new tracks. FX sends can be changed later by accessing the track properties (right-click on the track header for the FX send bus) and setting checkmarks next to the corresponding settings "AUX bus" or "Submix bus".

Effect as pre-master bus

If you select "Pre-master bus" when creating tracks, the new tracks will be added at the end of the project, and all project tracks will be routed to the input of the send track. The return track is automatically routed to the master. This enables the external effect to be used as a master effect.

Software Synthesizers

Software synthesizers work within your system as software, not as external devices that are connected to your computer. The software synthesizers can be broken down into two types:

VSTi

In the track editor and mixer you can access the available VST instruments (VSTi), which you operate via MIDI like external hardware instruments. The instruments are located next to the effects in the top plug-in slot in each track or channel.

For detailed information on VSTis, see chapter "VST and ReWire" (view page 275).

MAGIX Synths

MAGIX Synths refer to Independence, Revolta 2 and Vita Instruments, all included in Sequoia. These instruments are also VSTis.

The following is a list of the differences between object synths:

- MAGIX synths are track-dependent, meaning they are operated using a specific track. In contrast, synth objects can be moved to any track.
- All MIDI objects on a single track control the same instrument.
- All of the editing options in the MIDI editor are also available for the MAGIX synths.
- MAGIX synths can be played using a normal MIDI keyboard and recorded via MIDI (view page 99).

Object Synths

You can access more synths under "Object" > "New Synth Object". This object-based instrument group includes Beatbox2, LoopDesigner and Robota software synthesizers.

MAGIX Synths

The following software synthesizers and samplers based on VST plug-in technology are also integrated in Sequoia:

- **Independence**: A sampler with 12 GB of content for Samplitude Pro X5 and 70 GB Content for Samplitude Pro X5 suite or Sequoia.
- **Revolta 2:** A polyphonic analog synthesizer based on subtractive sound synthesis with frequency modulation. Sounds in Revolta 2 don't use samples as a foundation - they are calculated in real time from your computer processor.

Revolta 2 is particularly suited for electronic music, with lead, sequencer and pad sounds as well as bass and effects.

- **Vita**: A sampler with incredibly realistic-sounding, "classical" instrumental sounds like different guitars (Power Chords, clean electric guitar, acoustic guitar, bass guitar), different pianos, percussion, strings, brass, woodwinds (each as an individual set & as an ensemble set), and much more .
- **Vita Solo Instruments** is a collection of different sample players based on Vita sampler technology and with customized interfaces for each of the instruments.

Independence

The Independence sampler gives you access to hundreds of music-oriented software functions. The intuitive user interface and file management system, the ultra fast streaming integration, multi-core processor support and the Auto-RAM-Cleaner makes it possible to store and use a huge number of instruments in seconds.



Detailed information about "Independence" can be found in the PDF document.

Revolta 2

Revolta 2 is polyphonic and may be played with up to 12 voices. It includes a series of additional features, like an additional noise generator, a step sequencer, and an extra flexible modulation matrix. Its effects section of 9 different effects and presets, designed by a famous designer, make it a full-fledged synthesizer for all kinds of lead, sequence, and pad sounds.

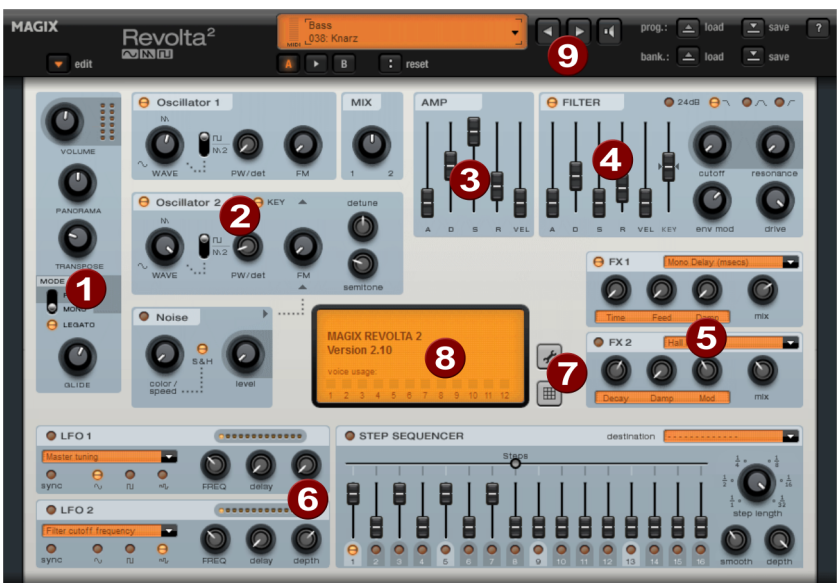
Revolta 2 has a whole array of presets. The sounds have been created by professional sound engineers and demonstrate the huge potential of this instrument right from the start. But first of all, we encourage you to try out the various control functions and to experiment as much as you like. The sky's the limit to your creativity.

Revolta 2 – the interface

The Revolta 2 interface may be displayed in two sizes. In "Rack" mode, only the elements necessary for preset loading are visible:



Clicking the "Edit" button opens the entire user interface.



1. Main parameter: The volume, panorama position, pitch characteristics ("transpose") and the play modes (POLY, MONO, LEGATO) may be set here. GLIDE controls the portamento time.

2. Oscillator section: Two oscillators are available with a smoothly adjustable curve form and a noise generator. Both oscillators may be tuned to each other and used to modulate frequencies.

3. AMP: This is the volume envelope. This influences the temporal progression of a track's volume. A(ttack) stands for the volume increase at the start, D(ecay) for the length of time the decrease in volume takes on a section set with S(ustain) at the maximum volume. R(elease) is the length of time it takes for the sound to ring out. VEL specifies how much the envelope curve depends on the velocity.

4. FILTER: Switch on a filter to influence the sound here. FILTER TYPE selects the filter type. "Cut-off" regulates the filter frequency; "Resonance" controls the level of amplification of the filter frequency. VEL" indicates how much the velocity influences the filter frequency, "KEY" changes the filter frequency depending on the note pitch ("Keytracking"). The filter envelope (ADSR slider) influences the filter frequency depending on the time. "Env mod" controls the strength of the filter envelope curve, and "drive" may be used to overmodulate the filter.

5. FX1/FX2: Mix in 2 different effects out of a total of 9 available effects.

6. LFO1/LFO2/STEPSEQUENCER: Two LFOs and the step sequencer may be used to modulate single parameters of Revolta 2.

7. Options and modulations matrix: The two buttons open Revolta's options page for general and preset-specific settings and modulation matrix. In the modulation matrix, modulation sources are connected with modulation targets. Simple modulations like the oscillator, where the pitch will be modulated via an LFO (vibrato), may be set quicker directly on the interface. Much more complex modulations are possible in the matrix, because the matrix offers more modulation sources (e.g. MIDI controller, oscillators) and the modulation source may influence more targets.

8. VALUE DISPLAY: The value display shows the exact value of the parameter that has been modified. You may also use this to find out the load of the twelve voices.

9. Preset section: Select different Revolta presets here. Every sound may be previewed, and an A-B comparison between two sounds is also possible (e.g. an edited and the original sound).

Vita Solo Instruments

The Vita Solo Instruments are contained in a sample player with customized interfaces for each of the instruments.

The basic controls are identical for all instruments.



One click on the arrow symbol opens a fold-out menu where you can determine the general sound of the instrument. If "ECO" appears in the description, this refers to especially performance-improving settings which may not sound so "smooth". You can also save the settings you changed in order to add them to your favorites lists for later use.



You can control the overall loudness of the instrument.



You can turn the instrument keyboard on or off with this controller.

Vita

The MAGIX Vita synthesizer specializes in the realistic reproduction of instruments. It works using sampling technology which combines short recordings (samples) of real instruments of various pitch, playing techniques and volume, which are then combined and reproduced.

The Vita synthesizer is controlled using MIDI objects.

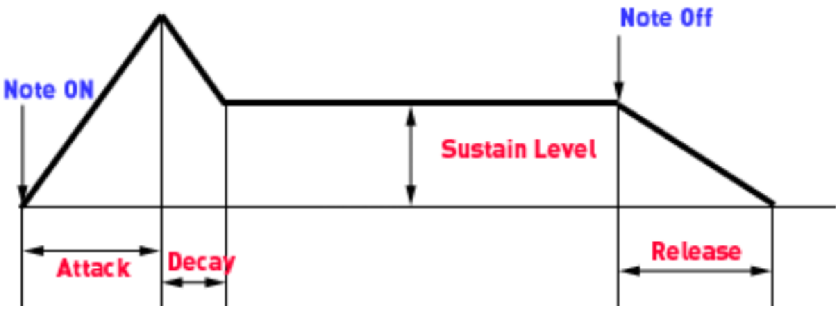
Vita – the interface



1. Layer selection/peak meter: The Vita sounds, also known as layers, may be selected here using the arrows. Right clicking on the display opens the layer menu.

2. Main parameter: Set the volume, panorama position, pitch characteristics ("transpose"), and the fundamental frequency ("master tune") here.

3. AMP: This is the volume envelope. This influences the temporal progression of a track's volume. A(ttack) stands for the volume increase at the start, D(ecay) for the length of time the decrease in volume takes on a section set with S(ustain) at the maximum volume. R(elease) is the length of time it takes for the sound to ring out.



4. FILTER: Switch on a filter to influence the sound here. FILTER TYPE selects the filter type. "Cut-off" regulates the filter frequency; "Resonance" controls the strength of the amplification of the filter frequency. "Velocity" specifies how strongly the velocity should influence the filter frequency, while volume can be balanced using the "Gain" controller. The filter envelope (ADSR slider) influences the filter frequency depending on the time.

5. DELAY: Switches on an echo effect; "Time" controls the delay time, "Level" controls the strength of the echo sound.

6. REVERB: Switches on a reverb effect, "Time" controls the delay time, and "Level" controls the strength of the echo sound.

7. VALUE DISPLAY: The value display shows the exact value of the parameter that has been modified.

8. DYNAMIC RANGE: Usually, the relationship between the created volume and MIDI velocity is proportional. Since some MIDI keyboards produce a velocity that is too hard for loud sounds or vice versa, this behavior may be compensated with "MIDI input curve". "Dynamic" and "dynamic curve" influence the dynamics of the sound, i.e. the relationship between the quietest and loudest sounds.

9. Voices: Control the number of voices played simultaneously here. If notes are no longer played during fast passages, increase the amount of voices here (performance will not be as good).

10. Keyboard: A preview of the Vita sounds. This only works during playback or recording.

Vita – MIDI specifications

Vita parameters	Control Change #	GeneralMIDI
ModWheel	1	ModWheel
Volume	7	Volume
PAN	10	PAN
Mastertune	14	Mastertune
Sustain	64	Sustain
Cutoff	71	Filter Cutoff
Amp Release	72	Release Time
Amp Attack	73	Attack Time
Resonance	74	Resonance
Amp Decay	75	Decay Time
Amp Sustain	80	Button 1
Filter Velocity	81	Button 2
Filter Gain	82	Button 3
Reverb Level	91	Reverb
Reverb Size	92	Tremolo
Delay Level	93	Chorus
Delay time	94	Delay/Vari
Filter Attack	102	-
Filter Decay	103	-
Filter Sustain	104	-
dynamics	108	-
dynamics curve	109	-
MIDI input curve	110	-

Vita MIDI Event Types

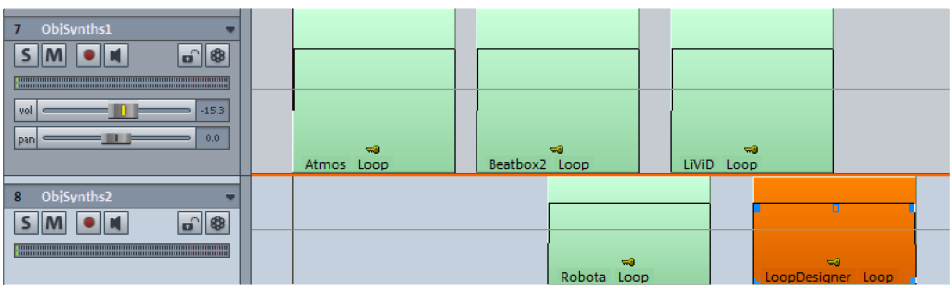
Pitchbend	x
NoteOn	x
NoteOff	x
ControlChange	x
Aftertouch	0

Object Synths

The Object Synths feature a particular form of synthesizers which form individual synth objects in Sequoia. You can move synth objects between tracks any way you want without affecting the synthesizer settings in the process. In this way, they differ from other software synthesizers, which are track-dependent and operated via MIDI like MIDI synthesizers.

The Object Synths are available as installation options. After installation they can be found in a special "Synth" folder within the program folder.

Synth Objects can be accessed through "Object -> New Synth Object". This creates a 4 bar loop object at the current playback marker position. The instrument interface for the synth object that is created can be accessed by double-clicking on the object..



The keyboard shortcut "Ctrl + Spacebar" plays back the selected synth object in "Solo" mode.

The following object synths are integrated into Sequoia:

- **BeatBox 2**, a virtual drum computer for drum kits with high quality effects
- **Loop Designer** for developing your own drum loops and bass lines
- **Robota** a complex, four-part drum computer that aids in the generation of sounds as well as working with samples and oscillators (analog sound synthesis).

BeatBox 2

BeatBox 2 is a 16-voice pattern-based drum computer featuring hybrid sound synthesis and a step sequencer. The interface includes a multi-effects section (one effect per drum instrument), auto copy, and a convenient velocity editing function.

A drum sound is created in BeatBox 2 using a sample, which is combined with a synthetic sound that can be created using up to three different synthesis models (hybrid sound synthesis). BeatBox 2 also enables detailed editing and automation of all sound parameters.

BeatBox 2's interface uses two states. While closed, you can listen to the included sounds and patterns or those you've made in BeatBox 2 without using too much window space.



Only the most important control elements are displayed when the program window is minimized:

1. **Volume controller**: Controls the volume.
2. **Peak meter and preset name**: The peak meter enables the output from BeatBox 2 to be checked visually. Clicking the triangle beside the preset name opens the preset list.
3. **Previous/next preset**
4. **Save preset**: The preset includes the drum kit in use, the pattern, and any possible automations (see page 249).
5. **Play/stop**: The playback controller in BeatBox 2 starts the BeatBox solo, i.e. without playing the arrangement.
6. **Edit button**: The edit button opens BeatBox 2 for you to program your own beats and sounds.

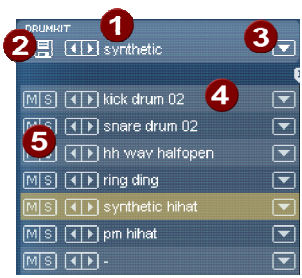


1. Maximized BeatBox 2 plus window:Pro**Drum kit**: This section is where drum kits (collections of different drum instruments) and the individual drum instruments are loaded.
2. **Selected drum instrument**: The settings in the synthesis section (5) and velocity/automation (4) always affect the selected drum instrument.

3. **Pattern editor**: Programs the beat sequence. In the top part, different patterns (sequences) may be loaded and saved as well as different settings for the view and function of the pattern editor. The matrix is where the beat is programmed: one line corresponds to a drum instrument, and a column matches a specific time position within 1-4 beats. If a cell is clicked, then the respective drum instrument will be triggered at this position.
4. **Velocity/control**: This section has two modes: velocity and automation. The "Velocity" setting displays the velocity levels for the selected instrument's beats as a column. The "Automation" setting allows automation to be set for sound parameters in the synthesis section (5).
5. **Synthesis**: The selected sound parameters and effects settings for the selected drum instrument may be edited here.

The following section explains the sections of BeatBox 2 individually:

BeatBox – drum kit



This section loads drum kits (collections of different drum instruments) and the individual drum instruments. You can also try out an already programmed pattern with different kits or exchange individual drum instruments.

1. **Select drum kit**: Use the <> buttons to cycle through the different drum kits. A drum kit is a collection of percussion instruments with matching sounds, e.g. rock kit or electronic drums à la TR 808. By changing the drum kit, you can add an entirely different sound to the rhythm you have already created.
2. **Save drum kit**: Use this button to save the current collection of drum instruments as a kit.
3. **Drum kit list**: Click the arrow to the right of the name to open a complete list of available drum kits.
4. **Select drum kit**: The arrow buttons function the same way as with the drum kit. The sequence of drum instruments in the kit may be resorted via drag & drop.
5. **Mute/solo**: The "Solo" button switches a drum instrument solo, i.e. all other instruments which are not "solo" will be muted. The "mute" button mutes a drum instrument.

New drum or effect sounds may by added to the current drum kit via drag & drop from Windows Explorer. Drag a wave file to a drum instrument to create a new drum sound based on this sample. BeatBox 2 copies the sample into the sample folder to

make sure that the instrument or kit created can be used again later. Drag a complete folder featuring wave files to the drum kit to create a kit based on those samples.

BeatBox – context menu

Right clicking a drum instrument always a context menu:

- **Copy/paste**: Copy an instrument from a track and paste it to another one.
- **Empty instrument**: An empty instrument is added. No sound is played, it has no name, and is used to clean up an unused track.
- **Default instrument**: Adds the standard instrument. This features the standard parameter for all synthesis shapes and serves as the starting point for your own sounds.
- **Reset automation**: Several of BeatBox 2's own presets contain automations. These are dynamic sound parameters like filters or pitch changes. This command allows these to be completely removed from the selected instruments.

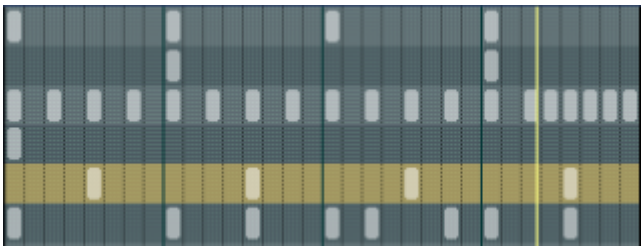
BeatBox – pattern editor buttons



1. **Pattern**: Use the <> buttons to switch through the different patterns. The arrow to the right opens a list of all available patterns, and the save button stores the current pattern.
2. **Clear track/all**: All events for the selected instrument (track) or all events for the pattern (all) are removed by clicking this button.
3. **Bar selection**: The bar you wish to edit may be selected via the corresponding number button. Use the "Follow" button to select "Follow" mode, i.e. the step display follows the steps of the currently played beat. "All" displays all bars of the pattern.
4. **"1>2-4" auto copy**: If more than one bar is set as the pattern length, "Auto copy" mode ensures that the drum notes set in the first bar are automatically placed into the next bars. This also makes it easy to create a continuous beat, even with a loop length of four bars. Notes set in the bars further back are not affected by the "Auto draw" function, e.g. making faint variations detectable only in the fourth bar.
5. **Bars**: The maximum length of a drum pattern is four beats. This length may be selected via the small scroll bar.
6. **Shuffle**: This controller changes the timing of BeatBox 2; if the controller is turned to the right, 1/8th notes in the rhythm will be played more and more as triplets. If that sounds too abstract, simply try it out, ideally with a pure 1/16 hi-hat figure; you'll soon see what the shuffle fader is capable of.

7. **Grid**: Set the time resolution for BeatBox here. Choose from 1/8th notes (only for very simple rhythms), 1/16 (default), and 1/32 (for more refined constructions).

BeatBox - Pattern editor Matrix



This is the heart of BeatBox. With a mouse click on any position in the drum matrix you can create and delete any drum note. Clicking and dragging draws in a series of notes. Together with the velocity editing options (see Velocity), you can easily create drum rolls.

If "Shift" is held when you click on notes in the range, a rectangle can be drawn out which selects the notes contained within this rectangle (lasso selection). Selected notes can be copied by dragging them to a new position. If "Ctrl" is also held down, then existing notes will remain at the target position. Delete all selected notes by right clicking.

Two special commands are available for selection:

Shift + double click: Select everything in the beat that has been clicked.

Ctrl + shift + double click: Select everything

A simple mouse click cancels the selection. The selection is automatically canceled after copying. If you want to keep your selection, hold down the Shift key while copying.

Many functions in BeatBox 2 can be controlled with the keyboard. For example, a beat can be triggered with the "Enter" key live in a running pattern. Here's a complete list of the keyboard commands:

General

E	Open/Close editor

Pattern editor options

A	"1>2-4" Auto copy

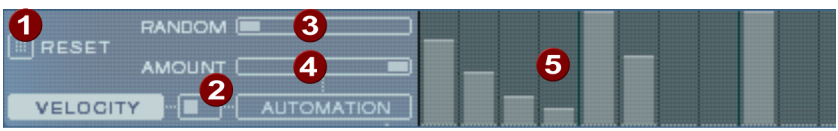
F	Follow
1-4	Display bar 1... 4
0	Show all bars
+/-	Grid finer/rougher

Selected drum instrument

Arrow up/ down	previous/next
P	Preview
Enter key	Live input
M	Mute on/off
S	Solo on/off

BeatBox – velocity

Use "Velocity" mode to edit the velocity/automation of the individual drum notes for the selected drum instrument.



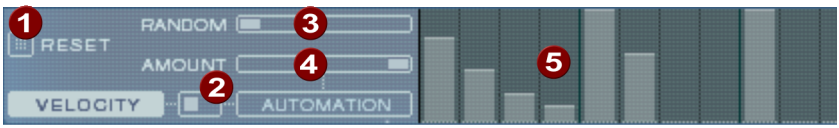
1. **Reset**: Sets all velocity levels to 100%
2. **Mode switcher**: Switches the range between velocity control and automation (see page 249).
3. **Random**: The random parameter adds random variations to the set velocities. This helps make your beats sound more natural.
4. **Amount**: No function in velocity mode.
5. **Velocity level**: Enables every set note for the selected instrument to be edited via the height of the column. Multiple columns may be edited at once; see "Editing velocity values and automations".

BeatBox – automation

Every parameter for a drum sound, including effects, may be automated, i.e. so that it changes during a pattern. For example, make your snare drums more lively by adding higher voices to loud hits or by setting accents on individual hits with reverb effects.



The synthesizer section below enables the selection of a parameter for automation via the small blue LEDs above a parameter controller. More about the parameter controllers is available in the "Synthesizers" section.

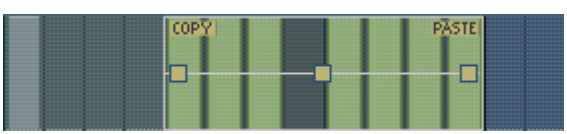


1. **Reset**: Sets all automation values for the selected parameter to "0".
2. **Mode switcher**: Switches the section between velocity control and automation. This happens automatically when selecting parameters.
3. **Random**: The random parameter adds random variations to the set automation. This helps make the beats sound more natural because each programmed hit will sound slightly different. The level of the random parameter is also influenced by the amount controller (see below), i.e. if the amount is at "0", then the randomness will have no effect.
4. **Amount**: The amount parameter regulates the complete influence of the automation values and of the random factor on the selected parameter. An amount of "0" does not have any effect at all on the automation; maximum level has the greatest effect. The effect of the amount controller on the automation values is made visible by a slightly brighter line in the value columns.
5. **Automation values**: The selected parameters may be drawn in as a column with the mouse. Automation values may also be drawn between the set notes; the sound of the drum instrument changes during playback. The automation values are added to the original parameter value.

Editing velocities and automation values

Hold down "Shift" and select any number of bars for velocity or automation with the mouse. Two special commands are available for selection:

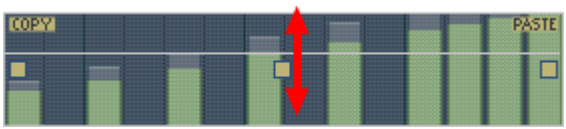
Shift + double click	Select everything clicked on in the bar
Shift + Ctrl + double click	Select all



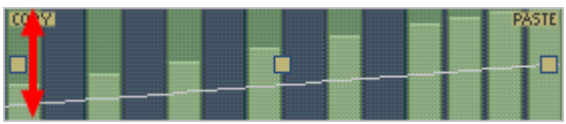
The "Copy" button copies the selection to the clipboard. If you select this or that track in another editor, you can paste the notes there now from the clipboard. If the target selection is larger than the contents of the clipboard, then it will be inserted again.

This lets you quickly add a short section throughout the complete length of the pattern.

The three handles allow the velocity or automation values to be edited together.

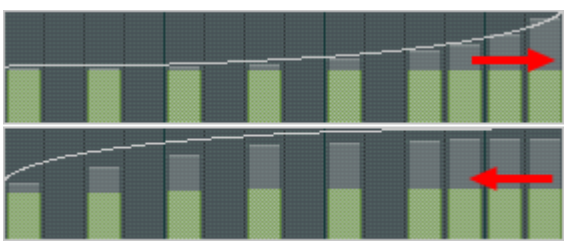


The middle handle increases or lowers the values together.



If the handles are moved horizontally, you can change the curve shape of the transition.

A single click in the automation section cancels the selection again.



If you move the handles horizontally you can change the curve shape of the transition. A single click in the automation section cancels the selection again.

Note: A selection of velocity values matches the selection of corresponding notes in the matrix editor track.

BeatBox – synthesizers

Use the lower section of BeatBox to set the sound for the selected drum instrument.

Synthesis in BeatBox 2 consists of a combination of a simple drum sampler and a synthesizer. There are three different synthesis models possible for the synthesizer: "Phase distortion synth" (FM synthesis), "Filtered noise", and "Physical modeling". The mixed signal of both components is then processed by a multi-mode filter. An envelope curve ("envelope generator") time-dependently controls modulations of in all components.

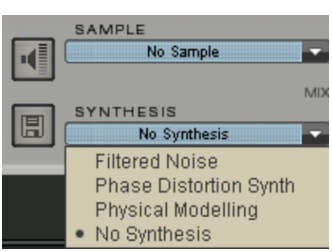


1. **Preview drum instrument**
2. **Save drum instrument**
3. **Select sample**: Clicking the arrow selects samples from the categories like kick, snare, etc.
4. **Select synthesis mode**: Switches between the three synthesis models.
5. **Mix**: Mix ratio between drum sampler and synthesizer.
6. **Parameter controller**: All six sound parameters for a drum sound may be set directly and automated via the parameter controllers. These parameters depend on the currently loaded drum sound. By clicking the name of a sound parameter, a menu opens for you to add parameter controllers to each drum sound's synthesis parameters.
7. **Automation**: This selects the controller's parameter for the automation.

BeatBox – synthesis models

The sound synthesis in BeatBox 2 consists of a simple sampler and a synthesizer which includes three different synthesis modules.

Sampler: The sampler plays short sound recordings ("samples") in different pitches. The sampler is intended for creating all kinds of drum sounds; the sounds are static and unnatural if the pitch is not altered. This is why the sampler is combined with on of the three synthesis models.



Filtered noise: White noise is filtered by two band-pass filters with separately adjustable frequencies and resonance. This algorithm is suitable for creating synthetic snares and hi-hat sounds.

Phase Distortion Synth: Two oscillators with regulated phase distortion and extremely variable frequency modulate each other (FM/cross-modulation/ring-modulation). Depending on the setting, this algorithm can be used to create kick, tom, or metallic percussion sounds; higher values for frequency and modulation level produce noisier sounds for hi-hats or shaker sounds. Since the oscillator frequency can be set exactly according to the musical pitch, this model can be used to produce bass lines or melodies.

Physical Modelling: This is a simple physical model of an "abstract" percussion instrument. A fed-back network of delays is oscillated by an impulse of filtered white noise (exciter). Depending on the setting of the exciter, the size of the model (surface), and the damping, a wide spectrum of natural sounding percussion instruments like cymbals, claves, gongs, or triangles can be created.

BeatBox – effects section

Each of BeatBox 2's drum instruments includes an effects unit which is fed in after the actual sound synthesis and editing. Each of these effects units includes a series of high-quality algorithms to add "audio polish" or to place the sound in a production-typical context.



1. FX on/off
2. **Parameter controller**: An effects module includes four adjustable parameters. The fourth parameter, "Mix", is always available and the remainder possess a function dependent upon the selected algorithm.
3. **Automation**: This selects the controller's parameter for the automation.
4. **Effects algorithm**: Click the arrow to select an effects algorithm.

The available **effects algorithms** are described in the following:

Mono delay (tempo sync/millisec.)

Simple, monophonic delay effect.

Parameters

Time: Delay time controlled by musical measure (sync) or free

Feedback: Repetitions

Damping: High damping of the repetition

Stereo delay (tempo sync / millisec.)

Stereophonic repetition, separately adjustable per side

Parameters

Left/right: Delay times, synced or free

Feedback: Compared to mono delay, no repetition takes place in the middle position of the feedback controller. When turned to the left, the type of delay is the so-called "ping-pong" variety, i.e. the signal is sent alternately so that it jumps from one channel to the other. When turned to the right, the delay effect is "dual mono", in which case left and right sides function as two independent time-delay units.

Chorus

Produces a typical "floating/shimmering sound" by modulated detuning of a signal to "thicken up" its sound or spread it across the stereo field. Detuning is achieved via a short delay, the length of which can be varied by the modulation. This produces the so-called "Doppler" effect.

Parameters

Time: Delay time in milliseconds. This may be understood as the "base" modulation that is stretched or compressed by the modulator.

Rate: Modulation speed

Depth: Modulation amplitude. Low values modulate only a little; higher values create a clear vibrato.

Flanger

Algorithmically similar to chorus, but different in that the delay time is significantly lower and delay works with repetitions (feedback). A flanger sounds more "cutting" and up-front than a chorus.

Parameters

Rate: Modulation speed

Feedback: Delay feedback

Depth: Modulation of amplitude

Phaser

A modulation effect just like chorus & flanger, but in this case no detuning takes place. Filter components periodically alter the signal's "phase response" (principle of the "phase shifter"). Characteristic notches are produced in the frequency spectrum response to create so-called "comb filter effects". The phaser effect is suitable for pads and "psychedelic" sounds.

Parameters

Rate: Modulation speed

Feedback: Feedback of filter steps

Depth: Modulation amplitude

Room reverb/hall reverb

In the case of reverb, there are two realistic simulations of natural reverberation. Sounds receive "atmosphere" to appear lively and "authentic". Room reverb simulates a small space with high echo density. Hall reverb mimics the typical reverb of large concert halls.

Since natural spaces never sound "static" (air molecules are constantly moving and the reflection process is quite complex), both algorithms include a modulation parameter which varies the delay time of individual echoes and thereby affects the liveliness of the reverb impression depending on strength.

Parameters

Decay: Reverberation length

Damp: Damping of highs to simulate absorption via air, wall materials, and objects

Mod: Modulation strength

Lo-fi

This algorithm gives the sound a little bit of "grit", or a certain measure of signal destruction depending on its setting. An ideal partner for creative sound design. The simulation of early digital synthesizes or samplers is equally possible, since their AD/DA converters were anything but "true" in the character of their sound. The sample rate from the output of the lo-fi effect can be continuously reduced and a generous measure of "aliasing" distortion may be produced alongside the unavoidable loss of highs resulting from "down sampling". Bit resolution is clearly changeable, too.

Parameters

Rate: Sample rate

Crush: Number of bits

Low-pass: Low-pass filter at the output to smooth out induced noise

Distortion

This overdrive effect works similarly to common guitarist effect pedals. Everything is possible, from light, bluesy signal saturation to hard "metal shred boards". In this case, a dual-band EQ works on the in and output signals and to provide a rich palette of sounds.

Parameters

Drive: Controls the internal level and the overdrive

Low: Bass portion

High: High portion

Analog filter 12/24 dB

An additional filter may be applied across the entire drum sound.

Parameters

Cutoff/resonance: Filter frequency and resonance

Type: Filter type

Drive: Filter saturation

Vintage compressor

A compressor with a special "analog" controlling behavior for more pumping bass drums.

Parameters

Input: Input level

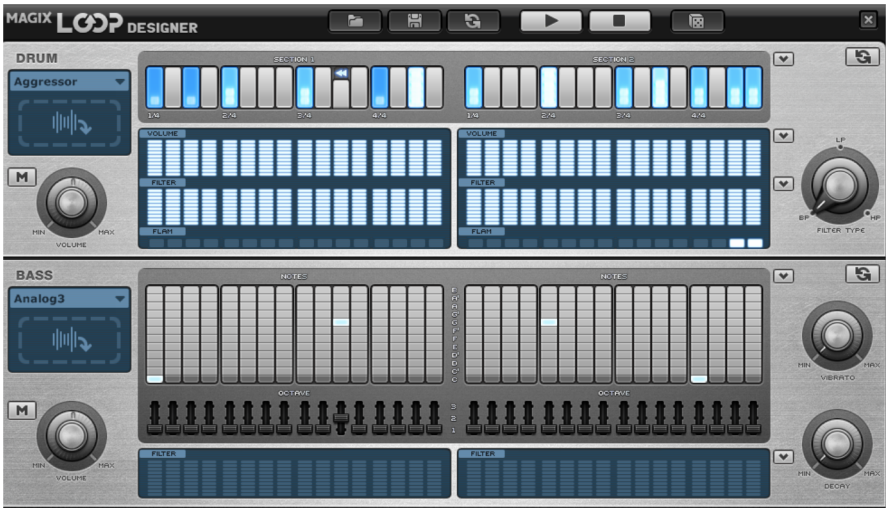
Attack/release: Time constants for compression.

Ratio: Compression ratio.

Loop Designer

The loop designer unites both stellar design elements of drum'n'bass style in a "device": turned-up beats and rumbling bass lines. With the Loop Designer you can create authentic drum'n'bass sounds without any specialized knowledge.

Loop Designer overview



The top half of the synthesizer controls the rhythm section, the lower half - the bass section. You can mute both sections using the M symbol on the left border. This way you can, for example, turn off the bass section in order to take only the drum section into the project. Only the drum section will then be added to the mixdown file when mixing down the project. Next to this are the volume controls, which control the volume of both sections.

You can preview and stop your drum'n'bass creations using the "play" and "stop" buttons.

Additional buttons:



Loads a saved pattern with all settings.



Saves a pattern.



Undo for all settings made in all sections.



Makes random settings changes in both sections. You can change settings later as you wish.

Drum section (top half)

Here complex, authentic-sounding Jungle-Breakbeats can be put together easily. In professional recording studios Jungle-breakbeats are made by chopping up arbitrary

drum loops and putting them back together in a different order. These laborious work steps are significantly simplified by Loop Designer.

The new sequence is determined in the top row, the so-called "step" row. The grey cells represent the individual sections ("counts") into which the loop is separated. Pick a different note or playback by left-clicking a grey cell. Various notes will be represented with a light bar which "grows" toward the top with each mouse click. This way, each time you click on the grey cell, the bar expands by one step.



1 of 4: Play Drum Loop starting from the first note.



2 of 4: Play Drum Loop starting from the second note.



3 of 4: Play Drum Loop starting from the third note.



Full bar: Play Drum Loop starting from the fourth note.



Reverse icon: Play backwards from this position



Stop icon: Stop playback

The right mouse key deletes settings of a step bar and the Drum Loop is played in its original order.



Pressing the double-arrow button generates a random step sequence. You can change this rhythm as you like.



Clicking into the blue field in the left part of the drum section opens a pop-up menu where you can pick the sound of the Drum Loop. When you select another drum loop it will load and be played in the way you programmed it.

Tip: You can also send a loop from Soundpool or a wave file to Loop Designer. To do so, select the desired loop or file and drag it into the field while holding down the mouse key ("drag & drop").

In the field under the "steps" row the sound of the loops is defined. The intensity of the settings is determined with a control similar to a peakmeter. The higher the bar, the stronger the influence on the loop is. With the help of the left mouse key, values can be smoothly adjusted. "Volume" controls the loudness (full = loud, empty = quiet), "filter" the filter strength (full = clear, empty = muffled). In the "Flam" row you have the option to make the note repeat itself twice in quick succession. This is how to program rolls and fill ins.

Using the top arrow buttons at the right edge you can load pre-defined pattern settings. The originally set loop will not be changed. The two lower arrow buttons offer presets in the form of standard curves for "volume" and "filter".



Filter type: selects the type of the filter sound: "BP" stands for "bandpass", "LP" for "low pass" and "HP" for "high pass".

Bass section (lower pane)

The bass section produces suitable bass lines.

- The first row of the "Notes" row, determines the order of the sounds i.e. the series of notes. Select a cell with the left mouse click, where the lowermost represents the lowest note and the topmost - the highest. A right mouse click deletes a cell.
- In the "Octave" row you can specify the octave of the bass. The slide control positions represent the various octave values. If the controller is down, a deep sound is produced, if it is in the top position, you will get a high tone. If the controller is in the middle, the pitch will be balanced.

Just like in the drum section, here you will also find arrow buttons on the right side for opening pre-defined patterns and and a double-arrow button for setting arrows. The filter can be set analogously to the drum section. The arrow button next to the filter area select presets in the form of standard curves.

You can determine the bass sound in the blue selection field at the left side. You can drag loops and WAV sounds into the field using drag & drop just like in the drum section.

Additionally, there are two slide controllers for sound changes to the right:



Vibrato: makes the bass tone's pitch change. When the control is to the far right, the sound vibrates more, if it's to the far left pitch isn't changed at all.



Decay: determines how long should it take to for the sound to subside. Set to the far right the sound is very fast (about 1 / 4 of a second), on the far left the sound lasts longer.

Robota

Robota is a four-voice drum machine with a virtual analog generator and a sample-based synthesizer. Virtual-analog sound generator means that the sounds are synthesized in realtime, i.e generated by a synthesizer. This gives you all of the typical analog sounds of classic drum machines like the Roland TR-808, TR-909 or more modern models like the Korg Electribe or the Jomox X-Base.



In the sample sound generator, samples of drum sounds or other recordings are loaded and used as a basis for creating the sound. After selecting the basic sound

generator, the sound characteristics of each of the four voices can be fine-tuned using various controllers.

Robota plays back through a step sequencer with light chaser programming. The four bars run as a loop in semiquavers (16ths) or two bars in demisemiquaver (32nds) steps. You can set the playback position from any beat position by clicking with the mouse. In Event mode, the instruments are arranged on the beat grid. In Snapshot mode you can adjust the various sound settings of the instruments.

Robota – sound synthesis

The four voices of Robota are built identically. The sound synthesis in Robota is quite complicated because each voice has to generate a large variety of drum sounds, ranging from a hissing hi-hat to heavy bass drum sounds.

The Robota drum synthesizer consists of an oscillator with a selectable waveform (sine, triangle or sawtooth) or a sample. You can also use a noise generator to add some noise to the mix. The oscillator features a pitch envelope and attack/decay controls. It also features frequency and ring modulation. The modulation depth can be controlled by means of an envelope parameter (FM/rng dcy). This is followed by a "Lo-Fi" section consisting of distortion (rectify, diode rectification), bitrate reduction (crush) and sample reduction (down sample).

The intensity of the Lo-Fi effects can also be adjusted by means of an envelope (lo-fi dcy). This is followed by a multimode filter (low-pass/bandpass/high-pass) with a roll-off of either 12dB or 24dB. A comb filter (a delay line named after its spectral view of peaks and dips that resembles a comb) can also be added. You can also modulate the filter frequency by using an envelope. There is also a compressor that can be adjusted for response time and intensity ('compressor' comp resp) and a tube amp simulation, both of which will provide the necessary final sound pressure.



To simplify this a little, when editing a sound the only **Parameters (1)** that can be changed are those which are appropriate for the selected drum sound (Snare, Kick, Hi-Hat etc).

There are four selected variable parameters, which are precisely tailored to the selected preset sound.

Oscillator Waveform (2): The fundamental waveform of the oscillators is selected here. You can choose between a sine, triangle, or sawtooth sample. If "Sample" is selected, you can use the controller to select a sample, i.e. a previously recorded drum sound. You can find these samples in the folder ./MxSynth/Robota/Samples. If you save samples in this folder, they will appear in the selection list.

(3): For each voice the Filter Cut-off, Resonance, Tube, Volume and Panoramaare always adjustable



With Select you can choose the instrument you want to edit in the step sequencer.



Click on “M” to set the instrument to mute, “S” for solo.

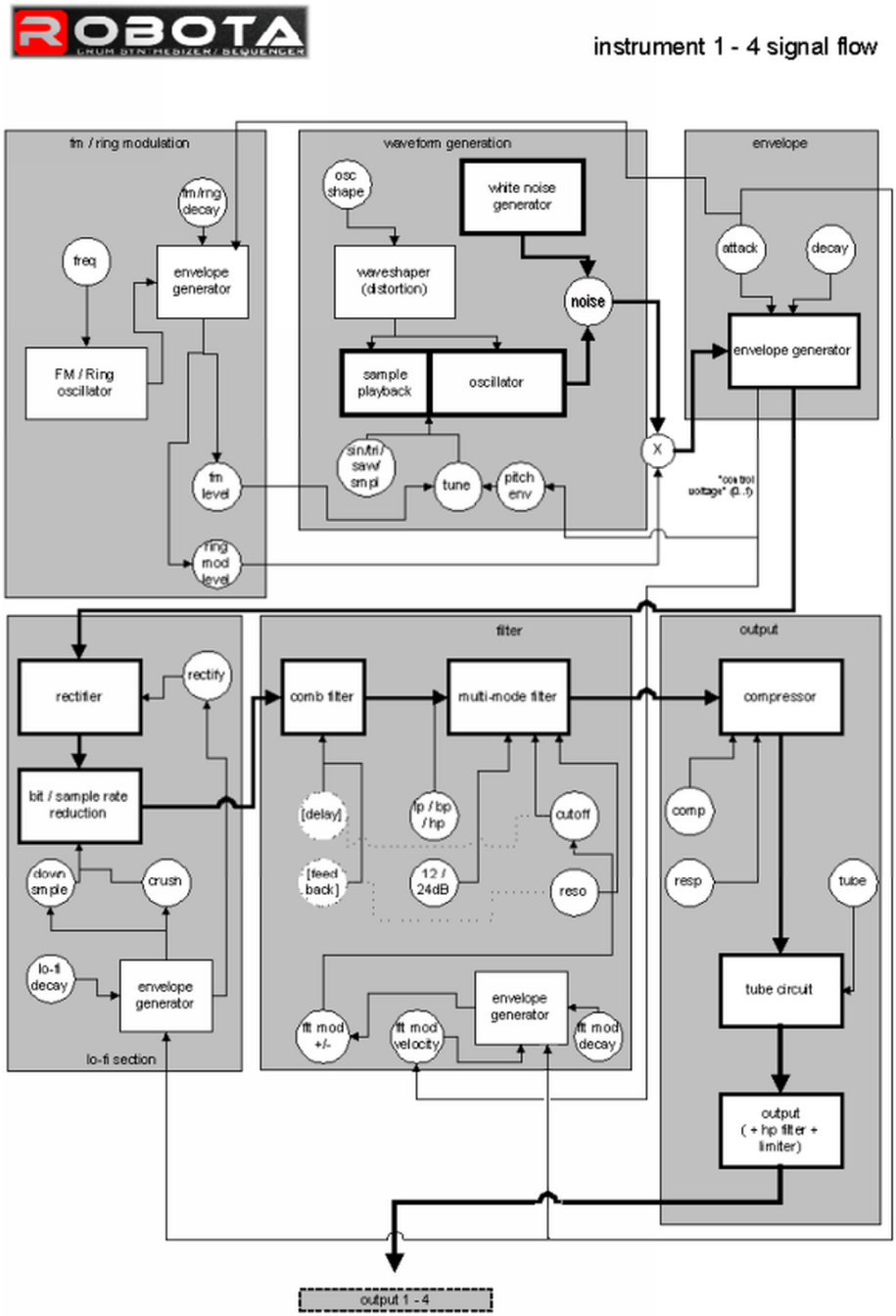


You can preview the instrument by pressing the loudspeaker button.

Master section

With volume you can control the master volume of the Robota. Distortion adds a controllable tube distortion to give the sound more power and make it sound rougher. The peak meter is used to control the output level. If this lights up in the red area, decrease the master volume.

Robota sound synthesis – schematic illustration



Detailed Circuit Diagram of a Robota Voice

Description of all control parameters:

Pitch Envelope (pitch env): Controls the pitch envelope level.

Tune: Tunes the instrument.

Oscillator Shape (osc shape): The shaper adds additional frequency components to the basic sound of the oscillator by artificially reshaping the waveform. A sine wave (shape = 0) can be reshaped up to a square curve (shape = max).

Oscillator Waveform: The fundamental waveform of the oscillators is selected here. You can choose from sine/triangle/sawtooth/sample. If "Sample" is selected, you can use the controller to select a sample, i.e. a previously recorded drum sound.

Noise: Here you can adjust the ratio between the oscillator sound and the noise generator.

Attack: This is the ascent time of the volume envelope curve. The higher the set value, the greater the length of time it takes for the sound to surge in. The attack value also applies to the LoFi and filter envelopes.

Decay: This is the descent time of the volume envelope curve. The higher the set value, the longer it takes for the sound to fade out.

FM/Ring modulation frequency (Fm/rng frq): This is the fundamental frequency of the frequency or ring modulation.

FM Level (fm lvl): At a low frequency FM first adds vibration to the sound, at high frequencies and low levels it creates bell-like sounds, as the level increases metal sounds, and finally noise.

Ring modulation level (rng lvl): Ring modulation creates typical auxiliary frequencies.

FM/Ring modulation decay (Fm/rng dcy): The time constant of the by-product of FM/ring modulation. Only the beginning of the drum sound is affected by the modulation.

Rectify: Distorts the audio signal

Crush: Bitrate reduction. Digital artifacts become audible with higher settings.

Down sample (dwnsmple): Sample rate reduction is useful for creating the "old school" sounds of older digital drum machines. The higher the set value, the more muffled the sound becomes.

Lo-Fi decay (lofi dcy): The time constant of the by-product of the three lo-fi effects for making the sound "dirty". Only the beginning of the drum sound of the lo-fi effects is affected if the decay is low. For instance, this makes the kick of a kick drum sound more interesting.

Filter modes (flt mode): The various modes of the filters:

- Low-pass (LP) - Sound portions above the cut-off frequency are filtered out.
- Bandpass (BP) – Sound portions above and below the cut-off frequency are filtered out.
- High-pass (HP) - All sound portions below the cut-off frequency are filtered out.

The mode is set as a preset and cannot be changed.

Filter frequency (flt freq): The cut-off frequency of the filter.

Filter resonance (flt reso): The sound portions are amplified at the filter cut-off frequency. If the resonance is high, the filter itself can also be used as an oscillator.

Filter-Modulation decay (flt mod dcy): Regulates how much and in which direction the filter envelope curve moves the filter frequency.

Filter-Modulation decay (flt mod dcy): Decay time of the filter envelope. Smaller values with high resonance create a "zapping" sound of the filter, greater values create the typical sweep sound.

Filter modulation velocity (flt mod vel): Specifies how much the filter modulation depth depends on the velocity. If this value is increased, louder beats will generate higher filter curves than quieter ones.

24 dB: The filter can operate with a slope of 12 dB or 24 dB. This mode is set as a preset and cannot be changed.

Comb filter (comb filt on): A comb filter can be activated here. This effect produces a feedback delay with a very resonant sound similar to a plucked string. The delay time and feedback levels are permanently linked to the filter parameters (frequency and resonance). This comb filter is set as a preset and cannot be changed.

Compressor: Controls the compressor strength. This lets you increase the "power" of the drum sound.

Compressor response (comp resp): Controls the compressor time. The lower the value, the faster the compressor readjusts the volume.

Tube: Controls the level of the tube amp simulation. This "saturates" the output signal of the voice and adds warmth to the sound if the settings are moderate. Increasing the settings makes the sound "dirtier".

Volume/Pan: Controls the volume and panorama position of the drum instruments.

Robota – sequencer



Classic light chaser programming, just like what you find in most of the classic drum machines and groove boxes, is used to control the drum patterns. The step sequencer consists of 16 individual step buttons with LEDs which correspond to the sections of a bar (16 or 32 beat, whereby only half a bar is displayed).

A flashing button corresponds to the drum sound at this point in the bar.

Clicking the button turns the step on and another click will turn it off again.



The maximum length of a drum pattern is four beats. The length can be selected in the small scroll bar above the toolbar.



The bar you want to edit can be selected with the corresponding "Edit" button. Use the “Follow” button to select follow mode, i.e. the step display follows the steps of the currently played beat.

„1>2-4“ **Auto Draw:** If more than one bar is set as the pattern length, "Auto Copy" mode ensures that the drum notes set in the first bar are automatically placed into the next bars. This also makes it easy to create a continuous beat, even with a loop length of four bars. Notes set in the bars further back are not affected by the "Auto Draw" function, e. g. making faint variations detectable only in the fourth bar.

Programming a new drum pattern:

- Use the slider to select the pattern length.



- Select “Event” mode.
- To make changes during playback, deactivate "Follow" and select the bar you want to edit with the "Edit" button.



- Use the “Select” button to select the instrument to be edited.



- The “Clear bar” button deletes all tracks in the current bar of the selected instrument.



- Turn on the corresponding step buttons. With the „Velocity“ slider you can set the velocity of the beat.

- Repeat this process for the other instruments.

Snapshots

It is also possible to automate the editable sound parameters of a drum sound with so-called “Snapshots”. This enables the sound parameters of an instrument to be saved on the step buttons of the sequencer.

Here's how to automate a drum instrument using snapshots:



- Set edit mode to “Snap”.
- To make changes during playback, deactivate "Follow" and select the bar you want to edit with the "Edit" button.
- Select an instrument and change the sound according to your preferences. You can check the sound of an instrument once playback is stopped by clicking the loudspeaker button or by triggering the sound by MIDI (requires activated monitoring in Sequoia).
- Save the sound as a snapshot to one of the step buttons.
- You can now change the sound of the drum instrument and save the settings to one of the other step buttons.

Note: Changes to parameters are not executed immediately but are blended in to avoid clicks and pops. When you play the pattern the drum sounds will sound different to what you expect if two snapshots with extreme parameter differences are too close to one another.

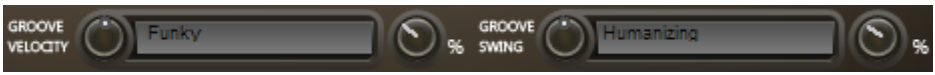


- You can switch between individual snapshots using the arrow keys when playback is stopped.

- Now activate the snapshot automation with "on".

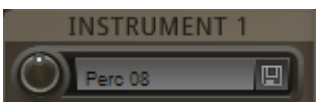
Groove-control

The secret to “groovy” beats is to play back the individual beats delayed or earlier according to certain patterns. For example, shuffle is used for house beats, whereby the even 16ths are delayed.

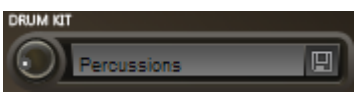


Robota features Groove Velocity and Groove Swing presets. Groove Velocity presets increase or decrease the original velocity of each step in a bar according to a particular offset for each step. Groove Swing presets apply a time offset to each step whereby the beat is played slightly earlier or later than the original step. The result is a considerably more lively-sounding drum sequence. The strength of each effect can be selected with the "**%**" controller.

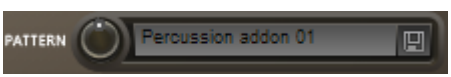
Robota – setups, drum kits, presets and patterns



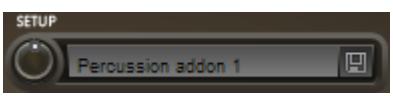
An individual drum sound is saved in a Preset.



The presets of the four voices can be saved together as a Drum kit



All note information and snapshots are saved in a pattern.



The combination of everything (instruments + pattern + further settings) is a Setup.

Load/Save: You can select presets, drum kits, patterns and setups using the knobs beside the display fields. To save click on  the "Save" button. At this point you can enter a new name in the data entry field. Press "Enter" to save.

Note: A drum kit only saves the names of the presets and not the actual parameters. This means that if you have prepared your own drum sounds by changing existing presets and would like to save these as a drum kit, you first have to save the drum sounds as new presets and then the new drum kit. The same is true for setups that only contain the pattern and drum kit name.

It is always best to save in the following order: Preset > Drum kit > Pattern > Setup. This only applies for creating your own "Templates". If you save your arrangement, the complete current status of Robota (synthesizer + sequencer) will be saved with it.

VST and ReWire

Sequoia provides the option of integrating software instruments into a VIP according to VSTi standards or through ReWire and then controlling them with MIDI.

VST is an interface for additional audio software that enables VST instruments and effects to be used as plug-ins in Sequoia. VST instruments create sounds virtually and can imitate real instruments and even synthesize entirely new sounds. They are controlled by MIDI and can be played using a MIDI compatible keyboard.

VST instruments can be accessed as audio track inputs and as mixer channels. In this way, the audio signals of a VST instrument can be further edited with all the options offered by the mixer, including the EQ, effects and routing. The maximum number of plug-ins is only limited by the performance of the system processor. You can increase the number of applied VSTs almost limitlessly by using the integrated freeze function

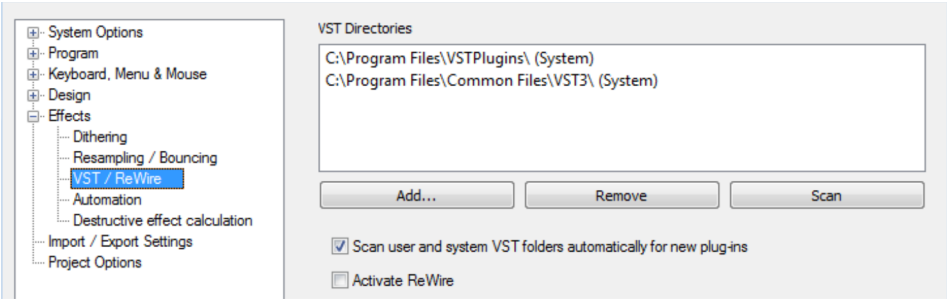
ReWire is a software protocol that allows you to remotely control another program from within Sequoia. For example, you can virtually connect a different DAW to Sequoia and exchange data between the programs directly. This does away with the need for importing audio and MIDI files.

ReWire is generally used for sample-exact, real-time transfer of audio channels between two programs. Both programs can be connected through the same sound card. The transport functions of the applications (such as playback and forward/reverse) are connected with ReWire. If ASIO drivers are being used, sounds from ReWire-compatible applications can be assigned to different sound card outputs.

Installation of VST Plug-Ins

The VST plug-ins provided by Sequoia are detected automatically and can be used immediately. To use third-party plug-ins in Sequoia you must tell the program in which folder they have been installed. To do this, proceed as follows:

1. Install each VST plug-in according to the instructions provided by the manufacturer. In most cases the plug-in installer suggests a default folder (e.g. C:\Program Files\VST): these System VST folders are included in the plug-in scan by default. You can also use any other folder. If you already have VST plug-ins installed on your system, it is best to use the existing folder.
2. Open the system settings (Key Y) and go to "Effects > VST/Rewire". Under the VST plug-in folder list, click "Add" and enter the folder path.



3. To start the plug-in scan, click "Scan" and choose "Scan selected VST folders" from the menu. This scan may take some time for many installed plug-ins and instruments. Not only are all the plug-ins imported, but they are also checked for usability within the program. Incompatible or incorrectly installed plug-ins and those that cause the plug-in search to crash are also included in the list and flagged as unusable, so that they are skipped during the next scan and cannot cause problems again. To scan these plug-ins again, use the menu item "Test failed plug-ins again".
4. The newly found plug-ins can be used immediately.

Hint For a complete reset of the VST settings and a new scan of all plug-ins, delete the file "VSTPlugins.ini" in the folder C:\ProgramData\Sequoia. All paths will have to be newly entered.

If you activate the option "Scan user and system VST folders automatically for new plug-ins", the list of plug-ins is automatically updated after every program startup when the track settings or plug-in browser are opened for the first time. This means that the program searches for newly installed plug-ins and removes uninstalled plug-ins from the list. This search is performed only once per working session, and the next time you access the track settings it'll open immediately.

You can also specify and scan any number of plug-in folders.

Hint: If you add VST plug-in subfolders into the main VST folder (e.g. "Equalizer", "Filters", "Modulation", etc.), these will also be displayed as subfolders in the plug-in browser.

Hint: The command for a new plug-in scan is also in the menu that you can open with an arrow button on each plug-in slot.

ReWire client application

ReWire-compatible client applications (e.g. Propellerheads Reason) can be integrated into Sequoia as synthesizers.

Activate ReWire functionality in the system options settings ("Y") under "Effects -> VST/ReWire". Afterwards, installed ReWire applications may be loaded as instruments into a MIDI out slot in Sequoia. All ReWire client applications appear as individual sections in the selection menu of the track's plug-in slot and are loaded as software instruments (VSTi). The client application should always first be started after launching Sequoia and closed again before Sequoia is ended. Some client applications may be opened automatically by right clicking on their name in the MIDI out slot.

The ReWire application may be controlled via MIDI, just like a software instrument. You can set up the ReWire client's individual output signals according to the multi-channel software instruments on multiple tracks. The client application runs, starts, and stops synchronously to Sequoia's time position.

The "classic" MIDI channel for MIDI notes and events is not important, it is replaced by the ReWire MIDI bus system. For each track, a MIDI object is able to control a ReWire MIDI BUS. This means that all of a MIDI object's events apply to the track for this ReWire bus, independent of the channel number the events originally had. Multi-timbral MIDI objects (like those created in the MIDI file import) cannot be played correctly with ReWire. However, you may access the ReWire client across multiple tracks on various ReWire MIDI buses.

ReWire supports up to 4,096 MIDI buses. A ReWire client only registers the MIDI buses that are actually available with the host (Sequoia). You can therefore select the bus within the track in Sequoia's MIDI channel menu (e.g. the receiving instrument for Reason).

Several ReWire clients may be opened directly via the host. If you start your client application manually, this automatically detects the presence of a host application and opens in a special client mode.

Detailed information is available in the section "VST/ReWire setup plug-in buffer (view page 390)".

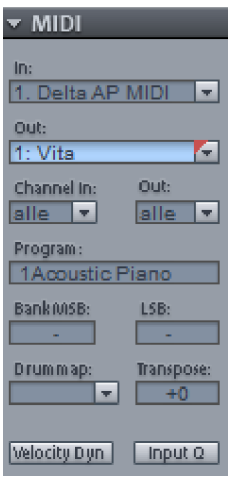
Load software instruments

You can also assign a software instrument to each track; the instrument is selected as a MIDI playback device. Selected instruments or their individual outputs are shown directly in the first VSTi plug-in/insert slot of the track header and track editor and can also be muted (left click) and opened (right click) from there.

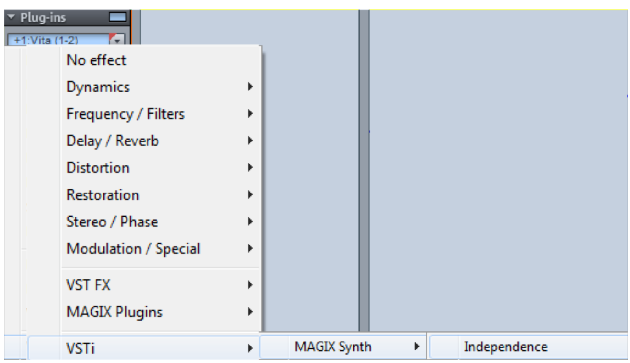
Load an instrument by selecting a "New instrument" in the project window as a track output. This can happen at various positions:

- In the MIDI section of the track editor: "Out" slot -> new instrument

This is the standard method for loading and routing a software instrument.



- In the playback device context menu, (right click "Mute -> MIDI -> New instrument)
- Plug-ins slot ->Plugin Browser in the track editor



Software instruments may also be created in the mixer:

- Insert effect slot of the mixer ->Plugin Browser

This option is located in the top insert slot of the individual mixer channels.

Alternatively, you can also install and manage software instruments in the VSTi rack. To do so, open the VSTi manager (view page 208).

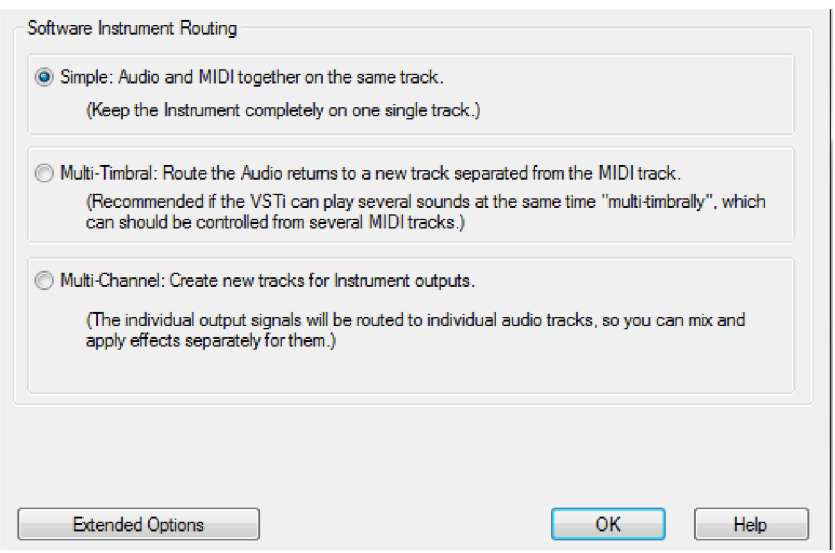
Note: You can load plug-ins In the master section of the mixer, just like in the mixer tracks. Click on the arrow to the right beside one of the plug-in slots and select the desired plug-in from the menu.

Loading routing settings with software instruments

An instrument may also be assigned to each track as a MIDI output device.

Audio output channels may also be routed from software instruments to any VIP track. MIDI (send) and audio (return) signals from a software instrument may be present together on one track (but this is not absolutely necessary). Sequoia is fully flexible in terms of individual routing configurations.

When installing a software instrument, the "Routing for multi-channel instruments" dialog opens. This is also available via the VSTi manager under "Routing settings".



Note: Press "Extended options" to specify that this dialog is always displayed when inserting a new software instrument into a track or when adding a track for individual output in the VSTi manager.

Simple mono or stereo software instruments may be installed completely in the current track. In this case, the track sends MIDI to the instrument and receives the audio signal from the instrument. Select the "Simple: Audio and MIDI together on the same track" configuration. Now all audio signals will be routed to the track that contains the instrument. Multiple outputs will be mixed together before the mixer effects. You may specify that only the first stereo output is routed to the current track via the advanced options. You may specify that only the first stereo output is routed to the current track via the advanced options.

Multi-timbral: Route the audio to a new track separated from the MIDI track

Several MIDI tracks are typically used for so-called "multi-timbral synthesizers", which may play several sounds on different MIDI channels simultaneously, in which case each device controls a specific sound program (part) on a specified MIDI channel. You can specify in the extended options that only the first stereo output is routed to the current track. The audio return track may also be hidden in the arranger. This setting is useful if individual outputs of an instrument are controlled by a single MIDI file, which therefore do not contain objects or information in the arranger window.

Multi-channel: Create new tracks for instrument outputs

Choose this option to automatically create new tracks for all VSTi audio outputs. The newly created tracks are named accordingly. The mono/stereo configuration is assumed from Sequoia (default).

Note: Individual outputs may also be routed to separate tracks. To display hidden tracks in the arranger window again, use the track manager.

Advanced options:

Stereo/mono (standard): Information delivered from the plug-in is used for routing.

All as mono: This option forces individual outputs to be treated as mono outputs.

All as stereo: This option forces individual outputs to be treated as stereo outputs.

Audio/MIDI combined (tracks also send MIDI): Activate this check box to automatically route the MIDI output for each individual track to this instrument.

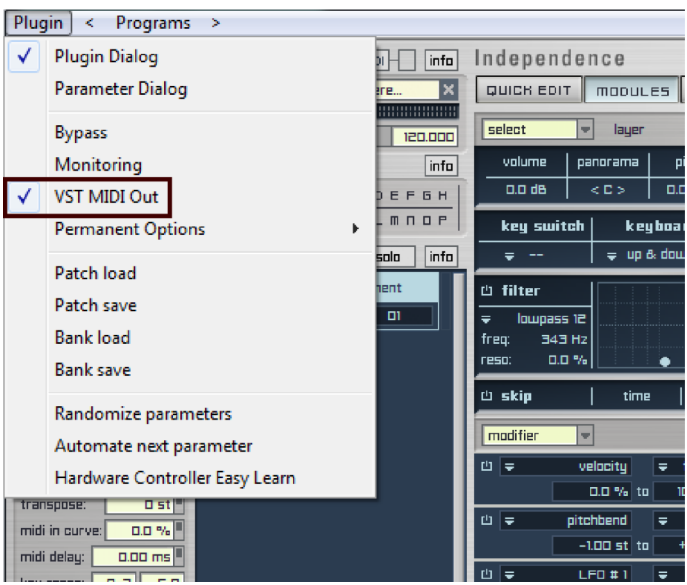
Don't show the audio return tracks in the arranger: All newly created output tracks for this instrument are hidden in the arranger but still appear in the mixer window. This setting should be used if an instrument's individual outputs are controlled by a single MIDI file, and therefore do not contain objects or information in the arranger window.

Note: A virtual synthesizer's audio output may normally be created, edited, and mixed in the same track as the MIDI data the instrument is receiving. In some cases, this results in duplicate use of the volume fader which, on the one hand, controls velocity or MIDI volume (CC7), and on the other, this controls the audio level. These are not identical parameters. For instance, you may include a MIDI instrument played with high velocity quietly in the mix and vice versa. Therefore, the volume fader may be assigned differently. To do this, right click the track's volume fader.

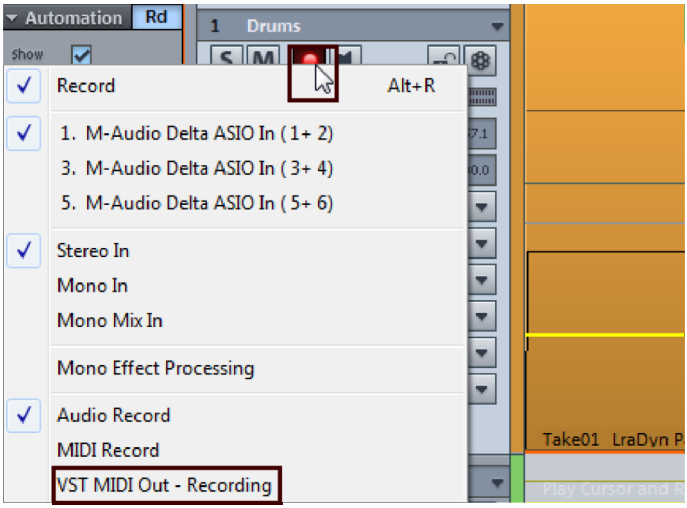
VST MIDI out + Audio out recording

(for "Mixer FX monitoring/Hybrid Engine" only)

You may record VSTi audio outputs for VSTi return tracks by activating recording in these tracks. "MIDI out" may be activated for every track VST plug-in or VSTi in the plug-in panel menu. This sends and records MIDI data from the VSTi in another track.



By activating the option "VST MIDI out recording" for a track (right click the record button), you can record all received VSTi MIDI out data in the corresponding track. Select the instrument output of the VSTi to be received.



If MIDI thru (view page 99) is activated, this signal may also be routed to another VSTi/VST-FX or external MIDI device. By right clicking the "Moni." button in the transport control, activate the "Automatic MIDI monitoring (thru) at recording start" function.

After recording, apply the track freeze function to visualize the waveform of the recorded track. If you remove the check mark again in front of the VSTi output with a right click on the record button in the track box, the routing connection to the sending VST instrument will be removed.

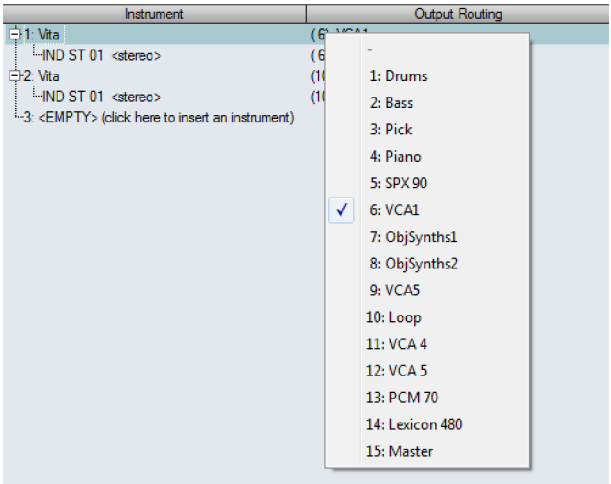
Routing VST instruments using the VSTi manager

There are many flexible methods for signal routing of instruments.

As described in the previous paragraph, an instrument may be fed from several MIDI tracks, and the outputs of an instrument may also be sent to several tracks.

Routing instrument outputs

Instrument outputs can be easily routed by using the VSTi manager. Simply open the VST instrument to view the single outputs and then right-click the "Output routing" column.



Now switch to the track that is to be fed with the respective single output. The assigned output then appears in the target track's plug-in slot.

Alternatively, you can also route directly to the track header of the arranger: To play the output of an already loaded instrument on a specific track, click with the right mouse button on the record button of the track, go to "Instrument outputs" and select the output of the software instrument available in the arranger you want to sound on this track by placing a check mark. Several outputs of an instrument can be routed simultaneously to a track. You can even combine the outputs of different instruments onto one track.

Deactivation of the instrument outputs works in the same way. Remove the check in front of the corresponding output in the "Instrument outputs" submenu.

Note that the routing options of the mixing console may also be used. A track containing the audio return of any instrument may be routed to any submix bus, AUX bus or master.

Routing instrument MIDI inputs

Most software instruments require MIDI data in order to generate an audio output. To route the MIDI output of a track to the input of a VSTi, click on the MIDI button in the Track Editor of the corresponding track and select the VST instrument in the "Out" slot that you want to activate. Already loaded VST instruments are listed in this menu.

To open a completely new instance of an instrument that works independent of any existing instruments within the project, select its name from the "New instrument" submenu. You can differentiate several instances of the same VST plug-ins by the number (index) in front of the name of the software instrument.

Previewing VSTi instrument outputs solo

To preview solo instrument outputs, switch the track to which signals are added by the corresponding instrument to solo. Even if the MIDI objects, which in effect are responsible for the sound, are not present in the track, you will still hear the selected track solo. Sequoia automatically recognizes which MIDI tracks have been routed to this track and plays the instrument output.

Instruments with multi-channel outputs

The number of available individual outputs generally depends on the instrument, and this may be adjusted via the instrument's settings. Outputs may be available in stereo or mono.

Here are two examples of typical handling of individual outputs:

Addressing an instrument on a MIDI channel and distributing the sounds across several tracks

This approach is ideal for drum samplers to address an entire drum set and simultaneously mix and edit the drum sounds individually.

Addressing an instrument on several MIDI channels

Several MIDI tracks are typically used for so-called "multi-timbral synthesizers", which may play several sounds on different MIDI channels simultaneously, in which case each device controls a specific sound program (part) on a specified MIDI channel. The advantage of this is that these instruments only require one instance of the plug-in for several sounds. Multi-timbral software instruments often have separate audio outputs. Don't forget to route the individual parts to the desired outputs within the VST instrument (panel).

Plug-in panel – graphical interface

The plug-in dialog (panel), the graphic interface, opens when an instrument is loaded and can be opened again later by right clicking on the plug-in slot which displays the instrument's name. For instrument plug-ins, you can also open the panel via "Track -> VST instrument editor".



The graphical interface opens by default. The ReWire client application will open when ReWire instruments are selected.

"Programs" features the presets for the corresponding plug-in.

Besides "Programs", you will also find "Sidechain input (view page 291)" for those plug-ins that feature the "Sidechain" function.

Plug-in Menu Functions

Bypass: Deactivates the instrument and mutes it. It is important to note that some instruments require processor power when they are bypassed. For this reason you should remove these instruments completely if you don't need them anymore.

Monitoring: This option must be switched on if you want to hear the instrument when it is being played live or recorded. If you want to have have monitoring active when the track's REC button is on, select the "Tape Monitoring" option under "System Options" > "Audio System".

VST MIDI Out - Recording: In the Plug-in Panel menu you can activate "VST MIDI Out - Recording" for each track VST plug-in or VSTi. This causes the MIDI data from the VSTi to be sent and recorded in another track.

By activating the option "VST MIDI Out - Recording" on a track (right-click on the record button), you can record all received VSTi MIDI Out data in the corresponding track. If MIDI Thru is activated, this signal can also be routed to another VSTi/VST FX or external MIDI device.

Permanent Options: These additional settings are valid for all instances of each plug-in.

Note: We only recommend changing the "Permanent Options" when compatibility problems occur. If you make changes here, reload the project to apply them.

Limit to only 1 CPU: All instances of a VSTi will be calculated only on one CPU. This avoids any multi-CPU conflicts between various instances.

If you apply this option to an effect plug-in, all track and object effects on the track that use this plug-in will now be calculated only on one CPU. As soon as you activate this option for an effects plug-in, the entire track and all its effects will be calculated only a single CPU.

Note: When using UAD cards, this option is activated by default.

Force calculation by silence input: When letting single tracks be processed as Economy tracks over the playback engine (view page 602), Sequoia normally doesn't process track signals if no audio signal is present. This decreases the load on the CPU.

With this plug-in option you can ensure that the entire track to which the plug-in is connected is calculated without interruption even when no audio signal is running through the effect. This is quite useful for delay plug-ins on Economy Tracks or for bouncing Economy Tracks, for example. This option is also recommended for plug-ins that produce their own audio regardless of the input.

Note: A continuous calculation, even of empty or silent tracks, takes place in the Hybrid Engine. This function can be deselected in the global Performance options (view page 674) (keyboard shortcut "Y") by checking the option "Deactivate FX on empty or silent tracks for ASIO".

No automatic copying: This option prevents automatic copying of plug-ins when you split or copy objects.

Load/Save patches/banks: Here you can save an instrument's settings and sounds. The standard formats for this are: *.fxp for patches or *.fxb for whole banks. Some instruments have their own patch/bank format. In this case, settings are usually loaded and saved directly through the instrument interface.

Set parameters randomly: Use this function to set all parameters of an instrument to a random value. For synthesizers this option can lead to surprising results and provides an interesting option for sound design thanks to the random generation of new sounds. However, please keep in mind that only the parameters that are accessible through the interface can be set randomly. With some very complex virtual synthesizers or modular systems, some parameters may not be adjustable using this option. It is important to note that due to the random placement of various parameters, very extreme sounds can be generated. This may lead to no sound being produced at all or extremely loud volumes and frequency ranges being reached. For this reason you should keep an eye on the monitoring volume while experimenting with this function.

Automate next parameter: The next parameter that you change during playback will be recorded with automation. You can also do this by holding "Ctrl + Alt" while clicking and changing the parameters with the mouse.

Hardware Controller Easy Learn: When you activate this function, you can set up your controller by

1. dragging and dropping the plug-in element that should be learned.
2. Move the element on the hardware controller you selected

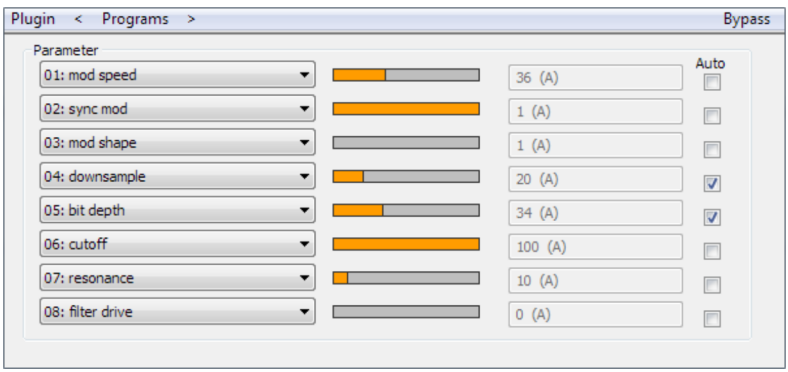
Pay attention to the following:

- When you are finished assigning the controls be sure to shut "Hardware Controller Easy Learn" off to avoid making any unwanted changes to the settings.
- The hardware controller element must first be learned in the hardware controller setup and cannot be left empty.
- Learned elements permanently modify the internal mode in the hardware controller setup. The assignment is also available later for other projects.
- It can be cleared again with the "Restore modes" button beside the internal mode in the hardware controller setup

Detailed information about adding hardware controllers is provided under "Hardware controller (view page 531)"

Plug-in parameter dialog

If the selected plug-in does not have its own interface, the parameter dialog will open. This display mode is limited to just a few control elements and can only be opened via the menu of the instrument window. In this dialog, eight freely selectable instrument parameters are clearly and numerically displayed. Bar control elements may also be used to adjust the parameter values.



Once the automation of the selected parameters is activated, the automation curve will be displayed in the project track for editing. You can automate parameters during playback via the corresponding bar faders.

The parameter selection is saved for each plug-in so that the next time you use the same plug-in, the same parameters are used. This provides the advantage that every frequently used parameter of the corresponding instrument is available immediately after it has been opened.

Play and monitor instruments live

Monitoring

Use the following settings for live monitoring:

- ASIO driver ("File > Program Preferences > System/Audio > Audio system > Driver system")
- Activate global record monitoring in the transport control ("Moni" button)
- Activate the track monitoring button (loudspeaker symbol)
- Software monitoring or Mixer monitoring ("Menu, File > Program preferences > System/Audio > System options > Audio system > Monitoring Setup")

Tip: You can also select this monitoring mode by right clicking "Record monitoring" on the transport control.

Latency: Please note that a system-dependent delay between pressing the key and hearing the sound occurs when you play virtual software instruments. This so-called latency time is mainly determined by the buffer size set for the ASIO drivers. To play in an acceptable manner, we recommend a buffer size of max. 1024 samples. This corresponds to 23ms at 44.1kHz. For many users, however, a latency of 3 ms, i. e. 128 samples, is optimum. Please note that the CPU load also increases at a lower buffer size. The "live" delay only occurs when an instrument is played; playback latency of an already recorded MIDI track via an virtual instrument is automatically compensated by Sequoia.

As software instruments are fully integrated into the audio engine of the program, the signals can be routed, mixed and equipped with plug-ins any way you like. There are restrictions when recording VST instruments, according to the selected monitoring settings.

Note: For monitoring effects during input, "Track FX monitoring" mode or "Mixer FX monitoring/Hybrid Engine" must be activated.

Recording and playing back an instrument

An instrument is recorded in much the same way as a normal MIDI track. Make sure that the MIDI recording is active in the desired track and then press the record button in the transport control. The recording begins immediately.

When playing back already recorded MIDI tracks, track monitoring should be switched off.

To be able to record MIDI data sent by a VST instrument in another track, activate the "MIDI out" function in the plug-in panel menu for the respective track VST plug-in or VSTi. This sends MIDI data from the VSTi.

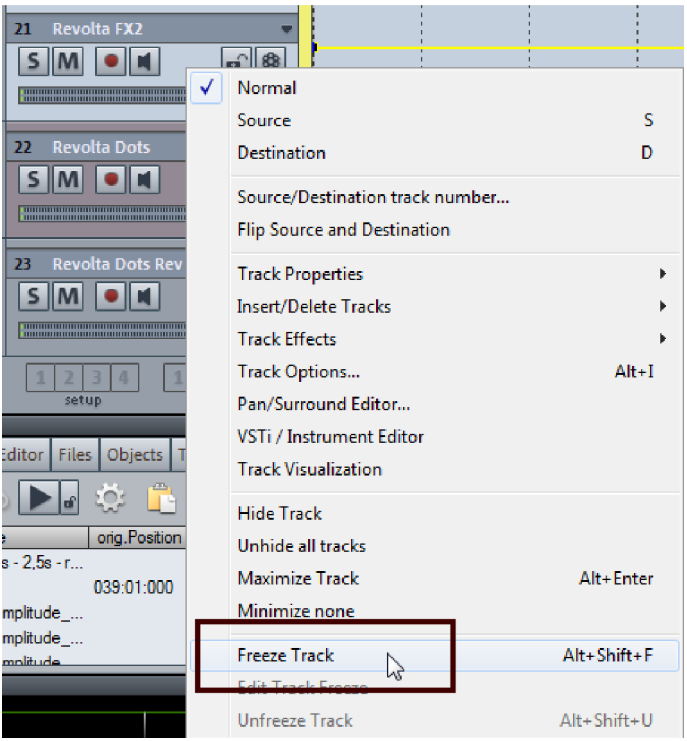
By activating the option "VST MIDI out recording" for a track (right click the record button), you can record all received VSTi MIDI out data in the corresponding track. Right click on "REC" in the track editor and set a check next to the corresponding context menu item.

You may record VSTi audio output for VSTi return tracks by activating recording in these tracks.

If "MIDI thru" is activated, this signal can also be routed to another VSTi/VST-FX or external MIDI device.

Freezing VSTi Tracks

1. Click on the small arrow next to the track name in the track header and select "Freeze Track"



Now you should see a wave object instead of the MIDI object. The new wave object is normally slightly longer because the reverberation time is included.

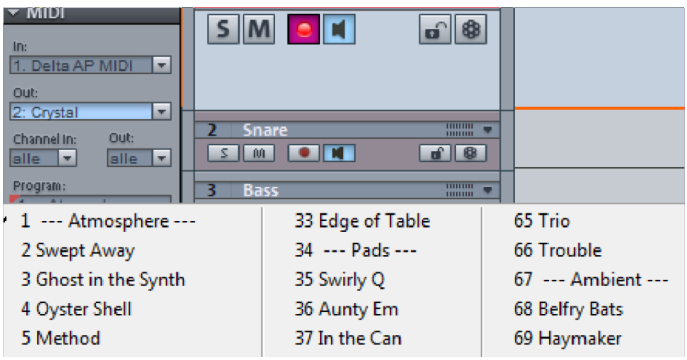
2. If you want to edit the MIDI files again, open the context menu and select "Unfreeze Track".

3. In the MIDI editor you can now make alterations, change the sound or experiment with a different VSTi.

4. When you are finished making changes you can "Refreeze Track".

VST instrument preset selection

If an instrument has presets, these are displayed in the "Program slot" ("Prg") with their internal name (if "MIDI" mode is activated).



This dropdown menu can be used to easily "step through" the presets of your instrument. These presets may also be selected using the instrument window's "Program" menu.

If your MIDI input device can send "Program change" commands, you can also directly select programs of this instrument. The program slot is automatically updated in the track to provide an overview of the program at all times.

Sidechain Input

At the track and master level of a correspondingly equipped VST/MAGIX plug-in (e. g. am-munition), as well as in advanced dynamics, you can activate the sidechain and specify one or more of the previous tracks as the sidechain input(s). In the tracks or mixer channels sought out as side chain signals the track name of the fed track appears in the AUX section just above the send level.Internally, a side chain bus (AUX bus) is created.

Right clicking on the AUX send button selects this as "Side chain send".

The "Side Chain Solo" function bypasses the corresponding effect and only the input of the side chain is played. When closing the plug-in dialog, the "Side chain solo" function is reset.

The command "Sidechain filter" opens a parametric equalizer that may be used to edit the sidechain signal.

Detailed information on the sidechain function can be found in "Effects -> Dynamics -> Advanced Dynamics -> Dynamic parameters (view page 872)".

MIDI in Sequoia

In this chapter you'll learn how MIDI data can be imported, played back and edited with internal editors in Sequoia. MIDI data is also used to control internal and external synthesizers and VST instruments (view page 275) as well as the MIDI timecode synchronization (view page 505). Remote control using external hardware controllers also takes place through MIDI control signals.

Tip: For seamless work with MIDI, always use the monitoring settings "Track FX Monitoring", "Hardware Monitoring/Hybrid Engine", or "Mixer FX Monitoring/Hybrid Engine".

Import, record, edit

Importing MIDI Files

Existing MIDI files can be imported into a Sequoia VIP project as objects.

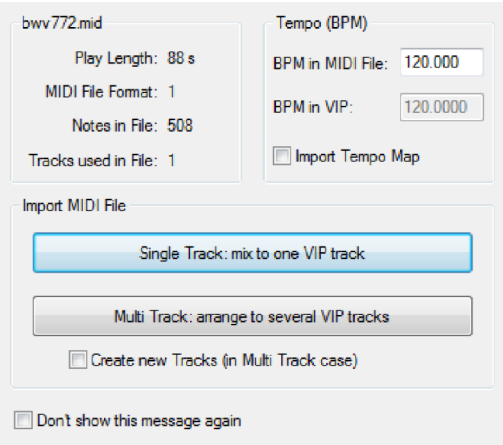
Menu: File > Import > Load MIDI File...

Keyboard shortcut:: Shift + M

Mouse: Drag & drop from file browser or Windows Explorer

When importing MIDI files, note that only type 0 and 1 standard MIDI files (SMF) are compatible with the import function. This corresponds with tracks featuring format 0 and multiple tracks featuring format 1. Each file should feature the file extension .MID in order to be detected as a valid import format.

The following dialog is displayed automatically:



Standard MIDI files often also contain tempo information. Sequoia displays this information in BPM (beats per minute) in a separate section of the dialog. At this stage of importing, select whether to adjust the virtual project tempo to the tempo of the imported MIDI file or not. Place a checkmark by the option "Transfer Tempo Map".

If you select the option "All MIDI channels into one VIP track", Sequoia inserts the MIDI object into the currently selected track of the virtual project. The MIDI object that is created contains all the tracks of the MIDI file.

To provide a separate track in the virtual project for each MIDI channel of a standard MIDI file (format 1), select the option "Each MIDI channel into a new VIP track".

Sequoia also offers the option "Create new tracks" for multitrack import of MIDI files. If this function is not selected, existing tracks will be used for importing.

Record MIDI Tracks

To find out more about MIDI recording and MIDI recording modes, see "MIDI recording (view page 99)".

Editing MIDI Objects

Editing MIDI objects in Sequoia follows the same principle as editing audio objects: MIDI objects can be copied, split and trimmed, have fade handles for fading in or out, and a volume handle that scales the MIDI velocity.

When MIDI objects are frozen („Object“ > "Freeze Objects (view page 804)“) the audio return signal of a software instruments replaces the MIDI object with audio objects.

Note: Make sure that the audio return signal of your MIDI receiver (VST or ReWire instrument) is routed to the MIDI track.

MIDI objects can also be edited using the MIDI object editor as well as in the different MIDI editors such as the Matrix editor, the drum editor, the controller editor, the event list and the note editor. The MIDI Object Editor will then only change the object properties and you can change the MIDI files yourself using the MIDI Editor.

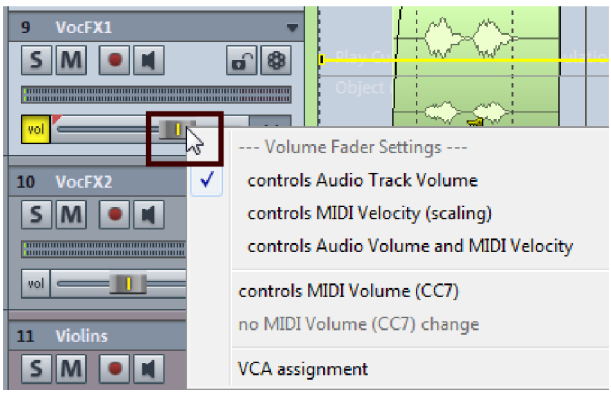
MIDI setup

All global MIDI settings are provided in the system dialog (shortcut: "Y") under "System options" > "MIDI".

Please read "MIDI settings" (view page 604) for more.

Volume Control Settings

1. Right-clicking on the volume control brings up the fader settings.



2.

- For example, if you use an external MIDI synthesizer, you should set the corresponding volume controller to "MIDI volume (CC7)".
- If on the other hand you don't use any external equipment, you should select the "Audio Track Volume" option.
- If you want the volume fader for MIDI tracks to reflect MIDI velocity scaling select "MIDI Velocity (Scaling)"

In any case it is worth checking these settings if you notice that a fader isn't having an effect on the track's volume.

MIDI object editor

Menu: Object/Object editor

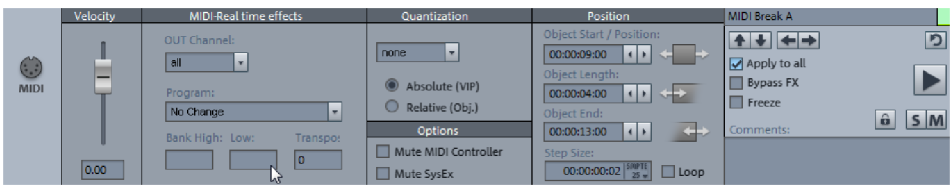
Keyboard shortcuts: Ctrl + O

Mouse: Shift + double-click

MIDI object parameters may be easily edited using the MIDI object editor which is structured similarly to the audio object editor. Among other options, you can set sound, length, volume, and program swapping for each MIDI object.

A Sequoia MIDI object can contain events in up to 16 channels. For example, it is possible to control a multi-output VST instrument using a single object on several MIDI channels.

Or select a specific MIDI channel for all events in the MIDI object editor.



You can also use the MIDI object editor to quantize MIDI files into MIDI objects.

Note: These settings influence the MIDI object in realtime and not destructively, making the changes invisible in the MIDI matrix or drum editor.

Important functions and settings in the MIDI object editor:

Velocity: Normally, every MIDI note contains a velocity value which determines how "hard" the note will be played back. Changing the velocity value in the object editor scales the velocity of the notes in the MIDI object to its physical maximum or minimum values (1 or 127). Changing the fader is identical to changing the top-mid handle of the MIDI object in the VIP track.

MIDI real-time effects: The settings for a MIDI channel can be found here:

OUT channel: Here you can redirect the MIDI data of a channel to another channel.

Program: The values preset here will always be sent if the object is played back again.

Bank High/Low: This sets the MIDI bank change bytes.

Transp.: This button transports all MIDI notes.

Quantization: While the quantization conducted in the MIDI editor has an immediate effect on the start position of the notes, a virtual quick quantization takes place. Absolute refers to the absolute resolution of the VIP project while Relative refers to the relative resolution of the object itself.

Options:

Mute MIDI Controller: Switches the MIDI controller off.

Mute SysEx: Prevents the sending of SysEx data.

Position: Define the object start/position, object length, object end and step size here. The setting is equivalent to moving the object in the VIP track or changing the length using the lower length handles right and left in the object.

Loop: If you set a check here, then you will set the MIDI object to "Loop" mode. Now drag the bottom right object handle to the right and you will see that the events of the MIDI object are arranged in a row, according to the new length of the MIDI object.

The object name is displayed in the upper text field and can also be edited.

In the square field to the side, you can select the object color.

Use the up/down arrow keys to "jump" up or down to the object of a neighboring track.

Use the double arrow buttons to jump to the previous or next object on the same track. If multiple objects are selected, these buttons will not be active.

If the "To all" option is active, all selected objects are updated when the object editor is opened. All recently applied settings that have been made in a selected object will be applied to all other selected objects.

Note: In a MIDI object - velocity changes are transferred relatively, i. e. they are increased or decreased relative to the velocity of other selected objects.

Bypass FX: Here you can deactivate all effects.

Freeze: If you click on the freeze function for an object, this will be calculated as a new wave file with all the associated effects. The new freeze object takes the place of the original object.

Play/Stop: This button corresponds to the normal playback function (spacebar).

Lock: The "Lock" button has the same function as the key button in an object. The object is then protected against any inadvertent horizontal movement. Additional locks against vertical moving, volume adjustment, fades, length adjustments, ripples or deletion can be set in the system options (shortcut: "Y") under "Program" > "Object lock definitions" (view page 786). Locking can be temporarily cancelled by pressing the "Alt" key.

Solo: During playback this button moves the playback marker to the start of the selected object and plays that object only.

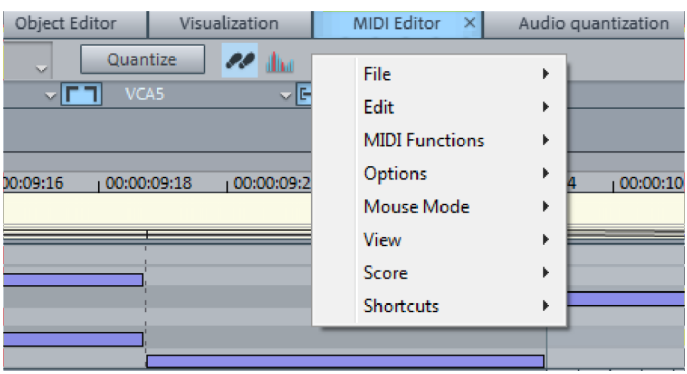
Mute: This button mutes the selected object. By right-clicking you can also mute the left and right channels individually.

Comments: You can type comments relevant to the selected object in this field.

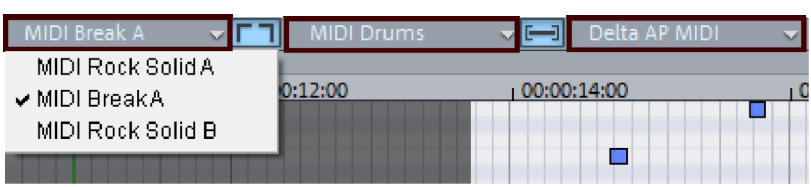
Opening the MIDI Editor

The MIDI Editor is opened by double-clicking on a MIDI object in the project. You can also start the MIDI Editor for all selected MIDI objects by selecting "Object > MIDI Editor ..." in the menu or by clicking on the "MIDI Editor" button.

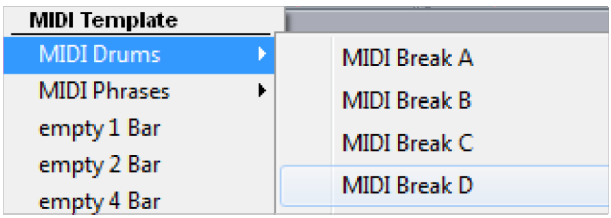
If you open the MIDI editor in the Docker (view page 177), the MIDI menu can be opened by right-clicking on an open area in the toolbar.



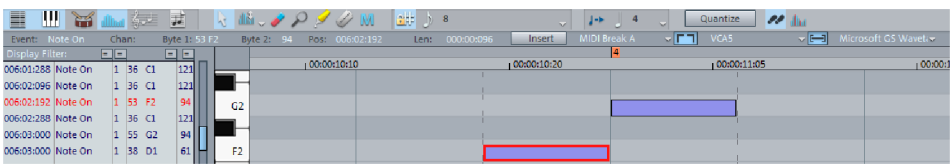
The MIDI Editor always displays all of the selected MIDI objects. In the field to the right you'll see the MIDI output and the software instrument that is activated for this track.



If no MIDI object is selected, a window appears which lets you create an object at the playback position. You can then choose between a menu of predefined MIDI templates from within the Sequoia “Templates” sub-folder.



Now you can use the Draw Tool to create events in the MIDI editor (view page 301) or record notes using your MIDI keyboard.



Detailed information about MIDI recording is provided in the chapter "Sequoia Quick Start -> MIDI recording (view page 99)".

Working with the MIDI Editor

MIDI data can be edited in the MIDI Editor in five sub areas:


Matrix Editor (Piano Roll)


Drum Editor

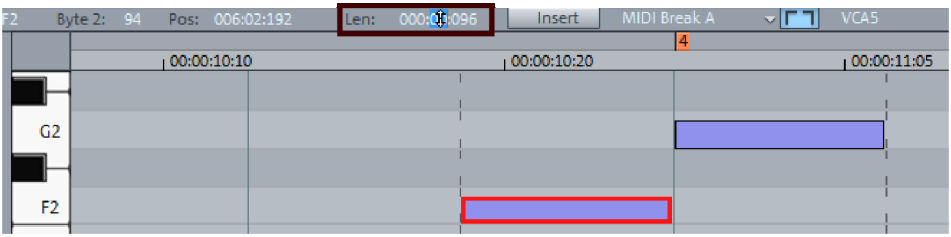

Controller Editor (for example, velocity, MIDI volume...)


Event List


Score Editor

Various tools are available for editing, e. g. the pencil or eraser tools.

The "current" MIDI events displayed in bright red can be very precisely adjusted in the editing fields above the editing window (view page 304).



Changes such as moving or deleting notes always apply to all selected MIDI events (displayed in red). Changes to the selection in an editor range always apply to every other editor range as well. For example, you can select a group of notes in the Matrix Editor and then change the velocity of these note groups in the Controller Editor which modifies all selected notes simultaneously.

MIDI Editor: Select events, delete events

(Matrix Editor, Drum Editor, Controller Editor, Event List, Score Editor)

- **Select events**: Left click the event or drag out a frame around the event while pressing the left mouse button
- **Add/remove event to/from selection**: "Ctrl" + left click on event or "Ctrl" + pull the frame over the new event
- **Set or change current events within a multiple selection**: Left-click on selected event
- **Set current event, deselect all other events**: Double-click on event
- **Selection of events within a range**: Left-click on the first event, then left click on the last event while holding down the Shift key. Another alternative is to drag out a frame around the events you want to select.
- **Selection of all notes of a pitch**: Double-click on the assigned key on the keyboard display.
- **Select all notes behind click position**: Shift + Alt + Double-click
- **Select all notes**: Ctrl + A
- **Freehand drawing and changing of note lengths without snap points**: Hold Alt key down while dragging.
- **Select previous or next note**: Left or right arrow keys
- **Delete events**: Right-click
- **Delete selected events**: Backspace or Del.

MIDI editor: Copy/duplicate events

You can copy selected events using "Ctrl + C" and then paste them at any playback marker position using "Ctrl + V".

Copying and pasting MIDI data is available not only within the MIDI editor, but also between different MIDI objects. Copied MIDI data is always inserted at the current position of the playback marker.

Duplicate: Press "Ctrl + D" to copy selected notes and add these to the next snap point behind the selection. Activate the quantization grid ("MIDI menu -> Quantization grid active") to do this.

MIDI-Editor: Move events

To move events, move the mouse over the event. The mouse pointer turns into move icon. Now you can move the event. If you have snap turned on, the events move horizontally according to the snap quantization values.

To temporarily remove a event from the snap quantization, hold the "Alt" key when moving.

If you only want to move an event horizontally, avoid accidental vertical movements by simultaneously holding the key "H". The move symbol will turn into a horizontal double arrow.

If you only want to move an event vertically, avoid accidental horizontal movements by simultaneously holding the "Shift" key. The move symbol will turn into a vertical double arrow.

MIDI Editor: Zoom use the scroll wheel

Just like in the project window, you can set the vertical and horizontal picture detail as well as the zoom factor in the MIDI editor using the scroll wheel:

Scroll wheel: Horizontal scrolling

Shift + Scroll wheel: Vertical zoom

Shift + Ctrl + Scroll wheel: Vertical scrolling

Ctrl + Scroll wheel: Horizontal zoom

You can change the scroll wheel settings at any time by going to "File > Program Preferences > "Edit Keyboard Shortcuts and Menu (view page 688)".

Synchronized image view in Arranger and MIDI editor:

If you change the horizontal display using the scroll bars, holding the Shift key changes the corresponding Arranger window.

MIDI Editor: Mouse Mode/Toolbar

There are many different editing tools available for event creation and editing within the MIDI editor. For every tool (except the eraser), editing events functions in the same way as described above. The tools behave differently only when clicking in free

areas. Access the mouse modes/tools in the MIDI editor "Mouse mode" menu and with the following buttons:

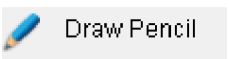
Selection (keyboard shortcut: 1)



Lasso: Hold down the mouse button to draw a selection window.

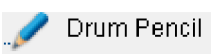
Clicking on a free range highlights an existing selection.

Draw (keyboard shortcut: 2)



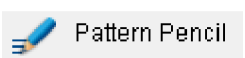
Draw an event by clicking the selected MIDI editor and dragging to the right with the mouse. The event snaps according to the current grid quantization values. If the "Alt" key is held down, the event can be drawn freely (without the snap function).

Drum Pencil (keyboard shortcut: 3)



A sequence of notes can be drawn with this tool. The length of the notes and the note intervals are determined by the current quantization settings. Moving the mouse back (to the left) while holding down the mouse button removes previously drawn notes.

Pattern Pencil (keyboard shortcut: 4)



Select the MIDI events to use for the pattern first, then save this pattern using the keyboard shortcut "Ctrl + P". Now you can draw a sequence of previously selected patterns with the Pattern Pencil. Moving the mouse back (to the left) while holding down the mouse button removes previously drawn notes.

Adjust Velocity (keyboard shortcut: 5)



When this tool is active, drag the mouse vertically and the velocity values of the selected events will be increased or decreased relative to each other. Holding down "Shift" changes all velocity values absolutely, i. e. every changed event has the same velocity value after editing. To recognize the velocity of events by their levels, switch on "Velocity" for a controller slot button in the Controller Editor Range.

Eraser/Delete (keyboard shortcut: 6)



This deletes the MIDI notes by left-clicking. If multiple notes have been selected, they will all be deleted. The Eraser Tool can also be used on specific MIDI notes to "rub them out" and delete them. The eraser is also available for all other tools (except the magnifying glass) by right-clicking.

Magnifying Glass (keyboard shortcut: 7)



Zoom into the horizontal lengthwise display using the left mouse button; use the right mouse button to zoom back out.

The combination of left mouse button + dragging zooms in on the range. The zoom mode can be temporarily accessed again by holding the "Z" key at any time. After releasing the key, the set mouse mode is reactivated.

Glue Notes (keyboard shortcut: 8)



If you click on a MIDI note with this tool, the note will join with the next MIDI note of the same pitch.

Split Note (keyboard shortcut: 9)



This tool separates a note on the matrix into two notes. The separated note snaps to the next snap point.

Mute (keyboard shortcut: M)



In this mode you can mute or reactivate individual notes or selected groups of notes by clicking on them. This function is also available as a command in the MIDI editor menu under "MIDI functions" ("Ctrl + M").

Select Velocity Controller: The "Mouse Pointer" button in the MIDI editor toolbar activates the Combi-Tool (view page 330) which you can use to select notes, change values, draw lines and perform freehand drawing.

Draw Velocity/Controller: This activates freehand drawing (view page 330). In the controller editor the mouse pointer turns into a pencil. Detailed envelopes, ramps, and grades can be created by dragging in the respective controller range.

Draw Velocity/Controller as a line: This activates line drawing (view page 330) In the controller editor the mouse pointer turns into a cross. Create envelopes, ramps, and grades by dragging horizontally.

Tips:

- "Shift to pencil": The "Shift" key is now the hotkey for "Draw" mode. This does not apply to the magnifying glass mouse mode, since "Shift" activates the vertical zoom in this case.
- The keyboard shortcuts to switch the mouse mode/tool can be freely defined in the MIDI editor menu "Shortcuts -> Edit shortcuts".
- Delete Mode can be activated anytime by clicking/dragging with the right mouse button. For instance, use the pencil to insert new notes by left-clicking, or remove already inserted notes by right clicking (without having to change tools).
- Notes created in "Draw" modes contain the MIDI channel and velocity of the editing fields under the toolbar.
- Select the preceding/following notes with the left/right arrow keys. Change the pitch of the selected notes gradually with the up/down arrow keys.

MIDI editor: Editing fields

Editing selected events (editing fields): The properties of every event in the piano roll, drum editor, controller, and listen editor can also be edited with the editing fields under the editing tools. The following buttons are available for each note:

- Channel
- Pitch (byte 1)
- Velocity (byte 2)
- Starting time in bars:beats:ticks
- Length in bars:beats:ticks

The display of the ticks has a resolution of 384 PPQ. 384 ticks correspond to a quarter note.

Hold the left mouse button in the desired field and drag upwards or downwards to increase or decrease the value. The value steps are greater if you press "Ctrl".

This value may also be edited numerically with the keyboard. Double click in the editing field you wish to change and enter the value via the keyboard.

A few special properties result when multi-selecting events:

For the "Pitch" and "Level" parameters, you can change the values relatively by dragging with the mouse, or by entering the values numerically and then confirming with "Enter". You can make absolute changes for all selected events by simultaneously holding "Shift" while dragging with the mouse in the editing field, or by finishing the entry process with "Shift + Enter".

For the "Time" and "Length" parameters, you can change the values relatively by double clicking the desired value field and then confirming with the mouse wheel.

Note: MIDI channel changes are always permanent for multiple selections.

Step Recording with a Keyboard or Controller Keyboard

In the MIDI editor, so-called "Step recording" can also be executed using the computer keyboard or a MIDI keyboard. Activate the corresponding button first.



A marker shows the scope of the current octave for the following entries. Now you can enter MIDI notes step-by-step using the keyboard. Note length and step length can be specified by the length quantization values. The most important keyboard shortcut for entering MIDI notes using step recording:

Tab	One step forwards (set pause)
Shift + Tab	One step backwards
Ctrl + arrow up/arrow down	Entry octave upwards/downwards
CDEFGAB	Note entry in current octave
Shift	Enter chords

Different notes can be entered without the playback marker jumping if you hold down the "Shift" key. This enables chords to be played.

Notes can also be entered in "Step recording" mode by using the controller keyboard.

Cell Edit Mode

Cell

This mode may be opened/closed manually by clicking the "Cell" button beside the horizontal scroll bar. "Cell edit" mode provides an alternative view of the MIDI events:

- Notes are displayed as cells. In this case, the actual note lengths for the individual events are no longer considered. You may specify a unified display width of all events a lot more by setting the snap quantization and length quantization values in the MIDI editor's toolbar.
- The intensity of color of the individual notes increases with velocity strength – for larger velocity values, the event is displayed in dark blue, and events with little velocity are highlighted light blue.
- "Cell edit" mode provides a better overview for display of MIDI percussion instruments (see drum editor (view page 324)), since these often appear as short, explosive events. Display is limited to the most important information, note starting points, and velocity.

Velocity Mode

Vel. If you click the "Vel" button next to the "Cell" button, the view switches to "Velocity" mode.

In this case, the velocity strength is no longer displayed like before (with different color intensity of events). "Velocity" mode keeps the events the same color. The respective velocity may be recognized by the height of the event. The event's velocity may be changed by dragging the mouse to the upper edge of the respective event. In all mouse modes (except for "Delete"), the velocity can be changed by directly clicking without having to switch to the controller editor.

Note: "Edit Shortcuts" in the MIDI editor menu enables the parameter "Event velocity" to be changed in the MIDI editor via a shortcut assigned by you. The functions "Velocity up" and "Velocity down" are available.

MIDI Functions

The commands in the "MIDI functions" menu of the MIDI editor always affect all note events in a particular range or all of the selected events. If no notes have been selected, the functions are applied to all notes.

Legato: Notes can be prolonged and played back in bundles.

Shortcut: Ctrl + L

Quantize Notes (standard): This command conducts standard quantization of the lengths of the MIDI notes of all selected MIDI objects according to the MIDI quantize settings (view page 313).

Shortcut: Ctrl + Q

Advanced Quantization:

Start Q: This command quantizes the start positions of the MIDI notes for all selected MIDI objects according to the MIDI quantize settings. The grid view corresponds to the start quantization value.

Start and Length Q/MIDI Quantization (Start and Length): This command applies quantization to the start and length (view page 314) of the MIDI notes of all selected MIDI objects according to the MIDI quantize settings.

Soft Q (quantize approximation): This command includes the current Soft Q level value in the quantization options (view page 317).

Q Length/Quantize MIDI Length: This command applies quantization to the lengths (view page 314) of the MIDI notes of all selected MIDI objects according to the MIDI quantize settings.

Quantize Note Ends to Grid: This command quantizes the ends of the MIDI notes of all selected MIDI objects according to the MIDI quantize settings.

Undo Quantization: Use this command to reverse all completed quantization steps.

Note: "Undo Quantization" is still possible after a VIP has been saved or loaded. During quantization, the distance to the next grid position saved together with the note and recalculated after "Reverse quantization" is selected. This way the original groove or phase of a note can be restored even after displacement or copying.

Quantization settings: In this dialog you can make detailed settings for the quantization (view page 313).

Quantize/Reduce controller: Controller values can be quantized or reduced with this function.

Humanize: This command factors in the current "Humanize" level value in the quantization options.

Mute Notes (Mute): Mutes and reactivates notes or selected note groups with a click.

Shortcut: Ctrl + M

Remove Overlaps (polyphonic): Notes can be shortened to avoid any overlaps. Chords (simultaneously played notes) are recognized and left uncorrected, i. e. chords are not split up.

Delete Doubles (remove overlaps monophonic): If necessary, notes are shortened to prevent any overlaps.

Transform sustain pedal into note lengths: You can transform sustain values of events into notes lengths and display them in the MIDI editor.

Transposition: Here you can transpose the pitch of selected events by half steps.

MIDI Time Stretching: With this function you can make various changes to selected events, e.g.

- double the tempo
- halve the tempo
- scale to range length.
- stretch manually. This allows you to enter individual values in the Stretch Factor field.

Reverse: With this function you can reverse the playback sequence. This means that the sequence of notes will be played backwards. The result is a mirror image at the vertical axis.

Mirror Melody: This function mirrors the selected events in terms of the pitch of the current note. The effect is that each interval in the note sequence that was directed upwards will now be directed downwards and vice versa. The result is a horizontal mirror image.

If you don't define any particular notes, the mirroring will affect the mid pitch of all notes.

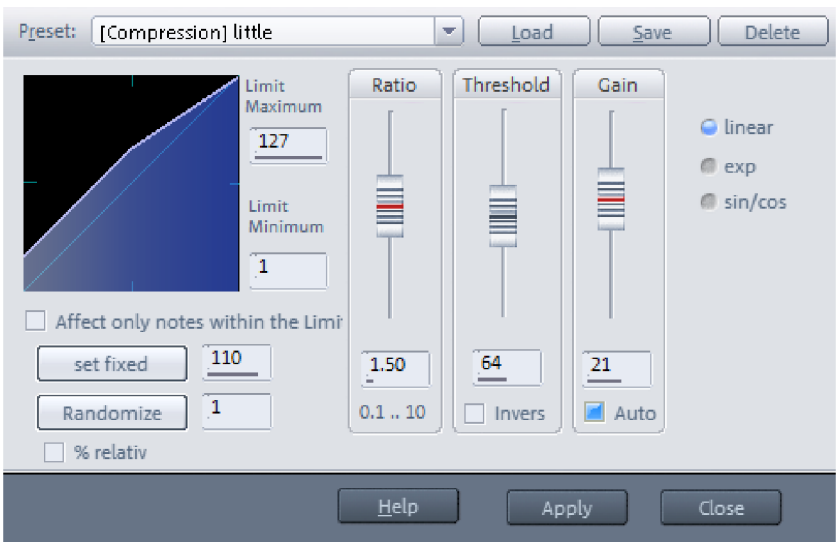
Velocity Dynamics

The MIDI velocity dynamics can be accessed as an offline effect via "MIDI editor MIDI Functions > Velocity dynamics", or as a real-time track effect in the track editor.

"Velocity dynamics" is a MIDI effect that edits the MIDI velocity dynamic of the selected MIDI notes. It can be used in real time as a track effect during playback and for MIDI-Thru (button in the MIDI section of the Track editor), and also offline via the corresponding menus.

Velocity dynamics makes it possible to adjust the MIDI velocity to the dynamics of the connected synthesizer, i.e. to adjust MIDI synths or VSTis, thereby compressing or expanding the velocity strength of the selected events.

Each input velocity value is assigned to a specific output velocity via the characteristic. The characteristic curve may be set linear like a compressor/expander, but other characteristics patterns are also available to produce exponential and sinus/cosine curves.



The Preset field contains a series of presets for different compressor or expander applications; these may be used to form the basis of individual adjustments.

Ratio (0.10 - 10.0): This parameter describes the level of compression/expansion as the MIDI input signal reaches the threshold.

Threshold (0 - 127): Threshold specifies the level at which MIDI events are adjusted via velocity dynamics.

Inverse: Velocity values above the thresholds aren't processed, but rather only those values below the threshold.

Gain (-128 – +128): The gain controller sets the amplification factor for events after they have been processed by other parameters

Auto: If this option is activated, the gain parameter will be adjusted automatically to reach a consistently full overdrive (Velocity 127).

Limit maximum and **Limit minimum** are used to limit the top and bottom velocity ranges by entering a minimal or maximum velocity value.

Offline mode

Offline mode makes the Set value button available. If this is pressed, then the selected MIDI events may be set to the velocity value entered in the field next to it.

Change velocity with the random function

With the Random Variation button you can assign several of the same velocity values to subsequently fluctuate around mean velocity values, e.g. in order to make a programmed sequence sound as if it were played live. In the field next to the button, enter the maximum deviation value allowed. The velocity values will be altered randomly within the range specified.

If **randomized variation should be displayed in relative %**, then this option must be checked accordingly. If the Randomize button is clicked now, the selected MIDI velocity values will change accordingly. At a lower dynamic in the original material, variation will be softer than positions with higher dynamics.

Only for notes in limit zone: A check placed here modifies notes with a velocity within the limit minimum and limit maximum. This enables all notes with a velocity of 100 to be set to 77 or only notes above velocity 100 to work with random variation.

Note: By setting the limit minimum and limit maximum to the same value, a fixed value may be set for the velocity for real-time use.

Quantizing

Irregularities during recording can be evened out with quantization. Sequoia features audio quantization as well as MIDI quantization.

Details about audio quantization are provided in the menu reference "Object" > "Quantize" > "Audio Quantization Wizard" (view page 217).

Recorded MIDI events can be set to the exact note start and note length values using MIDI quantization. Quantization can also be varied with additional parameters like "Soft Q", "Swing", or "Humanize", which adjust quantized MIDI notes to the musical properties of each song.

The quantization functions are featured in the MIDI Editor via the "MIDI functions".

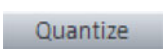
The settings for MIDI start and length quantization can be found in the "Menu Object > Quantize > MIDI Quantize settings" or in the "Grid quantization value" and "Length quantization" fields of the MIDI editor's toolbar. If you enter the value # in the length quantization values input field, the value will be linked to the respective snap quantization value.



The setting "Beats" means that time signatures based on quarter notes serve as a quarter snap unit, and time signatures based on eighth notes serve as an eighth snap unit. The snap therefore follows the time signature and observes any time signature changes.

The grid view in the MIDI Editor corresponds to start quantization value. If you also hold down the "Alt" key, the grid will be disabled temporarily.

Clicking the "Quantize" button at the top right in the MIDI editor quantizes the start positions of all selected MIDI objects according to the MIDI quantize settings (view page 313). If no notes are selected, all notes will be quantized.



Shortcut: Ctrl + Q

Right-clicking this button opens the dialog for the global grid/quantize options (view page 313).

The command "Extended MIDI quantize (start and length Q)" applies quantization to the start and length of the MIDI notes of all selected MIDI objects according to the MIDI quantization options.

"**Soft Q (approximate quantization)**" includes the current soft Q values in the quantization options.

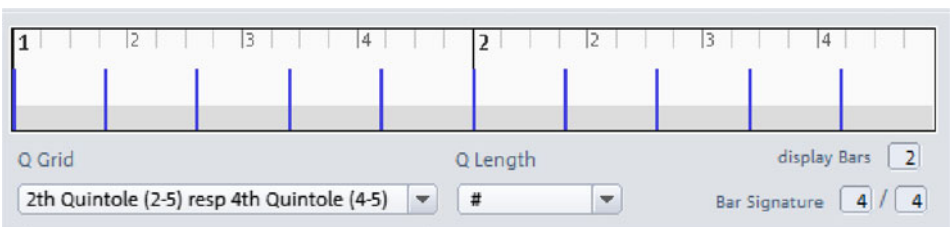
Quantize Note Ends to Grid This option can be used to extend the ends of the selected notes to the next snap point according to the set values.

Reset quantization: This function resets all quantization processes.

Note: Press the Alt key to temporarily deactivate the quantization grid so you can adjust notes freely. The same applies for moving objects in the arranger window as well.

Quantization - window view

The quantization snap grid (Q snap) is displayed graphically, and the Q range appears gray. Only notes or slices within the Q range will be changed.

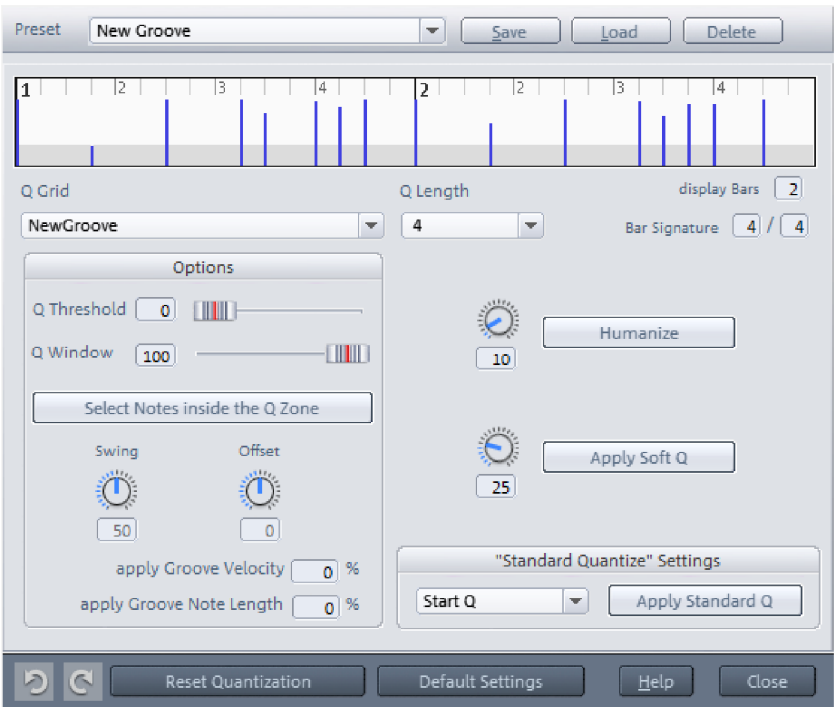


The beat names in the quantization grid depend on the beat signature that is set. The blue lines depict the effective Q snap as well the velocity level (bar height) for groove templates.

Note: The beat signature in the quantization dialog is independent of the tempo map.

Quantization settings

For precise specification of the global grid and quantization options, a separate dialog is available to you in the menu under "Object > Quantize > MIDI Quantize settings":



This dialog can also be accessed in the MIDI editor via "MIDI functions -> extended Quantize > Quantize settings" or by right clicking on the "Quantize" button. This dialog displays the main view depending on the settings in the quantization area and the respective quantization snap points.

Settings applied here also affect the behavior of the "Input Q (view page 69)" button in the track editor.

In this case, MIDI quantize settings changed in the VIP are applied as a preset for new VIPs.

Quantization presets

Here you can select from a number of presets:

- **5-tuplet:** Quantization occurs in fifths
- **Magnetic quantize:** The "Q window" value is set to "50", i. e. only 50% of quantization will be considered. Only events that are within a range of 25% of the grid width left and right of the grid point will be quantized.

- **Swing:** The swing parameter is set to "75", i. e. in contrast to the binary rhythm, which features a "swing" value of "50", inclined/un-highlighted counting times will be set to delay. This results in a swing feeling
- **Triplets:** Quantization occurs in triplets
- **16th offbeat:** The quantization grid's timing is moved back a 16th note
- **8th offbeat:** The quantization grid's timing is moved back an 8th note
- "**New groove**" and "**More life for hi-hat**" provide groove templates

Of course you can create and save your own settings as presets.

Q grid/Q length

Both fields for "Q snap" and "Q length" match the fields in the MIDI editor's toolbar. The settings for grid and length quantization are also located here. The value "#" in the length parameter couples the length quantization value to the respective snap quantization value that has been set.

Tuplets (8th, 5th, 7th)

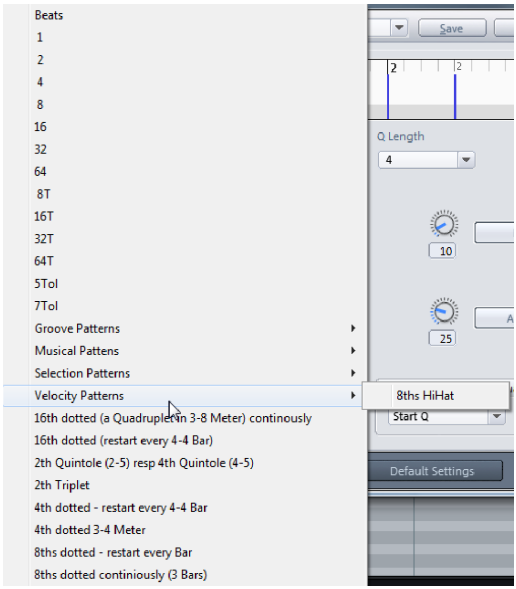
The Q grid also provides n-tuplet quantization values (for thirds, fifths, and sevenths). For example, if 7 is entered, the grid will be adjusted for septuplets. The count will be divided into 7 subsections for quantization. A thirds snap is indicated by a "T" behind the number.

Groove template

You can set a rhythmic pattern on all selected MIDI events or Audio objects with Groove Templates. Groove Templates adjust the selected range of a recording to a specific musical grid. So you can bring static MIDI patterns to life, quantize them according with available audio drum loops or create special metrics like dotted grid values. It is also possible to make a rhythmic pattern for the MIDI pen. As well as the position the velocity and note length can be also be changed with the Groove Template (only for MIDI).

A Groove Template enables a freely-definable grid. As a rule it is from one to four beats long and repeats itself in a loop. The length of the Groove Templates is in principle, however, freely-definable. You can also create a grid based on a track's bass drum, to draw a perfectly in-time bass using the MIDI pen. Length and start of the Groove Templates are continually set to all the beats.

You can select the grid from the Groove Template selection box, which puts binary, triplet and dotted note values at your disposal.



If a you have selected a groove template, the "Snap/Quantization value" field of the MIDI editor will indicate "Grv".

The MIDI editor may be used to create new groove templates yourself by selecting the MIDI events as the template and then accessing "Edit -> Create groove template from selection".

This quantizes the Groove Template to entire bar borders. So if you want to create a Groove Template that is 2 eighths long, choose a 2/8 time signature.

Additional to each note start position both the note length and velocity are saved in the Groove Template. In the file selection box you can give your Groove Template a name. Subsequently the Groove Template is immediately active and appears in the grid quantization list.

Groove velocity (appears only for groove templates): The groove velocity indicates in % how much the velocity information affects the groove template.

Groove note length (appears only for groove templates): The groove note length indicates in % how much the note length information affects the groove template.

Note: If a groove template is selected, then the parameters "Swing" and "Offset" will be deactivated and grayed out.

For audio objects, you can create groove templates by specifying the transients (AQ markers) for the audio object intended to serve as the template via "Object > Quantize > Extended audio quantize" and then accessing the "Create Groove Template from Transients" function. You can also reduce the range in which you want to create Groove Templates through another active range. Then only the AQ marker within the highlighted object range will be taken into account.

If you rename the new Groove Template and put it in the "fx-preset/Grooves" program folder, it will be available in the future in the Groove Template selection box.

Show Grid/Time Signature

Displays the desired beat signature and number of beats. The display window will change accordingly.

Q Threshold

The parameter "Q threshold" may be used to slightly vary quantization by excluding notes from quantization that are very close to the next quantization value.

Q Window

"Q window" refers to the interval to the left and right of a grid point; events will be quantized within this range. No quantization will take place beyond this, and for this reason, the events outside of this window will remain at their position. The quantization area is dependent on the values of the parameters "Q snap" and "Q threshold".

Example: Grid: 4 max. Window: 4

- 100: The Q range covers the entire area between the grid points on the quantization grid. All events will be quantized
- 50: The Q range covers half of the quantization interval. Events with gaps of ¼ of the grid width left and right of the grid point (1/16 note values in this example) are quantized
- 0: no Q range -> quantization off

Select notes within the Q range

This button activates a display of the selected events that will be quantized for the currently set Q window size of the quantization range (Q range). The affected events are displayed with a red outline. Click in an empty area in the MIDI editor to deselect.

The smaller the window size for the quantization area is, the fewer events will be included in quantization. The Q range and the graphical velocity display function help, e.g. to assign a lower velocity to all offbeat notes.

Swing

Set swinging, ternary playing with this value. This enables you to enter the division for an uneven/unaccented grid points.

- 50: "50-50/1:1" split. The unaccented eighths are exactly half way between the even eighth notes ("even", binary playing method)
- 67: "67-33/2:1" tuplets. The beat is split up into three counts, whereby the note is assigned 2 beats (67%) and the off beat note one count (33%)
- 75: "75-25/3:1" split. For example, a pointed eighth and a sixteenth is created from two eighth notes

Offset

The value range in this parameter stretches from -100 to +100. By changing the offset values, you move the whole quantization grid. Select a negative value for the offset, and the quantization grid will be moved by the corresponding value to the left, or forward in time. On the other hand, if you select a positive offset value, you will set the quantization grid by the corresponding value to the right, or backwards in time.

A value of -100 corresponds to a shift of half a grid length to the left; +100 corresponds to a shift of a half a grid length to the right.

Humanize

The "Humanize" parameter creates another variation option, i.e. notes are able to be assigned according to the randomization principle up to a specific interval to positions around the exact quantization value. The setting occurs in % of a 16th note. The value specified therefore determines the possible interval between the quantized notes and the exact quantization value.

Soft Q

This value sets the strength or "Soft Q" value of the quantization.

- "100" moves the event precisely to the quantize grid point,
- "50" shifts the event to the middle between the current position and the quantization grid point,
- "0" means no movement, i.e. quantization off

"Soft Q (quantize approximation)" (in the arranger menu "Object > Quantize > Extended") considers the current "Soft Q" value in the quantization options in contrast to "Note" and "Start" quantization.

In contrast, quantization commands "Quantize notes (start and length)" and "Start quantization" always proceed with 100% strength. This corresponds with a Soft Q setting of 100.

Note: Use the keyboard shortcut for the approximation of soft quantization and hard quantization. In this manner, you can always select between approximation soft quantization and harder quantization without having to switch quantization options every time.

Standard Quantization

Here you can choose what type of quantization is used as standard after pressing the "Quantize" button. You have the choice between:

Start Q: start quantization

Start and Length Q: start and length quantization

Length Q: length quantization

Soft Q: start quantize approximation

Apply standard Q: applies quantization according to the selection.

Note: If you give the "Soft Q" parameter a value other than 100, please take care that you also change the setting to the soft quantization mode. If the corresponding quantization settings should be considered, you need to set the "Soft Q start" command as the standard quantization. In contrast, quantization commands "Q start" and "Q start + length" always proceed with 100% strength (Soft Q).

Undo quantization/reset

With the help of the round arrow buttons, you can undo/redo the last quantization made. In this case, the counter-clockwise round arrow provides the "Undo quantization" function, while the clockwise arrow serves as the "Redo quantization" function.

Reset quantization: This function resets all quantization processes.

The "**Standard settings**" button provides the option of resetting the preset values:

Q threshold: 0

Q window: 100

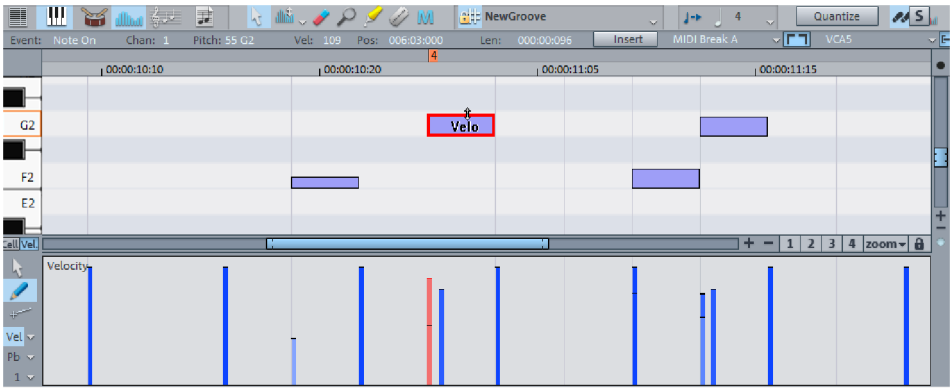
Swing: 50

Offset: 0

Humanize: 10

Soft Q: 25

Matrix Editor (Piano Roll Editor)



For all MIDI objects that show no entry in the "Drum Map" field, you can open the Matrix Editor with a double click. The button is selected in the MIDI editor if the Matrix Editor is active.



The Matrix Editor is easy to recognize via the piano icon on the left side of the editor. The MIDI events displayed are arranged according to their respective pitch.

Note display

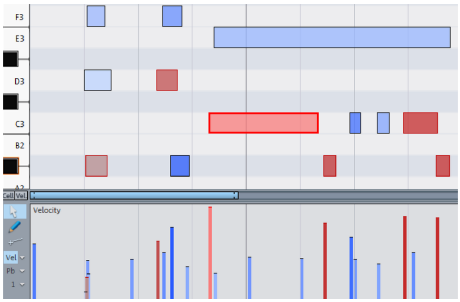
Notes which are not selected within the editor are displayed in blue. The intensity of the colour represents the velocity. The velocity increases while getting darker/stronger in color.

Selected notes are displayed in red. Here too, a more intense color symbolizes increased velocity

Current event: Selected events appear in bright red with a red border. The properties of the currently selected events are displayed in the edit fields above the Matrix Editor If an event is selected with the mouse, it turns into the current event.

Use Velocity colors (MIDI editor Options menu)

With Velocity Colors the velocity of events is displayed by different colors. The higher the velocity, the lighter the color.

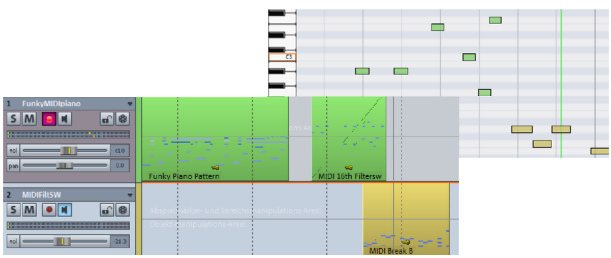


Velocity colors: Unselected note (blue), selected notes (red) and current events (red with border) in the Matrix Editor

You can also determine the colors of tracks or MIDI channels via the "Options" menu in the MIDI editor. If no previous track colors have been set, random colors will be assigned for track color display in the MIDI Editor.

Use Track Colors

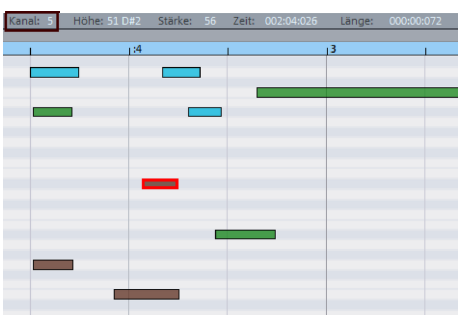
With Track colors the color of the events is set to the color of the corresponding tracks the MIDI objects are located in.



Track color display for MIDI events

Use MIDI channel colors

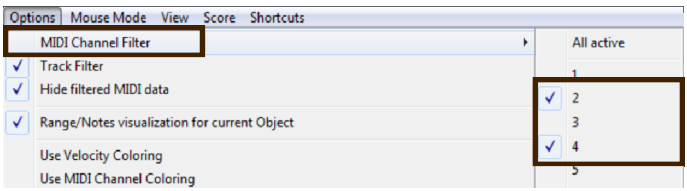
In the MIDI channel color display MIDI events are displayed in different colors, based on the MIDI channel settings.



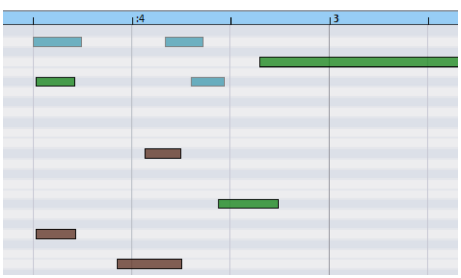
MIDI channel color display

MIDI Channel Filter: To get a better overview of the events of a MIDI object, you can filter specific events for the display.

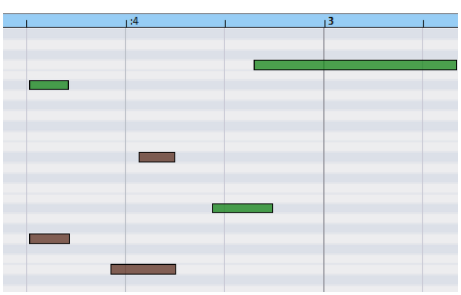
Example: The MIDI object features notes in the MIDI channels 1, 2, and 4. By selecting both MIDI channels in the MIDI editor menu "**Options -> MIDI channel filter**", you may make all notes in channels 2 and 4 available to editing tools.



All unselected notes in channel 1 will be filtered and will be displayed in gray/pale in the Matrix Editor and in the Event list.

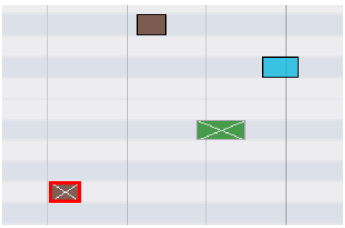


Events in the filtered channels may be hidden completely using the "Hide filtered MIDI data" in the "Options" menu.

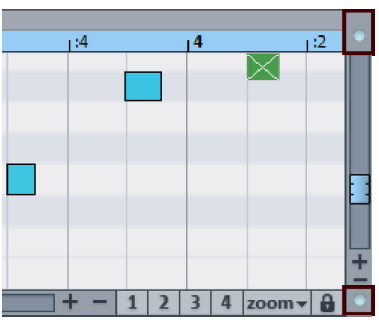


The Event list (view page 333) provides additional display filters that only function within the list.

Display of muted events: MIDI object settings (view page 295) (shortcut: "Ctrl + O") may be used to mute notes and filter other MIDI events. Muted events appear paler in the Matrix Editor and Event list.



Events above and below the current picture section: Two small red rectangles above and below the vertical scroll bar to the right-hand side of the MIDI editor screen show in red if there are notes outside the screen's display.



Matrix Editor: Special selection options

In order to directly select all notes of a certain pitch, double-click on a free section with this pitch in the Matrix Editor or in the keyboard. Hold down the "Alt" key to select notes with this pitch from this click position only.

Using the "Shift + double click" key combination on a free area, you can create a new event and simultaneously select all notes of this pitch.

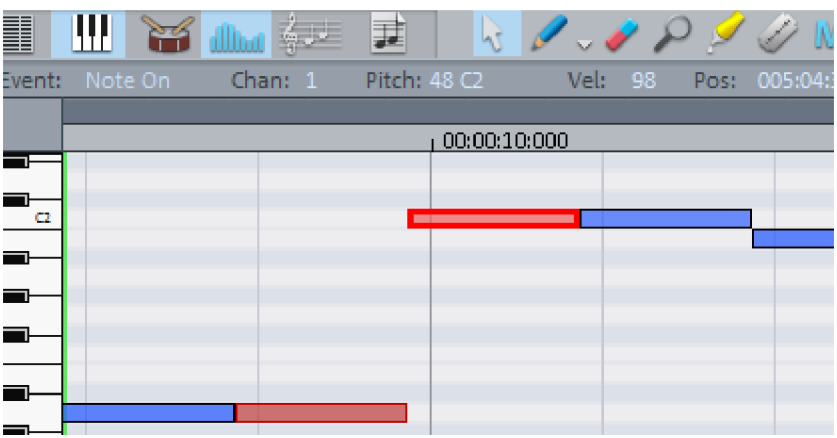
Note: If you additionally press "Ctrl" during any of these selection methods, the new selection is added to the current one (i. e. the previous selection is not canceled).

Matrix Editor: Edit events

If you move the mouse over an event the pointer will change depending on which event bar area the mouse is in. The following options are available:

- Change event start time: drag the event's front edge. The note end remains intact.
- Change event length: drag the event's back edge. The note start remains intact.
- Set fixed event length: for multi-selection, hold "Shift" and drag the currently referenced event longer or shorter. This function makes all event equal length.
- Scale event length relative: for multi-selection, hold "Ctrl" and drag the currently referenced event longer or shorter. The lengths of the other selected events change.
- Move events horizontally: move the mouse over an event whilst holding the "H" key. The mouse pointer will turn into a horizontal double arrow. Move the selected events horizontally in the timeline. The step size is defined by the set grid value.
- Move events vertically: move the mouse over an event whilst holding the "H" key. The mouse pointer will turn into a vertical double arrow. Move the selected events vertically in pitch.
- You can temporarily remove the snap and freely drag the event by holding "Alt".

Relative snap for moving operations ("Options -> Relative movement in grid"): If this option is active, the interval between events to the next snap position remains for event moving. This makes preferred implementation of instrument groups easier to rearrange.

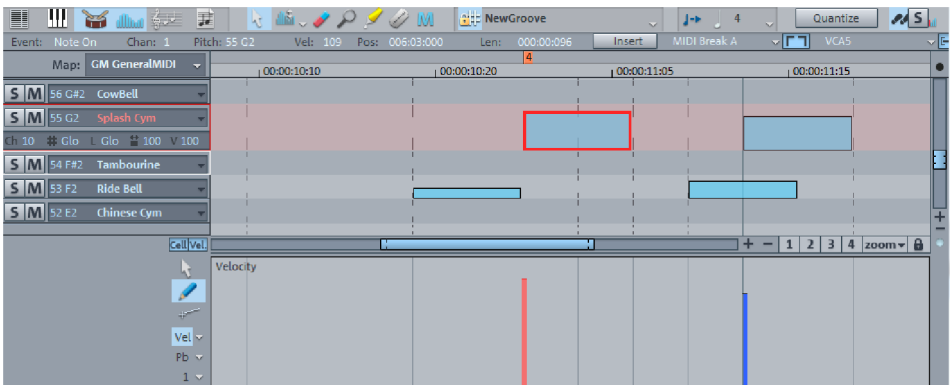


Drum Editor

After opening the MIDI editor (double click on the MIDI object), switch to the corresponding field



in the Drum Editor. Instead of the piano keys, a list of drum instruments is displayed.



Note: If a drum map is active in the arranger track (recognizable by the word "map" in the MIDI area of the track editor (view page 69)), the drum editor is loaded automatically when the drum editor is loaded.

Drum editor track box: In the individual track boxes of the drum editor, you can individually adjust MIDI channel (K), grid (#), quantization length (L), display width in cell mode (< >), and the velocity scaling in percent (V) for each drum instrument.



Display width in cell mode: You can determine the display width of the notes of the current snap in percent with this value. The setting has no effect on playback. At "100" the displayed note completely fills the cell.

Velocity scaling: The velocity value of each note is multiplied by the V value/100 and adjusted to the MIDI velocity between 1 and 127. Scaling is audible, but is not visualized further.

When the drum editor opens cell edit mode (view page 305) activates by default. Here, moreover you can turn Velocity Mode (view page 306) on.

Mouse modes

The following modes are available for editing events, just like in the piano roll:

Selection mode (shortcut: 1): Use this mode to select drum events, move, or change the length of one or several events simultaneously.

Draw tool (shortcut: 2): This mode enables you to draw in drum events manually. If the grid is active, the events will be quantized immediately.

Drum tool (shortcut: 3): This mode also includes a draw function. However, the quantization length is also included while the events are drawn.

Pattern tool (shortcut: 4): This mode allows you to draw entire drum patterns (or melody patterns). If you want to create a new pattern, you have to select it first in "Selection" mode and press "Ctrl + N" simultaneously (or go to menu "Edit" in the "MIDI/drum editor -> Create pattern from selection"). If you have created a pattern, you can start drawing at any position. The lowest note in the pattern is then the pitch you are drawing.

Velocity tool (shortcut: 5): This mode allows you to mark events and change the velocity values of all selected events in relation to each other. Absolute values are entered when you hold the "Shift" key, i.e. all changed events receive the same velocity value.

Eraser (shortcut: 6): This tool deletes events with a single mouse click.

Zoom mode/magnifying glass (shortcut: 7): Draw a rectangle. The tool will zoom into the rectangle. The left mouse button is used to zoom in, the right one is used to zoom out.

Velocity



If this button is active, the bar height is used when displaying the velocity value of the note.

In all mouse modes (except for "Delete"), the velocity may be changed by clicking directly on a note without have to switch to the controller editor.

Drum Maps

A drum map assigns specific properties to the various parts of a MIDI drum kit. The key assignment for the pitch, the output note, the MIDI channel and the quantization are all set here. The "General MIDI" map is used by default to distribute drum kits.

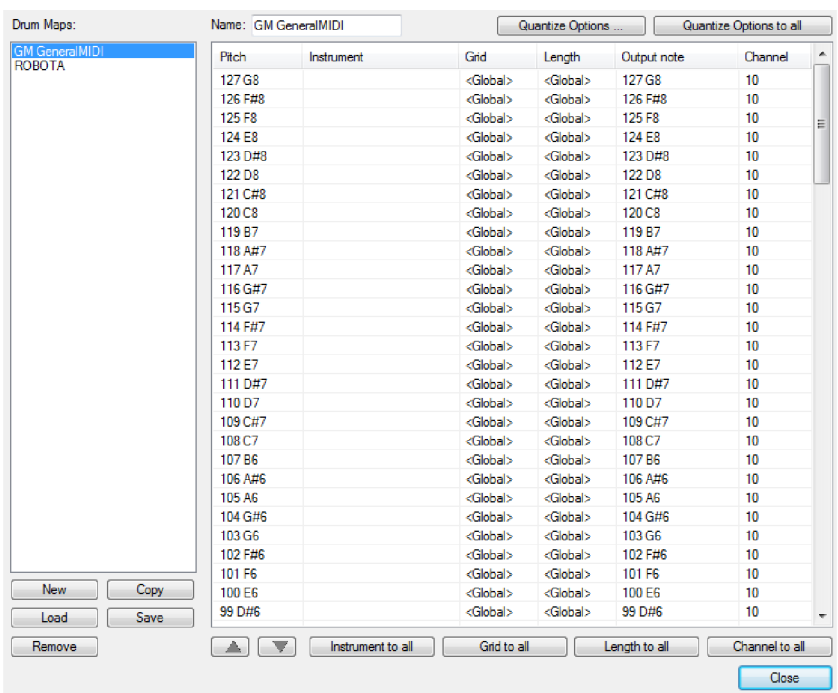
You can set the drum map in the drum editor by selecting the "map" field in the desired drum map. You can also open the drum map in the track editor under "MIDI".



It may be the case that your synthesizer, VSTi or drum machine uses a different mapping setup. If this is the case, you will hear a different sound than you expect when playing the drum events (e. g. a tom drum instead of a bass drum). Here we recommend creating a drum map that corresponds to the current playback device. To do this, select the command "Create new Drum Map" and assign the individual instruments to the corresponding pitch or key assignments of your MIDI keyboard. You can also specify the quantization settings and the MIDI channel. When the new drum map is saved it will appear in the selection menu.

Drum Map Editor

This editor enables you to route each played note to a different one and give them their own individual names. It's also possible to assign each individual instrument its own quantization and a new (MIDI) output channel.



Pitch: This parameter indicates the incoming MIDI note. The value cannot be changed, so the pitch always corresponds to the input note.

Instrument: Name each corresponding percussion instrument here.

Grid: Specify a grid for the start point of the drum events individually for each instrument. Leave the value set to "Global", and the globally set quantization value in the toolbar will be adopted.

Length: Specify a grid for each instrument for the length display of drum events. Leave the value set to "Global", and the globally set quantization value in the toolbar will be adopted.

Output note: Specify the note value each drum instrument or the incoming MIDI note in the "pitch" field" will be "mapped".

Channel: Specify a defined MIDI channel for each instrument. The value set here disables the channel setting in the MIDI track.

Note: In addition, you can set detailed quantization settings for each instrument in the drum map editor using the "Quantization options" button.

Quantization in the Drum Editor

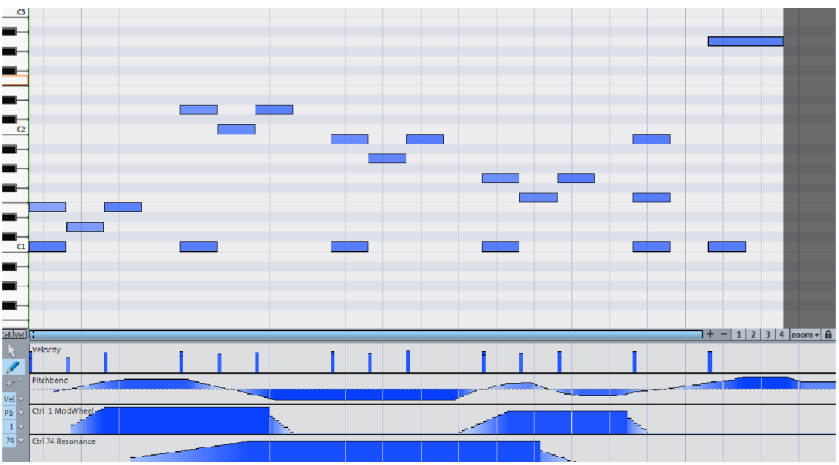
You may also assign individual quantization values to drum instruments. To make the necessary settings, select the arrow next to the track name of the instrument you want to edit and open the quantization options (view page 310).

Controller Editor

The controller editor is located below the matrix or the drum editor; this may be hidden or shown via the keyboard shortcut "Alt + V" or the button for the same function.

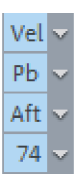


Pull the dividing line beneath the horizontal scroll bar up to enlarge display of the controller editor. The controller editor is used to display and edit up to 4 different MIDI controller curves simultaneously.



Note: To display more than one controller button, drag the dividing line between piano roll and controller editor up (the mouse pointer appears as a double arrow under the scroll bar for the piano roll). If you now click on one of the controller buttons, the corresponding controller line will be refreshed.

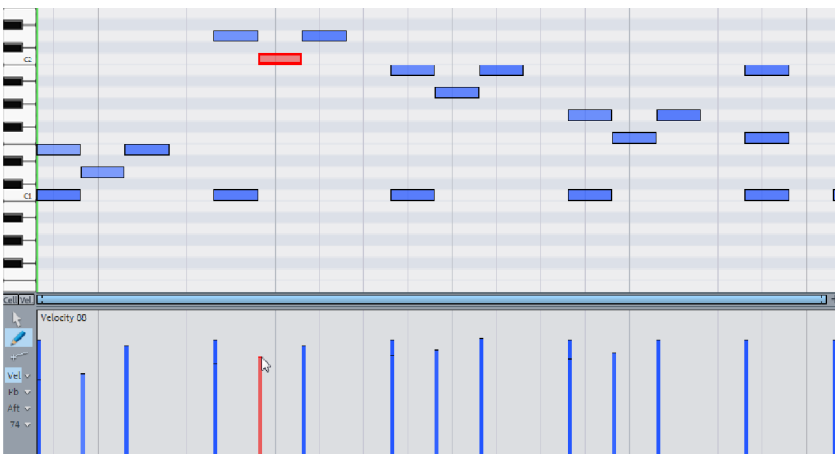
If you click on the arrow on the controller slot to select the displayed controllers, the following types can be selected:



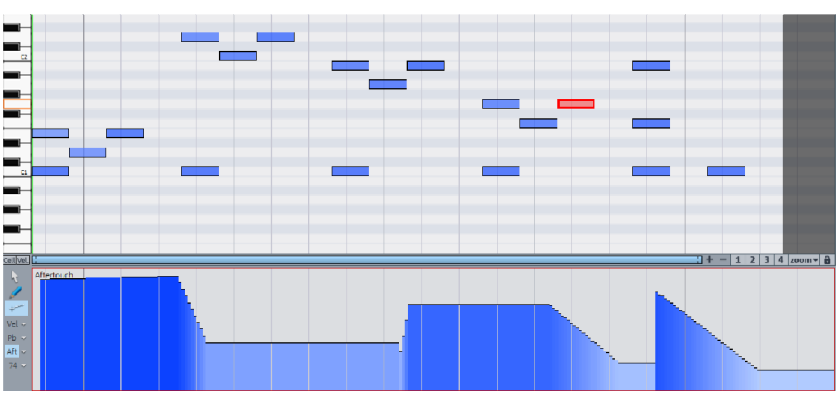
- **Velocity**
- **Pitch bend** (bend the note pitch, just like with the pitch bend wheel of a keyboard)

- **Aftertouch** (pressure on the key after the first strike to control additional parameters, specified via the MIDI playback device)
- **Program change** (switch program for assigning preset sounds to the corresponding MIDI device)
- with a continuous **controller** numbered 0-127. The controller selection features the controller types available to the respective object; a star behind the name of the controller type indicates that a controller curve already exists for this controller type.

The velocity values are displayed directly in the controller editor as vertical bars under the corresponding notes. The height of each bar represents the corresponding note velocity strength. As velocity increased, the color intensity of the bars also increases. Selected events appear red in the controller editor.



The values of all other controllers will appear in the controller editor as ramps. In this case, the height of the ramps and their color intensity also represent the last defined value of an event. The length of the ramps reaches to the next varying event. Selected event ramps also appear red.



Controller Editor: Selecting, editing and drawing events

Combi tool:



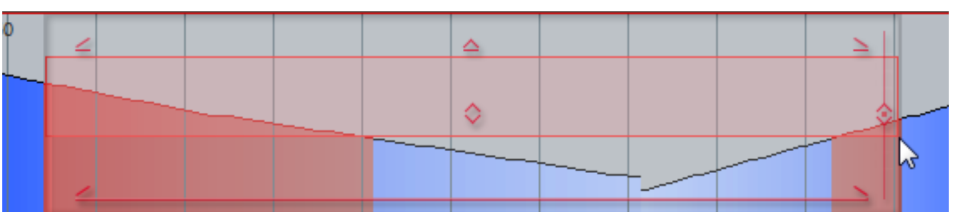
Keyboard shortcut: Ctrl + 1

The combi tool enables you to select events, edit values for the selected events and draw straight lines or freestyle curves.

Selecting and editing events

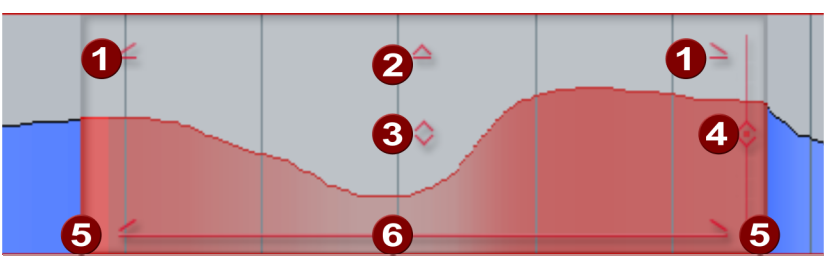
Individual events can be selected by clicking the respective bar. To modify the controller value, click and drag on the bar end.

To select multiple events click in the controller editor and drag to create a selection (lasso). The reference for the selection is the top edge of a bar. Since lasso selection is two-dimensional, it is only possible to select events in a specific value range.



Additional events can be added to the selection by holding down the Ctrl key.

The selected events can be grouped for editing using handles at the selection frame:



1. **Fade in and out:** Drag the handles to fade controller values in and out.
2. **Scale values:** Drag to handles to increase or decrease values in relation to each other.
3. **Increase/decrease values:** Drag the handles to increase or decrease all values to the same value.

4. **Increase/align differences in value:** Click a point on the vertical line and the differences in value on either side of the point will be increased or decreased. If you drag the handles right down to the bottom, this creates a straight line at the height of the clicked value and the "amplitude" of an existing curve will be increased in the other direction.
5. **Stretch/compress duration**: Drag at the edges of the selection for temporal stretching or compression of the curve.
6. **Move selected:** Drag down at the line to move the selected curve.

To delete selected events in the controller editor, press the "Del" key.

Other editing options with the combi tool:

Alt + drag (with the pencil): This activates freehand drawing, and the mouse pointer becomes the pencil. Detailed envelopes, ramps, and grades can be created by dragging within the respective controller range.

Shift + drag (with the crosshairs): This activates line drawing and the mouse pointer changes to crosshair. Create envelopes, ramps and grades, or set all of the events to the same value by dragging horizontally.

Draw freehand



Shortcut key: Ctrl + 2

This activates freehand drawing; the mouse pointer becomes a pencil. Detailed envelopes, ramps, and grades can be created by dragging within the respective controller range. Correct curves during drawing by dragging backwards. Any existing (multi) selection is ignored. Create crescendo or decrescendo using velocity curves, for example.

Single-clicking creates a new ramp which reaches up to the next controller event.

Shift + Drag with the crosshair to draw a line.

Note: When editing the velocity, no new notes are generated; existing velocity values are only modified by dragging and clicking the events.

Draw lines



Shortcut key: Ctrl + 3

This activates line drawing; the mouse pointer changes to crosshair. Create envelopes, ramps, and grades by dragging horizontally.

Single-clicking creates a new ramp which reaches up to the next controller event.

Note: When editing the velocity, no new notes are generated; existing velocity values are only modified by dragging and clicking the events.

Controller Editor Tips

Vertical zoom

To make it easier to edit very small curves (e.g. pitch bend), you can use Ctrl+mouse wheel to zoom vertically into the controller curve.

Copying controller events

You can copy selected events using "Ctrl + C" and then paste them at any play cursor position in the corresponding controller editor using "Ctrl + V". This also works from one controller to another. With Ctrl + A you can select all events of a controller.

Display filter function for velocity values

At points in the arrangement where there is polyphony the velocity bars for the notes are positioned on top of each other and it can be difficult to select the right note. To edit only notes of a certain pitch, e.g. all C1 notes in the controller editor, click the corresponding key on the keyboard. The key and the background of the selected pitch are highlighted. Now only notes of this pitch are displayed in the controller editor. Repeated clicking of the same key removes the selection.

You can also select several pitches for the velocity visualization filter by clicking on the desired buttons while holding down cmd or Shift.

A further possibility for selectively editing velocity bars located on top of each other is based on the fact that only the bar of the currently selected event can be edited using the mouse. First click on an event in the matrix editor or controller editor. Now navigate to the desired note with the "Arrow left/Arrow right" buttons and change the controller value by clicking the top third of the selected bar.

Quantize Controller Events

MIDI controller events can be quantized and thinned out. To do this select the "MIDI functions" menu and then the "Quantize/thin out controller" command. Quantization occurs according to the quantization settings.

Detailed information about quantization is provided in "MIDI editors -> Quantization".

Smooth controller events

It is also possible to smooth controller event curves. To do this, select "Smooth controller" from the "MIDI Functions" menu. This adds interpolated intermediate values between existing controller values, which creates a smoother curve gradient and avoids jumps between values.

Event list

Open event list: The MIDI editor has an integrated display of all events in list format. This List Editor can be opened with the button above the keyboard or by using the “Alt + L” keyboard shortcut.



When the event list is opened and activated for editing, it has a narrow red border. This is to make clear that certain functions, e.g. select next/previous event (arrow keys) or the “Select all” (Ctrl + A) command, refer only to the List.

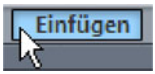
Working with the list editor: The list editor does not only display note events, but also percussion instruments (in drum editor mode), MIDI controllers, and SysEx messages. These controllers and messages may be hidden and even filtered by checking the appropriate "mute" box.



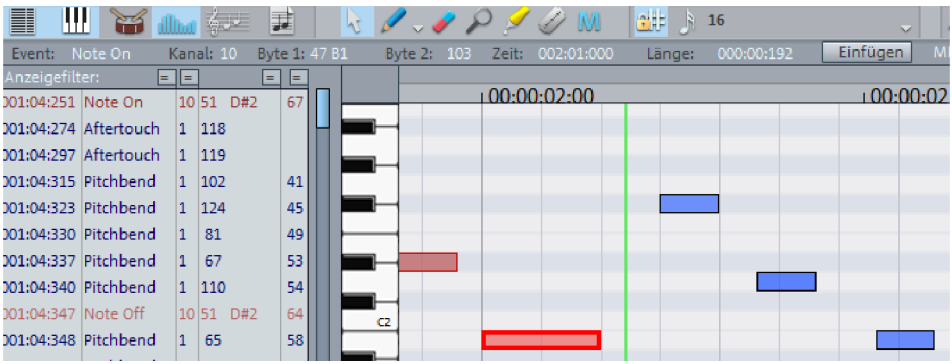
Double clicking on a SysEx entry in the list opens a simple editor for viewing and editing the SysEx information.

A note on and a note off event (or note on with velocity 0) belong to a note. These are always selected and edited in pairs. You can show and hide note off events by ticking the corresponding checkbox beneath the Editor.

You can create new events in event list directly at the playback marker position using the "Insert" button.

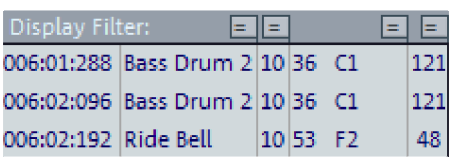


The values of the previously selected notes for the edit fields channel, byte1 (Pitch), byte2 (velocity), time and length are used at that.

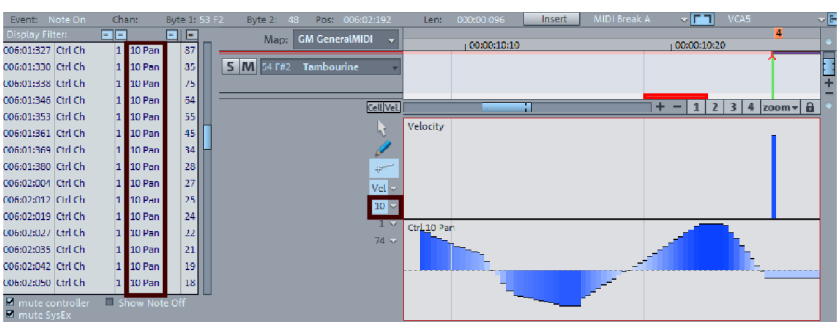


If you would like to edit only certain events specifically, the event list offers various display filters. These are small check boxes above the event list columns. If you click a box, it will change to blue and the respective display filter is active. The display filter provides information for the editing fields MIDI event, MIDI channel, pitch and velocity.

Example: Select a note and click a display filter of the pitch column to only display events of this pitch. All other events will be faded out:



Display filters can be combined with one another. For example, all control change events of type 10 (pan) to MIDI channel 1 may be filtered for display. Then, you can select all matches will "Select All" (Ctrl + A) and edit together in the editing field or delete with the "Del" key:

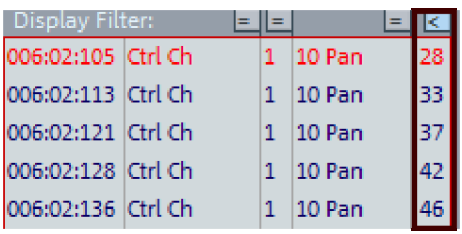


Advanced filter functions in the list editor

You can access the following filter functions by right clicking on the display filter buttons:

=	equal
!=	unequal
>	bigger or equal
<	smaller or equal

The filter functions listed above relate to the selected event kinds like "Note On" or "Ctrl Ch". For example, this function filters out all events with a value of 40 or lower:



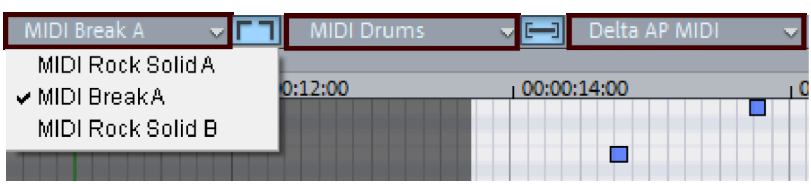
Multi object editing (MO-Editing)

You can also load several MIDI objects simultaneously in the MIDI editor. All MIDI objects selected in the arranger are added to the MIDI editor by clicking the "MIDI editor" button in the Docker (view page 177).



In an already-opened MIDI Editor you can include additional MIDI objects for Multi Object editing by clicking in the Arranger while holding down the Shift key.

The MIDI editor (and in multi object editing) always shows the current MIDI object and the current arranger track from which this object originates. In the field to the right you'll see the MIDI out of your sound card or the activated VST plug-in for this track.



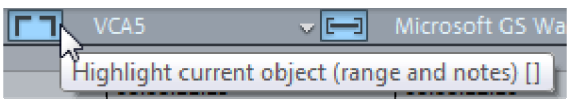
New MIDI events will always be inserted into the current object (i.e. the currently displayed object). The current object is also highlighted in the score editor.

If you click on the arrow next to the object/ track display, you will see all MIDI objects/MIDI tracks in MO Editing mode listed beneath one another. Here the current object or current track has a tick.

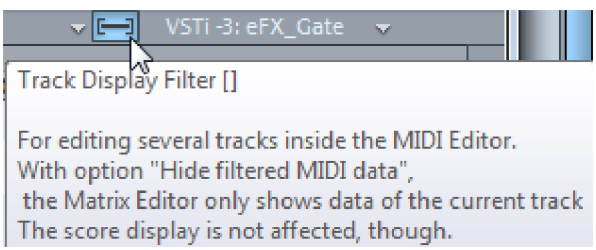
When notes or MIDI events are selected, the associated object becomes the active object. The current Note line will be automatically converted in the score editor.

You can also copy and insert MIDI notes between multiple objects.

The current object range can be highlighted by clicking the button behind the object selection field. Areas outside the current object will be grayed out and the notes of other objects will be paler but still selectable.



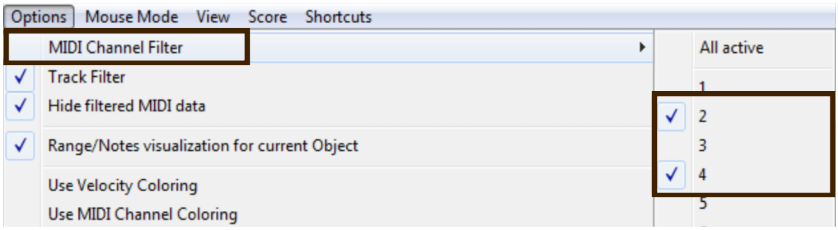
In addition, a display filter for the current track can be activated while in MO editing mode. Events on other tracks will then be grayed out. This also reduces the display in the controller editor to the velocity values for the current track. If the option "Hide filtered events" in the "Options" menu is also activated, MIDI events that are not current will be made invisible.



Note: When representing notes, the note system of the current track will be highlighted using blue note lines.

Filter by MIDI channel

With the "Options menu" > "MIDI Channel Filter" in the MIDI Editor you can select the MIDI channels that should be displayed.



Th disable the MIDI Channel Filter, just select "All active".

Explanations about the colorig of MIDI events can be found in "Matrix Editor“ > "Note display (view page 319)“.

Score editor

The score editor displays the MIDI data of a MIDI object as notes in real time and also offers all MIDI editing options in the note display. If you move or extend MIDI data, these changes are immediately reflected in the score. If you insert a new note in the score editor, a corresponding "MIDI note on" event is created immediately.

Each track may contain a maximum of 48 staves. During multi-object editing across multiple tracks, the scores from the systems of each track are compiled together. The staves of a track may be used as an instrument or instrument group within a score. The entire score may be revealed via the multi-object editing feature and by showing all MIDI tracks in the score editor simultaneously. A vocal excerpt can be obtained by showing the track of the desired instrument only or an instrument group in the score editor.

Open Score Editor

The score editor is integrated into the MIDI editor window. If the MIDI editor is opened, you can activate the linear score view by pressing the "Score editor (linear)" button.

 Score editor (linear) button

Score Editor Modes

Sequoia offers two alternatives for viewing the score: linear display and page mode.

The linear note display may be combined with the matrix display. This offers ideal MIDI editing options, since the detail in depth of the matrix editor and overview may be supplemented by several score systems. You can select the score in the score sheet and perform fine adjustments in the piano roll. Selection and zooms in all editor views (linear score, piano roll, event list, velocity editor) are always synchronized.

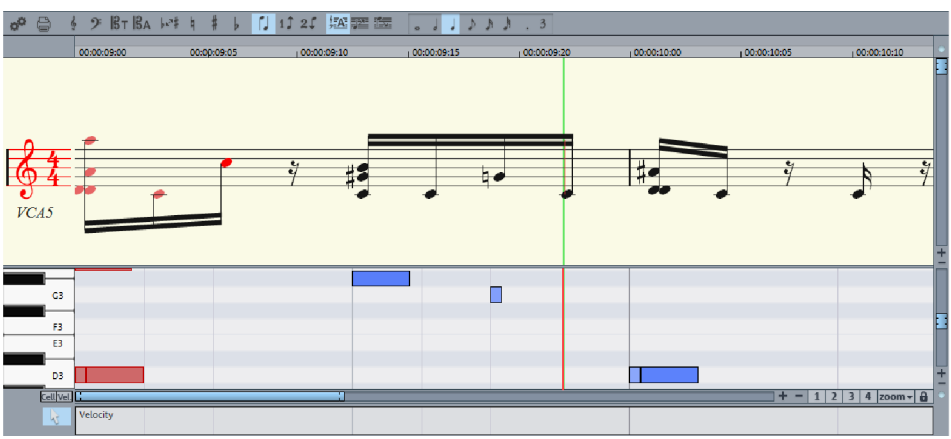
Note editor modes - Linear view



Score editor (linear)

In the linear view, not only the Score editor, but also the matrix and velocity display are available. For detailed editing of the MIDI data in the score, select the linear view. All parameters such as pitch, note length, and velocity are displayed.

Note: Only the notes of the currently active system are displayed in the velocity editor if notes and velocity are parallel. The red bar mark indicates the currently active system.



The linear height of the range of the note view may also be modified. To do so move the line between Score and Matrix editor with the mouse.

The horizontal section can be set above the scroll bar, parallel to the matrix view of the MIDI notes.

In "Linear" mode, details may no longer be displayed correctly if you zoom out horizontally, since the note symbols overlap. It may be necessary to reduce the

notation symbol size with the "-" button on the right side. Despite these details being displayed incorrectly, zooming out can still provide you with a good overview, particularly for large scores. Sequoia can adjust the notation symbol sizes automatically at the selected section. To do this, please select "Automatic zoom" in the "Score" menu.

Score editor modes - page view

The side mode displays the notes just like on a musical score, which also serves as a print preview. This enables you to arrange or delete notes or voices. The design of the score sheet can be used for editing longer passages. The stave break allows you to display considerably more bars than with the linear display.



Score editor - Page view

Scrolling in full view mode: Use the vertical scroll bar to scroll from page to page.

Page mode as print preview: You should first change the page format (in the Score Properties dialog) to the paper format of your printer before setting up the page for printing as the display depends on the page format. You will then see your score exactly as it will be printed.



Score sheet

The automatic conversion of MIDI data into score material is usually sufficient for editing MIDI data; in this case, efficiency is required instead of a perfect, ready-to-print display. However, it may still be necessary to adjust some of the display options for the musical material during editing. These display options are provided in the "MIDI score properties" dialog, which you can open by clicking the corresponding button.



Score settings: Open the score settings dialog for notation system and page properties

Presets are also located here. These are preprogrammed standard settings for certain instruments or instrumentation such as strings, piano, or orchestra. Selecting a preset changes the score display and its legibility:



Before...



...and after the selection of the "Piano" preset

Further explanations of the notation system (view page 346) are provided below.

Editing MIDI data in the score sheet

Select notes

As usual, you can select notes by clicking them. A group of notes (a chord) may be selected by dragging a frame over the notes with the mouse button. Select multiple single notes by pressing the "Ctrl" key.

Note parameters

Parameter pitch, velocity, and length can be changed for one or several selected notes. If one or several notes are selected, the data for the current note will appear in the info bar above the score view. Changing a parameter may have a comparable effect on all selected notes, just like in the matrix editor.

Note: In "Page" mode, notes may not be drawn with the pencil or moved or copied with the mouse. Use the corresponding menu commands and keyboard shortcuts. This limitation does not apply in linear display mode.

Move and transpose

To move notes, first select them, and then drag them to the desired position. The infobox can help you keep track of the pitch or position. The step size when moving is determined by the selected quantization raster in the MIDI editor.

Copy

Select the desired notes and copy them by holding down the "Ctrl" key and dragging the mouse. Alternatively, you can also use the copy function in the "Edit" menu.

Insert new notes

You can also insert new notes in the score editor with the pen. Click the desired position with the pen, hold down the mouse key and correct the position and pitch as required. If you let go of the mouse key, Sequoia will add a new note featuring the same length as the selected length quantization value.

You may only insert new notes into the active stave. For instance, to insert a note in the lower part of a piano system, click the lower staff in the system on the left first. Only notes that correspond to the current pitch are inserted. Non-scaled notes or chromatic intermediate steps are skipped. If you enter conventional music material with the mouse, diatonic insert mode increases the chances of hitting the correct note. If a new prefix is going to be added to the note, you can move the note chromatically with the arrow keys. This way, an inserted F in C Major may be transformed into an F# by pressing the "Page up" key.

Delete notes

Delete notes by

- selecting them and pressing the "Del" key or
- clicking the eraser tool or
- by clicking them with the right mouse button

The menu command "Score -> Selected notes: hide/show in score" enables the selected notes to be removed from the score, without influencing MIDI playback. Notes hidden in the scale are indicated in the matrix by a diagonal line. This function is practical for making trills easier to manipulated, or in order to to remove "control notes" from the score when switching between playing styles.

Insert notation symbols

Clef symbols may be inserted at the current cursor position by pressing the corresponding clef symbol button in the active system.

Delete notation symbol

Manually added notation symbols such as clefs and signatures may not be selected, since these are meta information for the notation display and no MIDI events have been assigned to them. Nevertheless, these may be deleted by clicking them with the eraser (or the right mouse button).

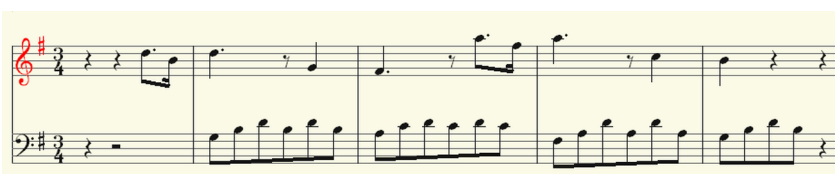
Adjusting and optimizing the score

Sequoia automatically generates a notation display from the MIDI events contained in the MIDI object. This is always correct with regard to pitch and position. However, this does not mean that the notation can be optimally read, since displaying note lengths also plays an important role in this context. In this case, the notation allows flexible interpretation, so the user usually has to intervene. This is a negative example of a poorly readable translation of a piano passage in notation:



Automatic notation with unsuitable display settings...

This representation may be correct; however, it is illegible and somewhat confusing. Why is that? The MIDI events contain very precise information about the start of a note and its length and pitch, which has to be taken into account during playback. It may influence the groove of a song if the notes are always slightly shorter than sixteenth notes. If this were to be displayed correctly in the notation, the score would be unreadable as in the example above. The MIDI events also do not contain information on whether the gap between two notes is a real pause, its harmonic correlations (pitch), and the characteristics of the dynamic sequences. This is why automatic processing of notation always differs from what would be ideal. Sequoia includes a number of automatically and manually controllable functions for making it easier to read the notation.The reworked version of the above score excerpt illustration shows how big the difference can be:



...and editing is finished.

Note allocation in multiple systems

The term "**System**" refers to an individual line in a staff as well as all staves of a score. In cases where it is important to be able to differentiate a score and a note line from one another, we use the term "System" for the score and "Staff" for a single system.

Mostly, what is meant by "system" is the result of the context. For example, a two-handed piano piece makes reference to the "upper" and "lower" systems.

Sequoia provides multiple systems, e.g. for piano notation or entire scores that are comprised of up to 48 individual systems per track.

To assign notes manually to a system, click the "Assign to upper system" button



to move the selected notes one line higher on a staff, and correspondingly on "Assign to lower system"



to move the selected notes a line lower. This results in the note being connected to the line (independently of the MIDI channel or pitch). This manually set allocation can be undone by clicking on

 Assign notes automatically

to ungroup them.

Note: You can only move notes within the staff of the corresponding track during "Multi-object editing".

When transcribing a MIDI piano recording, splitting the notes into a two-line piano system using the split point is recommended. This indicates that notes above the split point pitch belong in the upper system and that the other notes belong below it. The positions where individual notes are placed incorrectly can be easily corrected by assigning the notes manually by clicking the desired system.

Automatic assignment of notes to a specific staff is flexible. Either the MIDI channel of the note event, the pitch, or even a combination of the two can make up the criteria. This permits simpler and faster distribution of MIDI notes in the staff.

Example: For some standard MIDI files, the notes of a lower system of a particular piano piece may have a different MIDI channel than that of the notes of the upper system. Assume the notes for the right hand are on channel 1 and the notes for the left hand are on channel 2. In this case, set up two staves for the system in the note system settings. The easiest way to do this is with the "Piano" preset. For the first staff, set the MIDI channel allocation to "Ch. 1" in the "Channel" selection box and "Ch. 2" for the second staff (see MIDI score settings dialog (view page 346)).

The rules for allocation are as follows: If the "Automatic system allocation" option (preset) is set for the note, the note lines will be played until the MIDI channel corresponds with it and the pitch is over or equal to the split point.

Note: Some notes may not show at all; that's because they haven't been assigned to a system yet.

Multi-voice notation

Up to two independent voices may be annotated for each staff. The voices differ in the direction of the note stems: the first voice is always with the stem pointing upwards, the second with the stem pointing downwards. Pauses are displayed individually for each voice.

Multi-voice notation can simplify the score considerably and enable multiple instruments or parts to be displayed in one staff together.



Mono-voice display



Multi-voice display

You can set the voice by selecting the notes and clicking:



"1. Assign voice (stem direction up)" and



"2. Assign voice (stem direction down)". This sets the stem direction of the notes and therefore the voice assignment itself.

The set voice assignment can be ungrouped again by selecting

 automatic voice assignment

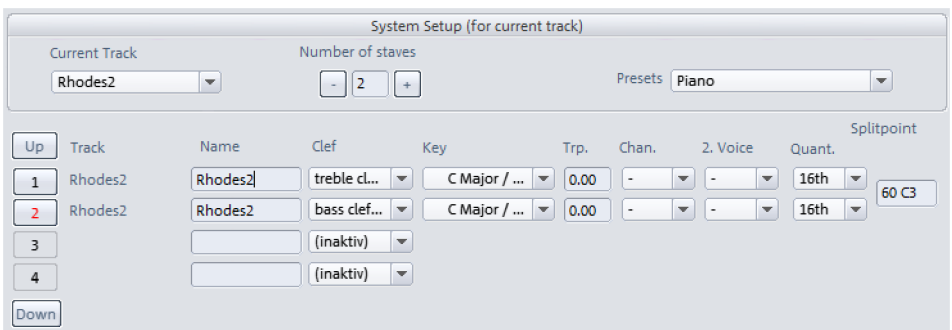
to ungroup them.

Tip: You'll find the corresponding commands in the MIDI "Score" menu; keyboard shortcuts may also be assigned to these.

For automatic voice allocation, the note's MIDI channel has to be analyzed.

To do this, set up a MIDI channel for the second voice in the "Note system properties" dialog. This can be set individually for each staff. All notes of a system featuring a MIDI channel that has not been assigned to a second voice are added to the first voice.

Voice allocation via the MIDI channel can be practical for displaying standard MIDI files with multi-voice piano pieces, e.g. if the right hand voices are set to the first and second MIDI channels and the left hand voices are set to MIDI channels three and four. Apply the following settings:



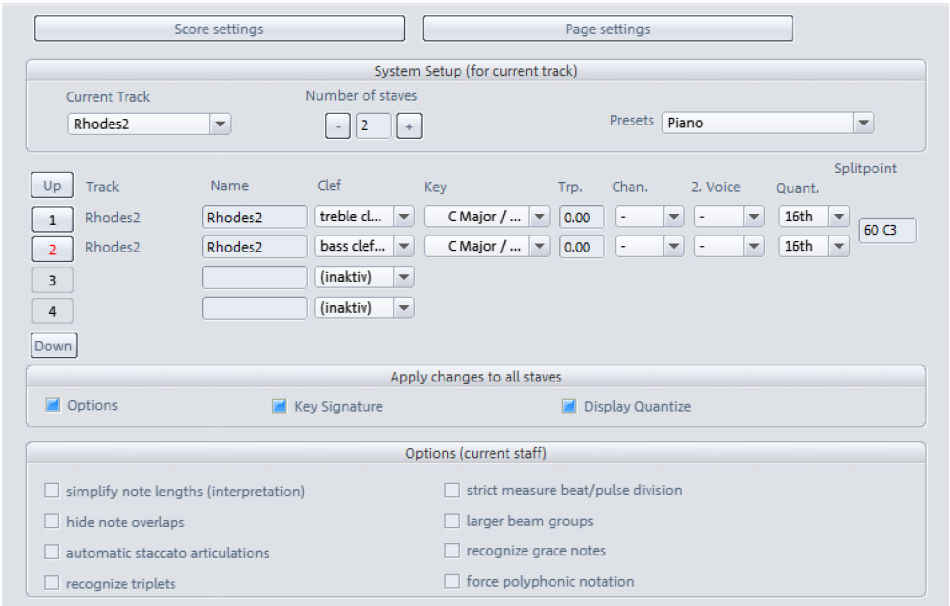
Note: If there is no MIDI channel selected for voice allocation, only one voice will be annotated (if manual voice allocation does not occur).

MIDI score settings



The score settings dialog opens the notation system and page format properties.

Apply the note system and page format settings here. None of the settings have any influence on the MIDI data itself, but rather only on the display of the score.



Note: The affected note system settings are always applied to all MIDI objects situated on the current track. The page format settings apply to the entire VIP project.

Note system settings

All templates for the system, the display parameters, and options are available on this dialog page. Specify the key, clef, and much more here.

Sequoia provides up to 48 staves on one track for MIDI data. Key, clef, display transposition (for transposing instruments like saxophone, etc.), and display quantization may be individually set for each staff.

An instrument's prefix ("Name") may also be set, plus the MIDI channel for automatic system/voice assignment.

Similarly, the split point provides automatic system allocation. Notes above the split point are added to the upper system, while those under the split point are added to the system beneath it (as long as the MIDI channel matches).

The active staff is recognizable via the index marked in red. The list of 16 staves can be scrolled vertically using the arrow buttons (up/down). The active staff's display options are displayed in the lower range.

Optionally, all changes to settings, display options, key, and display quantization are always transferred equally to all staves.

During multi-object editing across multiple tracks, the scores from the systems of each individual track are compiled together. A partial system of a track can be composed of multiple staves (e.g. 2 lines for a "grand staff" in piano notation). The system lines within a track will be grouped with struck-through beat lines.

The staves of a track may be used as an instrument or instrument group within a score. For this reason, consider early on how you would like to distribute multi-voiced pieces to multiple MIDI tracks to obtain a sensible score display. The entire score may be obtained by using the multi-object editing feature and showing all MIDI tracks in the score editor simultaneously. A vocal extract may be produced by showing the track of the desired instrument only or an instrument group in the score editor.

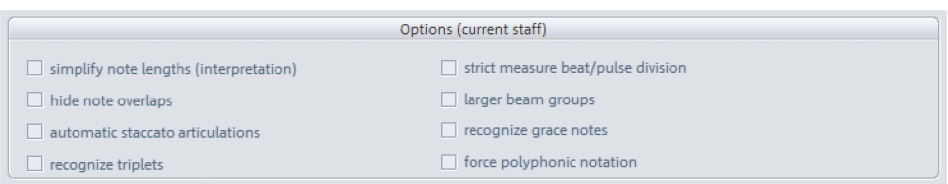
Display quantization

Using the display quantization ("Quant." column), you can set the rhythmic resolution of the note display independent of the actual quantization. This enables you to display a freely imported and unquantized track in sixteenth notes.

Set "Quant." to the lowest value in the sequence of the note value. For a sequence that contains sixteenth notes as the lowest value, select the setting "16th", not "64th" notes. A quantization display that is too fine may make the display illegible.

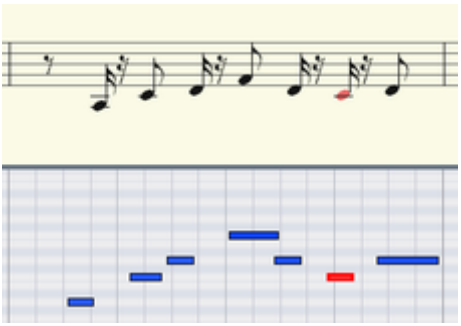
The display quantization does not have any influence on note playback, but rather adapts the note display to a raster. The actual recording (the MIDI files) is not changed, unlike the quantization function in the MIDI editor.

Interpretation Options

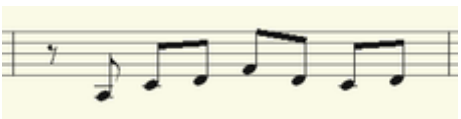


None of the options have any influence on the MIDI files or playback. An adjustment only occurs with regard to the note display.

Simplify note lengths (interpretation): Displays pauses and slurred notes in such a way that the score is as legible as possible without influencing playback.



Display without active interpretation options; Display quantization is set to one sixteenth.

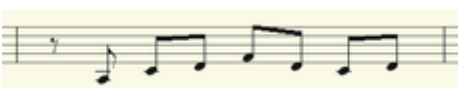


Display with option "Simplify Note Lengths (Interpretation)"

Hide Note Overlaps: With this option you can get rid of any overlapping of sequential notes which, for example, arises as a result of playing legato:

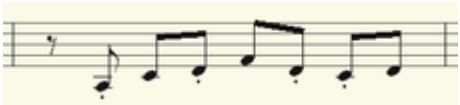


Original



Display with "Hide Note Overlaps" option

Automatic Staccato Articulations: A staccato symbol is added to a note whose value is considerably longer than the displayed MIDI note.



Display with additional option "Automatic Staccato Articulations"

Recognize Triplets: Activate this option if triplets are present in the MIDI object.

Note: The display quantization value always has to be one step finer than the smallest discernible triplet value. For example, to recognize one eighth triplets, display quantization has to be set to at least one sixteenth (or to 1/64 for 1/32 triplets).

Strict Measure Beat/Pulse Division: This option ensures that there are no note or pause values longer than a beat subgroup (pulse). In some cases longer notes are displayed as multiple slurred notes. This can simplify the legibility of the score.

Compile Bar Groups: Bar groups are partially compiled across beat subgroups. This can also simplify the legibility of the score.

Recognize Grace Notes: Note values which are much shorter than the display quantization value are annotated as grace notes as long as a base note is present.



Original



Display with "Recognize Grace Notes" option

Force Polyphonic Notation: The display of the notation always appears with polyphony.

Notation symbols

Clef



There are four clefs available in Sequoia: violin, bass, tenor, and alto clef. Sequoia differentiates between base clef and clef change. The base clef may be set up for every system in "Score settings" (and applies to all MIDI objects on the current track). A clef that you enter via the toolbar will be interpreted as a key change. Key changes are possible any number of times in a song and also within beats.

To add a clef, activate the staff where the key should be inserted by clicking the staff signature to the left.

Position the playback marker at the desired insert location and click the clef you want in the toolbar. The clef will be entered musically rather than graphically.

Example: For a key change at position 10:01:000 (the beginning of bar ten), the clef symbol will be displayed at the end of bar nine just as it corresponds with the notation rules.

Key changes may also be deleted by clicking the clef symbol with the eraser (or the right mouse button).

Beat signature

The metric and beat signature symbols are created automatically from the tempo markers of the VIP project.

Time signature changes are possible at full beat borders. Create a beat counter marker with the new signature (e. g. 6/8) at the desired position via "Edit > Tempo -> Tempo/Beat marker (view page 441)". If there are no beat count measure changes, setting the time signature of the piece (e. g. 3/4) in the transport control is sufficient.

Prefix



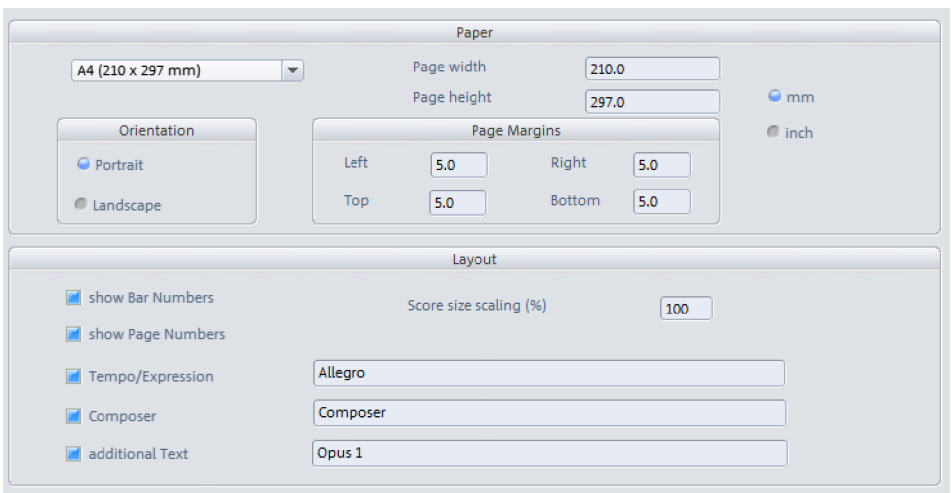
Enharmonic change

Sequoia sets the sharps and flats according to the clef selected. It is often the case, however, that enharmonic changes can considerably optimize the legibility of certain passages. In this case, you can work manually. To change one or more selected notes enharmonically, click the corresponding button. The function transforms flats into sharps and vice versa.

Page format settings

Page format settings may be accessed via "Score -> Score settings" in the MIDI menu. Click the "Page settings" button.

You may select the paper format independent of the printer settings so that it always has the same note layout, regardless of where you are working and independent of the printers installed in Windows.



Sequoia automatically creates the layout of the score for optimal division of staves and systems on the page. You only need to specify the page size, orientation, and page borders.

Scale score size (%): Scales the size of the note symbols for the printout. However, the score display influences the position where a line or page break occurs.

- Set a value smaller than 100% to fit more beats/systems onto one piece of paper.
- Set higher values than 100% to keep larger note symbols on the printout.

You can also make entries for tempo notation, composer, and an extra text field. Similarly, you can choose which layout elements should appear on the page (check boxes for beat numbers, page numbers, and texts).

Print notes

Activate the print process in the "Score -> Print score" menu or via the "Print" button. The print dialog window will appear which has been adjusted to your printer or printer drivers. Depending on the printer, there are various options like the selection of pages you wish to print and the number of copies. Make sure to set up the same paper format for the printer as for the page mode, otherwise the printout may be scaled and the page ratio changed.

The following items won't be printed; they are only visible on the monitor:

- The lines which highlight the page borders on the monitor
- Mouse pointer
- The color display of the currently selected notes and playback areas.

Note: When printing a file (e.g. as a PDF file with a special printer driver), note that you have to activate the option "Save fonts in document" in the printer driver so that the notation symbols will be displayed correctly in the document.

Score editor – tips

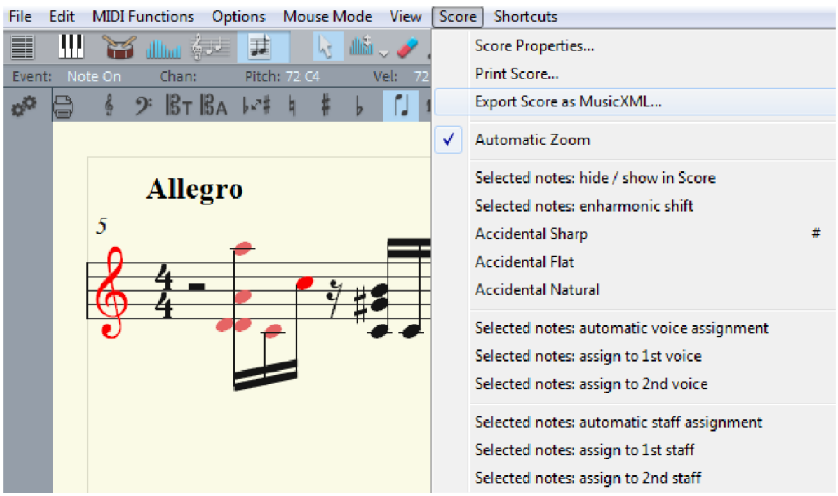
In complex arrangements, we recommend using two separate versions of a song for sequencing and for note print previews. This way, MIDI events for the note display may be changed in any way without influencing the playback. You may have to increase or decrease the length of a note in order to display the correct note value or the correct pause.

It may also be advantageous to remove trills, grace notes, and other ornaments in order to be able to print the score in a more simplified form. This manipulations, however, do affect playback. Compromises are not necessary when working with separate versions for sequencing and note print preview.

Score music XML export

The score may also be exported as a MusicXML file and then edited further in notation applications. Finale, Sibelius, Forte, plus many other MusicXML files can import music XML files. Converters are also available for transforming MusicXMLs into other formats. You can find out more at www.musicxml.org.

During the MusicXML export, all properties of the score (except for the page layout), like staves, notes and pauses, bar groups, keys, polyphonic voices, lead signs, enharmonic change, etc. are transferred. However, page layout, line breaks, and the distance between systems are not transferred. These may also be a component of MusicXML, but since these parameters in Sequoia may only be partially edited, they will be removed from MusicXML export.



Not every staff editing application processes all MusicXML elements correctly, so it may become necessary to manually correct struck-through bar lines later, or to regroup brackets for "parts".

If you create your composition in Sequoia, set score settings here in order to be able to edit and navigate more easily through the score view. You may then continue editing the score in order to optimize the layout and notation.

Converting to MusicXML format has the advantage (unlike standard MIDI file format) that many manual settings and optimizations of the score (e.g. number of scores, key, voice allocation, pitch, etc.) remain the same.

MPE

Sequoia supports the new MPE (MIDI Polyphonic Expression) expansion. MPE is a new control standard for synths which expands the range of possible expressions which can be produced with these instruments. Unlike OSC, MPE is not a new protocol; rather, it is standard MIDI which switches channels for each individual note.

When MIDI commands for controlling sound parameters (CC, NRPN), or pitch wheel commands for controlling the pitch, are sent to the synthesizer, these are normally applied to all of the notes on a channel. There is no way to assign different pitches or sound changes to individual notes when using multidimensional devices. With MPE, these limitations are removed. This is done by assigning each note to its own channel so that controller and pitch wheel commands can be applied specifically to the note in question.

This enables MIDI controllers which support MPE to produce a much wider range of expressions than a normal keyboard. In addition to aftertouch (the force applied to a key after it is pressed), MPE also offers the ability to "bend" each individual pitch separately to create vibrato and glissandi. An additional Expression Parameter (CC74) is also generated based on the position of the player's finger on the key. Some examples of MPE-compatible MIDI controllers include the Roli SeaBoard (or its mini version, Roli Blocks), the LinnStrument by Roger Linn, and Soundplane by Madrona Labs.

MPE also requires a compatible synthesizer, that is, one capable of generating separate, unique synth voices for each MIDI channel. There are many manufacturers who offer MPE-compatible VST synths. An extensive list is available on the website of keyboard manufacturer Roli Labs.

The key properties that MPE-compatible controllers and synths have are:

- For each Note On event, a new channel is created for the note. The channel is not made available for new notes until a Note Off event is sent for the current note. Since at least one other channel (Master Channel) is needed for global control commands such as switch program and global pitch, the maximum possible polyphony is 15 individual voices.
- CC74, pitch wheel and channel pressure (aftertouch) commands send to the assigned channel are applied to the note in question; CC74 is reserved as a third control dimension for MPE synthesizers (MPE Timbre).
- The pitch bend range that a MIDI note can be pitched up or down using the pitch wheel is set to 48 (+/- 4 octaves). Compared to the normal +/- 2 semitones, this seems like a huge range. But the idea is to give you the greatest possible pitch range to create glissandi from any note.

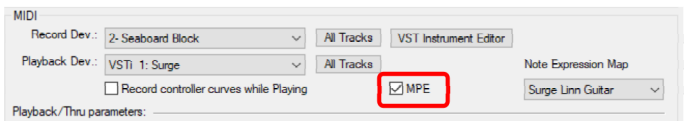
The MPE specification includes other rules, such as those for handling key pressure, for the Master Channel, and for what happens when the polyphony is exceeded.

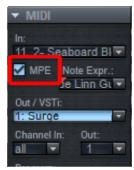
These rules, however, are not particularly relevant to MPE support in Sequoia. To learn more about them, visit
https://www.midi.org/articles-old/midi-polyphonic-expression-mpe .

MPE track mode

In order for Sequoia to be able to interpret incoming MIDI data as MPE, and in order to be able to edit it as such, the track must be set to MPE mode.

To do this, enable the option **MPE** in the track settings or in the MIDI section of the Track Editor.

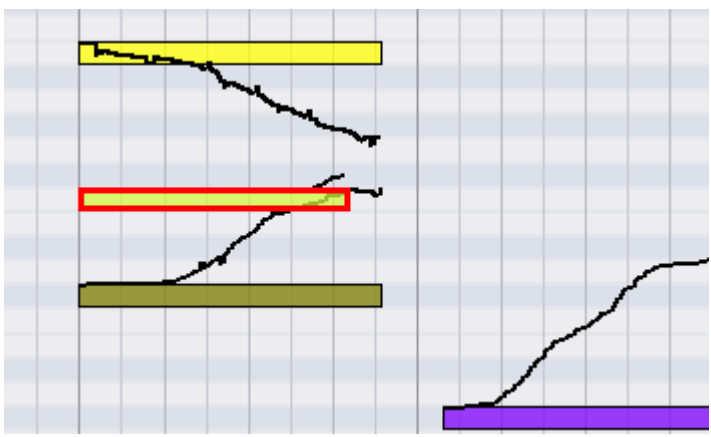




The MIDI editor in MPE mode

The MIDI editor's MPE mode has some special features which make it easier to edit MPE MIDI.

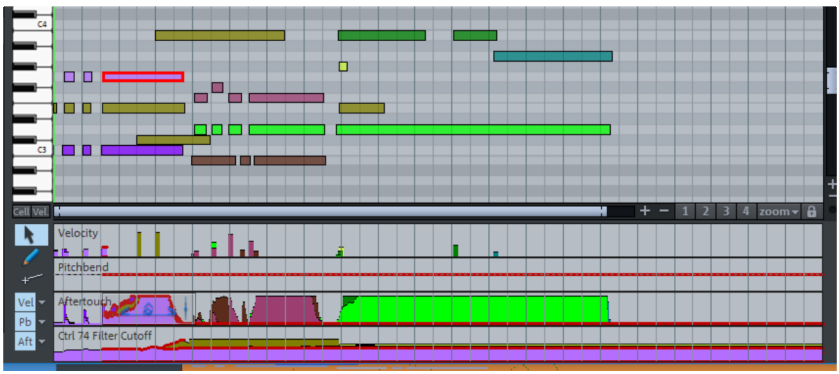
- Thanks to an extremely large pitch bend range comprising 48 semitones, even very small adjustments to the pitch wheel create audible changes to the pitch. Since these tiny changes are not easy to see in the Controller Editor below, the pitch envelope is displayed as an additional curve on the notes.



- When you move a note, the controller/pitch wheel data associated with that note (that is, the data located between a specific Note On/Note Off event on a specific channel) are moved as well.

Tip: To help keep track of which controller data belongs to which note, we recommend enabling the **Use MIDI Channel Coloring** option (located in the Options menu in the MIDI Editor).

- The channel in the foreground of the controller or note is always the one selected last.



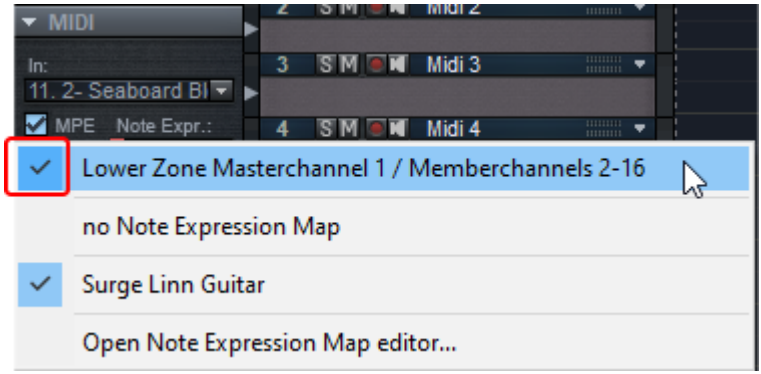
- To prevent this being moved when you drag a selection, hold down the **Ctrl key** when dragging the selection. The selected channel in the foreground is then maintained and only events on this channel are additionally selected. If you want to select events on other channels too, additionally hold down the **Shift key**.
- To make it easier to edit very small pitch bend curves, you can zoom in vertically on the controller curve by holding down Ctrl and using the mouse wheel.

Master Channel

A Master Channel is defined in MPE Standard as well. Controllers which are sent over this channel affect all notes, just like normal MIDI. Switch program commands must be sent over this channel as well.

MIDI Channel 1 is set as the Master Channel (MPE Lower Channel) in Sequoia by default. If you do not need a Master Channel, you can disable it, which will increase the maximum polyphony to 16 voices. You can also disable the Master Channel if you are using Note Expressions (see below).

You can disable the Master Channel in the Note Expression menu to the left of the MPE options in the Track Editor, or under "Note Expression Map" in the Track Options.



VST3 Note Expressions

Note Expressions are another way to control VSTi synthesizers with sound parameters note by note and hence to enable more expressive ways of playing. These are an extension to the VST3 standard. Note Expressions are currently only available for certain synthesizers by Steinberg.

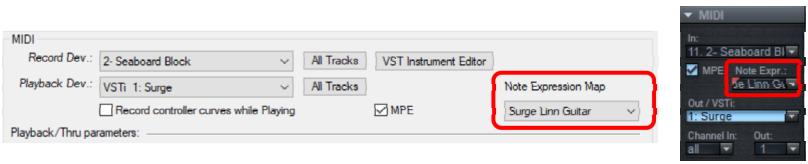
The plug-in provides a specific number of sound parameters which can be assigned as Note Expressions to MIDI controllers via the Note Expression Map in Sequoia, which then apply only to individual notes.

Differences between MPE and VST3 Note Expressions:

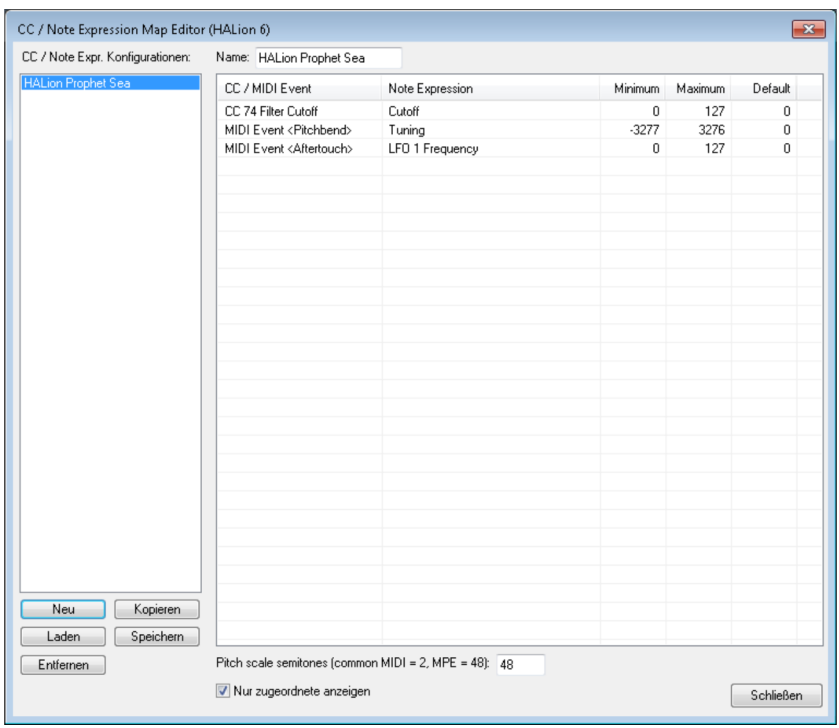
- MPE is a MIDI protocol in which notes played at the same time are assigned to different channels when using polyphonic playing. The synthesizer must be capable of receiving notes from different channels and of processing the MIDI controller commands for these notes separately by channel.
- VST3 Note Expressions are an extension to the VST protocol. Sequoia receives MIDI notes on various channels with corresponding controller values and sends the notes to the synthesizer on a channel. The Expression Values assigned to the controllers for each note are also sent over the VST interface to the synthesizer.
- As such, MPE cannot be used to control multitimbral instruments, but Note Expressions can.

Note Expression Map Editor

You can open the Note Expression Map Editor in the Note Expression menu to the left of the MPE options in the Track Editor, or under "Note Expression Map" in the Track Options.



1. Load a VST3 instrument which supports Note Expressions onto the track. Load a preset which has Note Expressions assigned in it as modulation sources, or make these assignments yourself in the VST instrument.
2. Now, open the Note Expression Map Editor and click "New" to create a new Note Expression Map. This map is now valid for the loaded patch in this specific plug-in.



3. Give the map a descriptive name (e.g. Synth_Preset).
4. Assignments are made on the right. The three standard MPE parameters CC74 (timbre), pitchbend and aftertouch are preset. If you want to assigned the Note Expressions to another controller, disable "Only show assigned".

5. Now, click on the corresponding controller in the Note Expression column and select a parameter from the list to assign it to the controller.
6. To create a new map based on an existing one, select a map from the list on the left and click "Copy". "Load" and "Save" let you load a map from a file or save a map to a file, respectively. "Delete" removes the selected map from the list.
7. Under "Pitch scale semitones", you can specify what pitch bend range your keyboard uses. MPE controllers use 48, normal keyboards usually use 2.

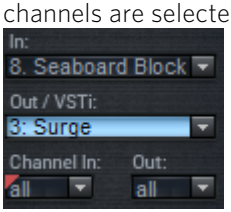
Note: The Note Expression parameters depend on the synthesizer's patch settings and are made available by the VSTi when the Note Expression Map Editor is opened (not before). It is therefore important to follow this order: first program the patch settings in the synth, then select the corresponding map!

A note about MPE and Note Expressions

- When using Note Expressions, be sure to switch the MIDI Out channel from "All" to a specific channel (e.g. Channel 1) if you are using a multitimbral synth such as Steinberg HALion which is capable of playing multiple patches at the same time, each of which respond only to specific MIDI channels.



- When using MPE controllers, it is absolutely important to make sure that all channels are selected for use under MIDI In.



- If you do not have an MPE-capable keyboard, you can still use the advanced capabilities of MPE and Note Expressions by manually changing the channels for notes and inserting the corresponding controller values in MPE mode when the MIDI Editor is in listen mode (view page 333).
- If you are recording with a normal keyboard which normally sends notes through Channel 1, be sure to disable the Master Channel or select a different channel on the keyboard.

MIDI Editor Keyboard Shortcuts

Apart from few exceptions such as "Space" for stop/playback, you can set up the keyboard shortcuts any way you like. To do this, open the editor for keyboard shortcuts, menus and mouse through the MIDI menu "Shortcuts > Edit Shortcuts". For some basic functions such as scrolling and zooming, all keyboard shortcuts from the VIP are used. You can explicitly define shortcuts for all commands that are available in the MIDI Editor menu.

Note: The mouse wheel settings are also applied from the arranger, as is the switch for temporary zoom mode (special) which is predefined with "Z".

Play/Stop	Space
Close MIDI Editor (discard changes)	Esc

File

Import MIDI	Ctrl + I
Export MIDI	Ctrl + E

Edit

Undo	Ctrl + Z
Redo	Ctrl + Y
Cut	Ctrl + X
Copy	Ctrl + C
Insert	Ctrl + V
Duplicate	Ctrl + D
Select all	Ctrl + A
Create pattern from selection	Ctrl + Shift + P
Delete selected MIDI events	Backspace, Del
Delete all MIDI data	Ctrl + Backspace, Ctrl + Del
Select previous MIDI event	Arrow left
Select next MIDI event	Arrow right

MIDI Functions

Legato	Ctrl + L
Quantize notes	Ctrl + Q
Quantization settings	Alt + Q
Mute notes	Ctrl + M

Options

Scroll mode	F
Play clicked notes	Alt + P
Quantization grid active	Ctrl + G
Show quantization grid	Alt + G
MIDI Object Editor...	Ctrl + O
Audition Panic-Played Edit Notes	Ctrl + F

Mouse Mode

Selection	1
Draw	2
Draw drums	3
Draw pattern	4
Change velocity	5
Delete	6
Zoom Tool	7
Compile notes	8
Split notes	9
Mute notes (Mute Mode)	M
Select Velocity/Controller	Ctrl + 1
Draw Velocity/Controller	Ctrl + 2
Draw Velocity/Controller as a line	Ctrl + 3

View

Event list	Alt + L
Velocity/Controller Editor	Alt + V

Notes

Accidental #	#

Keyboard Shortcuts

Select previous event (exclusive)	Arrow left
Select next event	Arrow right
Add previous event to selection	Shift + Arrow left
Add next event to selection	Shift + Arrow right
Event pitch up	Arrow up
Event pitch down	Arrow down
Snap event to left	Ctrl + Alt + 1
Snap event to right	Ctrl + Alt + 2
Select next grid quantization value	Alt + Arrow down
Select previous grid quantization value	Alt + Arrow up
Select next length quantization value	Alt + Arrow right
Select previous length quantization value	Alt + Arrow left
Move playback marker forward	Page Down
Move playback marker backward	Page Up
Playback marker to next bar	Ctrl + Page Down
Playback marker to previous bar	Ctrl + Page Up

Mixer

You can open the Sequoia mixer by going to "Window > Mixer"(shortcut: "M"). This is largely equivalent to a hardware mixing console. The integration of the Sequoia mixer into the digital environment is still much more flexible than an analog mixer.

All settings that you make in the mixer are calculated and edited in realtime. This includes track and master effects, panorama and volume settings, initial assignments for single tracks/busses as well as initial assignments for the master section.



Using the mixer

Every VIP track has a corresponding mixer track. However, you can hide any VIP or mixer track in the track manager (view page 194).

Alternative Mixer Skins

In the System Options (shortcut: „Y“) under „Design > Skins (view page 679)“ you can choose from various interface designs, a.k.a. "skins". A further option for changing the mixer skins is to click on the Sequoia icon in the top left corner of the mixer window.

Mixer - keyboard shortcuts

- **Arrow keys:** The arrow keys on your keyboard can be used to navigate through the individual mixer elements and to select the active mixer element (fader, knob or switch).
- **Page up/Page down:** Changes the value of the active mixer element. Hold the "Ctrl" button at the same time to change the value in large increments and hold down the Shift key for smaller increments.
- **Home:** Sets the mixer element to its preset value. If you press "Home" again, the element is reset to the last value, providing a simple comparison between the edited and preset states.
- **End:** Opens the assigned sub dialog of an element, e. g. the EQ window for one of the EQ knobs. This function corresponds to clicking on a knob with the right mouse button.
- **Enter:** Opens the numeric entry field for a controller. Switches are turned using the "Enter" key and "Page up/Page down" keys.
- **Del:** An activated plug-in slot can be reset with the "Del" key.

Mixer Operation with Mouse and Keyboard

Left-click: Selects a control element.

Hint: To move a single control with the mouse wheel you dont have to select it first. Just mouse-over the knob or fader and turn the wheel to change its value! for finer changes additionally press the Shift key.

Right-click: Accesses either a context menu for the corresponding control element or a dialog with additional settings options.

Ctrl + left-click: You can use this shortcut to select multiple control elements (multi-selection).

Shift + left-click: You can select all similar control elements between the previously selected and the current control element (multi-selection).



Note: After selecting several mixer elements, you can form these into one group (view page 377). Use the "Group selected controls" button to do this.

Ctrl + Shift + left-click on knobs and faders: This keyboard shortcut reverses the behavior of corresponding faders or knobs within a linked group (inverse selection).

For example, this feature is useful for moving two grouped faders in opposite directions with a single mouse movement, or for controlling the opposing movement of the panorama controllers on two tracks simultaneously.

Alt + Left-click: A single control element can be temporarily positioned within a group by pressing the “Alt” key and operating the element.

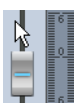
Double-click on numbers: Opens the data entry field to allow manual entry of numeric values.

Double-click on knobs: Resets the control element to its preset (initial) value. Double-clicking again sets the value back to the altered value again.

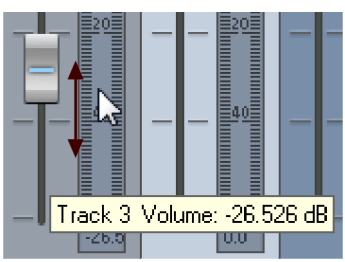
Clicking on value range: Clicking on the area below the faders or on the left of knobs decreases the values. If the mouse button is held down, the value is automatically reduced until the button is again released or the end of the scale has been reached. (Keyboard shortcut: Page down).



Clicking on the area above the faders or on the right of the buttons increases the value. If the mouse button is held down, the value is automatically increased until the button is again released or the end of the scale has been reached. (Keyboard shortcut: Page up).



The level faders can be adjusted very precisely. If you click on them while keeping the left mouse button pressed and moving the mouse pointer left or right from the fader and moving the mouse in a vertical plane, you can fine tune the settings. The step size of the settings become smaller as the mouse pointer is moved further away from the selected fader.



The step size of the fader movement also becomes smaller if you hold down the "Shift" key while moving a fader.

Knobs (potentiometers) can be set in two different ways. After clicking on a knob you can move the mouse around the knob or move the mouse up and down as for the

faders. This makes it possible to reduce the step size value by holding down the Shift key. The way in which the knobs are controlled can be set in the "System Options" dialog (keyboard shortcut: Y) under "Keyboard, Menu & Mouse > Mouse".

Drag & drop in the mixer: You can rearrange individual channel strips in the track number field or name field using drag & drop. The mouse pointer turns into a hand during this operation. The overall settings of a channel can be transferred to another channel by dragging & dropping the "FX" button. The equalizer settings can be transferred by dragging & dropping the "EQ" button.

The plug-in slots offer drag & drop functionality to copy effects to a different track or between tracks and the master. The sequence of effects for VST and DX plug-ins can be changed within the track using drag & drop.

Note: If a certain effect is already available, a repeated drag and drop of the same effect onto the same track performs a reset of the effect to the values transferred with the effect and does not create an additional copy of the effect; the original values are overwritten by this process.

If all you want to do is move effects between the channels in the mixer instead of copying them, simply keep the Shift key pressed while dragging & dropping.

Channels

Each track in the VIP is assigned to a specific mixer channel. The sound settings of the respective mixer channel are applied to the objects on the track.

Every channel strip offers the following settings options whereby the individual sections can be opened and closed by clicking on the corresponding arrow symbol:



Input: Shows the audio input for the recording (see below).

Gain: Controls input amplification on the mixer for the channel.

Aux Sends/Outputs/Sidechains: In this section the AUX components for various AUX buses (view page 383) can be activated and set. Open a menu by right clicking for advanced settings such as AUX pan editor (view page 384). You can also switch to output display (view page 385) or sidechain send display in this context menu. The button at the top right turns all signal paths either on or off, which effectively makes it an bypass switch.

A newly added AUX bus is always displayed in the arranger at the bottom and in the mixer to the right with the highest channel number.

Plug-ins: Here you can insert effects into the channel. Clicking on the empty insert slot opens the plug-in browser (view page 222), where you can load a plug-in to the slot. Click on an occupied slot to activate/deactivate the plug-in. Right-clicking on the slot opens the interface of the plug-in. The arrow next to the respective insert slot opens a menu with various functions. You can ope the plug-in browser again to switch out the plug-in or remove it ("No effect"). The "Plug-ins" button activates or deactivates all effects in the channel. A visual indicator (*) is displayed for plug-ins that were previously active and are active the next time "Plug-ins" button is pressed.

Equalizer: Here you can adjust the frequency response of the track using the EQ 116, a fully parametric 6-band equalizer.

You can numerically edit the gain and frequency of the corresponding filters.

For fine-tuning, right-click one of the knobs. Sequoia shows the equalizer dialog for you to specify more exact settings.

Panorama: This adjusts the signal alignment in the stereo panorama. The settings of the knob have a different effect on mono and stereo tracks. In a mono track, depending on the control setting you can place the signal more left or right in the stereo panorama. In contrast, in stereo tracks, this lets you control the balance between the left and right channel. If you are working with a surround track, the position in the surround field can be set here.

Beside the panorama dial, you'll also find a switch to reverse the phase.

Right-clicking on this opens the stereo panorama dialog (view page 392) where you can make adjustments to the panning laws and the stereo width.

Link: This button connects the corresponding channel with the one to the right of it. All faders, panorama, input, AUX sends, as well as EQ settings and adjustments now affect both channels. Both panorama faders are linked through inverse selection. With the keyboard shortcut Ctrl + Shift + left-click on knobs you can invert the behavior of corresponding panorama knobs within a linked group.

Automation: This button gives you access to the channel's automation functions

Detailed information about automation is available in the chapter "Automation (view page 457)".

Solo: The solo button mutes all channels with the exception of the selected channels.

Keyboard shortcut: Alt + S

More specific Solo button behavior can be set by right-clicking on it:

Solo exclusive: This setting switches the active track to "Solo exclusive" mode, i.e. only the active track is audible. All other tracks are muted in "Solo" mode.

Keyboard shortcut: Shift + Alt + S

Solo safe: This setting ensures that all channels that are switched to solo are automatically monitored with the AUX Return channels that they feed.

Detailed information on the global solo modes can be found in "Global buttons -> Global solo modes (view page 382)".

The output assignment for the channel also appears in the same context menu.

Recording: This button activates the track. Right-click to specify the recording mode for the input signal (see above "**In (Input)**").

Mute: The mute button mutes the selected channel.

Keyboard shortcut: Alt + M

More specific Mute button behavior can be set by right-clicking on it:

Mute/Inactive: Mutes and also deactivates the selected track. This increases performance as caching and FX processing of this channel are not necessary.

Keyboard shortcut: Ctrl + Alt + M

Mute bus inputs: Use this function to mute the bus inputs routed to this channel.

The output assignment for the channel also appears in the same context menu.

Volume fader: Controls the track levels. The behavior of the MIDI track's volume fader is preset to controller 7 (MIDI volume). Alternatively, you can also specify that volume fader setting does not change for MIDI tracks or that the volume fader for MIDI tracks should match the MIDI velocity scaling (view page 295). To do this, right-click the track's volume fader.

Note that a virtual synthesizer's audio output can usually be created, edited, and mixed on the same track as the MIDI data the instrument is receiving. In some cases, this results in duplicate use of the volume fader which, on the one hand, controls velocity or MIDI volume (CC 7), and on the other, controls the audio level. These are not identical parameters. For instance, a MIDI instrument played with high velocity can be added to the mix at low volume and vice versa. For this reason, the volume fader can be assigned differently.

Note: Levels higher than 12 dB may also be entered numerically. To do this, double-click on the volume value display at the bottom end of the channel fader scale and type in the desired value.

Monitoring: Click the loudspeaker symbol to activate the monitoring function. This will route sound card input signal through to the output.

If the "Hybrid Engine" is set up for monitoring, all mixer channel effects for the output can be used. This requires ASIO drivers for the sound card. By this, Sequoia can be used as a live effect device, for example.

More details on Monitoring can be found in the chapter "System settings >Monitoring settings (view page 596).

VCA: This button activates the VCA Mode for the channel. Here you can set the fader as the VCA master or assign the fader to a new VCA group ("VCA Groups (view page 372)").

FX: This button activates the Effect Routing dialog (view page 388) for the corresponding track and can be used to configure and edit the effects chains. This determines the sequence of the effects.

More information about this is available in the chapter "Buses and routing -> Order of effects calculations and signal manipulation (view page 387)".

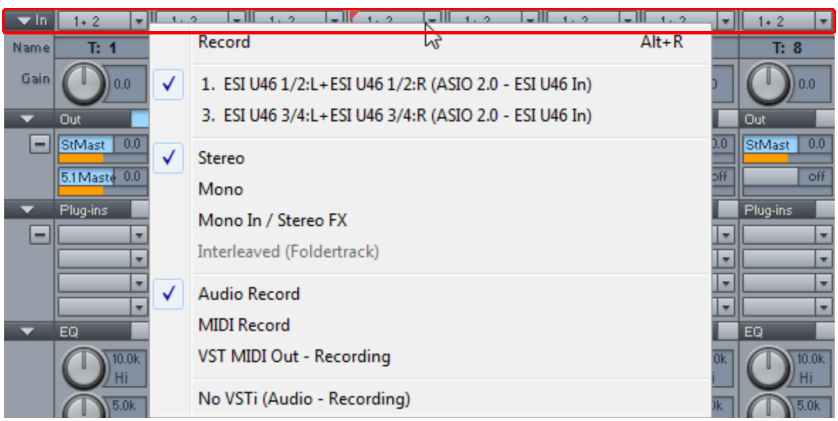
Right clicking the "**FX**" button opens a context menu. Here you can open the effects routing dialog for the channel and the preset effects settings. These can be copied, inserted, reset, saved, or loaded. You can save your own track effect settings in the folder „Program Files“ > „MAGIX“ > „Sequoia“ > „fx-preset“ > „Track FX“. You can also create new subfolders here.

Track name: You can enter the name of the VIP track here. This name can be edited by double-clicking.

Out: Sets the output for the channel. This can be a stereo master channel, another stereo audio output, a submix bus and/or a surround bus. For MIDI channels, you can set the MIDI (VST) output device here. The option "No Output" means that the channel is not assigned to any output.

In (Input)

Determine the recording mode for the incoming signal up top on the channel strip.



- The "Record" option activates the track.
- Below you will see all the available inputs for your audio/MIDI interfaces. Select the input where the signal you want is located.
- If you select "Stereo In", the channel input will be switched to stereo and the signal will be recorded on two channels.
- If you select "Mono In", the channel input will be switched to mono and the signal will only be recorded on one channel.
- If you choose "Mono In / Stereo FX", the channel input is switched to mono and a single-channel input signal is recorded, but the track itself remains a stereo track in which the effect calculation is performed in stereo.
- You can select whether you want to record in audio or in MIDI and - if using MIDI - whether you need a VST MIDI output.
- In addition, if you have selected "Audio recording" above, the program will display "No VSTi (audio recording)".

Apply effect to multimple channels simultaneously.

You can apply multiple instances of the same effect to multiple mixer channels:

1. Select the channels that you would like to apply the effect to. First click on the channel number of the first desired channels to activate it. Then, holding the Shift key, select the number of the last channel from the row. This will select all channels in between. If you press the Ctrl key instead of Shift, you can add any additional channel to the selection.

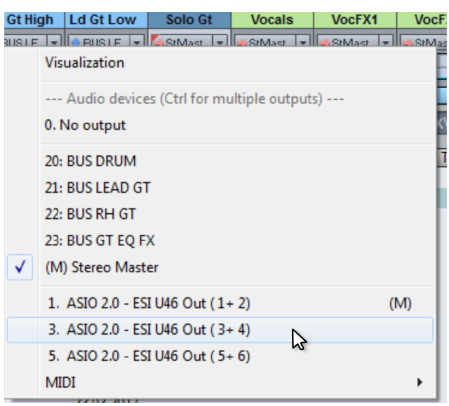
2. Now, select an insert effect for one of the selected channels and make your adjustments. As soon as you close the effect dialog window, the effect will be integrated into the insert slots of all selected channels with the exact settings made. Additional parameter changes will be applied to all instances of the effect found in channels selected at that time.

3. To remove the effect from all selected channels, simply take it out of one insert slot. All other instances of the effect will also disappear.

Assign track to multiple outputs

You can also route the output signal of a channel to several hardware outputs, buses or masters simultaneously, for example to create a stereo mix in the same project in addition to a surround mix, or to listen to a mix on different speaker systems.

Click on the corresponding field in the output section, hold the Ctrl key and select additional audio outputs.



The stereo master output is labeled with and (M) and is the default output.

If you set the AUX section of the channel to "show outputs" with a right click, you will get a quick overview of the outputs used.



To revoke assignment to a stereo output, click to the corresponding field in the output section, hold the "Ctrl" key and select the stereo audio output you would like to deactivate.To revoke assignment to a stereo output, click to the corresponding field in the output section, hold the "Ctrl" key and select the stereo audio output you would like to deactivate.

In the Matrix display of the Routing Manager (view page 208) you'll find a good overview of all assigned outputs.

VCA Groups

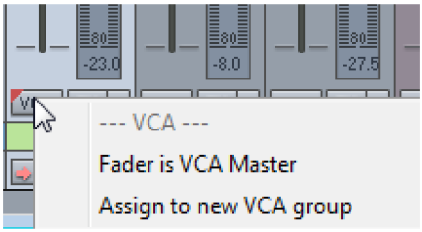
VCA is the abbreviation for "Voltage Controlled Amplifier". In analog mixers VCA are found as voltage controlled amplifiers. This makes the fader influence the music signal indirectly, i.e. through a decoupled voltage stabilizer. VCA groups are a type of a remote control that makes it possible to control many individual channels and mix them together. No audio channel flows through VCA groups themselves.

A major advantage of VCA groups is that, the way the VCA Master Fader regulates both the assigned channel signals and all post fader AUX send feeds are down regulated too. In this way the balance between the direct and effect signal remains the same.

In contrast to audio groups, VCA groups have their own audio output, meaning they have audio outputs at their disposal, and offer the possibility to further edit the sum total signal with insert effects.

You can also work with VCA groups in Sequoia. You can be access the functions using the VCA buttons of the individual channels:

By clicking on the VCA button in the channels a context menu will appear which shows you which VCA Master channel (VCA group) controls the fader.

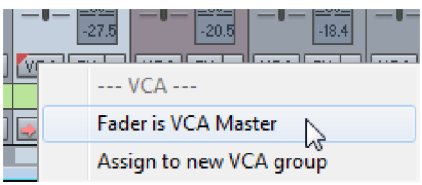


Create a VCA group

1. Create a new channel in the mixer by right clicking on the AUX channel number and selecting "Insert tracks" > "Insert empty track".

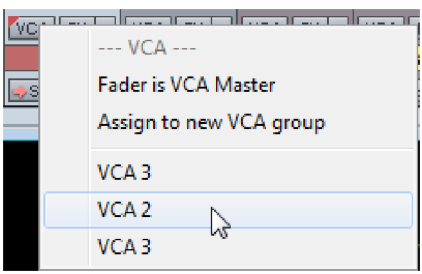
2. Double click on the name field of the new channel and name it "VCA1".

3. To properly turn this channel into a VCA group, press the "VCA" button and select "Fader is VCA Master".



Assigning channels to an available VCA group

Click on the VCA button of the channel that you want to control using a VCA group and select the VCA group which should control the channel.



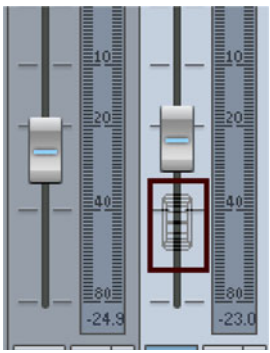
Assigning channels to a new VCA group

Here creation of a VCA group and channel assignment to a VCA group are united in one option. If you want to integrate a specific channel into a new VCA group, select the VCA option "Assign to new VCA group" for the corresponding channel in the mixer. At the back end of the mixer a new VCA group channel is created, to which the selected channel is assigned.

Ghost Fader

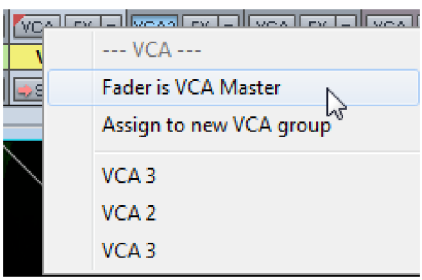
By assigning individual channels to a VCA group "Ghost Faders" are created which display the resulting volume in the track according to the set VCA Master Controller. This way the VCA master control also influences the volume of assigned channels. Relative volume levels of channels in relation to each other remain unchanged.

This enables continued control of the volume of every individual channel assigned to a VCA group. The ghost fader value will change accordingly.



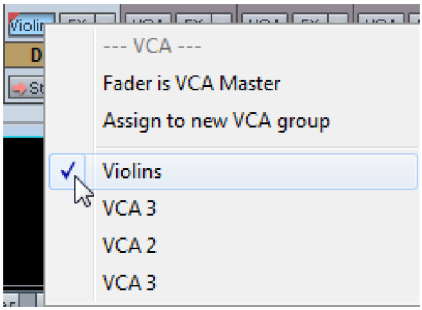
Change channel fader into VCA Master

Each fader can serve as VCA master for other channels. As soon as you select the "Fader is VCA Master" option for a channel, this channel will appear as a VCA Group in the dialog selection.



Remove VCA mapping

Remove a channel's VCA assignment to a VCA Group by switching off the selected VCA group in the respective channel. To do this remove the corresponding tick in the VCA button's context menu.



When deactivating the "Fader is VCA Master" option in a VCA Master all assignments for this VCA group will be revoked.

Assigning and deleting VCA assignments can be done in the Routing Manager (view page 211).

Master section



The master section can be found to the right beside the channel strips.

Master plug-ins: This section can be used to set effects inserts, MAGIX plug-ins or VST plug-ins for the master out signal. Right click on the right mouse button to show the plug-in dialog (view page 388). To switch the plug-in effects on or off quickly, click the control button.

Master equalizer: This equalizer affects the complete signal.

SE stereo enhancer: The multiband stereo enhancer is used to modify the entire signal's stereo picture. Three bands are available for the bass, mids, and high frequencies.

Mono: This button plays the entire signal in mono. It's useful for temporarily checking mono compatibility.

Normalize (N): This is the master normalization. If this key is selected, the output level will be adjusted in such a way that the loudest signal sections reach 0 dB levels. The maximum level reached during the last playback will serve as the basis during this process and will be displayed on the peak meter.

Hint: Click the level display in stopped state, and the playback marker will jump to the position indicated by the level.

Automation: This button gives you access to the channel's automation functions

Detailed information about automation is available in the chapter "Automation (view page 457)".

Link: This key connects the left and right channels of the master signal.

Faders: These faders adjust the left and right master signal. Reset the faders to the 0 dB position by double clicking them.

FX: Opens the effects/routing dialog for editing and configuring any number of effects chains. This specifies the sequence of the effects.

More information about this is available in "Order of effects calculations and signal manipulation (view page 387)".

AAC Preview/MP3 Preview

In the master plug-ins section under "Modulation/Special", there is a preview function which can be used to prepare MP3s or AAC files for export. This makes it possible to hear in realtime what the generated file will sound like after encoding. You can change the export settings directly in the dialog window. During the mastering process, you have the option of taking the idiosyncrasies of the encoder into account and adjusting the quality relative to the requirements (e. g. "Mastered for iTunes").

Mix to file while recording

The Mix to file and On buttons can be used to mix down in realtime and to change any of the mix's parameters during playback. At the end of the playback process, the mixdown created will be written to a wave file.

1. Click the "Mix to file" button to open a dialog to specify the name and place where the created wave file should be saved. When the "On" button is active, the master output of the mixer can be written to a wave file during playback.
2. Now start playback of the VIP
3. On playback, any parameter may be changed in order to record any live adjustments that are made.
4. At the end, stop the playback of the VIP.

Important: Make sure that the "Mix to file – on" button is deactivated if the master signal should not be mixed into a new file. Otherwise, the system creates a wave file every time the file is played.

The "Mix to file" function does not necessarily need to be used to record dynamic mixer movements during playback. Effects may be controlled via AUX busses, and volume and panorama curves may be automated.

Note: The "Mixdown" button is available in some mixer skins, e.g. "Multitrack mixer" or "Recording mixer". This function also mixes down and saves the entire VIP, including all settings, to a single file. In contrast to "Mix to file", no parameter changes during playback are taken into consideration, but the track is merely "bounced". Detailed information may be found in the menu reference under "Menu tools -> Track bouncing (internal mixdown)".

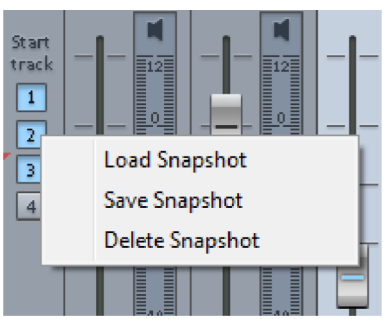
Device/master out: Enter the playback device for the stereo signal. Select "Master inactive" if the master playback device should not be used in the project (e.g. multiple I/O setups).

Note: If the settings for the master playback device are changed, all tracks routed to the master will also automatically change settings. If this automatic process is not helpful, select "Master inactive" before setting up a new device.

Global Buttons

Some additional buttons for adjusting the global settings of the mixer window and mixer handling can be found to the right side of the mixer window.

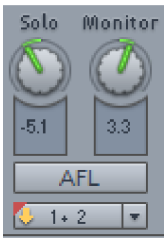
This includes (counter-clockwise from bottom left):



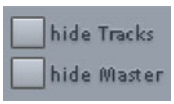
Start track: If the arrangement has many tracks, only a section of the corresponding mixer channels are generally displayed in the mixer window. The visible section can be moved using the scroll bar at the bottom of the mixer.

Hold the "Shift" key down and click on one of the "Start track" buttons to save the current view (a left click is sufficient the first time). The previously saved section can

be recalled by clicking the "Start track" button again. Right click the "Start track" button to open a context menu to select between "Load snapshot", "Save snapshot", and "Delete snapshot".



Solo/Monitor/PFL/AFL/Solo in place (view page 380)



View functions: Here you can hide or show the individual tracks and the master section.



Group selected controls: These two buttons are used to group and ungroup elements of the mixer.

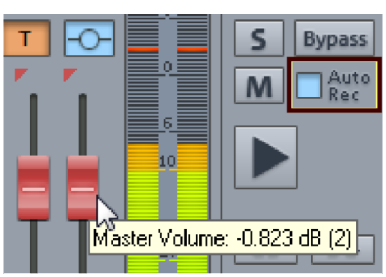
To group controls, select the desired elements while holding down "Ctrl" and then activate "Group selected controls". To ungroup, select one of the elements in the group and click on "Ungroup controls".

Note: If you have previously selected multiple tracks, this function is also available to group selected controls if the same control elements are involved. More information about grouping tracks in Sequoia is available in "Working in the project window -> Selecting/grouping multiple tracks (view page 130)".

Multi-channel Selection: Multi-channel selection in the mixer functions exactly like multitrack selection in the arranger, i.e. by clicking the track number/name while holding down the "Ctrl" or "Shift" key. In order to clear a group of channels, click a channel before or after the selection and activate a control element.



Play/Stop: Playback of the VIP can be started and stopped with this button. Right-clicking opens the transport console.



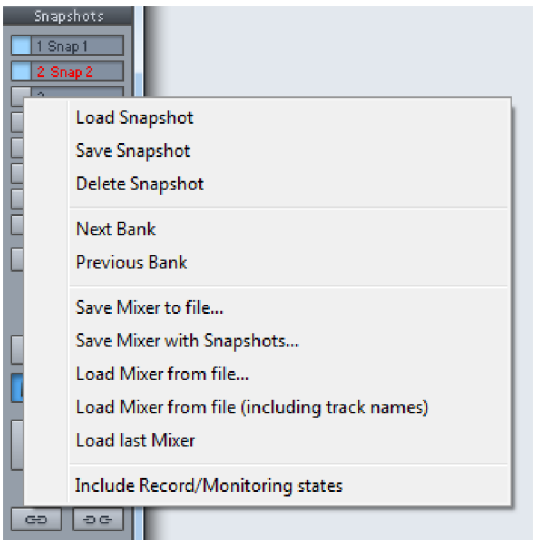
AutoRec: This display will light up if the level automation is written in the master.

Bypass: Using this button you can switch off active effects in the project for comparison purposes. You will now only hear the audio with the volume and panorama settings for the mixer channel.

Solo/Mute: All solo/mute functions can be activated and deactivated globally using these buttons.



Load mixer setup/save mixer setup: These buttons enable the current mixer settings to be saved as a preset (without snapshots) or previously saved mixer setups to be accessed.



Snapshots: The snapshot function is particularly useful for comparing various setups. You can easily switch between setups. Naming snapshots is possible by double-clicking the name field. Right-clicking a snapshot opens a context menu to save, load, or delete snapshots or to switch snapshot banks. Using this process, up to 32 mixer snapshots can be saved.

Save mixer to file: Save mixer settings to a separate file using this option.

Save mixer with snapshots: This option enables mixer settings to be saved including snapshots. When these mixer settings are loaded in the future, a dialog will ask whether the saved snapshots should also be included. If this is confirmed with "Yes", you will overwrite any snapshots used up until this point.

Load mixer from file/load mixer from file (including track names): Complete mixer settings can also be accessed with or without track names.

Load last mixer: When loading a snapshot, the current mixer settings will be saved temporarily and these can be recalled with the command "Load last mixer". This makes an A-B comparison between the snapshot and the settings possible.



Reset (mono): Resets all mixer settings to default settings for mono tracks (when using mono or LR audio files). This reset function refers to the track panorama settings (view page 392).

Reset (stereo): Resets all mixer settings to the default settings for stereo tracks (when working with stereo audio files). This reset function refers to the track panorama settings (view page 392).

Reset AUX: Resets all AUX portions of the channel strips to their default values (= no AUX portion).

Reset EQ: Resets all equalizer settings.

Reset peaks: This resets the LED peak meter (peak hold display).

Reset FX: This enables you to remove effects from the signal path for all channels.



Setup: The mixer setup dialog can be accessed by clicking the "Mixer setup (view page 658)" button.

?: Click this button to open the help file, which includes detailed information about the mixer.

Solo / Monitor Volume Control

In the master section there is a solo level and a monitor level available.



The solo knob controls the monitor volume of the solo mode, while the monitor knob also influences the level on the monitor output. The value corresponds to the Volume setting in the Monitoring section (view page 397).

Furthermore, the audio monitor position can be set using the AFL/PFL switch.

The selected monitor output corresponds to the first output device of the monitoring section (view page 395).

The signal can be accessed at two different positions:

PFL: PFL (pre-fader listen) intercepts the signal before the faders and effects, but after the input gain.

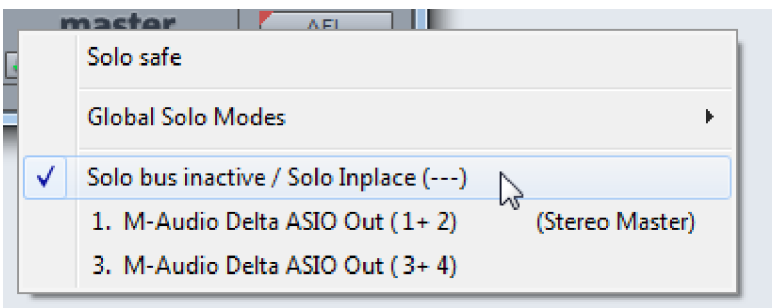
AFL: AFL (after-fader listen) takes all panorama changes/fader movements into account and influences the monitoring signal accordingly.

In general, the monitor bus behaves like "Main to Monitor", meaning that the content of the master can be heard at the same time on the monitor bus. As soon as solo is activated, only the solo state is played over the monitor bus.

Solo in Place (default)

This solo button behavior has been in use in Sequoia since the beginning. "Solo in place" sets channels to solo in the mixer at their corresponding positions within the stereo picture of the mix. Simultaneously, all other channels are muted. This solo mode is typically used during the mixdown to identify single instrument tracks and edit them individually.

You can switch to "Solo in place" mode and simultaneously deactivate the solo output bus by clicking on "Solo bus inactive/Solo in place" under the AFL/PFL button.



Attention: When using the monitor fader in this mode, it is possible to influence the audible master level and thereby the level of external devices connected to the stereo master. This means you can set up the monitor level in the mixer without using any external volume controllers. The audible result is different to that of the peak meter display in the mixer (which is the reason the numerical level values in the mixer are displayed in red). Additionally, the solo fader influences the volume of the solo tracks that are played through the master section. However, the internal level for bouncing and burning remains unchanged.

A further feature of this mode in Sequoia is that you can toggle between Pre-Fader Listen and After Fader Listen in Solo in place mode by clicking on the AFL/PFL switch.

Solo/monitor bus (in the "Hybrid Engine" only)

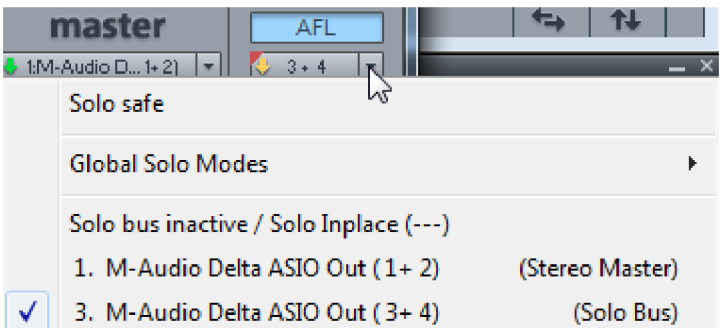
Use this bus to monitor tracks individually by pressing the corresponding solo buttons without influencing the stereo sum signal. This also affects mastering or live mixing.

The signal can be accessed at two different positions:

PFL: PFL (pre-fader listen) accesses the signal before the faders and effects, but after the input gain.

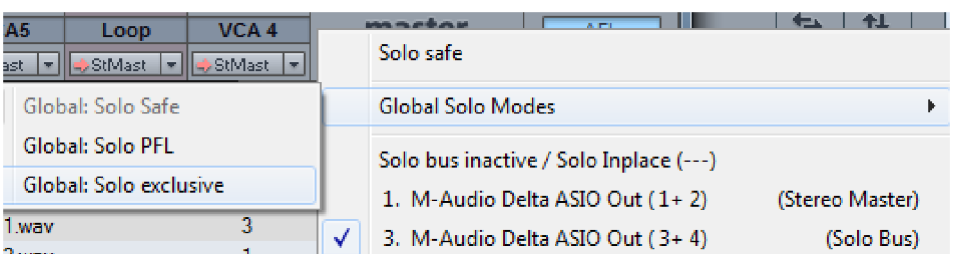
AFL: AFL (after-fader listen) includes all panorama changes/fader movements and influences the monitoring signal accordingly.

To set up an AFL/PFL monitoring line, switch the button to AFL or PFL by clicking it, and then select an output in the slot under your sound card as a solo bus.



Global solo modes

You can use the solo bus selection to set the global presets for using solo. This can also be accessed by right-clicking the solo buttons.



Global: Solo is not explicitly listed as a mode. If no check is set for a solo mode, then global solo will be active and you will hear the solo channels without AUX return channels.

Global: Solo safe: While in "Solo in place" mode, every channel switched to solo may be heard including the AUX return channels automatically.

Global: Solo PFL (available only in hybrid mode) switches solo tracks to PFL mode. The signal is accessed before the fader and effects, but after the input gain. When deactivated, the solo tracks are accessed after the fader.

Note: Economy tracks cannot be monitored with PFL.

In Global: Solo exclusive mode, only one channel may at a time may be switched to solo by clicking the solo button. In this case, the solo state is deactivated for all other tracks. If you activate the "solo" button for another channel, only this will be switched to solo. Solo is deactivated for the channel previously switched to solo. This enables you to quickly "walk through" individual channels.

To listen to the AUX return channels on the channels switched to "solo", select "Global: Solo safe" mode as well.

Note: While in "Global: Solo exclusive" mode, to still be able to listen a single channel in "Solo exclusive" mode, click the channel's "solo" button while holding down the "Shift" + "Alt" keys.

If you have already set "Solo exclusive" as the global solo mode, the same key combination deactivates "Solo exclusive" mode in order to hear multiple channels "solo" at the same time.

Buses and routing

AUX buses

An AUX bus combines all signals of the corresponding AUX sends from the individual channels. AUX buses are usually used for controlling real-time effects via the volume curve. For this purpose, part of the signal of the desired mixer channel is sent to the AUX bus ("AUX send") where effects are added. The fader of the AUX bus represents "AUX return".

Note: AUX tracks normally don't contain objects in the arranger view. They provide objects on other tracks with effects.

Submix buses

A submix bus combines several tracks. It controls the volume, panorama, and effect settings for the sum of all tracks that are "routed" to the submix bus. For example, all drum tracks (hi-hat track, bass drum track, etc.) may be combined as one submix bus so that the entire drum kit may be controlled via the volume controller of the submix bus.

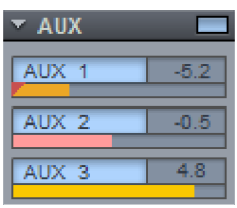
Creating submix/aux buses

Submix or AUX busses are always meant for stereo busses. The outputs may be routed for any channel to output devices or other submix buses.

- To create an AUX bus, click with the right mouse button on the number of the channel in the mixer. In the context menu, select "Insert tracks -> New AUX bus". Another option for adding an AUX bus is to raise the volume of an unused slot in the AUX area of the track editors or the mixers. This creates a new AUX bus automatically.
- If an AUX bus has already been created, you can the feed the signal of the previously created AUX section into the the newly created AUX bus.
- To create an Submix bus, click with the right mouse button on the number of the channel in the mixer. In the appearing context menu, select "Insert Tracks -> "New Submix Bus". The submix channel is integrated behind the channel that is selected. All channels, whose outputs are routed to this bus will be edited as a sum in this channel.
- Right clicking the number of a channel provides a context window for selecting between the properties "AUX bus", "Submix bus", or even both for the respective channel strip.

AUX Routing

In the AUX section of the Mixer and Track Editor you can route AUX sends and control their levels.



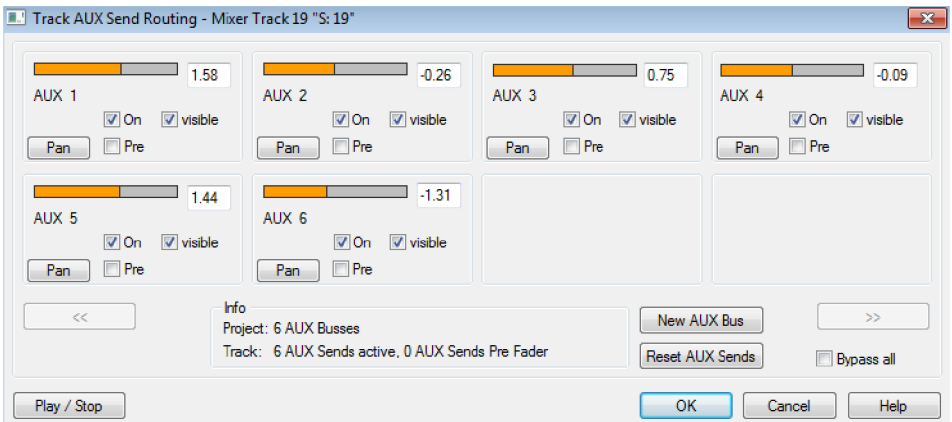
In the context menu of the Aux section you can set the AUX modes Direct out , Pre-Fader or Post-Fader.

Direct out: Audio is sent to AUX directly after gain. The color of the AUX send fader changes to red.

Pre-fader: Audio is sent to AUX after gain and all fx processing but beforethe volume fader. The color of the AUX send fader changes to yellow.

Post-fader: Audio is sent to AUX after all effects and the volume fader. This is the default setting. The color of the AUX send fader is orange.

Right-clicking on a mixer channel's AUX button opens the AUX routing dialog.



You can also open this dialog by right-clicking on the channel number and selecting "Track Effects > AUX Sends...". You can enter the intensity of each AUX bus in numeric values or by moving the orange bar (the bar is not visible in Off mode).

By default all AUX buses in Sequoia are set to "Post Fader". To switch them to "Pre-Fader", place a checkmark in the "Pre" box. You can use the effects routing dialog (view page 388) to specify the precise position of the "pre" and "post" AUX buses within the effects chain.

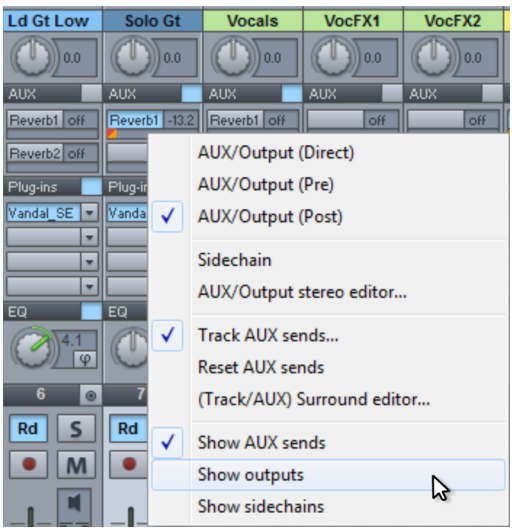
The "Pan" button applies a complete panorama section for every AUX send, which operates analogous to the stereo panorama dialog (view page 392) of a track. For example, this allows you to adjust the stereo width or invert the phase for the AUX send.

New AUX Bus: Creates a new AUX bus.

Reset AUX Sends: All AUX parameters are reset.

Displaying Outputs

In the AUX section of the Mixer and Track Editor you can also display the output assignments of the channels with a right click. This is especially helpful if you use multiple outputs (More information can be found in the section "Assign track to multiple outputs" on page 371).

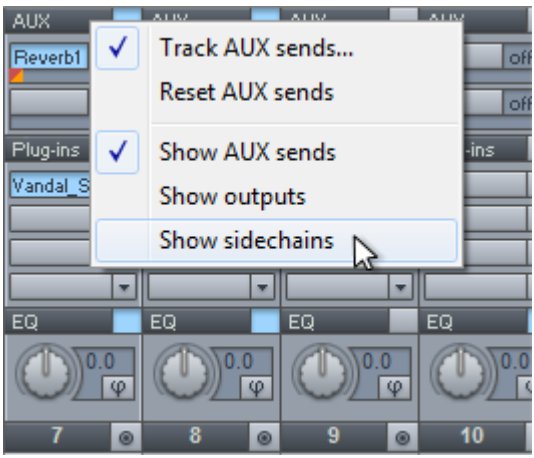


By right-clicking you can define the selected output as a "Direct out" (red fader display), a "Pre-fader out" (yellow fader display) or "Post-fader out" (orange fader display) or open the stereo panorama dialog to set the panorama (view page 392) for the respective outputs.

In the Matrix display of the Routing Manager (view page 208) you'll find a good overview of all assigned outputs.

Show sidechain sends

With a right-click in the AUX section of the mixer and Track Editor you can also display the output assignments of the sidechain sends.



In tracks that are used as sidechain signals or mixer channels, a new sidechain bus will be created and a sidechain send fader will be assigned for the target track. To differentiate from the "normal" AUX buses, a "**>**" is added. "Pre-fader" is set to catch the side chain signal (yellow fader display). This means that the sidechain remains independent of the fader position of the sending channel.

By right-clicking you can define the selected sidechain as a "direct out" (red fader display) or "post-fader out" (orange fader display), or open the stereo panorama dialog to set the panorama (view page 392) for the respective sidechain send.

A typical example for using a sidechain send is to control a compressor (view page 872) for the bass using the kick drum signal. This way, every time the kick drum sounds, the bass will be lowered.

You can get a good overview of all assigned sidechain sends in the matrix display of the Routing Manager (view page 208).

The sequence of effects calculations and signal manipulations

The sequence of gain, panorama, and filter manipulations has no effect, at least when working with 32-bit float. In this case, it is not important whether you filter first and the change the gain, or vice versa. Most other effects are also independent on the input signal. This includes the Room Simulator, Stereo Enhancer, Delay, Resampling, and Time Stretching or Pitch Shifting.

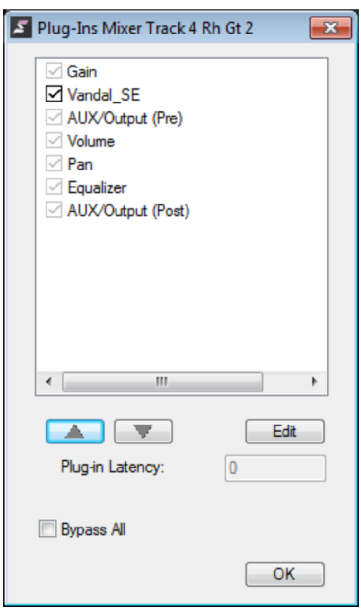
All dynamics functions and those that remove or reduce noise (De-hisser and noise reduction) are dependent on the input signal level and, in some cases, on the frequency response of the input signal. Once the optimal settings for a parameter have been found, other effects located in front of the effects should not be adjusted. For example, use the De-hisser and the multi-band dynamics section in the mix master only after all other changes have been applied to the individual objects and the mixer channels.

Effect Routing/Plug-ins Dialog

In the Effect Routing/Plug-ins dialog you can make all the important settings for realtime effects and plug-ins. This dialog is available at object level ("FX" in the object editor), channel/track level ("FX" button in the track editor), and at mixer level ("FX" button in the stereo channel strip).

Effect Sequence for Plug-ins

All used effects are listed here in the respective context (track, object, or master). The sequence of the effects can be configured in the effects routing dialog.



You can use the arrow keys to change the sequence of selected effects. For example, you can position an FX insert before or after the EQ.

Note: Object AUX sends still remain in the last position.

The movement of effects creates a corresponding change in the sequence of effects in the insert slot of the track editor or mixer channel.

Internal effects (view page 219) can now be used multiple times per track/object.

Exceptions: Distortion.

Note: The first instance of each internal effect can be loaded into older program versions. Otherwise, this new functionality is not compatible with the characteristics of older program versions.

Plug-ins Pre or Post

The plug-ins dialog enables you to insert effects "Pre" or "Post". Highlight "pre" or "post" in the plug-in list before loading a plug-in. The plug-in is then inserted "pre" or "post" fader.

Pre fader means that the plug-in is added to the signal before it reaches the fader. The result is that the input level of the plug-in is not dependent on the volume control setting.

In contrast, post fader means that the plug-in is added to the signal after the fader. The result here is that the input level of the plug-in is directly dependent on the level setting of the volume control.

Example: If you insert a reverb plug-in to a channel "post fader", a higher channel volume level will lead to a higher input level in the plug-in. The channel fader determines how much of the signal will be sent to the reverb. If the fader setting is high, more reverb will be produced. If the fader setting is low, less reverb will be produced.

With a "pre fader" setting the input level at the plug-in is not influenced by the channel fader setting, but instead by the input control of the reverb effect itself. A change in the channel fader setting has no effect on the reverb level.

Effect Routing / Plug-ins Dialog Parameters

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

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.

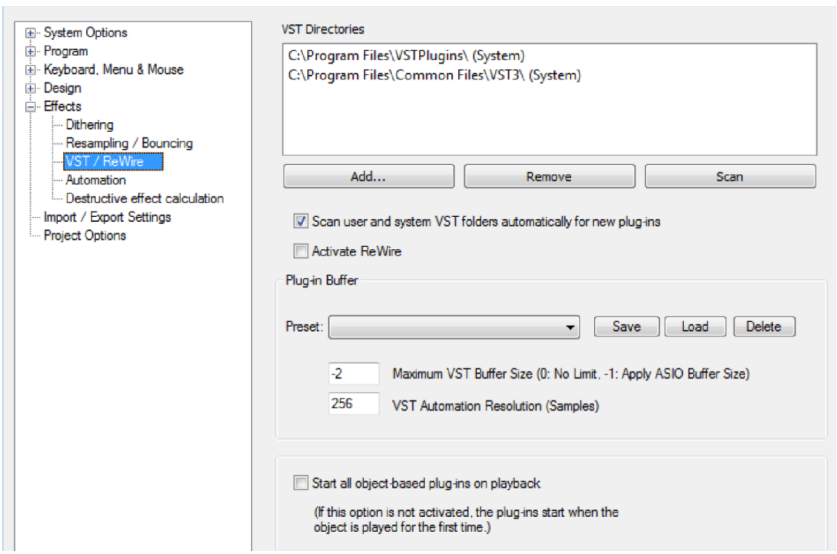
Detailed information about the plug-in dialog is available under "Software instruments/VST plug-ins/Rewire -> Plug-in panel - graphical interface (view page 285)".

Copy/Paste: Use these buttons to quickly apply the settings to different tracks or objects.

Plug-In Latency/Force Latency: Sequoia works with latency compensation for all plug-ins. Plug-ins report their latency to Sequoia and this value is transformed into a time shift in the audio material. This prevents audio delay while the plug-ins are used. If the values reported by a plug-in are incorrect (i. e. time delays result from the use of a plug-in in spite of this), you can force a specific latency value to compensate.

VST//ReWire

Keyboard shortcut: "Y" > "Effects" > "VST/Rewire"



VST Plug-in folder list: Set the paths for your VST plug-in effects and VST instruments here. You can add or remove search paths for VST plug-ins with "Add" or "Remove". With "Scan" Sequoia will scan the selected folder for VSTs. Not only are all the plug-ins imported, but they are also checked for usability within Sequoia. More information regarding this dialog and general information on installing VST plug-ins can be found under Installation of VST Plug-Ins (view page 275).

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.

Activate ReWire: If this option is activated, rewire-compatible client applications can be integrated into Sequoia as synthesizers.

ReWire is generally used for sample-exact, real-time transfer of audio channels between two programs. Both programs can be connected through the same sound card. The transport functions of the applications (such as playback and forward/reverse) are connected with ReWire. If ASIO drivers are being used, sounds from ReWire-compatible applications can be assigned to different sound card outputs.

To integrate ReWire applications into Sequoia, find the ReWire-compatible instrument that you want to connect with Sequoia in the "MIDI out" slot of the track editor by going to "New Instrument > ReWire" The instrument outputs of the ReWire application can be opened in the "Audio in" slot of every track.

The tempo is always configured according to the settings specified in Sequoia, i.e. Sequoia is the master. The tracks/channels connected with ReWire-enabled software can be edited with the equalizer, effects, and other plug-ins and then routed to the available buses.

Note: Make sure that the set sample rates of the two applications connected with ReWire match one another so that playback at the correct pitch can be guaranteed.

Additional information about ReWire is available in "ReWire client applications" (view page 277).

Plug-in Buffer

The default presets usually cover most applications when working with plug-ins. Try out various presets if you are experiencing problems with plug-ins or DSP cards.

You can enter the maximum buffer size that will be applied to VST plug-ins. The following special values apply in this case:

0: The program sets the buffer size.

-1: The ASIO buffer size is used

The VST buffer size is equal to the standard VIP buffer size. To use the ASIO buffer size in the Hybrid Engine, enter the value "-2" or select the "Forced ASIO buffers (Hybrid)" preset. The "Forced VIP buffers (use UAD & Powercore in economy engine)" setting diminishes latency with UAD/Powercore plug-ins.

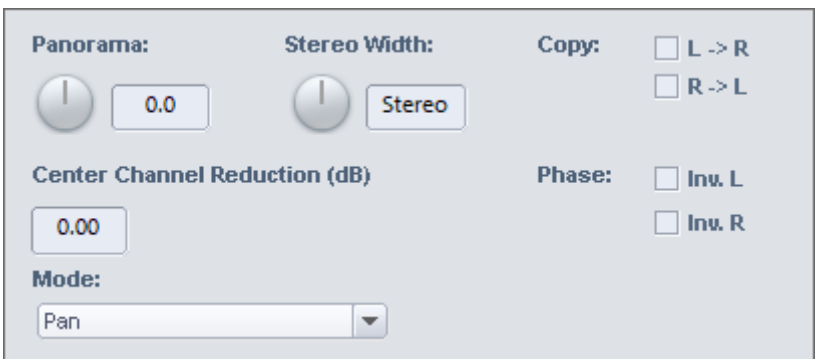
VST automation resolution

The Hybrid Engine uses the ASIO buffer size. To achieve shorter automation times in the "Economy Engine" (which works with VIP buffers), enter a corresponding lower value here.

You can also activate the "Start all object-related plug-ins at play start" option. Some plug-ins cause delays when switched on; in this case, these plug-ins should be activated before starting playback, otherwise errors may occur when the corresponding objects are played back.

Stereo Panorama/Stereo Editor

Right-clicking on the panorama control in the arranger, mixer, track editor and object editor allows you to change the panning between the left and right channel for each track/object in the **Panning Dialog** dialog, as well as other settings that affect the stereo image and phase.



With mono tracks, the signal is converted from mono to stereo in the signal flow at the position of the panorama controller. Plug-ins or effects can still be added at this position. The routing positions can be changed in the Effects routing dialog (view page 768).

"Stereo Panorama" Dialog Options

Panorama: Here you can set level distribution between left and right.

Stereo width: Adjusts the stereo base width for each track. No change occurs in the middle position. Towards the left, the stereo signal is gradually replaced by the center signal (mono sum of L+R), thus reducing the stereo base. If the slider is set to the leftmost position, you will only hear the mono signal. To the right, the side signal (difference from L-R) is mixed in, thus increasing the stereo base.

Detailed information about this is available in "Effects -> Stereo / Phase -> Multiband Stereo Enhancer (view page 963)" With the Multiband Stereo Enhancer you can edit the stereo bandwidth in three different bands frequency-selectively.

Copy: Here you can change the channels assignment. If you activate "**Copy L > R**", only the left channel will be audible on both sides; "**Copy R > L**" has the same effect for the right channel. Activate both options so that the left and right channels are swapped.

Center channel damping (panning law): In order to balance volume fluctuations while panning (particularly for automation) it is quite common to additionally dampen the track volume at the center position, while no additional damping occurs if the pan controller is moved to the left or right limit. The setting depends on the audio signal. Usual settings are:

- **0 dB**: This setting is recommended for stereo material. No damping occurs at the center position, i. e. if panned to the right, the right channel volume remains the same, if panned to the left the left channel remains unchanged. The audio material is not changed if set to the center position. For mono signals, this value may result in an increased volume if the signal is placed in the center.
- **–6 dB**: This setting can be used for mono tracks. If set to the middle, the level of the right and left channel is reduced by half.
- Common setting for automating stereo signals are **–3dB** or **–4.5 dB**.

Phase: Here you can invert the phase.

Mode: Here you can determine whether the panorama should be calculated after - Pan - or before - Pan (StEnh post pan) - the stereo width. In addition you'll also find the options "2 Channel Panorama" and "2 Channel Volume":

- **2 Channel Panorama:** If you activate this mode, you can use both knobs to control the panorama of the left and right channels separately.
- **2 Channel Volume:** If you activate this mode, you can use both knobs to control the volume of the left and right channels separately.

The parameters are processed in the following order: Copy -> Stereo Width -> Phase Invert -> Panning/Panning Law

Additional options for track panning



Use the + button to assign additional outputs (hardware or buses) to a track. The outputs can be switched on and off individually. For each of these outputs there is a separate panorama processing. This allows you, for example, to create a stereo and a surround mix in the same project. Click on one of the tabs to open the corresponding panorama editor. If the track is routed to a surround bus or master, the Surround Editor (view page 483) is displayed in the dialog accordingly.

In the Stereo Editor there are some additional options:

All Tracks: Use this function to apply the panning law to all tracks/channels of the project. In the menu you can select whether the panning law should be applied to all tracks or to all tracks except buses. Also the option " Affects primary output only" (see below) can be transferred to all other tracks with this button.

Affects primary output only: If the track is routed to multiple outputs (e.g., stereo and surround master), and this option is active, the panorama is applied to the primary output only. This prevents panorama processing from being done twice if a track is routed to another output. In FX routing, panorama processing is moved to the very end of the effects chain, even behind the AUX/Output Post, the standard output for additional outputs.

Follow track panorama: If the previously mentioned option is active, the signal of the track is sent to the additional output(s) without panorama processing. If panning is also to be applied to the additional output, you can use this option to make the settings in the Panorama Editor follow those of the primary output.

MS-Processing

"MS" stands for "Middle/Side" and describes the process by which the channels are separated not into left and right, but into middle ("M") and side ("S").

MS recordings: With MS microphone setups files will be received after recording with separate middle and side parts, rather than a left and right channel. With the stereo panorama dialog's presets provided you can edit MS recordings (here we assume that these are available as a stereo file with M left and S right).

MS processing of mono original material: To extract the correct stereo picture from an MS file, duplicate the output material and save it as a new track. For the first track, select the "Left channel only" preset. This results in the M signal being played in mono

only. On the second track, use the preset Side signal (Stereo) (from M/S source) to only play the side signal in stereo, i.e, left +side and right -side. Both tracks can be mixed together each at 0 dB

MS processing of stereo original material: Another requirement is the independent editing of mids and side parts even if the output material is available in stereo format. To do this duplicate the output material and save it as a new track.

"Mono (get mid signal from stereo source)" extracts the mid part from the output material as the first track, while "Side signal (Stereo) (from stereo source)" extracts the side part in stereo as the second track. Here you can subsequently mix both tracks each with 0dB.

Advanced MS processing: In Sequoia there are additional options for processing stereo signals in mono tracks.

Mono (get mid signal from stereo source): In this case, only the mid component of the stereo signal is used in the mono track.

Left channel only: In this case, use only the left component of the stereo signal used in the mono track.

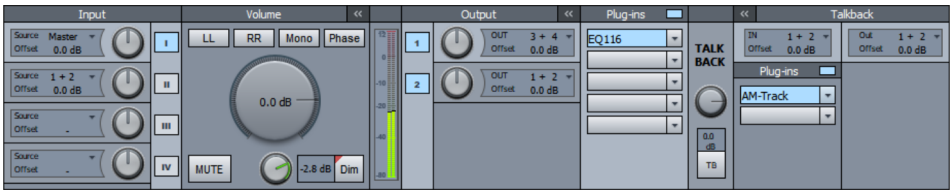
Right channel only: Use only the right component of the stereo signal used in the mono track.

Side signal (mono) from stereo source: Use only the mono side component of the stereo signal used in the mono track. This way, you can apply all subsequent object and track effects to the side part as mono parts.

Side signal (mono) from stereo source: Use only the mono side component of the stereo signal used in the mono track. This way, you can apply all subsequent object and track effects to the side part as mono parts.

Convert side signal (mono) to stereo: Converts the mono side component to stereo.

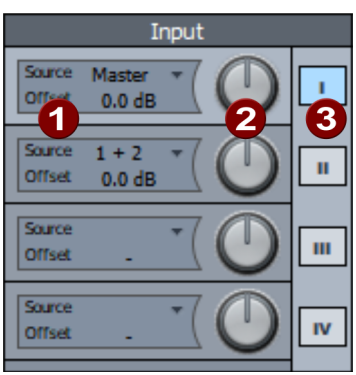
Monitoring section



The monitoring section provides a separate workflow for monitoring an input, bus or master. It is possible to switch between four different inputs and two different outputs, each with separate level settings and effect chains, for example to realize monitoring on different pairs of speakers (home system and studio monitors). There is also an independent talkback path for communication between the recording booth and studio.

Note: Despite the similarity in its name to track monitoring (view page 596), this is another function independent of it. Track monitoring means listening to an input signal of a track, the monitoring section is used for general playback control of the program.

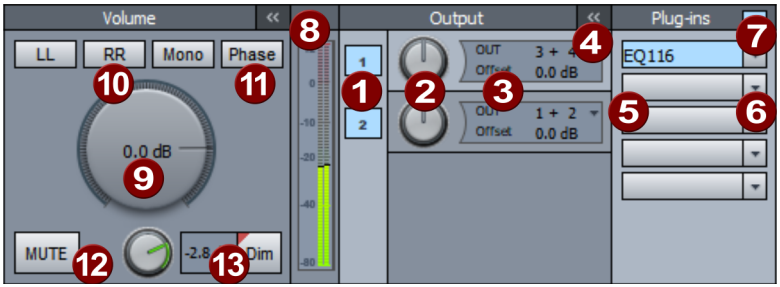
Input



1. **Select Source:** Default source is the Master, but you can also select any input pair or (stereo) bus as input source.
2. **Input Offset:** Here you can adjust the level of the input signal independent of the selected output.
3. **Switch Source:** You can switch between 4 different sources.

If the input is the Master or a bus whose output is routed to the same output device as the output of the monitoring section, the signal is not reproduced via both ways, but the monitoring section has priority.

Volume/Output/Plugins

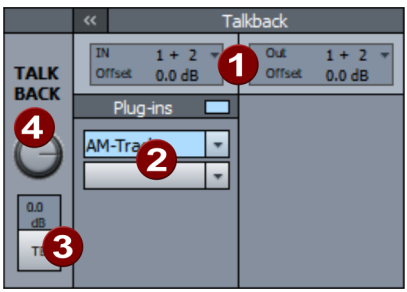


1. **Output selection:** A separate output device can be set for each output. All settings in the Volume and Plugins panels can be adjusted separately for both outputs.
2. **Output level offset:** The output level can be trimmed here per output.
3. **Output selection:** Select the output device here. The first output device corresponds to the device that can be set in the Monitoring section of the mixer (view page 380).
4. The plug-ins panel can be hidden with this button.
5. **Plug-in slots:** The plug-in slots work like those in the mixer, track or object editor. A left click on an empty slot opens the Plug-in Browser, a left click on a loaded plug-in deactivates the plug-in, a right click opens the plug-in window.

   You can even load more than two plug-ins, but only the first two are displayed in the slots. The other ones can be reached via the FX routing dialog.
6. Use the arrow to open the **plug-in menu**.
7. A left click on this button deactivates the entire plug-in chain, a right click opens the **FX routing dialog**.
8. The **Peak meter** shows the level behind the large volume control, but before the output offset.
9. **Volume control:** Here you set the monitoring volume.
10. **LL, RR, Mono:** Use these buttons to adjust the mono stereo signal distribution. With **LL** the left input signal is reproduced on both output channels, with **RR** the right signal. **Mono** sums right and left to a mono signal, which is then also reproduced on both output channels.
11. **Phase:** Inverts the phase of the right input signal to compensate for possible phase shifts.
12. **Mute**
13. **Dim:** A click on the button reduces the volume by the amount set on the knob.

Talkback

With the talkback function, you can set up an additional communication path between the recording booth and mixing console.



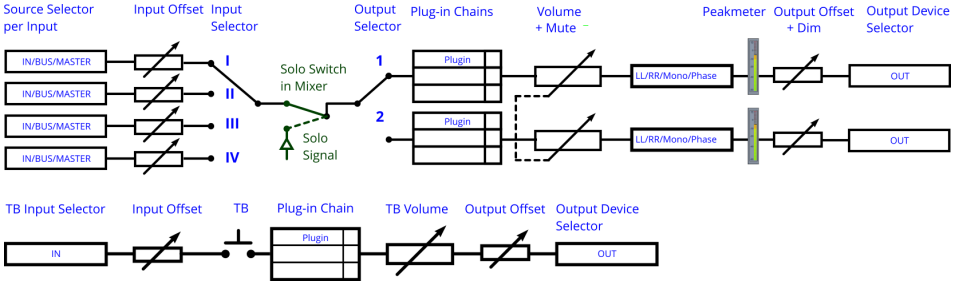
1. **Input/Output:** Here you define the input and output device for the talkback.
2. **Plug-ins:** The usual plug-in slots are also available for Talkback.
3. **Activate talkback**
4. **Talkback Volume**

Keyboard shortcuts for the Monitoring section

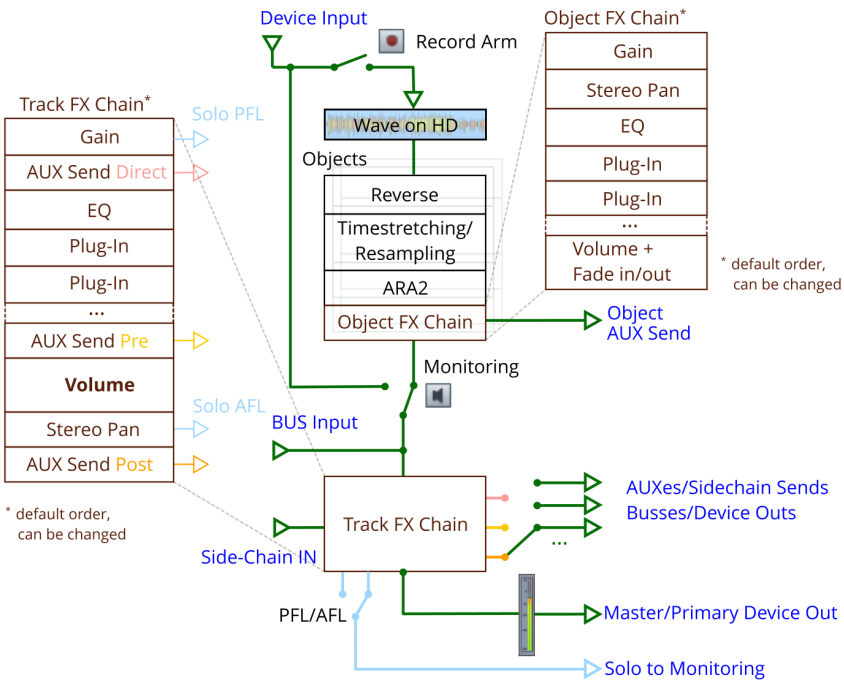
To be able to execute the most important monitoring functions quickly without mouse clicks, there are some keyboard shortcuts especially for the monitoring section. These only apply when the monitoring window is open and focused. To use them, you must therefore have clicked into the window at least once after opening it.

Alt + 1..4	Select Input 1-4	**H**	Invert phase
1, 2	Select Output 1 or 2	**B**	Open Plug-in browser (Monitor)
Alt+M, 0	Mute	**Shift + B**	Open Plug-in browser (Talkback)
D	Dim	**P**	Plug-ins on/off (Monitor)
L	LL (Input L to Output LR)	**Shift+P**	Plug-ins on/off (Talkback)
R	RR (Input R to Output LR)	**T**	Talkback (momentary)
Alt+N	Mono (Input L+R to Output LR)	**Shift + T**	Talkback on/off

Monitoring section signal flow



Mixer Signal Flow



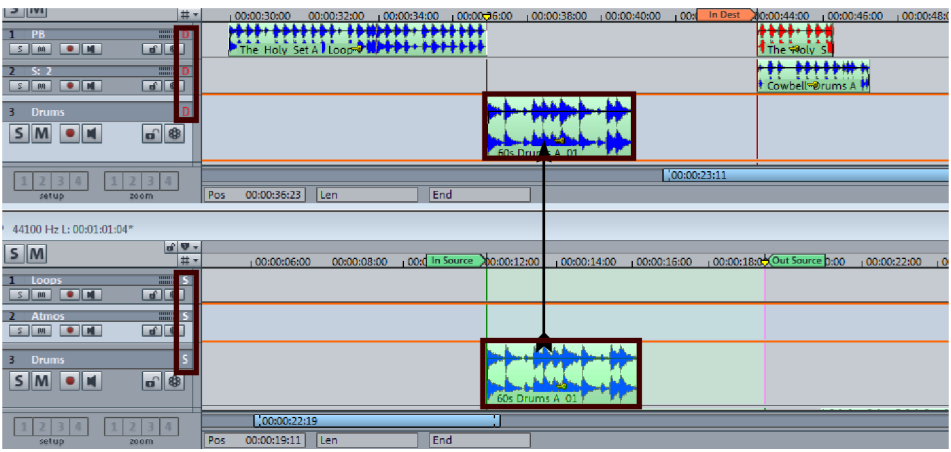
Remarks:

- A track can have any number of AUX send, sidechain send or output routings to busses or hardware outputs, for each routing one of the three positions in the track effect chain can be specified.
- The Record Arm and Monitoring switches are dependent on each other depending on monitor switching behavior (view page 602) and playback state.
- The bus input of a track is only active when the channel operates as an AUX/submix bus and always receives the sum of all aux sends or routings of the other tracks, regardless of the playback state.

Source Destination Cut

Source Destination Cut is an important editing function in Sequoia and is particularly useful for classical music productions, installations and current productions in which editing operations need to be carried out during the running recording.

The Source Destination Cut workflow is especially effective and intuitive because it connects the advantages of a linear audio editing studio with those of a non-linear hard disk system. Two display areas let you view and play back source and destination tracks simultaneously and independently of each other. The lower monitor area (source) displays the source material, i.e. all of the tracks recorded (including all takes).



The upper area contains the target tracks (destination). Here the edited material can be arranged using the mouse or the keyboard. Both areas can be navigated, zoomed and played independently. Control of the whole project is retained by being able to navigate to source material while playback is running in the destination area.. The actual editing takes place with markers.

In and out markers in the source material are used to indicate that the material following the edit command at a marker position should be copied into the destination track.

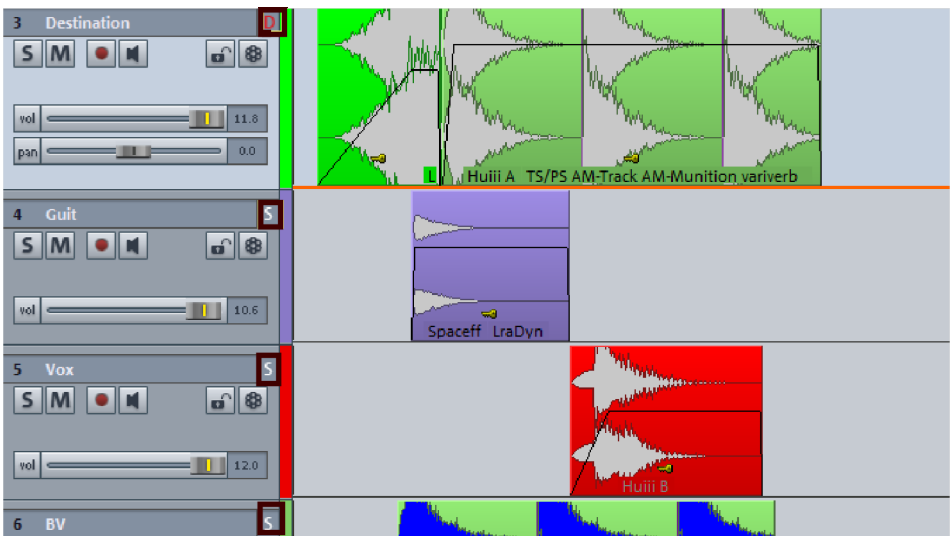
This function can be used to quickly perform editing processes that would otherwise be very time-consuming. For example, you can use 4-point editing to replace a section of your multitrack project with a slower version of the arrangement. With this process the subsequent audio material is placed on the new cut borders synchronously over all tracks.

You can use Source Destination functions in any multitrack project. There are also additional modes available, i.e. S/D Special Mode (view page 409) and Multi-Source Session (view page 407) and Multi-Synchronous Cut (view page 411, view page 422).

Source Destination Cut - Overview

The source track indicates the audio track with the source material. In this track you can set In-point (Source) and Out-point (Source) markers.

The destination track contains the audio material arranged from the source tracks. Here you can set an In-point (Destination) and an Out-point (Destination) to insert cut passages between the respective markers.



The definition of a track as the source or destination track is done by clicking the black arrow to the right of the track name in the track editor or track info box. In the context menu that opens, you can set the number of bundled Source Destination tracks under "Source Destination track number".

Any track can be the source and destination simultaneously, e.g. for cutting audio from a later location to insert into an earlier segment, or to select many takes at once.

An In-point marker defines the starting point of a cut. An Out-point marker defines the ending point of a cut.



The In-point and Out-point markers can be set in the toolbar; in this case, the following keyboard layout applies:

In-point (Source)	Ctrl + Home
Out-point (Source)	Ctrl + End
In-point (Destination)	Shift + Home
Out-point (Destination)	Shift + End

Delete existing cut markers with the "Delete destination points" and "Delete source points" commands, or reset the markers to overwrite the existing markers.



Note: If playback stops automatically after each object, you are probably in Cue Mode (view page 828). To clear this playback mode, click the corresponding button or go to the menu "Play/Rec -> Playback Mode" to deactivate this option.

General Cut Commands

Insert cut: This command executes the cut process. This function overwrites the existing audio material in the destination track.



Shortcut key: F9

Insert with ripple: This command also executes the cut process. This function moves the existing audio material in the destination track over to the right.



Shortcut key: F10

Delete and ripple: This command deletes the cut range in the destination track and closes the resulting gap by pulling the audio material following it backwards.



Shortcut key: F11

Delete silence: This command deletes the cut range in the destination track. Audio material found following it is not shifted.



Shortcut key: F12

You can couple the functions indicated above with various crossfades or special insert cuts. Cuts containing a fade-in or a fade-out at the start or end of an object are called "blackfades". These blackfade cuts are found in "Edit -> Source/Destination Cut" (view page 737).

Other important edit commands:

Insert with time stretching: In this case, the source material is stretched between the destination in-point and destination out-point. Existing material in the destination track is replaced with the material from the source track.

This enables the timing of drum loops to be adjusted to the existing arrangement, without the need to apply complicated operations. Time stretching is applied non-destructively.

Keyboard shortcut: Alt+F9

Insert at source position: You can use this cut command in synchronous cut, if the absolute time position is already set in the source material. You don't need to set any Destination markers, markers that were previously set will be ignored. The cut retains the time position of the source material.

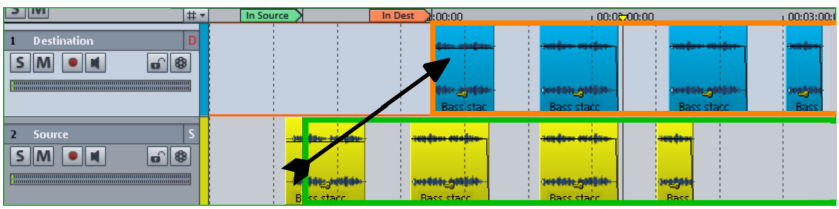
Shortcut: Ctrl + Alt + F9

2-Point Cut

In a 2-point cut, one cut marker is set for the source track and one for the target track. The source material will be transferred according to the set cut marker into the target track. The 2-point cut offers you three different options:

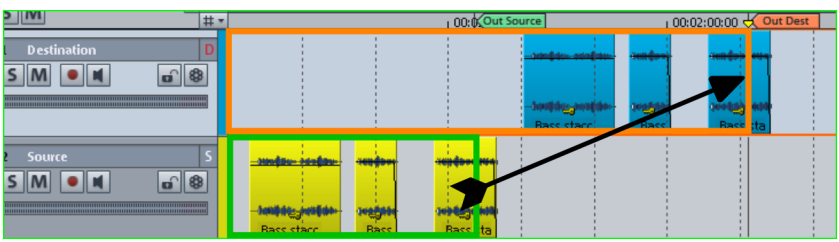
1. You define the Source-in-point and the Destination-in-point.

The cutting command ("Edit -> Source destination cut -> Execute insert cut"; shortcut: "F9") inserts the complete audio material after the source-in point of the source track after the destination-in point on the destination track.



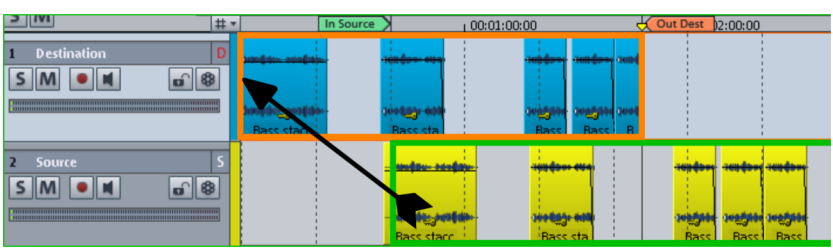
2. You define the Source-Out-Point and the Destination-Out-Point.

The cutting command ("Edit -> Source destination cut -> Execute insert cut"; shortcut: "F9") inserts the complete audio material before the source-out point of the source track before the destination-out point on the destination track.



3. You define the Source-In-point and the Destination-out-point.

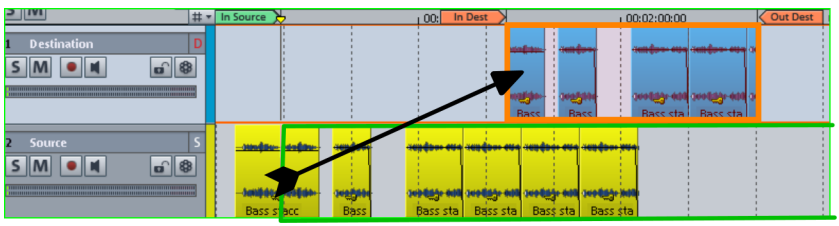
The cutting command ("Edit -> Source destination cut -> Execute insert cut"; shortcut: "F9") inserts the complete source material after the source-in point of the source track beginning at the start of the destination track to the destination-out point of the destination track.



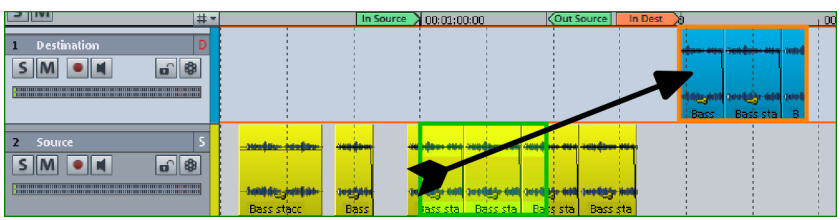
3-Point cut

In a 3-point cut, a range is defined by setting an in and out point on either the source or target track, while the third cut marker is set on the other track. One of the four cut markers is not set. This cut is applied using the length indicated by the in/out points.

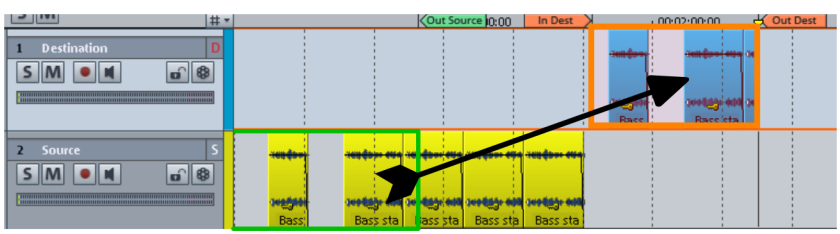
Case 1: If you have set the in and out points in the destination track, but only the in point in the source track the source material is cut from the source in point into the destination track between and according to the length set by the destination in and out points.



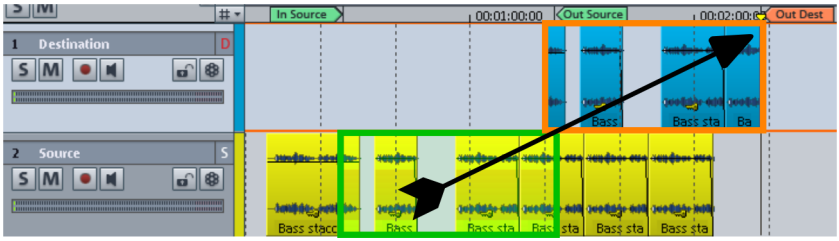
Case 2: If you have set the in and out points in the source track, but only the out point in the destination track, the source material between the source in and out points is cut after the destination out point in the destination track.



Case 3: If you have set the in and out points in the destination track, but only the out point in the source track the source material in the source track before the source out point is cut to the destination track between and according to the length set by the destination in and out points.



Case 4: If you have set the in and out points in the source track, but only the out point in the destination track the source material between the source in and out points is cut before the destination out point in the destination track.



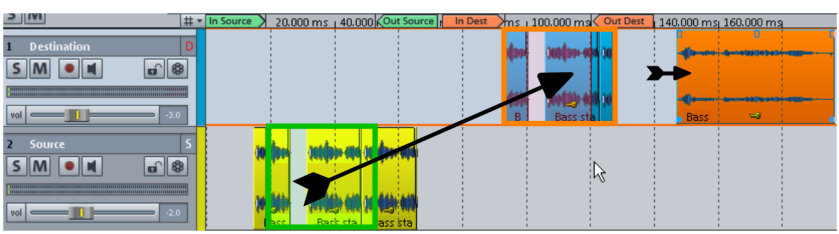
4-Point Editing

"4-point edit" allows you to set all 4 cut markers. While doing this you work with two ranges, the source range and the destination range. The following applies:

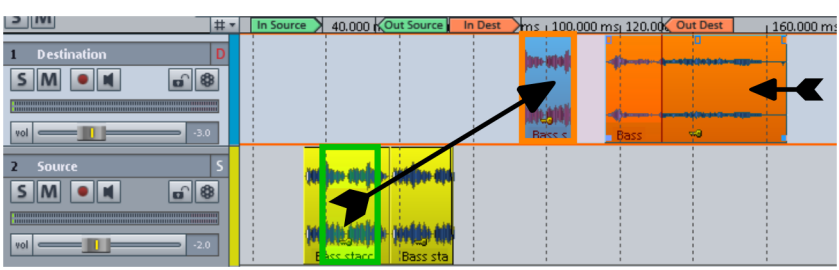
- The length of the cut is determined by the In-point/Out-point pair of the source cut marker.
- The range between the In-point and Out-point of the destination cut marker is replaced by the source cut material.

Please note:

- If the selected destination range is shorter than the source range, the existing destination material after the destination out-point will be shifted to the right so that the cut material fits in completely and the subsequent destination material will not be overwritten.



- If the selected destination range is longer than the source range, the destination range will be cut out completely; the audio material following the destination out-point will be shifted to the left so that it connects directly with the inserted cut range.



Note: 4-point editing can also be performed as a stretch cut ("Edit" menu, "Source Destination Cut -> Insert with time stretching"; Shortcut: Alt + F9). In this process the contents of the source range are stretched or compressed to the length of the destination range. The main application of this is the synchronous cut. Please note that time stretching has its limitations and can result in audio artifacts. Sequoia features a preset stretch factor range of 0.5 to 2.0. If you want to apply more severe time stretching for the inserted cut range, you can always do this in the Object Editor or in Pitch Shift/Time Stretch Mode.

Source Destination Cut - Special Function

You can also define source and destination markers in various projects, e.g. both destination markers in the current virtual project and both source markers in an audio file. In this case an audio file can only be used as the source.

If you want to perform editing between different VIP projects, set up a Multi-source session first or add source projects in the Source List Manager (view page 207).

Multi-source session (S/D cut with multiple projects)

The Source Destination Cut for multiple projects (multi-source session) enables cuts from one project to be applied to another with the help of Source Destination Cut functions.

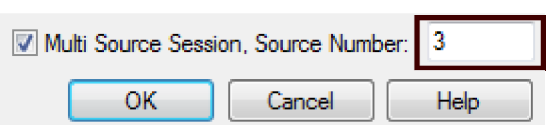
Multi-source sessions feature the following properties:

- Multiple projects exist; a destination project and at least one source project.
- Partial ranges of the source projects can be applied to the destination project with the help of the conventional Source Destination Cut tools.

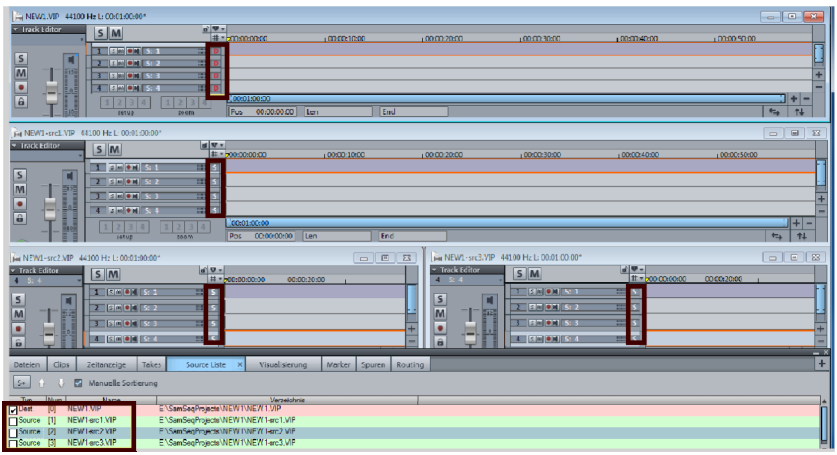
In a Multi-source session it is easier to sort audio material, e. g. source projects for cues, dialogs, MuSyC VIPs or several virtual projects for a CD compilation.

Multi-source session – Create new project

In the "File" menu, select "New Virtual Project". Now set a check next to "Multi-source session" and enter the number of desired source projects. Confirm this by clicking "OK".



A destination project and the desired number of source projects are then created by Sequoia.



Notes: The individual windows can each be maximized. "Page down" and "Page up" let you switch between the last active source project and the destination project. You can also switch between individual source projects and destination project via the "Window" menu.

Open new source project or create one retroactively

To integrate a new source project into an existing multi-source session retroactively, select the entry "Source" in the context menu of the track box overview or track editor for the track. Assign the project to an opened destination project via source list (view page 207).

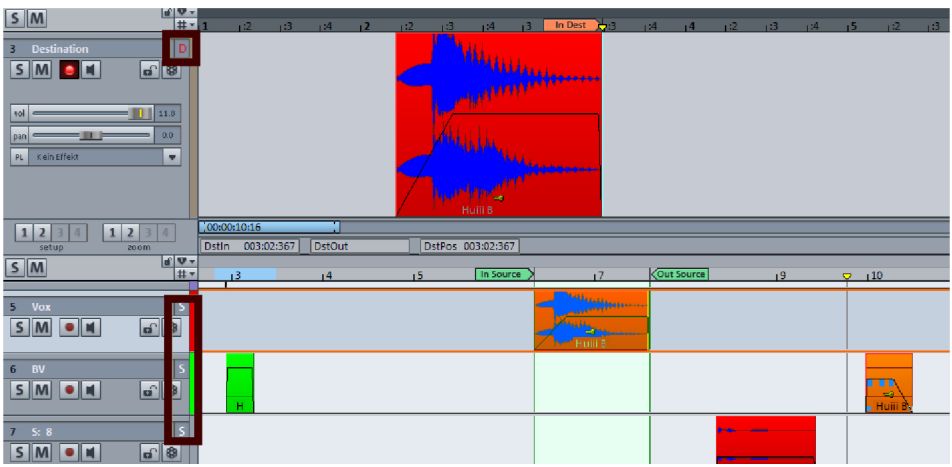
The available source tracks in the newly created project enable the cut to be combined as usual and then added to the destination project. If multiple source projects are opened simultaneously, it is possible to compare the different cuts prior to application. The command to apply to the destination project may be accessed via "Edit -> Source Destination Cut (view page 737)".

Select source for the cut

Normally, material is cut from the most recently active source project. In all other projects, the source markers will appear gray. You may also select the source project where the cut should be made via the "Source list (view page 207)".

Source Destination Cut Mode (S/D special mode)

The "Source Destination Cut mode" function from the 4-point edit bar switches the project window to a special mode consisting of two separate track ranges, whereby the destination track is located at the top and the source tracks are at the bottom. This mode is primarily intended for editing ready-mixed stereo and surround projects. In order to record an already available audio track from your project as a source/destination track, mark this track as the source/destination before switching to the special mode.



The source destination cut mode has the following features.

- If you switch a project to S/D special mode for the first time, a new track range is attached which then becomes the source track range.
- In both track ranges, you can change scroll positions and zoom steps independent from each other.
- The recording and loading of new audio files takes place automatically in the lower source track range.
- The playback marker and the range markers can be selected independently from one another in both track ranges.
- The currently active track range is displayed highlighted. You can select a track simply by clicking it or by using the "Page up" and "Page down" keys.
- Playback is always activated for one track. This way you can switch to the track range and make changes without influencing playback.

Combining raw material in the source

The raw material can include several recording takes that have been edited together. Recording may be done directly in the source track or an existing audio file can be loaded into the source track.

Select important positions in the raw material

Select the instances of an instrument in the various recorded takes so you can compare them more easily.

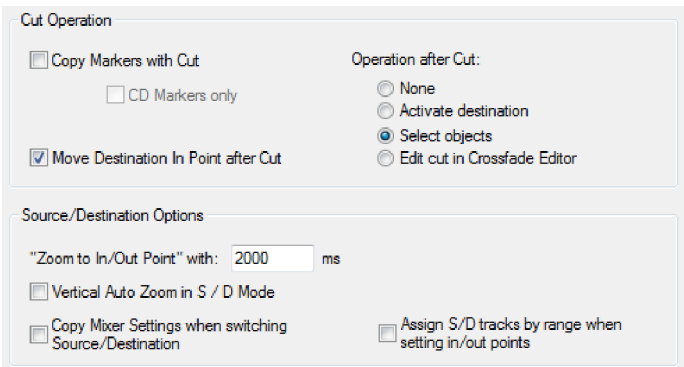
Project markers on the source track timeline and audio markers coupled directly with the audio marker can help with the selection.

Cut commands/Setting In-points and Out-points

All cut commands (view page 402) and the setting of cut markers (view page 403) also apply in S/D special mode.

Source Destination Cut Settings

The settings dialog for Source Destination Cut can be reached through "Edit" > „Multi-synchronous cut" > „Source Destination Cut Settings" (view page 742).



You can apply the following cut operations here:

- Copy markers with cut: The set markers will be applied to the destination during cutting.
- CD markers only: Only the CD markers in the destination will be applied.
- Move the In Point destination after the cut: This option has the effect of placing the In Point destination for each cut at the end of the current cut. You can cut additional material immediately without having to reset the destination in point each time.

Action after cut:

- No action: This option avoids switching the view to the destination, which takes the focus away from the source.
- Activate destination: This option activates the destination after the cut.
- Select objects with cut: After a cut is made, the object cut into the destination is selected.
- Open in crossfade editor: After a cut is made, the last object inserted into the destination opens in the Crossfade Editor (view page 425).

Source Destination Settings:

- The setting „Zoom into In/Out-Point with ... ms“ refers to the command „Edit“ > Source Destination Cut > Zoom into In/Out-Point (view page 739)“. Specify the time in milliseconds for the zoom display for the in/out-point. The shortcut "Ctrl + Alt + Shift + Page up/Page down" moves the playback marker in the currently selected track by the zoom factor to the in/out-point. The preset value is a time of 2000 ms.
- Vertical auto zoom in S/D mode: This option activates the automatic vertical zoom function in Source Destination Cut mode. In this case, the tracks are optimized vertically depending on the window size, i. e. the visible wave section is displayed normalized to 100%.
- Copy mixer settings when switching Source Destination: This option ensures that changes made to the mixer settings for a source or destination project within a multi-source session are applied to all other multi-source session projects. Make sure that the projects contain the same number of tracks.
- Assign S/D tracks by range when setting in/out-points: This option includes tracks that have not been defined as the source or destination tracks in the cutting process. If you drag a range into a track that was not previously defined as the source or destination track, the placement of a source or destination marker at the edge of a range will define this track automatically as the source or destination track (depending on the marker), and this will be indicated accordingly in the track header.

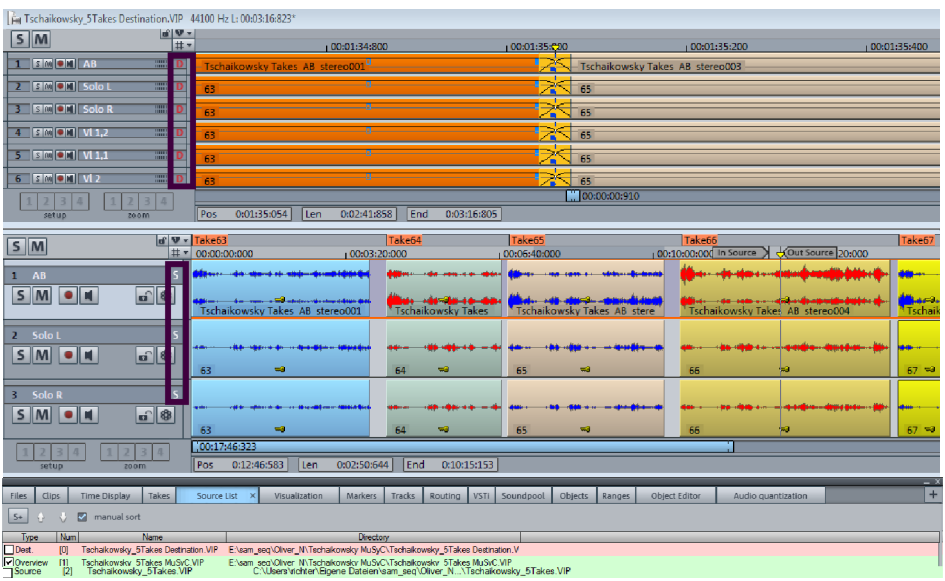
MuSyC – Multi-synchronous cut

The multi-synchronous cut (MuSyC) makes navigating and cutting complex multi-source sessions (view page 407) featuring numerous takes quick and easy, which is usually the case with classical recordings.

The principle of the multi-synchronous cut is this: After the audio material is analyzed, all takes with the same content are displayed directly one below the next, and can be compared and cut without manually moving the playback marker. A purely visual type of time stretching process takes place to display the take's waveform synchronously. Next, cuts from different recordings can be made at the same musical position.

MuSyC Overview

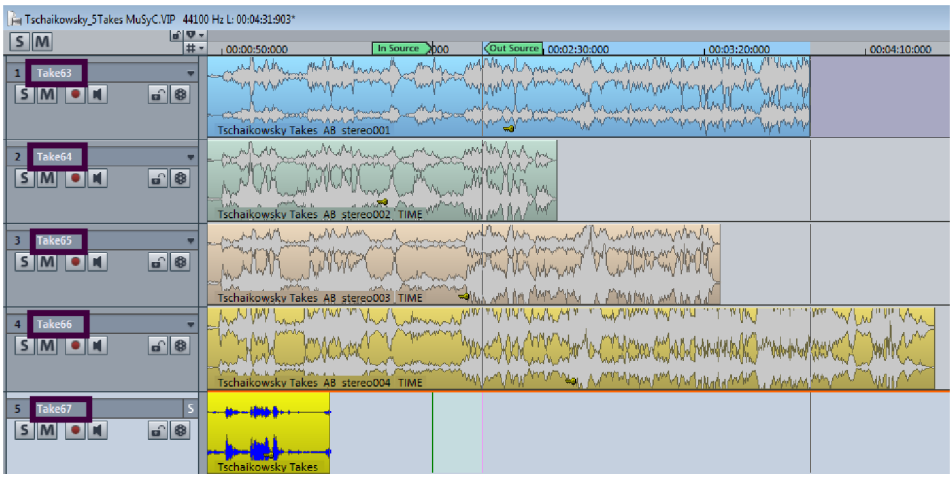
Multi-synchronous cutting, or "MuSyC", completes and supports the conventional method of editing an audio recording, especially for classical music. Imagine MuSyC as a wizard for the editing process. You can work with Source Destination Cut as usual,



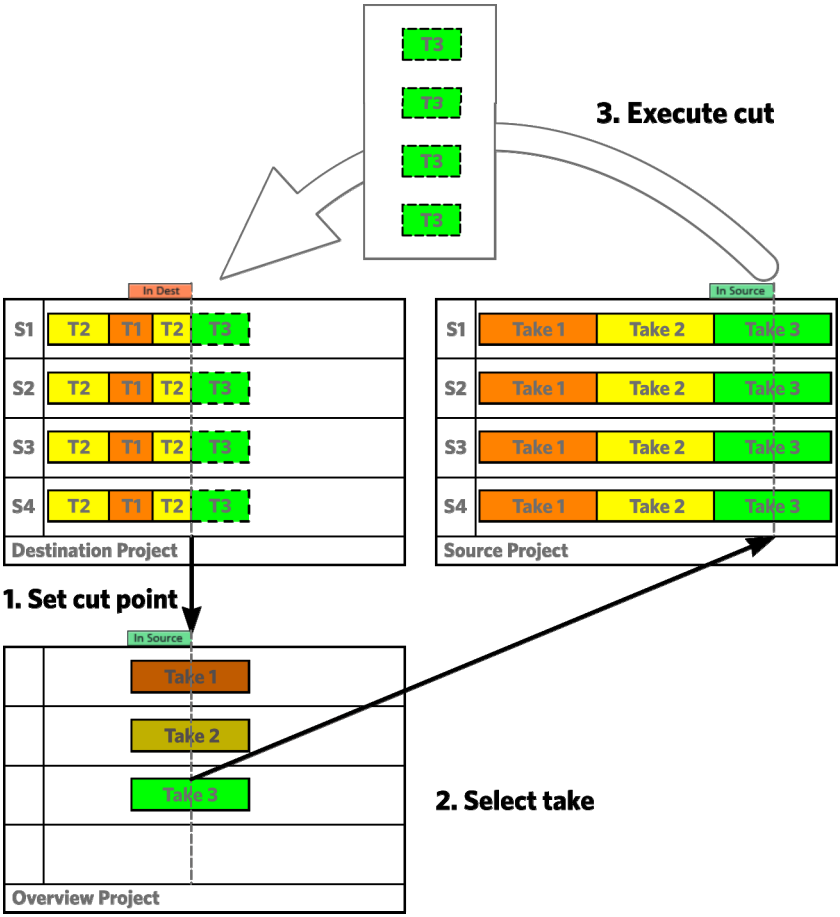
but the MuSyC functions are also available to you. The goal of MuSyC is to minimize the time spent searching for similar passages of music and accelerate the editing process.

MuSyC expands working with source and destination projects with an overview project. If a section of music has been recorded multiple times, then this will normally be available in the source project (or just "source") as different takes.

The overview project provides the primary navigation in the source. The takes are represented by one object each. Musically equivalent content is arranged in ascending order and displayed in a temporally ideal manner. This allows you to switch between takes exactly according to the music to keep an overview of alternative recordings.



If the active track (i.e. the active take) is switched in the overview project, the playback position and the "source points" will be automatically synchronized in the respective source project. If you make a cut, the audio material will be inserted from the source project into the destination project.



Pay attention to the following when using the multi-synchronous cut:

1. Work takes place in a multi-source project (multi-source session) (view page 407). The concept of destination and target project refers to the fact that the project in which the edit is created is a different one from that which contains the raw recorded material.
2. The recording is divided into individual takes. If a section of music has been recorded multiple times, then this will normally be available in different takes. If this isn't the case, open the take assistant to help split up the recording (Edit -> Multi-synchronous cut -> Take Assistant (view page 423)). The cuts are purely virtual and will not alter the raw data in any way.

MuSyC Reference Track

This reference track is synchronized with all of the recorded takes in the overview project. It provides a kind of master track for the multi-synchronous cut.

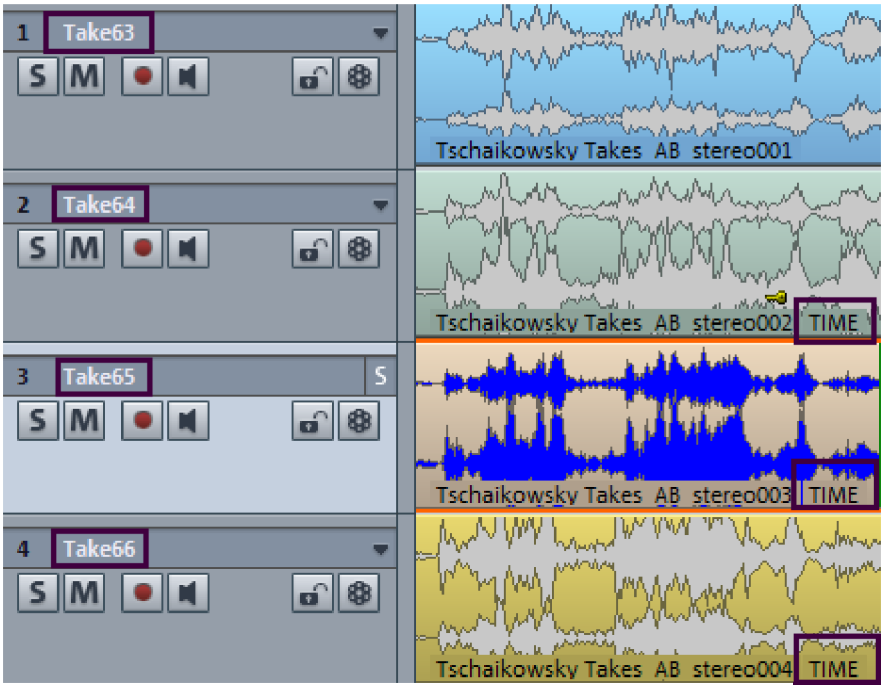
The reference track may contain any mono or stereo track from the source project. Typically, the track for the main microphone will be used as the reference track. A rough mix can also be created in a new track and this can be selected as the reference track.

Note: Make sure that the reference track contains mostly continuous material, i. e. material with no long pauses.

MuSyC Overview Project

This is not a regular VIP, and for this reason the MuSyC overview project doesn't show the time sequence determined by the take recording. The MuSyC overview project represents an overview mode for better management of existing takes. Objects are displayed on tracks in order to optimize take sorting and selection.

In the overview project, takes from the source projects are displayed as reference objects. In this case, takes with musically identical content are displayed above each other. Temporal shifts are balanced visually by the time alignment. Adjusted objects are indicated by "TIME".



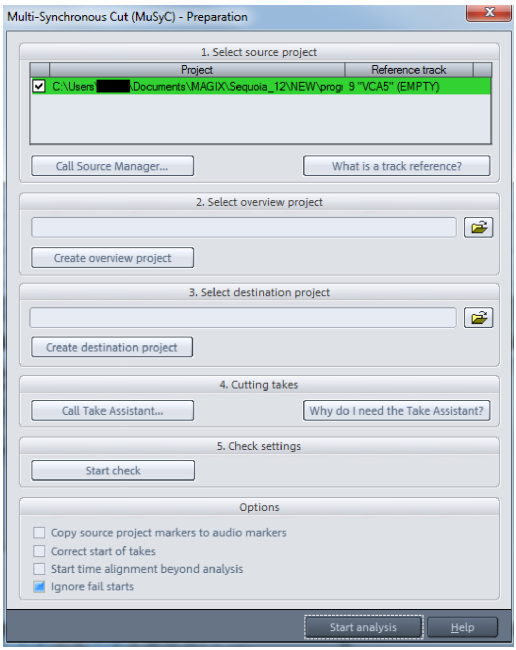
In the overview project, only one track will be active. All positions refer only to the takes in this track. If playback is started, only the take at the playback position in the active track will be played. The synchronized elements (cut markers, ranges, playback marker) in the source project also refer to this particular take only.If the track is switched, the elements in the source project will be included.

Note: You can also switch the active track during playback. This enables you to switch to another take in real-time while maintaining smooth and musical playback.

The synchronization between the source project and the overview project refers at all times to the respective object and is constantly updated. If necessary, the object can also be moved and edited.

Note: If objects are moved, their time references should be observed. Typically, time alignment must be performed again after a move is made (Edit -> Multi-synchronous cut -> Time alignment -> Align selected objects). Note that the the top object always qualifies as the time reference.

MuSyC Preparation



If you open the dialog for preparing MuSyC (Edit -> Multi-synchronous cut -> Multi-synchronous preparation), you can open the source projects and the source manager, select or create an overview and destination project, cut takes, and start analysis. The overview project applies takes from the previously selected (reference) tracks in the source projects. The "Project" field features the source projects with the respective names of the reference tracks. Double-clicking a source project field activates its project window.

To complete the analysis, press the button "Start analysis". After the calculation, the tracks in the overview project will be arranged so that cuts from takes that belong together are stacked above one another and takes that follow each other are placed in sequence. Empty tracks are deleted from the overview project.

Differences in playback position and tempo fluctuations between the individual tracks are compensated for visually. Nevertheless, the playback speed does not change; the adjustment only affects the graphical display.

- The first step is to create a multi-source session (view page 407).
- Open all of the projects that should be used as sources for the multi-synchronous analysis.
- Open the "Source List" manager and include all of the desired projects as sources by selecting the respecting project and pressing the button "Insert active source project to the list". Source projects that are not required may be deleted from the source list by selecting them and pressing the "Del" key.
- Open the dialog for preparing the multi-synchronous cut by going to Edit -> Multi-synchronous cut -> Multi-synchronous preparation.

To prepare the multi-synchronous cut, proceed as follows:

1. Activate your source projects and select the track of the respective source project that you want to serve as the reference track. The active track in the corresponding project applies as the reference track.

2. Click on "Create overview project" to open a new overview project.

3. Open an existing destination project or create a new one by clicking "Create destination project".

4. If your recording has not yet been cut up into takes, open the take assistant (view page 423) and place new cuts at the desired positions.

5. Click the Start check button. The settings will be tested. If all of the necessary settings have been entered, a green OK will appear.

6. Now start the analysis by clicking "Start analysis". Additionally required tracks will be created in the overview project

Options

Copy source-project markers to audio markers: Use this option to convert source-project markers into audio markers

Correct start of takes: This option automatically shortens the start of takes after analysis. Improved analysis results may be achieved if this is done prior to analysis. To prevent the time alignment at the beginning from being imprecise, the start of the takes cannot include seconds of silence at the beginning or unwanted sounds. Support for this task is provided by the take assistant (view page 423).

Start time alignment after analysis: This option starts time alignment automatically after analysis. If this option is left activated, the overview project can be checked and

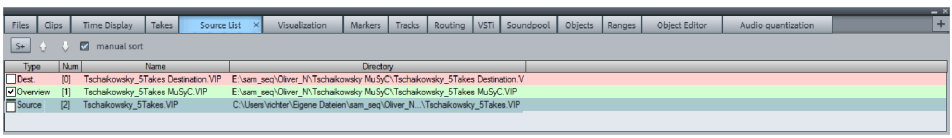
corrected as required after the analysis and before time adjustment is initiated (Edit -> Multi-Synchronous Cut -> Time Alignment -> Adjust All Objects).

Ignore false starts: With this function you can make sure that false start takes are removed from the analysis.

Multi-synchronous cut – Application

In terms of use, MuSyC corresponds with the Source Destination Cut (view page 400).

All editing functions that are used to copy the audio material from a source project into the destination project may also be used from the overview project. The overview project is essentially treated as a source project. In the source list manager, the overview project is referred to correspondingly.



The difference from classical Source Destination Cut is that the user doesn't work with the actual source projects, but rather with the overview project. The connected synchronization mechanisms make many manual navigation steps unnecessary.

Note: If playback stops automatically after each object, you are probably in Cue Mode (view page 828). To clear this playback mode, click the corresponding button or go to the menu "Play/Rec -> Playback Mode" to remove this option.

MuSyC Take Display

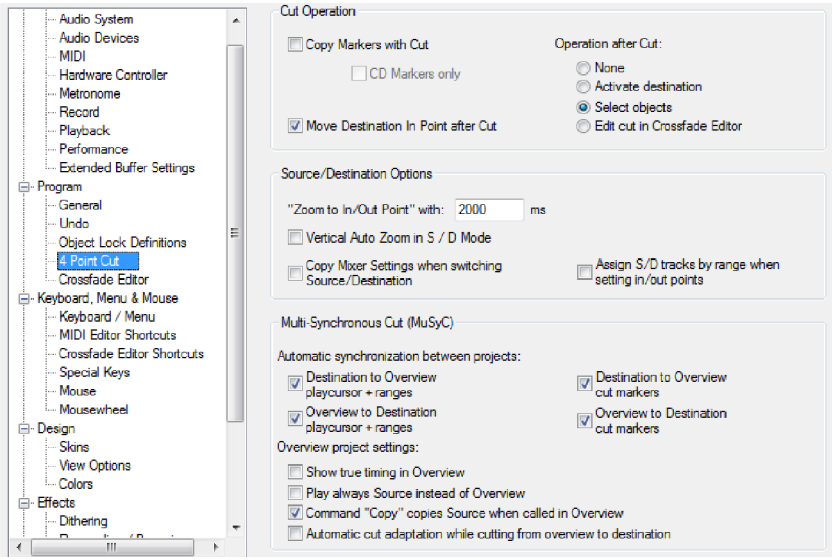
After analysis has been completed, you can see the take display of the reference track in the overview project. The following particulars apply:

- At all times, all matching alternative passages/takes are organized one above the other. This makes it unnecessary to search for equivalent passages of music.
- Changing the active source track also changes the takes during playback. Click in the track header or on an object to do this. This makes an A/B comparison possible for different takes. Normally, only the reference take is played back. In "Edit" > "Multi-synchronous cut" > "Play source" you can play back the source project instead. This mode can also be set in the system settings ("Edit -> Multi-synchronous cut -> MuSyC settings".
- If the playback position in the overview project is changed, the playback marker will move to that location in the source project automatically. This makes it possible to work with a direct connection to the source project at any time.

- If a range is marked or a source-in or source-out marker is placed, this will be reflected in the source project automatically. This removes the need to search for the exact position of the cut. If you make a cut, the source material will be copied from the source project into the destination project. The actual edit is therefore completely independent from the overview project and also functions as a cut marker via simple copying and insertion of a take range. In particular the option „In an overview project the 'Copy' command applies to source project“ means that, instead of the range from the reference take, the corresponding range from the source project will be copied to the clipboard. This behavior can be switched off in the system settings.
- The cut itself may also be improved with an automatic cut adjustment, which automatically opens the crossfade settings so that the cut cannot be perceived. To make these settings more precise, please use the crossfade editor (view page 425).
- If you change the range, the playback position, or a cut marker in the destination project, you can also adjust this in the source project and overview project. This synchronization function may also be turned off in the MuSyC system settings (view page 421).
- Manual correction of the assignment is also possible by simply moving the corresponding object in the overview project. The exact synchronization may also be repeated for the individual takes. Select the corresponding objects in the overview project and select Edit -> Multi-synchronous cut -> Time alignment -> Align selected objects. This ensures that objects from other equivalent musical sections remain synchronous.

MuSyC System Settings

The following settings can be adjusted by going to "Edit" > "Multi-synchronous cut" > "Source Destination Cut Settings...".



Automatic synchronization between projects:

Destination to overview playback marker + ranges: Synchronizes the playback marker and the range selection of the destination project in the overview project.

Destination to overview cut markers: Synchronizes the in and out point marker of the destination project in the overview project.

Overview to destination playback marker + ranges: Synchronizes the playback marker and the range selection of the destination project in the overview project.

Overview to destination cut markers: Synchronizes the overview project's in and out point markers in the destination project.

Overview project settings:

Display original length in overview project: If this option is active, the reference objects will be displayed in the overview project without time alignment, but the waveforms will be synchronized with the playback marker. This enables the actual time of the takes to be observed.

Play back source project instead of overview project: Always plays the source project instead of the reference object. "Edit -> Multi-synchronous cut -> Play source" plays back the source project just once.

In an overview project the 'Copy' command applies to source project: This option means that, instead of the range from the reference take, the corresponding range from the source project will be copied to the clipboard. Use this to move sections with "copy & paste".

Automatic cut adaptation while cutting from overview to destination: The crossfade settings for a cut from the overview project into the destination project are applied automatically.

MuSyC commands

The following commands are available via "Edit -> Multi-Synchronous Cut":

MuSyC preparation: Opens "Multi-Synchronous Cut – Preparations".

Take assistant: Opens the take assistant dialog.

The following commands may be accessed via "Object alignment":

Recalculate object position: This command arranges the selected object differently. If a range is also selected, then only this range will be considered for the search.

optimize project overview: This command provides a more compact display of the overview project. In this case, longer reference objects are arranged in the upper tracks.

Find similar range: The musical content of a range is searched for and marked in the overview project. If this command is executed without changing the limits of the range, then other located passages will be marked. This function is used to locate musical repetitions.

The following commands are located under "Time alignment":

Align all objects: With this command, display of all objects in the overview project is adjusted to reflect the uppermost object. Objects that are adjusted are indicated with "TIME".

Align selected objects: With this command, display of all selected objects in the overview project is adjusted to reflect the uppermost object. Objects that are adjusted are indicated with "TIME".

Reset selected objects: This command resets the temporal adjustment.

Show true timing: This corresponds with "Show true timing in overview" in the MuSyC system settings. More information is available under "MuSyC system settings (view page 421)".

Fast insert: "Fast insert" accelerates range-oriented work. A range must be opened up in the overview project first, or an in point marker must be placed and the cursor set to the end of the range that is to be inserted. If a fast insert cut is executed, then an insert cut will be made first. Next, the in point marker will move to the end of the range automatically.

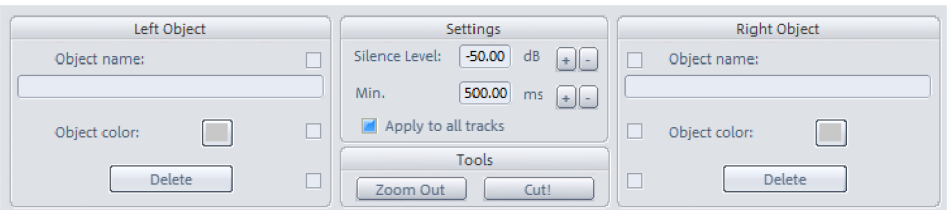
Play source: Plays the source project

Play destination: Plays the destination project

Activate overview (MuSyC): Activates the overview project

MuSyC settings: Accesses the MuSyC settings in the system settings (see below)

MuSyC Take Assistant



The take assistant makes it easy to navigate objects, place cuts, or delete objects.

- First, place the playback marker in front of the object that you would like to edit and then open the assistant. The playback marker will now jump to the next analyzed cut position in the object.
- Confirm with "OK" to place a cut at the playback marker location. All settings will be applied, i.e. silence level, minimum duration, object name, object color, etc. for the left and right object. The "left object" is the object component to the left of the playback marker and the "right object" is the object component to right of the playback marker. If you press "Cut" at this position, a cut will also be placed, and the settings for the left and right objects, i.e. object name, object color and the "Delete" function will be ignored.
- "**>>**" and "**<<**" provide navigation to the next or previous position without making a cut. All of the parameters will be preset automatically.
- The active track in the active project applies as the analysis track. The track and the project are displayed in the titlebar of the dialog window.

Silence level: This specifies a threshold value in dB below which the signal will be considered silence. The playback marker jumps according to this value to the potential cut positions when ">>" and "<<" are clicked.

Min.: This value specifies in milliseconds how long an object cut must be below the specified silence level before it is suggested as a potential cut position.

Overview: This makes the entire project visible in the selected section.

Cut: Places a cut at the playback marker location according to the settings made for silence level and min. duration.

Object name: Enter the name of the object here. This will only be applied if the check box is activated and confirmed with "OK".

Object color: The object's background color may be defined via this setting. This will only be applied if the check box is activated and confirmed with "OK".

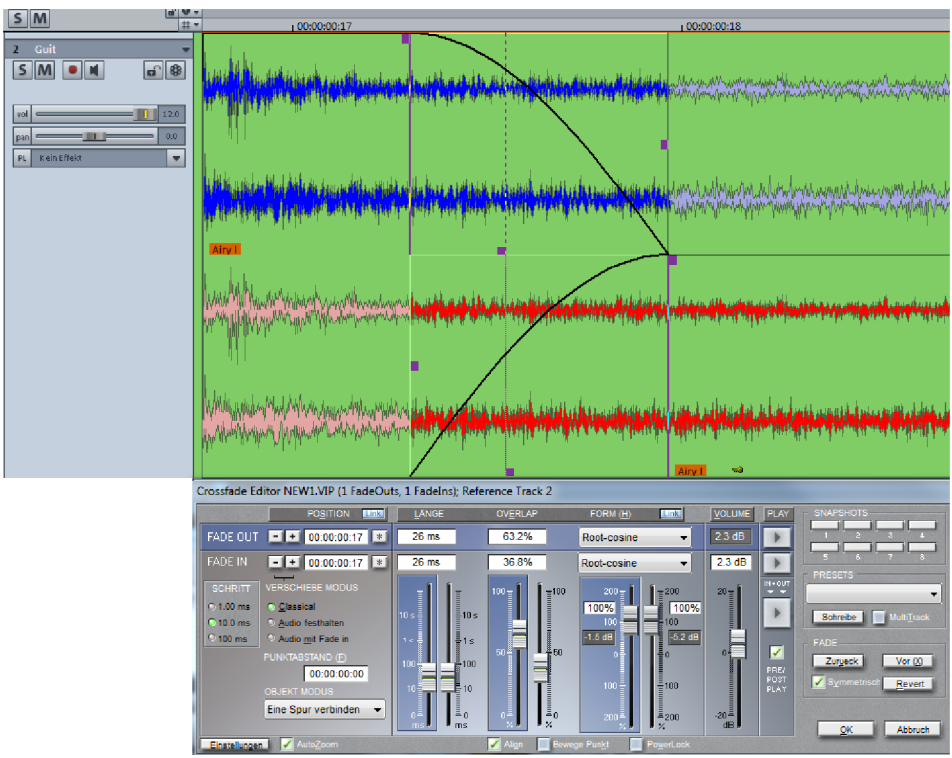
Delete: This commend deletes the selected object to the left or right of the playback marker position. The "Delete" command for the "Left Object" deletes the object range left of the playback marker and the "Delete" command for the "Right Object" deletes the object range to the right of the playback marker. This will only be applied if the check box is activated and confirmed with "OK".

Crossfade Editor

The crossfade editor is a powerful tool for working with fades in your project. It offers a wide range of options for creating transitions between objects and defining the precise alignment of the specific parameters of a cut. All of the function settings can be adjusted with the mouse or the keyboard.

Together with the Source Destination Cut (view page 400) it enables a completely cut point-oriented workflow.

The crossfade editor is opened by clicking on the toolbar button or through "Edit -> Crossfade editor". Alternatively you can also use the corresponding button in the toolbar. The preset keyboard shortcut for this is "Ctrl + F".



Activate an object or drag out a range (view page 81) over a fade before opening the crossfade editor.

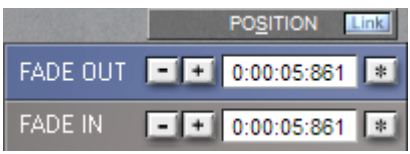
The crossfade editor is closed after the settings in the Crossfade Editor dialog are confirmed by clicking "OK" or discarded with "Cancel".

By opening the crossfade editor, the VIP is also switched into a special state. Those objects selected in the section around the cut point will be opened with a zoom display. Every crossfade to be edited will be divided into two vertically arranged areas, the upper containing the fade out, and the lower containing the fade in. In this way even several active objects positioned on top of each other can be displayed.

The crossfade editor is also suitable for multitrack editing (view page 437). For synchronous editing, special movement modes have to be set to help avoid unwanted offsets between tracks.

Crossfade position settings

The position of the "Fade out" reference point may be set freely using the minus and plus buttons. If the "Link" button is activated in "Lock audio" move mode, the fade in and fade out curves move synchronously. The "*" button places the fade at the cursor position. The audio material's fade out material is generally locked on the timeline to prevent unwanted offsetting of objects.



In "Classic" move mode, the "Fade in" reference point may be moved along the time axis and along the audio material relatively freely.

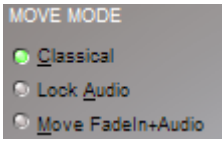
The positioning link button (shortcut: "I"): causes a fade-in and fade-out to be moved together. By pressing the "Alt" button while editing a fade, this can be temporarily deactivated. An active position link causes the space between "In-point" and "Out-point" stay constant.

Note: The behavior of the fade positions is determined by the position link button state and the current move mode.

"Classic" mode (shortcut: "C") is optimized for operations that occur during classic cuts; "Lock audio" mode (shortcut: "A") is especially for synchronous cut (e.g. during film dubbing). "Audio with fade in" mode (shortcut: "M") is only used in special cases.

Moving a crossfade

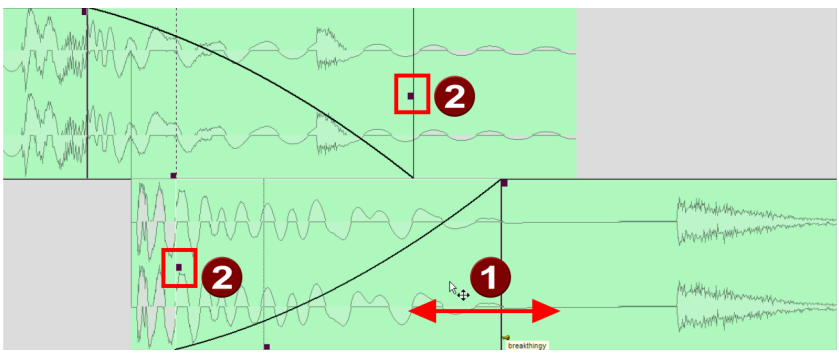
	General	Classical Mode	Keep Audio	Audio with Fade-in
Move Fade-out for Active Position Link	The end position of the object is moved relative to the audio material in all modes.	In addition, the fade-in objects (i. e. the audio material and fade) are moved as well.	In addition, the fade-in reference points over the audio material while the audio material is held in place.	The position-link state is ignored. (see Crossfade Editor Preset: Ignore Link Status in "Audio with Fade-in" Mode")
Move Fade-out for Inactive Position Link	In all modes, the out-point is displaced in relation to the audio material and the fade-ins remain unchanged.			
Move Fade-ins for Inactive Position Link		The fade-in audio material is moved on the timeline; the time position of the fade-in reference points and the fade-outs remain unchanged	Reference points for fade-ins and fade-outs are moved over the audio material	The fade-in objects (i. e. audio material and fade) are moved together; the fade-out does not change and the link state is ignored (see Crossfade Editor Preset "Ignore Link Mode in Audio with Fade-in Mode")
Move Fade-ins for Inactive Position Link		The fade-in objects (i. e. audio material and fade) are moved together	The fade-in reference point is moved over the audio material	Fade-in objects (i. e. audio material and fade) are moved together



The fades can be moved incrementally using the keyboard. The keyboard shortcuts for the fade-out are "+" and "-" on the numeric keyboard. Holding down "Ctrl" assigns these commands to the a fade-in.

The "*" key moves the fade reference point to the current playback marker position (even during playback).

Moving with the Mouse



1. By dragging the used audio material in Classical Mode you can move the entire "Fade In" object while leaving the "Fade Out" object where it is.
2. By dragging on the fade position handles you can change the fade position relative to the audio material.

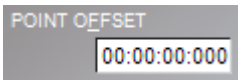
Step size

3 step sizes may be set for the "+/-" buttons, and the values may be adjusted via the settings.



Position editing fields

The time position of the fade reference point may also be edited directly by double clicking on the corresponding entry field or via the corresponding shortcut ("Shift + S" for the upper position, "Ctrl + S" for the lower position, "F" for the point offset).



The point offset places the difference of the time positions for fade in and fade out reference points.

Object mode

The crossfade editor has its own object mode. This enables fast, context-related handling for applications in the conventional source-destination cut as well as when working in the arrangement. "Connect track" object mode is loaded automatically when the crossfade editor is opened.

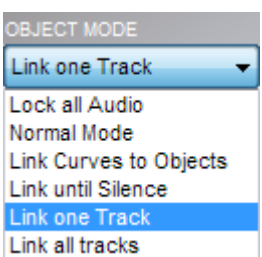
To use the same object mode as in the arranger, select "Use global mode" under "Preset object mode" in the crossfade editor settings.

More information about Sequoia's object modes is provided in "Screen elements -> Toolbars -> Object modes (view page 89)".

When the crossfade editor is closed, the active mode in the VIP is restored again.

To maintain cuts during crossfade editing, select "Connect track" mode.

For synchronous editing, use "Lock all audio". In this object mode, only the "Lock audio" move mode is available.



Crossfade parameter settings

The Shift key increases the precision of the crossfade fader by a factor of 10.

If the fader is active, then the current value can be increased with the "Page up" button or decreased with the "Page down" button. Alternatively, the top or bottom of the fader may be clicked.

Double clicking a fader switches between the current value and a standard value.

Defaults are: Fade length 100 ms, overlap 50%, curve parameter 100 %, level 0 dB.

If the fader is active, the default values may be accessed via the "Home" key.

The values may also be edited numerically by double clicking the value display and pressing "Enter" when the fader is active. The length parameter may also be adjusted via the shortcuts "Shift + L" and "Ctrl + L".

The values changed may be applied by pressing "Enter" and with "Escape".

Fade length/overlap/shape

The shape link button (shortcut: "N") specifies whether both fades should be shaped together or individually.

To edit the fade, the link state can be temporarily deactivated with the "Alt" key.

Tip: "Shift + N"/"Shift + I" operates the position and shape links simultaneously.

Fade length

The fade length may be set via the length fader. A logarithmic scale is available in this case.

Shortcut for fade out length value: "L" or "Shift + L"

Shortcut for fade-in length value: "Ctrl + L"

The fades in the project window may be changed via the fade length handles.

Overlap

Here you can specify the position of the reference point within the fade.

For the fade out, the positioning of the reference point specifies the end position of the fade out object.

A 0% fader position means that the fade out takes place completely before a defined end position, i.e. the fade is completely within and to the left of the reference point.

A 50% fader position means that half of the fade out takes place before a defined end position and half after it, i.e. the fade is 50% within and 50% outside the reference point.

A 100% fader position means that the fade out takes place completely after a defined end position, i.e. the fade is completely outside and to the right of the reference point.

Shortcut for fade out overlap value: "E" or "Shift + E"

For the fade in, the positioning of the reference point specifies the start position of the fade in object.

A 0% fader position means that the fade in takes place completely before a defined start position, i.e. the fade is completely within the fade in object and to the right of the reference point.

A 50% fader position means that half of the fade in takes place before a defined start position and half after it, i.e. the fade is 50% outside the fade in object and 50% within the reference point.

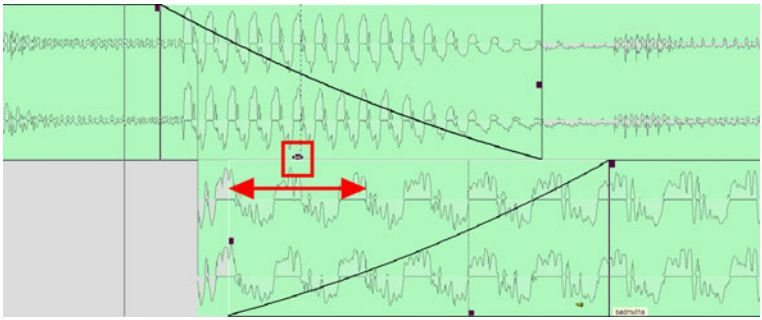
A 100% fader position means that the fade in takes place completely before a defined start position, i.e. the fade is completely outside and to the left of the reference point.

Shortcut for fade in overlap value: "E" or "Shift + E"

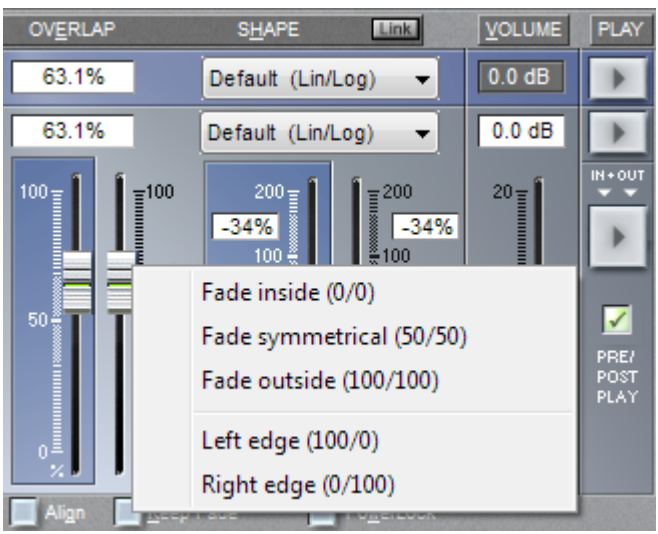
If the button "Move point" is activated, the reference point can be moved relative to the rigid fade. If this option is not selected, use the "Overlap" controller to move the fade relative to the fixed reference point.

Regulate the overlaps of linked fades with the align function so that the fade in and fade out values add up to 100%.

In the crossfade editor project window, use the "Overlap" handles to set the fade relative to the reference point.



Right-click on "Move point" to open a context menu for setting the default values for the reference points of fades.



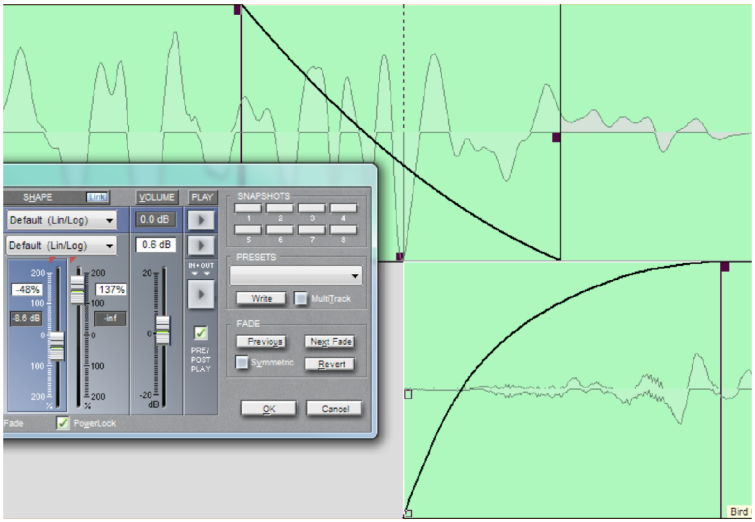
Curve Form

The selection menu "Form" contains mathematical curve functions that can be used to define the display of various crossfade progressions. In addition to the linear form of a straight line there are also various sinus, cosinus as well as exponential and logarithmic progressions available.

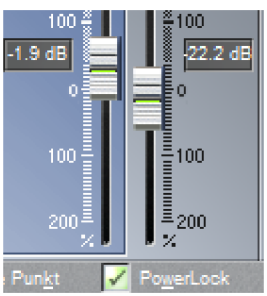
With the curve form controllers you can change every fade from a linear curve (0%) on up to an original form (100%). A setting of -100% creates an inverted display of the curve. Parameter values larger than 100% or smaller than -100% create additional distortions in the curve.

Shortcuts for fade out values: "H" or "Shift + H"

Shortcuts for fade in values: "H" or "Ctrl + H"



The "PowerLock" button is also included in the parameters for linked fades. This function ensures a constant amplitude level in the parameter range between -100% and +100%. If you open a new curve form, the complimentary parameter value for the linked fade will automatically be applied.



The dB display beside the faders shows the relative level of the object at the fade reference point.

Volume/level

The level fader (shortcut: "V") may be used to set the relative change of the object level of the crossfade objects.

Auto Zoom

Activating the "Auto Zoom" function adjusts the crossfade according to the values specified via the "Settings" button.



If changes have been made to the position, fade lengths, and overlap changes that have slightly moved the transition out of the visual project area, then activating the "Auto Zoom" function restores the visible section across the crossfade.

Holding down the "Shift" key at the same time causes the "Auto Zoom" function to jump to the crossfade position without changing the zoom factor.

Symmetrical adjustment of the fade

The "Symmetrical" option adjusts the fade in to the fade out. Position and parameter links are switched on.



Opening the crossfade editor with a symmetrical fade activates the button. The state of the button indicates whether the requirements for a symmetrical fade have been met. These are:

- Point interval: 0
- Equal fade length
- Overlaps add up to 100%

Playback Functions



The audio material that is currently being played back is visible at the playback marker and the lit up playback button:

- When the upper playback button lights up, playback takes place until the end of the fade out (including pre-roll time).
- When the second, middle playback button lights up, playback takes place from the beginning of the fade in (including roll-out time).
- When the lowest playback button lights up, playback takes place over the entire fade (including pre-roll and roll-out time).

All crossfade playback functions can also be found in "Playback -> Play section (view page 825)".

Overview of the playback functions

- "Alt + Spacebar": Plays the audio material of the entire project, including all fades. The playback marker may be positioned freely.
- "Pre/post play + Spacebar": The entire fade is played back including the pre and post-roll times (lower play button).
- "Shift + Spacebar": Plays the audio material of the fade out objects including pre-roll time (upper play button).
- "Ctrl + Spacebar": Plays the audio material of the fade in objects including post-roll time (central play button).
- "F7": The fade-out objects are played up to the out point without fading.
- "F6": The fade in objects are played from the in point without fading.
- "F8": Unused audio material of the fade out objects is played from the out point.
- "F5": Unused audio material of the fade in objects is played up to the in point.
- "Shift + Alt + Spacebar": Plays the upper audio material (tail) from the cursor position.
- "Ctrl + Alt + Spacebar": Plays the lower audio material (head) from the cursor position.

Snapshots

Different variations of the currently edited crossfade may be saved into the 8 snapshots. This is done by pressing "Shift" and left clicking the corresponding button. In case a snapshot has not been specified for the respective button beforehand, then left clicking is enough. The snapshots may also be saved by holding down "Ctrl" and pressing the number block keys (1-8).



Saved snapshots may be loaded simply by left clicking the assigned button. This allows different crossfade variations to be compared quickly.

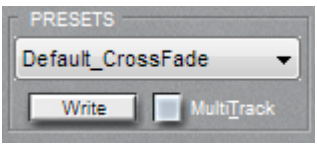
As soon as the crossfade editor is closed or the next/previous fade is activated, the snapshots will be discarded. If the "Revert" function is activated, they will be maintained.

A snapshot may also be loaded, saved, or deleted via the corresponding button.

Crossfade Presets

Current settings in the crossfade editor can be saved as crossfade preset files (.cfp). Click the "Write" button to do this; enter a new name in the save dialog that appears. The saved preset will now appear in the preset list.

You can open, change and overwrite presets. When opened, the "Fade out" reference point ("Out-point") of a crossfade and the reference point of a single fade in or fade out remains intact.



The presets include:

- Point offset
- Fade length
- Overlaps
- Curve types and curve parameters
- Symmetrical or asymmetrical behavior of the fade outside of the crossfade editor

They do not include:

- Time and audio position of the "Fade out" reference point ("Out-point") and "Fade in" reference point ("In-point")
- Volume of the fade in object

Using crossfade presets

"**Default crossfade**" is applied to all object-related cut operations in "Auto-crossfade" mode (cut, trim, etc.). By default, objects are "cropped", i. e. the fade is completely within the object and fade ins/outs do not overlap.

"**Default_Crossfade_Alt**" is linked to the "Cut objects with alternative crossfade (view page 724)" menu command (shortcut: "Shift + T"). This fade is predefined for the "Level " use case. This corresponds with a linear curve, which is symmetrical to the intersection with an overlap of 50%.

Default_FadeFromBlack and **Default_FadeToBlack** are linked to the black fade source-destination commands. This involves a default fade-in and fade-out that may then be applied in context. Adjust these presets to suit the most frequently used areas of application.

Note: If the crossfade is saved internally, changes to the fade length will not correspond with changes made to the object borders.

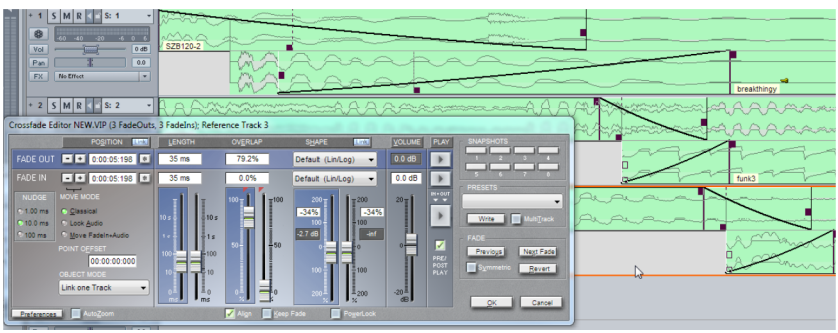
"**Default_CrossFade_SD**" corresponds with the default application for source-destination commands. This crossfade is optimized for fading inaudible similar signals, i.e. as a root-cosine 50/50 fade. Adjust the default length to a particular project's conditions.

Multi-user operation (view page 711) enables a complete set of presets to be defined for every user. These are used automatically when the user is located in his or her specific subfolder. If no individual user subfolder is available, then Sequoia will load the general set of presets.

Multitrack Crossfade

In Sequoia you can also edit several object crossfades in different tracks at the same time. To do this, the objects must first be grouped. Objects created during a multitrack recording are already grouped by default.

- Select the objects to be edited (e. g. with the object lasso shortcut "Ctrl + Alt + L" or with the mouse and "Shift" held down in the corresponding VIP tracks
- Form a group (shortcut: "Shift + G")
- Open the crossfade editor (keyboard shortcut "Ctrl + F")
- Zoom out of the VIP until all of the tracks containing grouped objects with crossfades are visible.
- Now when you change a fade, the changes will also be applied to the transitions on the other tracks relative to the changed fade.



Lock tracks/crossfades

If the fades on a specific track should not be included in processing, then this track may be locked. Use the "Lock" button in the track header. The crossfade editor switches the locked track automatically to "Lock all audio" mode so that objects are not moved by accident.

Multitrack Presets

Loading presets: When loading crossfade presets, a choice can be made about whether they should affect all crossfades or only the selected crossfade in the current track. This can be set by checking the "MultiTrack" preset in the crossfade editor. (Shortcut: T).

Note: The check box for loading to all crossfades has to be activated before loading a preset.

Saving presets: The "MultiTrack" check mark can also be used during writing if the preset should be saved as a multitrack crossfade.



If you have activated the "MultiTrack" checkbox, the complete multitrack crossfade will be saved. The preset receives the extension "MT02.cfp" for a two-track crossfade, and a three-track crossfade is named "MT03.cfp" (for example).

For instance, if the "Crossfade_MT03.cfp" is loaded onto a 2-track crossfade after saving it, an error message will be displayed indicating that the number of tracks does not match. If you continue loading, only the crossfade settings of the first two tracks will be loaded.

General crossfade functions

OK: (shortcut: "O") confirms all changes to the current fade and closes the crossfade editor.

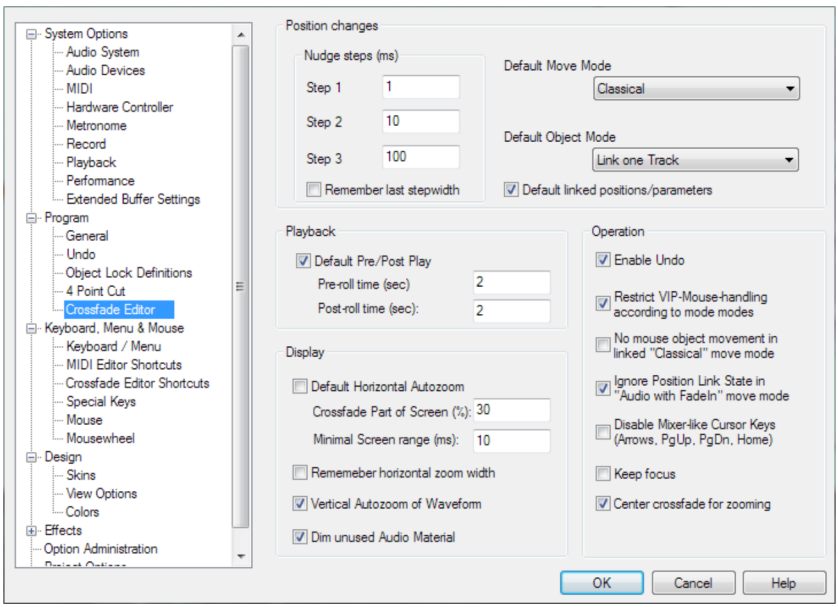
Cancel: (shortcut: "Esc") discards all changes to the current fade and closes the crossfade editor.

Revert: (shortcut: "R") discards all changes to the current fade and opens the crossfade again with the fade's original parameters. Saved snapshots for the current fade will be retained.

Back/forward: (shortcut: "U" and "X") Accepts all changes made to the current fade and opens the crossfade editor again for the previous or next fade in the current track. For multitrack fades, the reference track displayed in the title list will be used.

Crossfade editor settings

Clicking on the "Settings" button opens following dialog:



Changing positions in the crossfade editor

Step sizes: Select up to three step sizes for the crossfade editor. Millisecond values such as "0.2 ms" may also be entered.

Remember last step size: If a check is placed here, the last step size selected will be applied to the next edit.

Preset mode and object mode: The mode set here will be used when opening the crossfade editor.

Preset link status for position/parameter: This setting enables the position link button for the fade-out and fade-in to be set by default.

Playback

Pre/post play: Decide whether the set pre-roll and post-roll times should be applied automatically when the crossfade editor is opened.

Lead-in and lead-out time: Specify the lead-in and lead-out time for the corresponding playback function in seconds.

Horizontal screen section in the crossfade editor

Horizontal auto zoom preset: Set the status of the "Auto Zoom" button here when the crossfade editor is opened.

Section of screen for crossfade (%): Specify how much of the screen the crossfade editor should occupy when it is opened or after autozoom. If "0" is entered, this value will be ignored and the following absolute value in milliseconds will be used for the minimum screen area.

Minimum screen area: Determines the minimum screen section in milliseconds after opening the crossfade editor or after autozooming.

Waveform display in the crossfade editor

Remember horizontal zoom step: Select this option, and the last horizontal zoom step that was selected will be set when the crossfade editor is opened again.

Vertical auto zoom for waveform: Resizes the graphic display of sample data of each object to maximum independent of each other. This enables inputs to be evaluated better, but evaluating relative level ratios will be more difficult.

Gray-out unused audio material: The depiction of unused audio material may be activated here to make it easier to recognize objects.

Crossfade editor operation

Enable undo: Activates the "Undo" function.

Restrict mouse handling according to move modes: This function allows only those mouse operations in the project window that correspond to the current position link and displacement modes to avoid errors.

No mouse object movement with mouse in linked "Classic" mode: Blocks the displacement of linked objects in "Classic" mode.

Ignore position link state in "Audio with fade-in" mode: Turns off the link status in the "Audio with fade in" mode.

Disable mixer cursor keys: Deactivates the "Page up", "Page down", "Home", "Cursor" buttons for the active faders, making these buttons available for entering commands into the project window.

Keep focus: This function keeps the crossfade editor constantly in focus, even after mouse operations in the arranger window.

Center crossfade for zooming: If a check is placed for this option, then the crossfade will be displayed at the center of the zoom section.

Tempo editing

Tempo and beat change

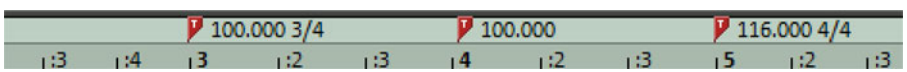
Tempo changes can be defined by means of the tempo marker in the marker bar or the



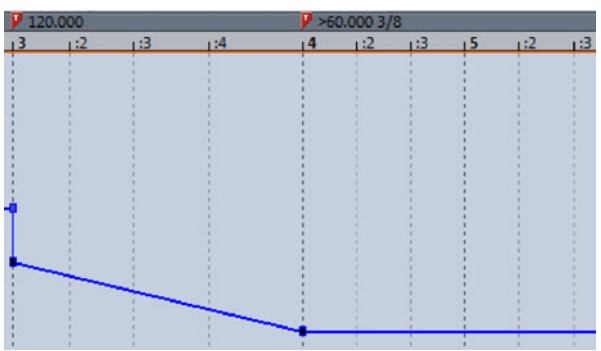
marker button. Tempo changes are displayed in the marker bar.



The Tempo Map (view page 444) can be opened through "Edit" > „Tempo" > „Tempo Map". The tempo map displays all tempo markers and thereby determines the musical position for each time position. The tempo map defines the bar grid.



This means that the bar grid is made up of all the tempo markers. Sequoia offers you convenient options for adjusting the bar grid to existing material (audio/video/MIDI). You can also display and edit a graphic display in the tempo track (view page 450). A tempo track contains the project tempo map as an automation curve. Each automation point of the curve corresponds to a tempo marker.



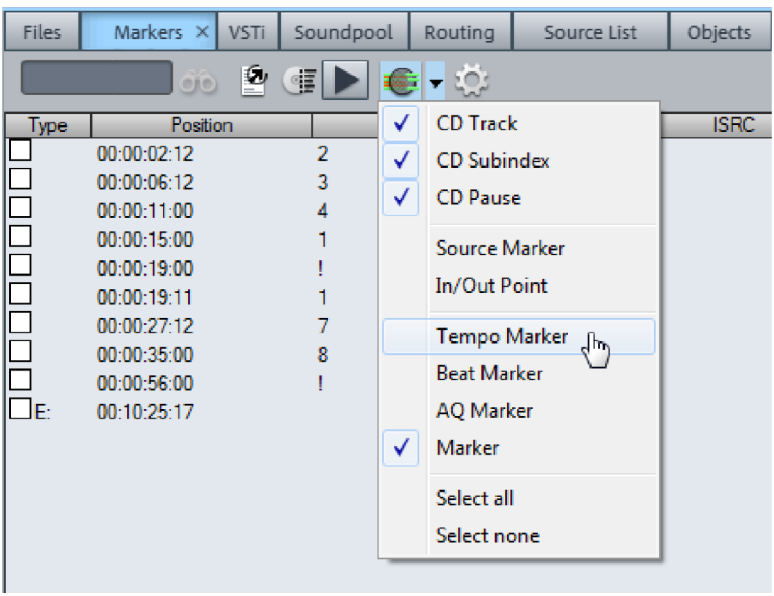
BPM marker, tempo curve (>) and beat change (3/8) - under the tempo track.

In addition to the display in the tempo map dialog and the tempo track, the current values for tempo (BPM) and the beat type at the playback marker position are displayed in the transport console (view page 65).

Tempo Markers

Tempo markers signify a tempo change at a specific position in the project. They appear either as BPM markers or as Grid position markers according to the Tempo Map mode (view page 448).

You can also change the time signature when creating BPM markers and grid markers. Set the Tempo marker directly using the menu or indirectly by manipulating the tempo in the tempo track with the mouse. Another way of accessing and editing the Tempo marker is by using the Marker Manager (view page 196).



BPM Markers (red)

You can define tempo changes for each musical position directly in the project window using BPM markers. When setting a BPM marker, a new tempo is defined at the corresponding position. A new project first has a single “Master” tempo which can be defined in the transport console or in the project settings (keyboard shortcut: “I”).

This master tempo affects either the entire project or the start of the project up to the first BPM marker.

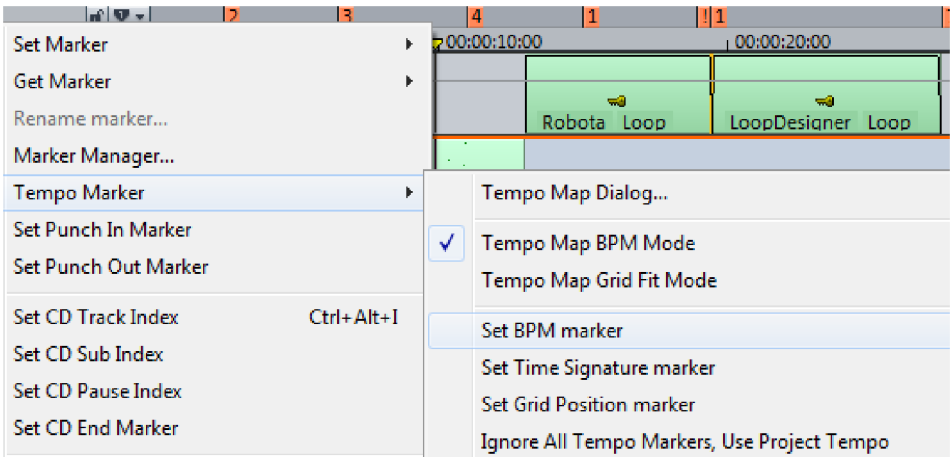
Exception: If the tempo should be interpolated to the first tempo marker, the master tempo is only effective directly at the project start in order to go linearly to the tempo of the next BPM marker.

You can define tempo changes at any time and for each musical position (even between beats) directly in the project window using BPM markers or the corresponding tempo curve points:

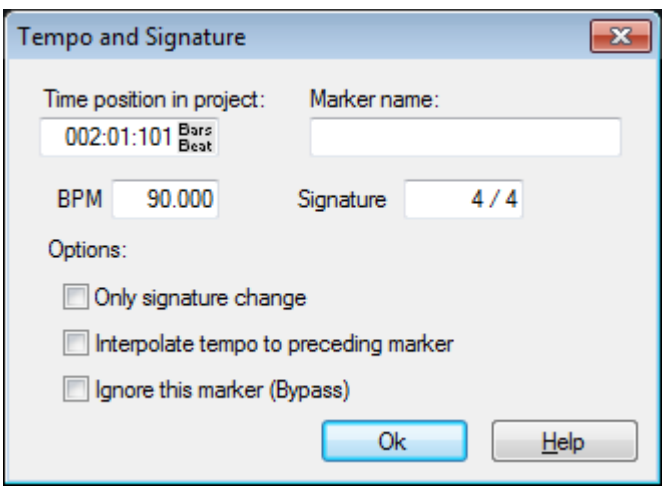
1. Place the playback marker at the position where the tempo change should take place.

2. Right-click on the marker bar to open the marker menu.

3. Go to "Tempo markers" > "Set tempo change"



4. Enter the desired tempo change in BPM (beats per minute) in the "Tempo and Beat Type" dialog and confirm by clicking "OK".



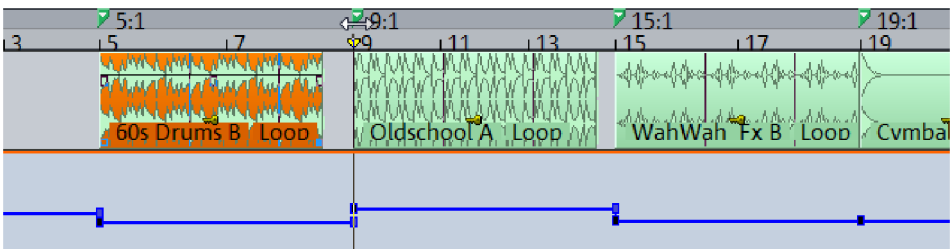
The tempo markers are set to the next snap point to the playback position when the grid is active. By dragging the tempo markers and holding down the "Alt" key, you can temporarily deactivate the grid function.

Note: In order to use tempo markers to adjust audio objects in your arrangement to match tempo changes, musical tempo adjustment (view page 454) needs to be activated for the object.

Grid Position Markers (green)

Grid position markers assign a specific musical position to a specific time position.

If you move a grid position marker, you'll see how the beat grid is immediately adjusted. In the tempo track (view page 450) you can apply the effects of the beat grid changes to the tempo.



A grid position marker defines the tempo indirectly, causing the tempo before the marker to be adjusted in such a way as to reach the desired musical position exactly at the marker position. This way, the bar frame/grid and MIDI events can be easily synchronized with existing audio material.

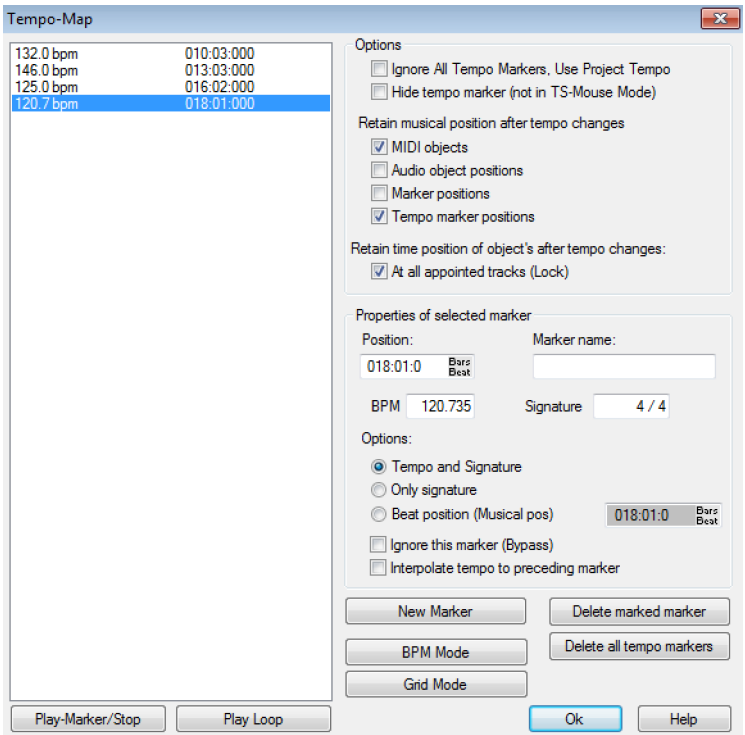
Time Signature Markers (blue)

Bar markers change the time signature from the marker position, e. g. 4/4 beat to 3/4. This means you can only insert bar markers at the borders of bars. They must also be positioned at bar borders when moved.

Tempo Map Dialog

1. Set the playback marker to the position at which the tempo or beat changes should occur.

2. Open the tempo map dialog with the command in the marker menu (right-click on the marker bar) or through "Edit" > "Tempo".



3. Now click on "New Marker". A marker is inserted at the current position.

You can now use the marker options to set the properties of this marker. Now make the new entries, e. g. the tempo in BPM for a tempo marker (view page 442) or the new time signature for a bar marker (view page 444).

Note: Be aware that the settings are only valid for the markers currently selected in the marker list. For example, you cannot set marker properties such as type or tempo after a new marker has been created.

A bar marker is always placed at the start of a new bar. If the playback marker is at a different position, the marker position is automatically moved to the beginning of the next bar.

Ignore all tempo markers, use project tempo

This option allows you to ignore any previously created tempo markers in your project so that only the project tempo is observed.

This occurs automatically for grid position markers if illogical bar positions are created while moving, e. g. if grid position markers are exchanged (bar 20 before bar 19) or are moved in such a way that this bar position originating from a previous tempo marker cannot be met by tempo interpolation.

Hide Tempo Marker: This option ensures that all tempo markers in the arranger are no longer shown. If you are in the pitch shift/timestretch mouse mode, this option is ignored.

Retain musical position after tempo changes

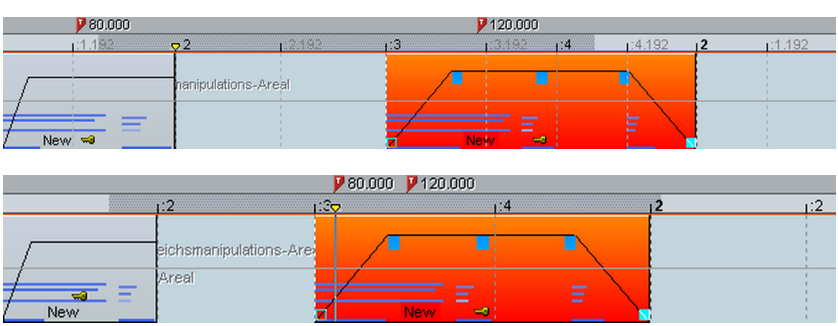
If a tempo marker is changed or a grid position marker (view page 444) is moved, this influences the following markers and audio/MIDI objects in the virtual project.

There are two main possibilities: The time position either remains constant, or the musical position remains. The time position in the virtual project is the absolute position, the musical position is flexible and is displayed by the grid.

Maintaining the musical grid position means that objects or markers are adapted to the newly created grid and their absolute time position in the project changes. If the time position is maintained, the musical position changes accordingly.

When changing the tempo markers, the behavior of markers, audio and MIDI objects is controlled separately. By default, the musical grid position of MIDI objects and tempo markers is retained and the time position of markers and audio objects remains constant.

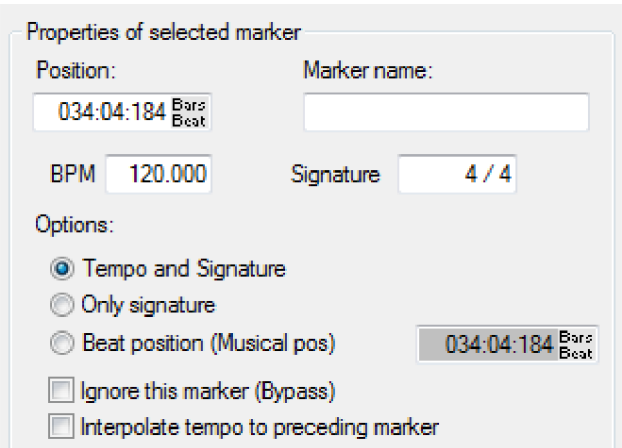
If you drag a tempo marker while holding down the Alt key in Grid Adjustment Mode, the time position of the object remains unchanged.



In this example, the first tempo marker in the second image has been changed. The MIDI object and the second tempo marker were moved correspondingly so that the musical position (3:01:000) was maintained.

If you generally want to keep the time position of objects during a tempo change, select the option „On all locked tracks“ and activate the button in the track header  to lock all the objects on the track.

Properties of Selected Markers



Here you can determine the time position of the selected markers in the project as well as the marker names, the desired tempo in BPM and the time signature.

Options

Here you can specify whether the entries will be applied only for a beat change or for tempo and time signature.

The option "Beat position (musical position)" transforms the selected BPM Marker into a Grid Position Marker (view page 444). The position can be adjusted in the editing field.

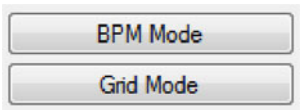
You can also have the selected markers ignored.

Here you can also choose whether the tempo should be interpolated to the previous tempo marker (Ritardando or Accelerando). During playback the tempo constantly gets closer to the defined tempo at the current marker until it finally reaches this tempo at exactly the current marker position.

Tempo map modes BPM mode/Snap mode

In the Tempo Map dialog (view page 444) you can choose between two modes: BPM mode and Snap mode.

- In BPM mode you work exclusively with BPM markers and Time signature changes.
- In Snap mode you work exclusively with grid position markers and time signature changes.



You can toggle between BPM markers and grid position markers at any time. The tempo markers remain the same, but change their type. The effective tempo map and beat grid remain unchanged.

For example, first adjust the bar grid in grid mode to a MIDI recording with many tempo changes. Now switch on the BPM mode and insert musical tempo changes for single sections.

Note about working in BPM mode: By holding down the "Alt" key in the time stretch mouse mode, you can also move a temporary grid position marker while dragging with the mouse in the timeline.

Grid Tapping

You can set grid position markers in the tempo map grid fit mode during playback.

1. Assign the menu command „Insert grid position marker“ (view page 728) to a keyboard command of your choice, e. g. „+“.

2. Now press „+“ in rhythm with the music during playback to "tap in" the beat grid and tempo map.

The following will happen: At the time of pressing the key, the nearest bar position of the selected bar grid will move to the current playback position and it will be marked with a grid position marker.

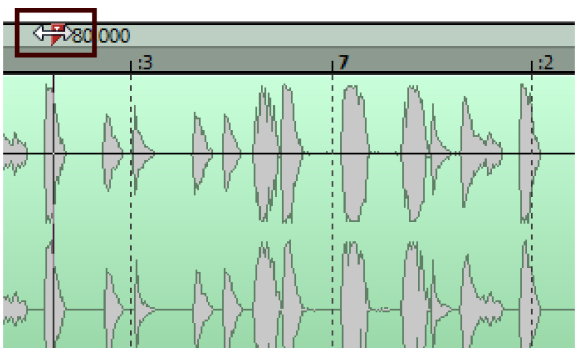
Tip: To detect the tempo, work in the bar grid (entire bar borders) first. In a second step, you can edit slight tempo fluctuations in the "beat" grid.

Manipulating the Tempo Marker in Timestretch Mouse Mode

The timestretch mouse mode simplifies working with tempo markers. Select "Pitchshift/Timestretch" mode from the toolbar.

Create BPM markers by directly clicking the desired playback position and holding down "Shift". You can adapt this tempo by moving the mouse vertically while holding down the Shift key.

You can create grid position markers in "Timestretch" mode by clicking the desired playback position and holding down "Alt". You can move them using "Alt" + horizontal mouse movement, e. g. to manipulate the bar grid and adapt it to existing audio events.



Tips - Examples of Use

- Tempo markers can be placed when composing in the MIDI Editor. Subsequent MIDI Objects and tempo markers retain their musical position (advanced setting).
- The grid can be adjusted to available audio events, e.g. assigning beat numbers to certain time positions. You can either use the "Set new grid position marker" menu command to create a marker at a position within the project and assign the corresponding bar position, or use "Alt" + mouse click on the respective beat grid position and, while holding down the mouse button, immediately move it to the required time position (e.g. to the beginning of an object).
- If the project includes MIDI data, this will be automatically adjusted (preset). The newly created tempo grid is used in the MIDI Editor for subsequent editing.
- When working with MIDI files and complex tempo changes, you can switch off the tempo map before starting to record the new MIDI data ("Ignore all tempo markers, use project tempo"). After the recording has finished you will be able to switch on the tempo map once again, which makes the newly recorded MIDI data adjust itself automatically.

Tempo Track

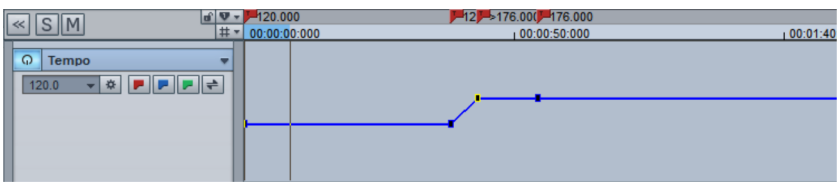
The Tempo Track can be used to display an overview of the tempo and time changes in your project in a track of its own. The Tempo Track is a graphic display of the project's tempo map (view page 444) as an automation curve (view page 457). Each automation point of the curve corresponds to a tempo marker.

The Tempo Track header provides quick access to the most important commands for placing and editing tempo markers.

Note: *You still have the option of displaying the "tempo" automation curve as one of many automation curves in any other track, instead of displaying it in a separate Tempo Track. When you create a Tempo Track, the tempo automation curve in the other track is removed.*

Creating Tempo Tracks

To create a Tempo Track, select "Track" > "Insert new Tracks" > "New Tempo Track".



Changes to tempo markers are automatically transferred to the Tempo Track and vice versa. Keep in mind that the tempo track can only be edited in BPM mode; in snap mode it serves solely as a display.

The tempo automation behaves differently than track-based automation curves. Automation points are coupled with evenly increasing/decreasing tempo changes and the pencil of the automation drawing mode doesn't create any new automation points as you drag, only when you click.

Note: In the tempo track the beat grid is constantly active (regardless of the system option "Program > General > Use snap for automation curve points"). As usual the snap can be temporarily disabled using the "Alt" key.

Tempo Track header



1. **Tempo automation on/off:** You can disable tempo automation entirely with this button. This corresponds to the option "Ignore all tempo markers, use project tempo" in the Tempo Map dialog (view page 444).
2. **Current tempo:** The dropdown menu under the arrow corresponds to the one in the transport console and contains a selection of preset tempos, as well as the tap tempo feature.
3. **Tempo Map dialog**
4. Insert new **tempo marker**
5. Insert **signature change**
6. Insert **grid position marker**
7. Switch between Grid mode/BPM mode (view page 448)

Edit in the Tempo track (only BPM mode)

Insert tempo changes

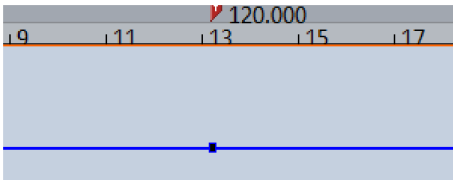
- Double-click on the tempo curve in Universal Mouse Mode:
- In Curve Mode: Click (and drag) on tempo curve.
- In Automation Draw Mode: Click

Change existing curve points

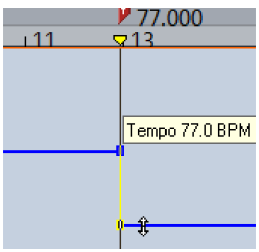
- BPM. value and positions can be changed by clicking and dragging curve points. For tempo levels there are always two curve points for one tempo marker, one for position and and one for tempo.
- Alt key: With this modification key you can temporarily deactivate the grid and convert tempo jumps into tempo curves.
- Shift key: With this modification key you can change BPM values and make fine adjustments to them (horizontal lock).
- Ctrl + Shift + Click: This key combination can be used to switch between tempo curve and tempo step.
- Right-click: Right-clicking on a tempo marker opens the „Tempo and Time Signature" dialog (view page 454) where numerical values can be changed.

Inserting tempo changes and tempo curves into the tempo track:

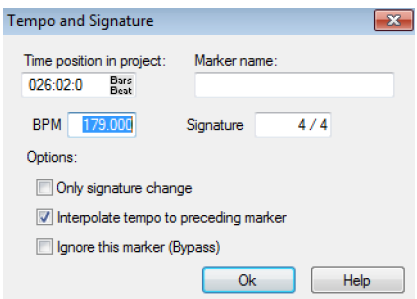
1. Double-click (in Universal Mode) or simply click (in Curve Mode or Automation Draw Mode) on the desired position in the tempo curve. In addition to the new curve point a new tempo maker will appear in the marker bar.



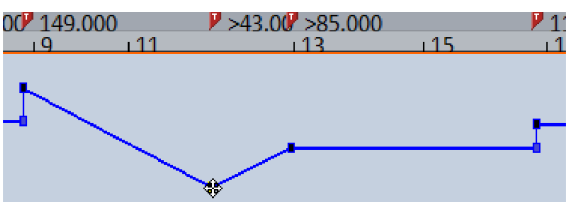
2. Starting at the new curve point, drag the mouse pointer upwards or downwards to change the tempo.



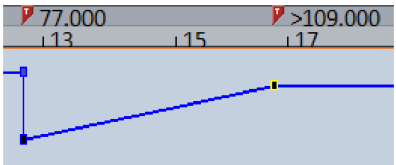
3. Right-click to the open the "Tempo and Time Signature" dialog where you can change the values numerically.



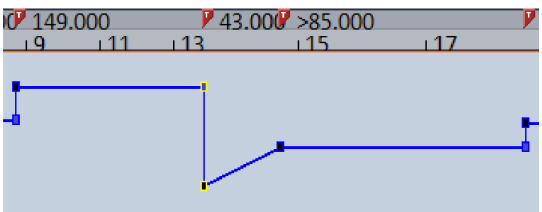
4. If you hold down the Alt key while clicking on a tempo marker, the grid is deactivated and the tempo jumps to the neighboring tempo markers are converted into tempo curves.



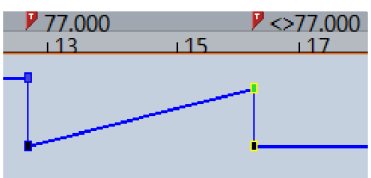
5. Hold down the Ctrl key while double-clicking (in Universal Mode) or simply clicking (in Automation Draw Mode). Now drag on the created BPM marker to set a tempo curve to the previous BPM marker.



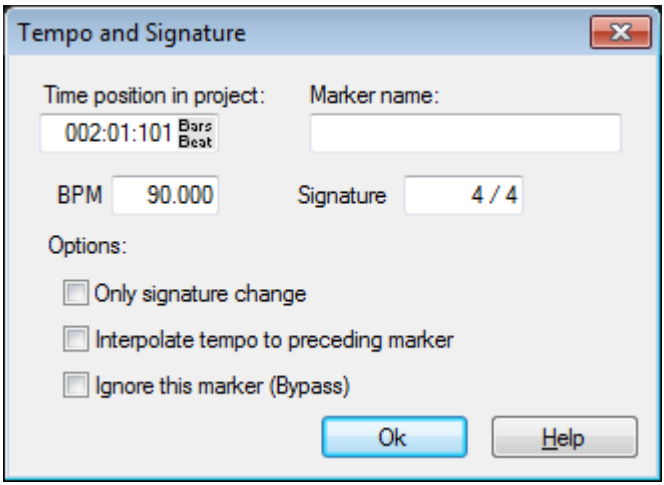
6. To switch back and forth between the tempo curve and the tempo step to the previous tempo marker, click on a tempo marker while holding down the Ctrl and Shift keys.



7. This creates a Ritardando or Accelerando. While holding down the Ctrl key, create a new tempo marker by double-clicking (in Universal Mode) or simply clicking (in Curve Mode or Automation Draw Mode) on the tempo curve. Hold down the mouse button and drag upwards on the created tempo marker for Accelerando, or downwards for Ritardando. This will generate a tempo curve to the previous tempo marker and the corresponding BPM marker will be displayed as a double arrow in the marker bar. The previous tempo is applied again from the newly created tempo marker position.



Tempo and time signature dialog



This dialog appears by double-clicking on an available tempo marker or curve point or if you set a new tempo or bar marker via menu command ("Edit" > "Tempo" > "Insert tempo/beat change").

As in the tempo map dialog, you can also determine the time position in the project as well as the marker name here, enter the desired tempo in BPM (not for grid position marker) and determine or change the time signature. Also you can choose here whether the tempo should be interpolated to the previous tempo marker (Ritardando or Accelerando) and whether this marker should be ignored.

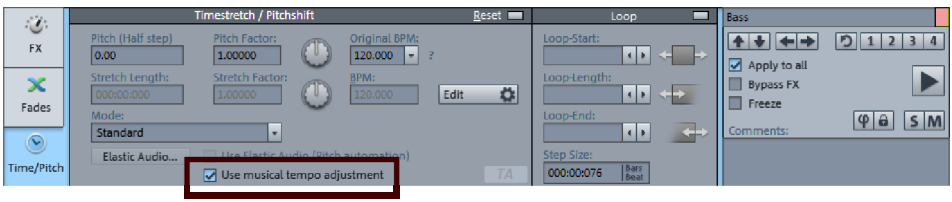
Change Global Tempo

Using the command „Edit“ > „Tempo“ > „Insert Tempo Change...“ You can scale all of the BPM values on selected tempo curve points by a set factor simultaneously.

Musical tempo adjustment

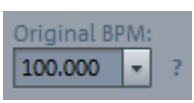
With musical tempo adjustment, audio objects are adjusted to match any changes made to the tempo of the project. This means that when you change the tempo of the project, the tempo and time positions of the objects are adjusted accordingly using tempo markers or a tempo curve. The bar position remains untouched while the start positions of objects are moved, and time stretching is applied.

You can activate musical tempo adjustment in an object by placing a checkmark next to it in the Time/Pitch section of the object editor.



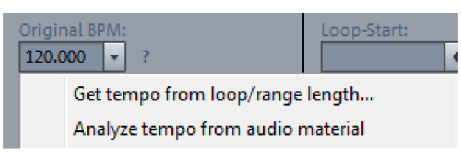
Note: Musical tempo adjustment is automatically activated when you load Soundpool samples into a project. If you activate the **BMP sync** preview option (view page 188) when you load audio files via the file manager, tempo adjustment will also be activated for the object.

The **Original BPM** setting used as a basis for the adjustment must be a valid value.



A question mark next to the entry field indicates that the displayed value is a default value and may not be correct.

You can either enter the original tempo manually or let it be calculated. To do this, click on the arrow next to the value.

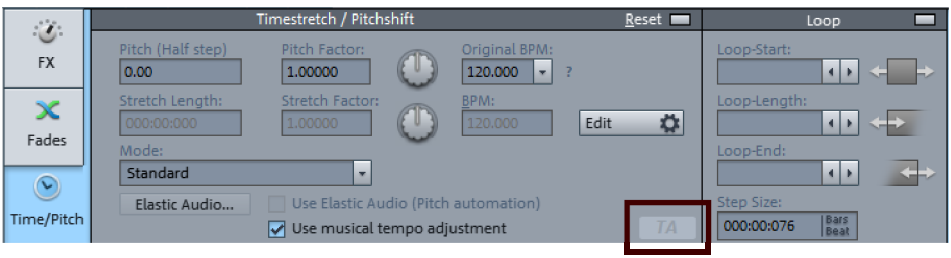


Get tempo from loop/range length: A range section above the object is used to calculate tempo. Select the option and enter how many quarter notes should be included in the selection.

Analyze tempo of audio material: Melodyne will calculate tempo. Melodyne must be installed for this. You can learn more in Melodyne integration (view page 234).

Musical tempo adjustment cannot be used at the same time as Elastic Audio (view page 914) (pitch automation), ARA (Melodyne integration) (view page 234) or object resampling.

The indicator field in the object will show you which of these options is currently being used:



PA Elastic Audio (pitch automation).

TA Musical tempo adjustment: The length and position of the object are changed to match the tempo map of the project. The original BPM value will be used as a reference tempo for the object.

SRA Sample rate adjustment: Adjusts the sample rate of audio objects by using object resampling if the sample rate of the Wave file is different from that of the project. If sample rate adjustment has been carried out automatically, you will be asked if you want to remove it when you activate tempo automation.

ARA Object tempo and pitch are controlled by the Melodyne plug-in (view page 234).

Note

- The positions of object automations and audio quantization markers (view page 217) will be taken into account and changed accordingly.
- In the current program mode, non-relevant control panels (such as stretch length, stretch factor and BMP) are grayed out.
- Remix Agent (view page 805) cannot be run.
- Tempo automation does not work with looped objects or objects played in reverse (view page 171).

Automation

When mixer settings such as volume, panorama and effect setup for a project are modified to enable (for instance) the fading in of tracks, audio signal movement in panorama or panorama changes for effects, this is known as automation.

The display and drawing of this automation is done using automation curves. An unlimited number of automation curves are available on each track. You can draw automation movements on track and master level during playback in automation curves. All automation curves (track, master and object automation) can be edited after the fact or directly drawn in.

- The following dynamics automation settings are available at track level: volume, panorama, surround, AUX sends, EQs, and plug-in control elements and MIDI controllers.
- At object level, you can automate the volume, AUX sends, and VST plug-ins.
- At master level, you can automate the volume, EQs, VST plug-ins, and MIDI controllers.

Create automation curves

There are specific control elements on the track header for activating the most important track automation curves – volume and panorama.

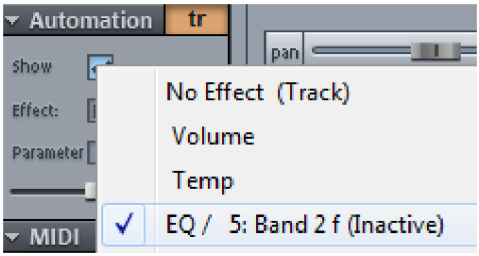
You can select the other parameters for automation in different ways:

- Select parameter in automation context menu or menu automation
- Via keyboard shortcut Ctrl + Alt + Control element in the mixer or the interface of a plug-in
- Via the VST parameter dialog

Automation context menu or menu automation

In the automation menu or automation context menu (right-click on "Effect" in section "Automation" in the track editor) you can select a automation parameter. Below in the menu, all available parameters are listed in sub-menus according to loaded plug-ins and effects. Select the parameter you want.

All available active or inactive automation parameters are listed in the top part of the menu. Here, select an automation parameter that you can then manipulate during playback with the automation controller in the track editor. The curve of the selected parameter can be seen in the track in the foreground. Via the context menu, you can opt to have all other curves displayed as thin lines or you can also have them faded out (See Automation – Context menu for more details).



Ctrl + Alt + Control element

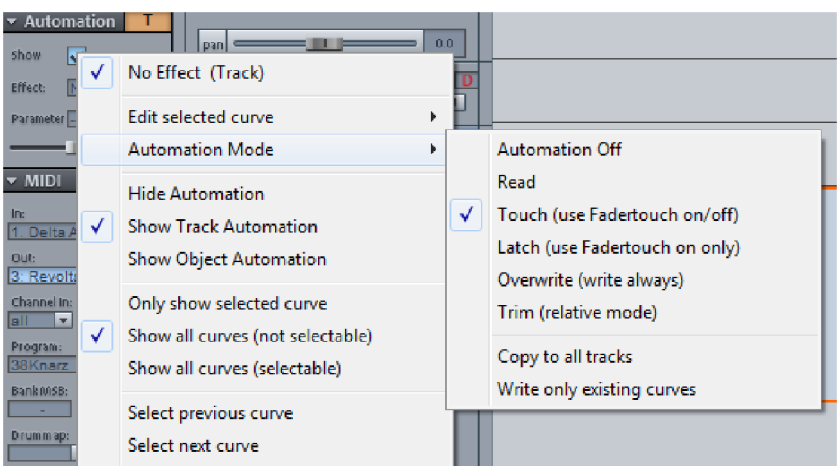
This is the fastest and smoothest way to automate a plug-in parameter. In the mixer (volume, pan, aux sends, EQ) or interface of a plug-in, click the control element of the parameter you want to use while holding down Ctrl + Alt. In stopped state, this then creates a curve. During playback, a change to the parameter is drawn directly (regardless of the selected automation mode (view page 458)).

Note: This works only at track level, and not for the automation of object effects (view page 465).

An alternative method of automating a parameter is to select "Automate next parameter" in the plug-in menu of the VST plug-in dialog and then click the corresponding control element.

Automation modes

You can draw automation using various modes.

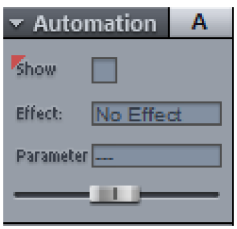


The automation modes define how automation data should be written or played back. You can set the automation mode for each channel/track. Various automation modes

may be selected by right clicking the automation button in the mixer channel or track editor. Clicking this button toggles between the last activated mode and "Read" mode.

Off mode

All automation functions are deactivated in "Off" mode, all previously written automation sequences are ignored.



Read mode

All recorded automation parameters are played in "Read" mode. The automation button is usually set to this state.

Touch mode

In "Touch" mode, automation data is only recorded as long as you touch the selected control element with the mouse or touch it from your external controller. As soon as you release the control, the automation recording stops. Once released, the control will move back to the previously recorded position. You can determine the fader return time in the System Options (Keyboard shortcut: „Y“) under „Effects“ > „Automation“ > „Automation Release Time“.

Latch mode

In "Latch" mode, automation files are written starting only from the first touch of the control element or the external controller. As soon as the corresponding button is released, the automation is written to the last value until playback ends or another mode is switched on.

Overwrite mode

In this mode, automation data is recorded immediately as soon as playback begins, irrespective of whether you touch the fader. The automation continues to be written until you end playback or switch into another mode. This allows you to quickly overwrite a previously recorded automation curve.

This mode only affects the previously selected curve, regardless of whether other curves are active, since overwriting all automations at the same time is not useful.

Trim Mode During Playback

"Trim" mode produces a relative change of values during simultaneous playback for existing volume automation.

Note: For other automation parameters, trim mode during real-time drawing behaves exactly like touch mode (view page 459).

In Trim mode you can move pre-existing automation data. The channel's volume fader will initially be set to 0dB, regardless of the value of the current automation data. The fader display that shows the value relative to the changes (fader value box) will show the output value as -0.0 dB.



This mode is useful if you want to maintain the existing automation movements, or increase or decrease the overall level.

When trimming a range, new automation points will be set at the range edges. If you move the fader for a relative volume adjustment during playback, the relative

adjustments will be displayed. You can see the current difference to the previously recorded value in the fader value box as well as in the tool tip which appears.

If the option "Use snap for automation curve points" (System Options -> Program -> General) is selected, automation points can be set while moving set values on the grid.

This mode is useful if you want to maintain the existing automation movements, or increase or decrease the overall level.

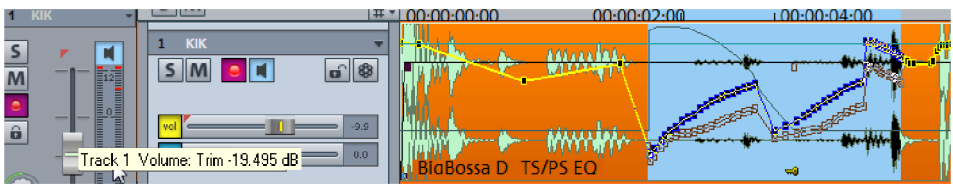
In "Trim" mode, automation data is only recorded as long as you touch the selected control element with the mouse or touch it from your external controller. As soon as you release the button, the automation recording stops. Once released, the control element will move back to the previously recorded position.

You can determine the fader return time in the System Options (Keyboard shortcut: „Y“) under „Effects“ > „Automation“ > „Automation Release Time“.

Trim mode with stopped playback

You can also make relative changes to all volume automation points within a certain area when playback has been stopped.

Select a range for the respective volume events in the track and drag the channel fader in "Trim" mode up or down. You can see the level adjustment difference value in the Tooltip or Fader box.



Note: You can of course activate any number of automation points for all other Automation curves (e. g. EQ, Panorama etc), by clicking whilst holding the Shift key, then change the values of the selected Automation points, again by holding Shift.

Apply to all tracks

"Copy to all tracks" applies the currently selected automation mode to all tracks/channels.

Automate existing curves only

Select the option "Automate existing curves only" to apply the automation data to existing automation curves only. This enables you to experiment with parameters that have not been recorded yet, without having to write automation for these parameters directly. This option should be removed once the new parameters are ready to be committed.

Preview automation (all tracks)

This option temporarily overrides all write automations.

The background is as follows: You have a project that already contains automations. Now you want to add a new section at the end of the project and make changes to the mix there. This creates a problem: When a track is in Read mode, you cannot change existing parameters because they are controlled by the automation curves. With a write automation mode (Touch, Latch, Overwrite), new automations are written immediately and transitions are created between the new parameter values and the existing ones, which you may not want. In addition, new curves are created for parameters that were previously not automated, which means that these parameters are also changed in the existing section of the project.

Follow these steps to resolve this issue:

1. Activate a write automation mode (Touch, Latch, Overwrite)
2. Activate the option **Automation Preview**.
3. Start the playback and make your changes in the mix. Existing automations are not changed. However, if you change a parameter that was not previously automated, a new automation lane is created that retains the existing value (not the changed one!).
4. Stop playback, the play cursor jumps back to the beginning of the new section.
5. Then choose **Set parameter value(s)** from the Automation menu and one of the options to create jumps or transitions.
6. Depending on the option selected, jumps or transitions from the existing to the new automation values are inserted at the cursor position.

You can now deactivate the option **Pre-listen to automation** again and continue working with the automation modes as normal.

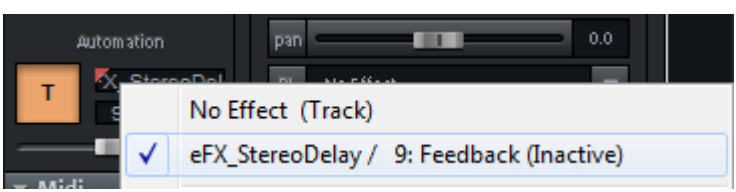
Track automation - Basic procedures

1. For the individual tracks/channels, select the recording mode for creating automation curves. You can select either "Touch", "Latch", "Overwrite", or "Trim" mode for recording. "Overwrite" mode only applies to the last-activated automation curve, and "Trim" mode is intended for volume automation.

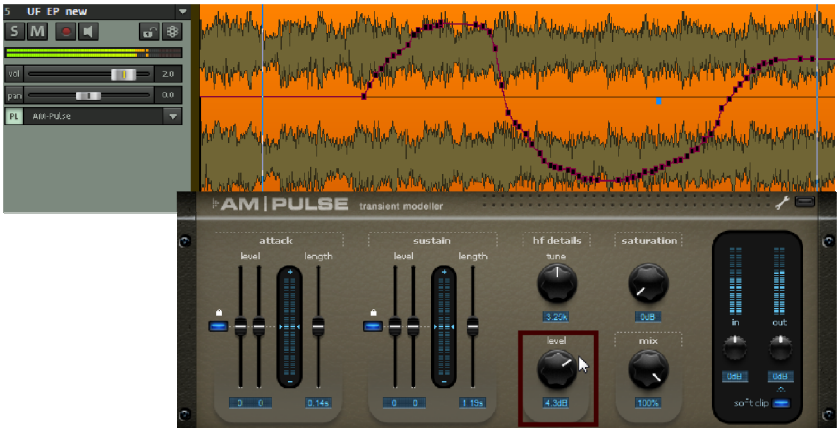
2. If you want to automate a plug-in control element, select "Automate next parameter" in the plug-in menu and then click on the button for the feature you want to use in the plug-in dialog.



If you checked "Show" in the automation section of the Track Editor, you will now see an (inactive) automation curve for this parameter in the Arranger. You can also select the plug-in parameter you want to control by right-clicking on the automation display in the Track Editor.

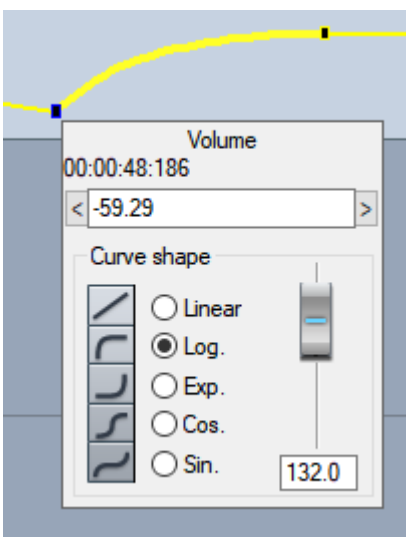


3. Press "Play" to start the automation recording. Whatever changes you made to the selected control element will be recorded in real time during playback.



You can overwrite, edit and copy the automation data at any time. You can draw fine steps in the curve by holding down "Shift".

In universal mouse mode, you can edit the curve points by right-clicking on them. A numerical entry field will appear where you can enter the values you want, or you can use the mouse wheel to change the value in 0.1 dB increments. Holding the shift key while doing this will adjust the value in increments of 0.01 dB. Holding down the "Ctrl" key, in turn, will change the value 1 dB at a time.



You can jump to the next automation point using the Tab key.

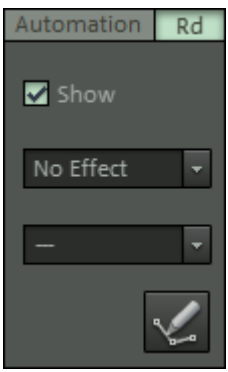
Note: Afterwards, if you want to change the vertical positions of the curve points—that is, their values, not their positions—hold the shift key while moving them. Holding the "Alt" key in addition to this will only move the curve points horizontally. (So you'll be changing their positions and not their values).

Object automation

At object level you can automate the volume, AUX sends, and VST plug-in parameters.

Note: Drawing object automation during playback like such as with the tracks in not possible. You can only create automations as curves and edit these curves.

For automating object parameters, select an object and open the object editor (view page 165).



In the **FX** tab, click "Display" in the section "Automation".

Automation lanes are opened and existing track automation curves are displayed in automation lanes, which frees up space for drawing over the objects in the track.

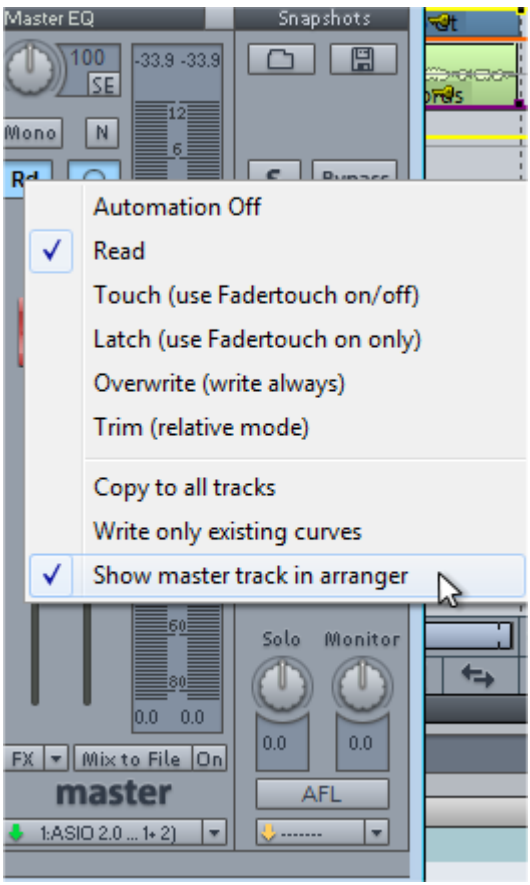
Click on the arrow button on the drop-down menu and search for the parameters you want to automate in the bottom section of the context menu.

Switch to automation draw mode and draw the parameter changes directly in the object.

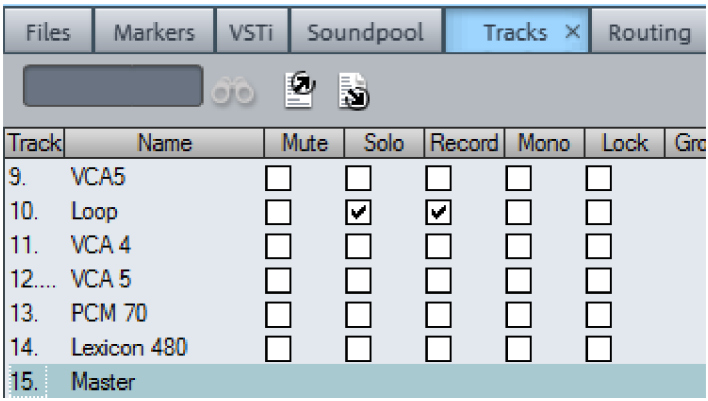
To transfer volume and panorama automation from a track to an object, select the option "Automation" > "Edit Selected Curve" > "Transfer Track Automation to Object". You can reset the object automation to track automation at any time using the command "Switch object automation to track automation".

Master automation

Volume and effect settings for the master track can be automated. To edit these track curves, you must display the master track in the arranger. Right-click the automation button in the master channel and select the option "Display master track in arranger". In the arranger, an additional track with the name "Master" appears, which displays the master automation track curves.

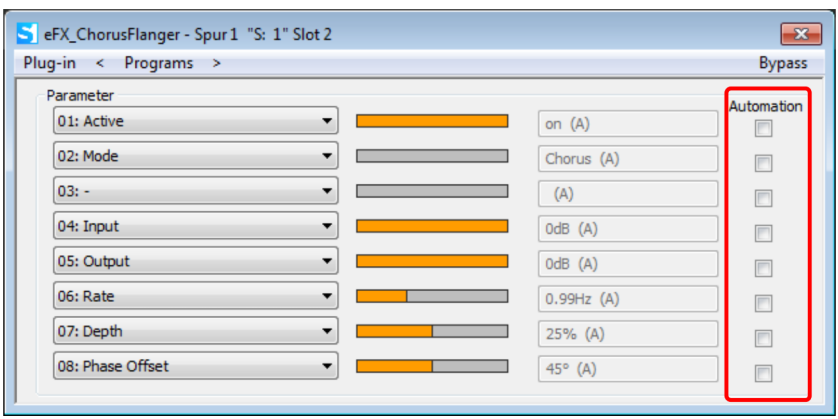


You can also make the master track visible in the arranger window via the track manager (view page 194). To do this, open the track manager and place a check next to "Arrangement" for the "Master" track name.



Automation via VST plug-in/VST parameter dialog

You can also select and activate all parameters you would like to automate in the parameter dialog of the VST plug-in. You can open the parameter dialog in the plug-in's menu via "Plug-in" > "Parameter dialog".



Select the parameter you want from the drop-down menu at the front (the first eight parameters are selected by default) and activate the box under "Automation".

Tip: To display the new parameter in an Automation Lane, select "Lanes for all curves" from the Automation menu or the Automation context menu.

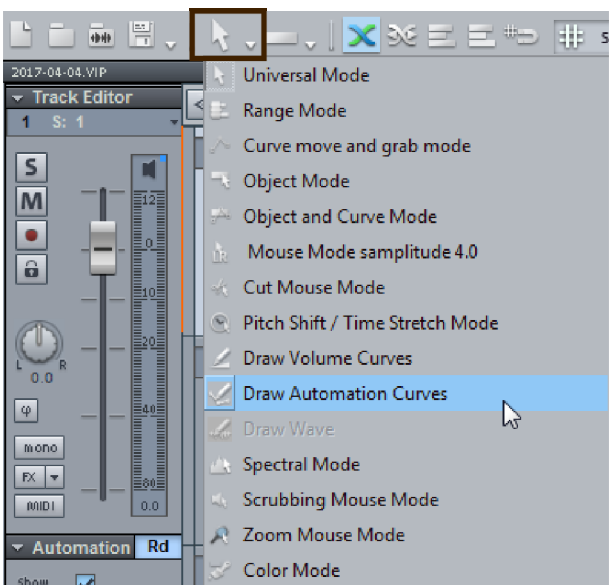
Detailed information about the VSTi parameter dialog is provided in the chapter "Plug-in parameter dialog (view page 288)".

Recording automation in "Read" mode

The easiest way to write automation in "Read" mode during playback is via the "Ctrl + Alt + switch/controller" shortcut, while working directly in the plug-in/instrument/mixer interface. This way allows an automated recording to be made at any time. As long this key combination is held down, you may also automate the selected parameters in "Read" mode. If a mixer/VST control element is clicked with the same key combination "Ctrl + Alt" while stopped, a corresponding inactive curve will be created automatically for editing in "Draw" mode.

Automation draw mode

Select automation draw mode in the toolbar to draw automation curves for a parameter or a MIDI controller.



You can edit an automation curve for the selected parameter using the pencil tool. Simply click anywhere in the track display and then draw the automation curve by dragging with the mouse cursor.

If you press the Shift key while drawing, a curve will not be drawn. Instead, this enables you to drag an individual curve point to the desired location and the curve progresses in a straight line from the previous curve point.

VST parameters are shown in % while drawing. Volume and AUX send parameters are indicated in dB.

Editing Automation Curves

There are various options for editing curves:

- In universal mouse mode, you can create and edit individual curve points.
- With curve mode or combined object and curve modes, you can edit curves in detail
- With automation draw mode you can freely draw curves

Move automation curve with audio/MIDI data

To move the audio material to a new position on the timeline along with the automation curve, the automation curve will have to be connected to the object. This is possible via "Link curves to objects" mouse mode. Click this button in the mouse mode toolbar to get started.



Objects may now be moved or copied together with your curve points.

Note: If you only want to move or copy curve points, follow the instructions above and simply delete the objects once they have been moved/copied. The curves remain intact.

To copy the curve points, simply use "Curve edit" mode and drag out a section by moving the mouse from right to left over a curve while holding down the mouse button, copy the curve points via "Ctrl + C", and then insert them into a different track at the current play cursor position via "Ctrl + V".

Automation – context menu

A number of options are available to open the automation context menu:

- Access the "Automation" menu item
- Right click the "Automation" button in the track editor
- Right click the automation controller in the track editor
- Right click the "Vol" button in the track box

The following functions are available:

The first entry indicates which parameter has been automated, e.g. "Volume" or "Pan". Checking the corresponding entry activates the corresponding automation curve. If an element hasn't been automated, then "No effect (track)" appears.

Editing curves

Create new point at playback position

A new point is created on the selected curve at the playback position. It also opens the editor for the value and waveform (view page 79).

Shortcut: Shift + A

Edit next point at playback position

For the next point from the play cursor position, the editor for the value and waveform (view page 79) is opened.

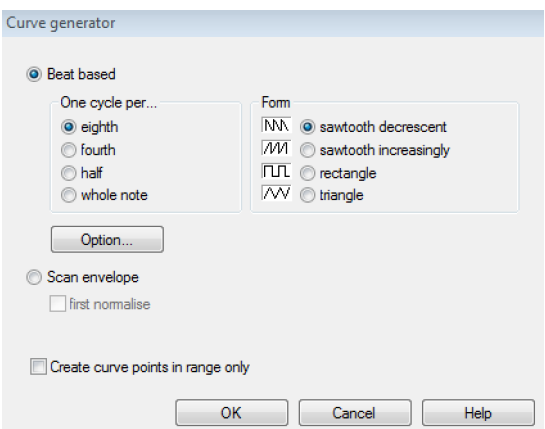
Tip: As long as the editor is open, you can use the Tab key to jump to the next item, and Shift+Tab to jump to the previous one.

Keyboard shortcut: Alt + A

Curve generator

The curve generator will create a volume curve for you that is beat based and pulses according to eighth, quarter, half or full notes. You can enter a minimum and maximum value or define delay values as an option for the beat-based envelope curve calculation.

The option "Create..." in the automation context menu opens a selection dialog to define the shape of the automation curve more accurately.

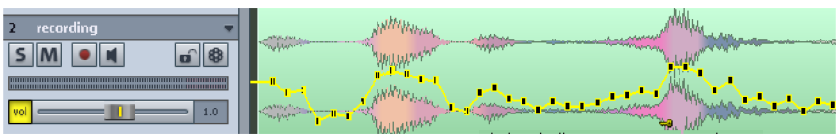


Beat-based: This option causes the form of the automation curve to follow the beat in the arrangement.

There are 4 basic patterns available for beat-based automation curves: 2 saw-tooth shapes, rectangle, and triangle. These shapes run depending on the beat and control the intensity of the activated effect. The left side of the dialog enables the automation curve to be set to activate once per eighth, quarter, half, or whole note.

Options: This opens an additional dialog to control the style and intensity of the influence of the beat on the automation curve (see below).

Scan envelope curve: Allows the volume process to be displayed as an automation curve as an alternative to a beat-based automation curve.

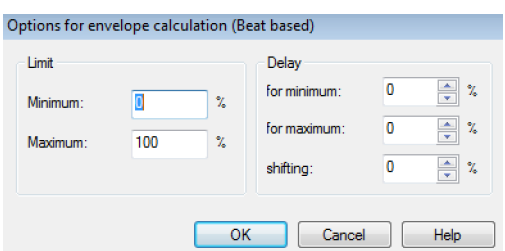


First normalize: This function normalizes the level of the audio object before the object is scanned to create an automation curve.

Create curve points in range only: If this option is selected, curve points will be generated only in the area previously defined in the arranger.

Curve generator - options for envelope calculation (beat-based)

Limitation: Determine the minimum and maximum value of the envelope. Via Delay Minimum/Maximum the basic forms can be altered further.



Shift moves the entire curve. This lets you create interesting off-beat effects.

Edit curve – invert

This command inverts the activated automation curves.

Edit selected curves – thin out

While recording, the automation events are placed in very short intervals. The command "Thin out" reduces the number of curve points. The course of the automation is then displayed and reproduced even more accurately.

Edit curve – transfer track automation to object

You can use this function to transfer volume and panorama automation to objects. If the track in question already has automations, you will be asked if you want to overwrite existing track automations or overlay them with a new curve.

Edit curve – transform object into track automation

You can use this function to transfer volume and panorama automation from objects to tracks.If the track in question already has automations, you will be asked if you want to overwrite existing track automations or overlay them with a new curve.

Edit curve – inactive

Use this command to deactivate the selected automation curve. The curve remains intact, but will be displayed as a dotted line and will no longer affect the corresponding parameter.

Edit curve – curve color

This command displays a color palette for you to specify the color of the active curve.

Logarithmic

This option lets you switch volume curves to logarithmic display.

Edit curve – copy, paste, delete

The activate automation curve may also be copied, pasted into other tracks, or deleted.

If the option "Use snap for automation curve points" (System options -> Program -> General) is selected, automation points may be set while moving set values on the grid.

Delete all curves

This command erases all automation curves on each track automation or each object automation.

Automation Mode

Specify the Automation mode here.

You can find detailed information in "Automation modes (view page 458)".

Set Parameter value(s)

This command creates new automation points in all active and inactive automation curves of the selected or all tracks according to the currently set parameter. This allows, in connection with the option **Automation Preview (view page 462)** , to explicitly write the automation of the entire project at a certain point, in order to prevent existing parameters from being changed by new automations and unwanted transitions from occurring. The following options are available:

- **Jump to current value:** New points are inserted as jumps in all automation curves.
- **Jump to current value:** New points are inserted as jumps in all automation curves of the selected track.
- **Transition to current value:** New points are inserted in all automation curves as a transition to the previous value.
- **Transition to current value (panorama only):** A new point is inserted in the panorama curve of the selected track as a transition to the previous value.
- **Transition to current value (selected track):** New points are inserted in all automation curves of the selected track as a transition to the previous value.

Keyboard shortcut: Shift + Alt + #

A popup menu appears at the mouse pointer, select the corresponding option with another keystroke (underlined letter of the option).

MIDI Controllers

For each track you can also draw MIDI controller curves automatically and send them to a VST plug-in or external MIDI synthesizer via the track MIDI output device. In the menus select the desired MIDI controller for the current track.

To make the automation curve visible, activate the "Show" option in the automation section of the track editor. Now you can edit the displayed automation curve Automations Draw Mode (view page 86, view page 468).

Do not display automation

With this option, you can deactivate the display of the automation curves for all tracks.

Display track automation (default setting)

If you select this display option, only the track automation will be displayed.

Show lanes and object automation

If you select this display option, only the object automation will be displayed for all tracks. Simultaneously, this option opens the automation lanes and displays the existing track curves in the lanes.

Show Only Selected Curve

Views only the selected curves for the corresponding track. This helps to maintain a clear overview if several automation curves have been created.

Show unselected curves (not selectable)

The unselected automation curves are also shaded, but may now be activated with the mouse.

Show unselected tracks (selectable)

The unselected curves are also shaded, but may now be activated with the mouse in universal mode/curve editing mode/automations drawing mode.

The last section of the context menu lists the ranges that automation parameters are available for when automating the corresponding track or the corresponding object.

Select previous curve

With this you can select the last edited curve.

Select Next Curve

With this you can call up the next curve from the selection list.

The last section of the context menu lists the ranges for which automation parameters are available during automation of the corresponding track.

Surround Sound

Sequoia offers several possibilities for mixing audio material in various multi-channel formats - from LCR (Left-Center-Right), to Quad (Left-Right-Left Rear-Right Rear) or various 5 and 6 channel formats and even up to 22.2 3D Surround.

A prerequisite for this is a mixer configured with at least one Surround Master corresponding to the target format. Examples of such so-called target formats are, for instance, 5.0 ITU, 5.1 ITU or 7.1 SDDS.

In the „Surround Settings“ you can assign Surround Master channels to available playback devices. The audio material of the tracks in an arrangement that are routed to a Surround master can be distributed across the available surround outputs in the Surround Editor. In the mixer, the panorama knob in these tracks is replaced by a Surround panorama display.

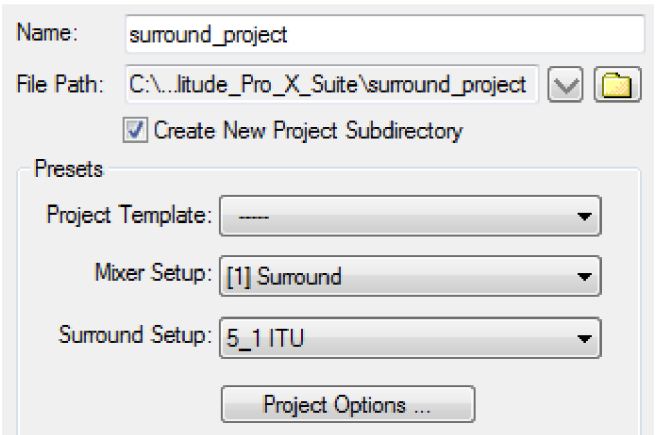
You can also route individual objects directly to the surround master and distribute them in the surround panorama, regardless of whether the track on which they are placed is routed to a stereo or surround master.

The Surround Editor provides seven different modes for routing the signal into the Surround panorama. Various surround effects are provided in the mixer for mixing in multi-channel formats.

You can create any number of additional master buses, e.g. in addition to a 5.1 mix you can create a 7.1 mix in the same project. A master bus can also be routed to any other surround or stereo master, while the required upmix or downmix is performed.

Create new Surround Project

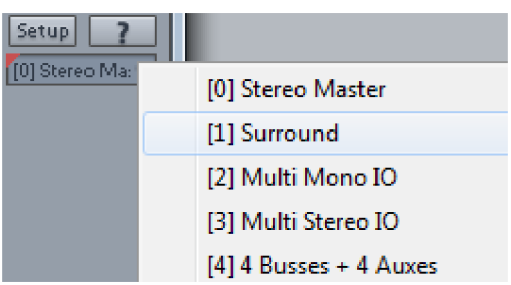
In the "Setup for new Project (VIP)" dialog, select the option "Surround" in the "Mixer Setup". Select the desired format under "Surround Setup" (e. g. 5.1 ITU).



In addition to the audio track channels, the mixer in the new VIP now also contains the Surround Master in the selected format (e. g. 5.1 ITU L, R, C, LFE, Ls, Rs). All mixer channels for the corresponding project tracks are automatically routed to the Surround Master. The mixer's Stereo Master will be hidden initially, but it can be displayed by clicking the "hide Master" button in the mixer. Routing of the individual mixer channels to both the Surround Master and the Stereo Master is possible.

Converting an Existing Stereo VIP into a Surround Format

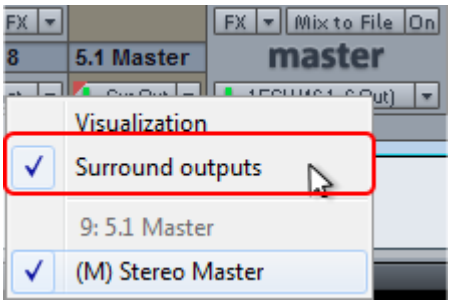
In the mixer of an existing VIP click on the mixer "Preset" selection button at the bottom right corner under the "Setup" and select "Surround". This opens the „Surround Setup“ (view page 477) dialog where the you can select the surround format (e. g. 5.1 ITU).



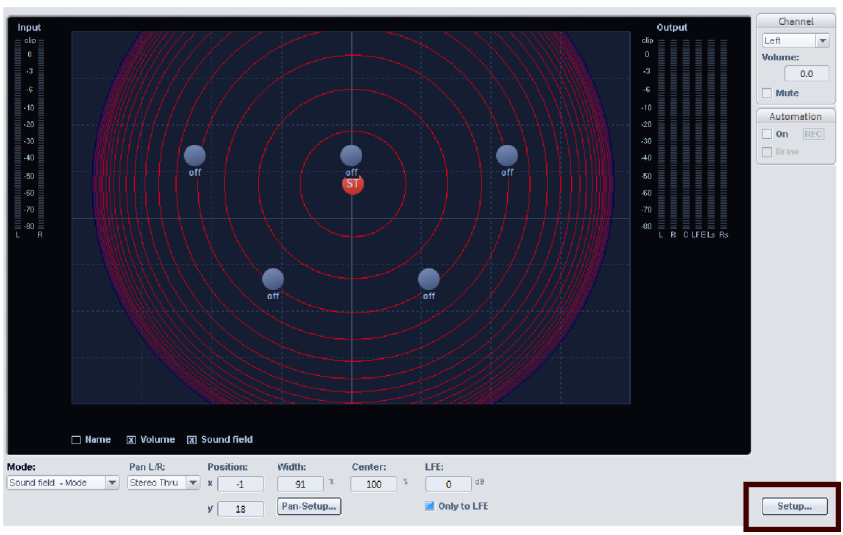
Surround Settings

Setting the Physical Outputs of the Surround Buses

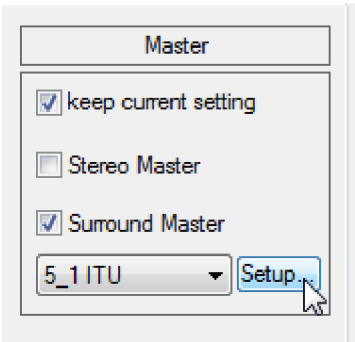
The "Surround Setup" dialog window opens when you click on the outputs of the of the Surround Master in the mixer and select "Surround outputs..."



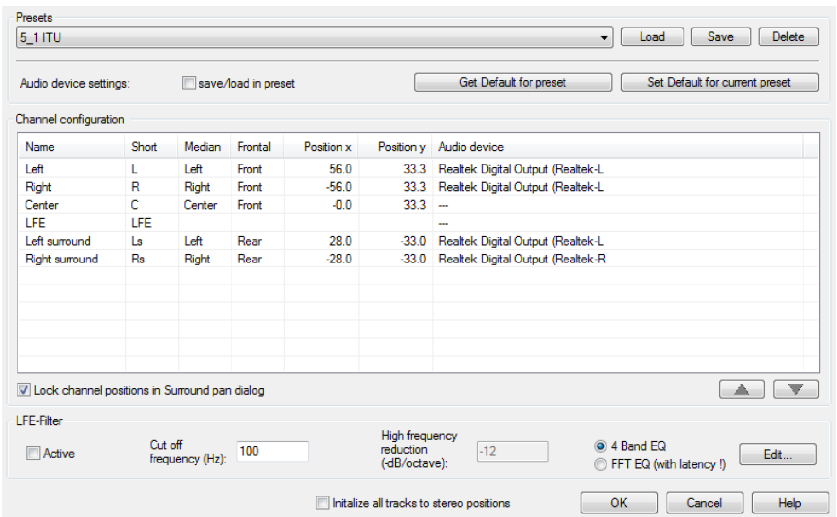
or when the "Setup..." button in the „Surround Editor“ is clicked. (view page 483)



It can also be opened by clicking the "Setup..." button in the master section of the „Mixer Setup“ (view page 658).



The surround format for mixing is set in the "Surround Setup" window. Here you can open various saved presets (e. g. 5.1 ITU, DD, DTS, LCR, LRS, Quad...) or edit and save your own settings. The filter settings for the LFE (Low Frequency Effect) channel are also set here.



Presets

Previously added Surround formats may be accessed and new formats may be saved here. A preset features the Surround buses, their name as an abbreviation, the sequence, and the position coordinates of the speaker. When loading a preset, the playback device settings will feature the default values set for the preset.

Loading Standards for Presets

Clicking on the buttons loads the default configuration of the playback devices belonging to the currently active preset.

Setting the Default for the Current Preset

Pressing this button saves the current assignment of the Surround buses to the playback devices as a configuration of the currently active preset. This configuration is then always loaded when the preset is used.

Loading & Saving Audio Device Settings with Presets

Activating this option saves the current playback device configuration during playback, independent of the default configuration in the preset.

Channel Configuration

Name Determines the name of the individual surround buses. The abbreviation in the following column is generated from this name. The first letters of each word (or capitalized letters) are used for this abbreviation.

Left = L

Right = R

Center = C

Left surround = Ls

Right surround = Rs

Low Frequency Effect = LFE

These abbreviations are used to label the corresponding channels in the mixer and in the Surround Editor. During Surround track bouncing the abbreviations are automatically added to the resulting WAV files as the file name.

Example: Select the name "SurroundMix" when Surround bouncing a 5.1 Surround mix as 6 mono files. This will generate the following WAV files: „SurroundMix_L.wav“, „SurroundMix_R.wav“, „SurroundMix_C.wav“, „SurroundMix_Ls.wav“, „SurroundMix_Rs.wav“ und „SurroundMix_LFE.wav“.

Median / Frontal These columns describe the position of the respective loudspeaker in relation to the median and frontal levels.

Position X / Position Y: This describes the precise position of the speaker in a system of coordinates. You can edit the coordinates manually. The settings refer exclusively to the loudspeaker arrangement in Sound Field Mode.

Audio device: Assignment of a physical output on an existing sound card for the respective Surround bus. For each Surround preset you can save a default device setting (view page 479).

Lock channel position

If this is selected, the position of the loudspeakers cannot be changed using the mouse in the Surround panning module. This function is always initially activated and should only be switched off in special cases (e.g. variable speaker positioning).

Up arrow/down arrow buttons

If a line has been selected in the configuration table (corresponds to a Surround channel), this line may be moved up or down within the table.

The order of the Surround channels in this table determines the order for display of the mixer, Surround Editor (e.g. peak meter), and the Surround effects windows.

LFE Filter

You can activate a filter for the LFE channel in "Surround Setup". You can choose between a 4-band EQ (default setting) and an FFT filter. These filters can be adjusted by clicking on "Edit...".

Cut-off frequency: Here you can set the cut-off frequency for the low-pass filter.

High Frequency Dampening: The dampening of signal above the cut-off frequency is set to 12 dB/octave for the 6-band EQ.

Edit EQ: A dialog for the filter settings opens here.

Set All Tracks to Stereo Position

The downmix is factored in during a stereo master bounce project, i. e. you can stereo bounce from a surround project when the downmix (view page 496) is set.

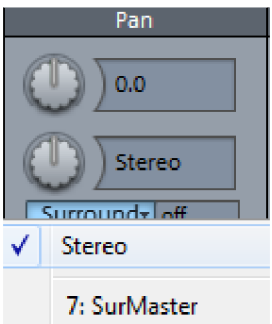
Surround panning

Now that the basic settings have been made, proceed to the actual mixing. There are two methods for mixing audio material displayed in the VIP window into the Surround master:

Track-based Surround Panning

In the Surround Editor (view page 483) of the corresponding channel strip in the mixer, each audio track in the arranger can be assigned to a position in the Surround panorama. All of the objects on the respective track will then be positioned at this location in the Surround panorama.

The Object Editor of this object displays the following the panorama setup:

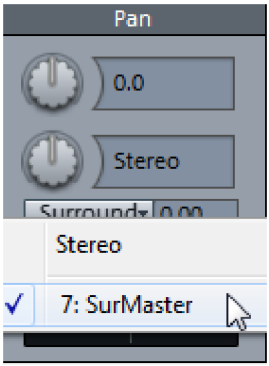


The selection "Stereo" under "Surround" allows you to route the object to the corresponding track, i.e. to use track-based Surround panning. In this case the audio signal runs through the channel of the respective mixer track. This is the default setting for Surround projects in the Object Editor.

Right-clicking in the Surround Panorama Field opens the corresponding Surround Editor (view page 483). Here you can set the positioning in the Surround panorama.

Object-based Surround Panning

In the Object Editor, each object can be routed to the Surround Master or another Surround submix bus. To do this, open the Object Editor of the object to be edited with a double-click and set the "Surround" button in the "Pan" section to "SurMaster".



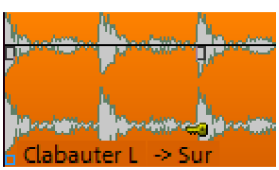
The setting "SurMaster" provides access to direct routing of the object to the Surround bus, i.e. to object-based Surround Panning.

An additional controller appears in the object editor which can be used to adjust the object's send level to the Surround Master.



Right-clicking in the Surround Panorama Field opens the corresponding Surround Editor (view page 483). Here you can set the positioning in the Surround panorama.

Object-based Surround panning is displayed in the object view in the arranger with the appendix "Sur".



Note: Object-orientated Surround panning will not route the audio signal through the channel strip of the corresponding mixer track. All settings made here (AUX send, EQ, etc.) have no effect on this object.

Track-based and object-oriented Surround panning gives you the freedom to choose between one of the two processes. If you hold down the "Ctrl" key while making a selection under "Surround" in the Object Editor, you can even select "Stereo" and "SurMaster" simultaneously and combine both types of Surround panning.

Surround Editor

In the Surround Editor the panning of a track or object on the Surround bus can be set and edited.

The following options are available for opening the Surround Editor for track-based Surround panning:

- Right-click the Surround Pan Display of the corresponding channel strip / track editor or
- Right-click on the "pan" button / panorama controller in the corresponding track header or
- Go to "Track" > "More" > "Pan/Surround Editor"

The Surround format and the track currently being edited are shown in the title bar of the Surround Editor. Example: "Surround Editor: 5_1 ITU - Track 4"

To open the Surround Editor for object-based Surround panning, double-click on the respective object to open the Object Editor. Right-clicking on the Surround Pan Display opens the Surround Editor for the respective object.

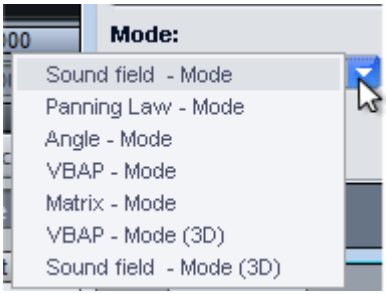
The Surround mode of the track currently being edited is shown in the title bar of the Surround Editor.

Display Elements in the Surround Editor

According to concepts of stereo tracks, a two-channel input meter is located on the upper right in Sequoia. Mono input signals are displayed in both scales of the input meter.

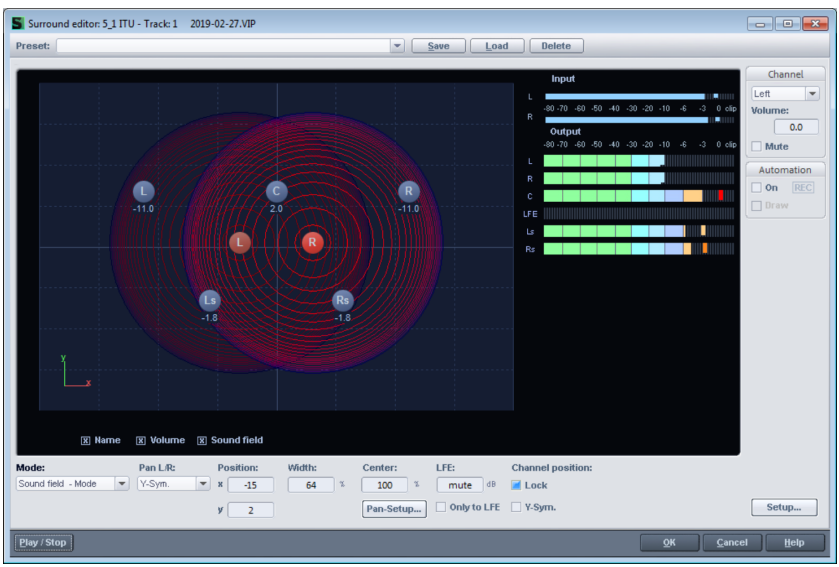
The level meters of the individual channels in the Surround Master are shown below it. Only the level components from the material currently being edited in the Surround Editor are displayed.

Under "Mode" you can choose from seven different display modes: Sound field mode, Panning law mode, Angle mode, VBAP mode, Matrix mode, and the 3D modes VBAP (3D) and Sound field (3D).



Note: The following remarks about the general display elements apply to all modes except the Matrix Mode. In addition, mode-specific parameters are explained in „Panorama Mode in the Surround Editor“ (view page 488).

The loudspeaker channels of the Surround setup are displayed as blue dots in the main panorama window. Their position in the Surround Panorama Field is dependent on the selected mode. If a channel is shut off using "Mute", it is displayed as a gray dot. The sound source to be positioned in the Surround panorama is displayed as a red dot. Depending on the "Pan L/R" setting, a stereo sound source can also be displayed with two dots.



The following information is also displayed under the main panorama window:

Name: The loudspeakers are labeled with the abbreviations specified in the "Surround Setup" window.

Volume: Level information is displayed for each loudspeaker. This displays the value of the signal component that is passed on to this bus by the sound source. The sum of the levels of both stereo sources is displayed for the stereo panorama setup under „Pan L/R“ (X-Sym., Y-Sym., XY-Sym., Parallel). If you hold down the "Shift" key and simultaneously click one of the two sound sources, only the level for this source will be displayed for a short time.

Sound field: Depending on the mode being used in the Surround Editor, the sound field is displayed either as a red surface or as concentric circles..

You can shut off the display information for "Name", "Volume" and "Sound Field" by deactivating the corresponding fields.

Pan L/R: In the Surround Editor, both mono and stereo sources can be positioned in the panorama. The function "Pan L/R" determines which way the mono and stereo signals will be arranged. For more information, please read the section entitled "Stereo and mono signal processing in Surround projects" (view page 493).

Position: In the Surround Editor, you can move the sound source with the mouse to the desired position in the panorama. Enter the position directly in the numeric fields "x" and "y" or change the numbers by moving the mouse pointer vertically on the numeric field or with the scroll wheel.

You can also move the sound source directly on the interface using the mouse. By holding down the following keys you can coordinate the position setting:

x + **Mouse movement:** Only a position change parallel to the x-axis is possible.

Result: L -> R movement

y + **Mouse movement or** Z + **Mouse movement:** Only a position change parallel to the y-axis is possible.

Result: Front -> Rear movement

c + **Mouse movement:** The distance from the sound source to position 0 (Px=0, y=0) remains the same. The result is a circuit.

Result: Circular movement

a + **Mouse movement:** The sound source can only be moved diagonally. A line running from the initial position of the sound source through point 0 of the system of coordinates determines the movements.

Result: Diagonal movement with consistent angles.

Pan Setup: (view page 487)

Center: This parameter controls the portion of the center channel for assignment of the sound source to the front channels. For certain applications (e. g. film sound) it is normal to reserve the center channel exclusively for dialog and to position music and background noises outside of it. A signal placed directly in the center is played back only by the center channel in 5.1 format if the center = 100%; at 0% this is only a phantom sound source that is output by the left and right channels. This parameter is often referred to as "divergence".

LFE(Low Frequency Effect): In this field you can determine the level component of the signal that will be routed to the LFE channel.

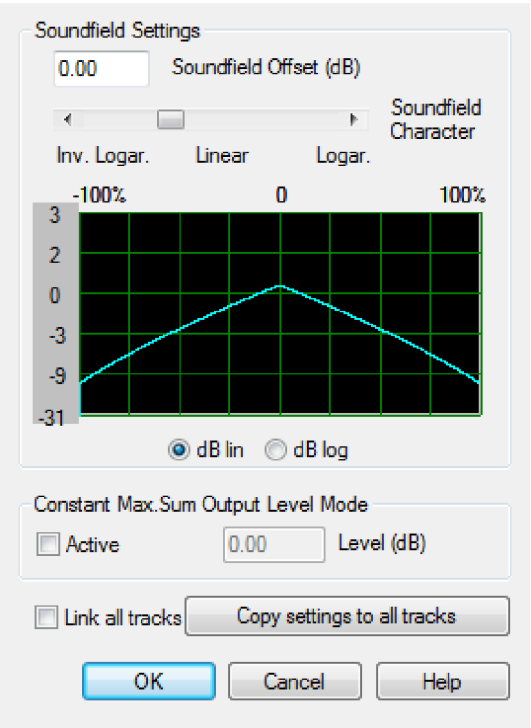
LFE only: The source is routed only to the LFE channel.

Channel: Mutes or changes the level of the bus outputs of the Surround Editor to the Surround buses. Deactivated bus outputs/speakers are displayed as gray dots in the panorama field.

Automation (view page 497)

Pan Setup

In the "Pan Setup" dialog window you can make more settings for the Surround Editor. The dialog window can be opened by clicking on "Pan Setup" in the Surround Editor or right-clicking on the sound source.



Sound field offset (dB): The set offset is included for sound sources that are directly positioned and routed to a single Surround channel/speaker. For example, signals traveling from directly routed positions to panning between individual channels can be compensated if they are too strong.

Sound field characteristic (only available in the Surround Editor in Sound Field or Angle Mode): The sound field characteristic can be adjusted smoothly. "Invers. logar." produces a quick drop of the sound field and short fades between loudspeakers. "Logarithmic" provides a slow reduction of the sound field and longer transitions between speakers.

Maximum sum output level constant: If this function is activated, the maximum level of all channel outputs of the Surround Editor will not exceed the set value. This means that unintentional fluctuations of the max level can be counteracted as positions

change. This function is particularly useful when you want to move certain sources through the move using automation.

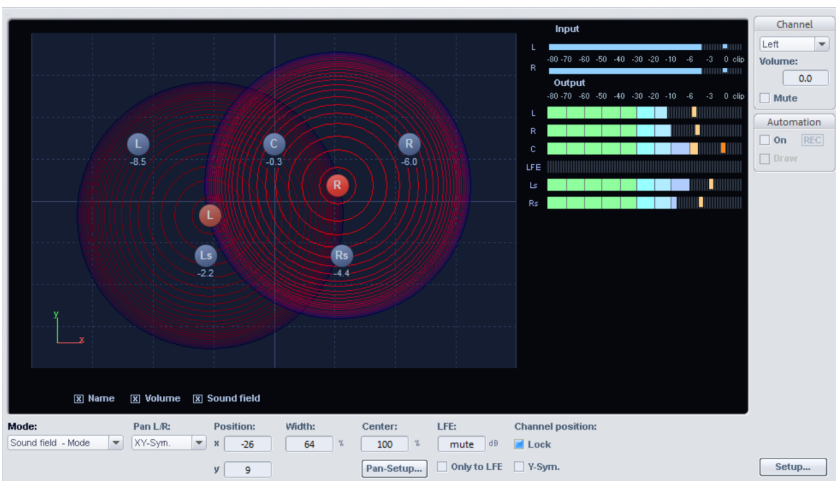
Link all tracks: Changes in the following settings within the Surround Editor are always transferred to all tracks in the arranger: Sound Field Offset, Level Drop Characteristics, Output Volume Sum Constant, Center, Pan-law/Width, LFE level and settings, and channel settings such as level changes or mute.

Copy settings to all tracks: The settings made in this Sound Panorama Module (sound field offset, level drop characteristics, output volume sum constant, center, pan-law/width, LFE level and settings, channel settings such as level changes or mute) are copied to all other tracks once.

Panorama Modes in the Surround Editor

Sound Field Mode

In Sound Field Mode the input signal is displayed as a concentric sound field. Each red line corresponds with a 3dB drop in the sound field level. The loudspeakers are positioned in such a manner that the distance from a single loudspeaker to a neighboring speaker remains constant. This positioning permits a uniform distribution of the sound source across all channels. Level ratios develop between the channels, which could not be achieved in other modes.



Example of use: Exact position localization, particularly during movements.

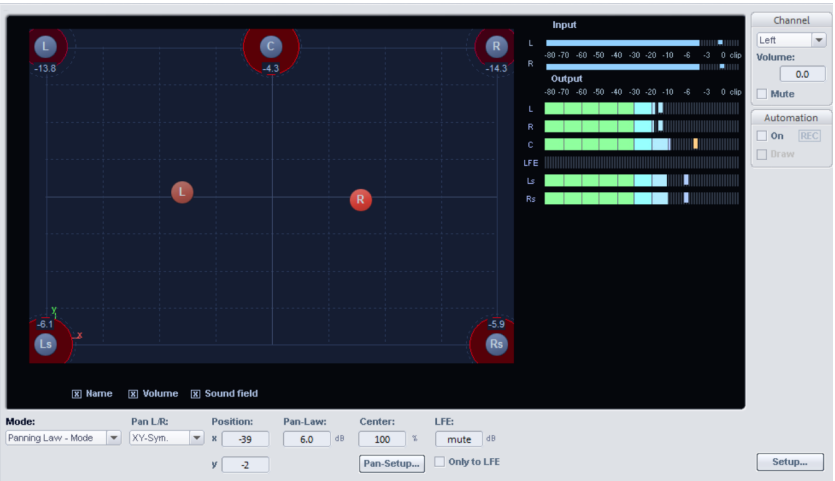
Level conflicts can occur for movement in a constant direction (e. g. a jet flying overhead). In this case, Angle Mode is better.

Width: This parameter influences the sound field width of the loudspeakers.

Pan Setup (view page 487) – Sound Field Characteristic: The sound field characteristic can be adjusted smoothly. "Invers. logar." produces a quick drop of the sound field and short fades between loudspeakers. "Logarithmic" provides a slow reduction of the sound field and longer transitions between speakers.

Panning Law Mode

This mode works with one of many digital mixing consoles in the usual Surround panorama display.



The position of the loudspeakers are displayed on the outer borders of the useable panorama. By clicking the "Sound field" option, you can visually display the graphical levels displayed across the Surround buses. The level distribution between two neighboring loudspeakers follows the -3 dB law, which means that a sound source directly in the middle of two neighboring loudspeakers will be emitted at a level of -3 dB.

Example of use: Static 2-dimensional panning, basic rough localization.

Because Panning Law Mode does not enable 100% exact localization, it is not suitable for dynamic panning (e. g. Automation).

Pan Law: To compensate for volume fluctuations during panning you can also lower the volume of the pan mid position here.

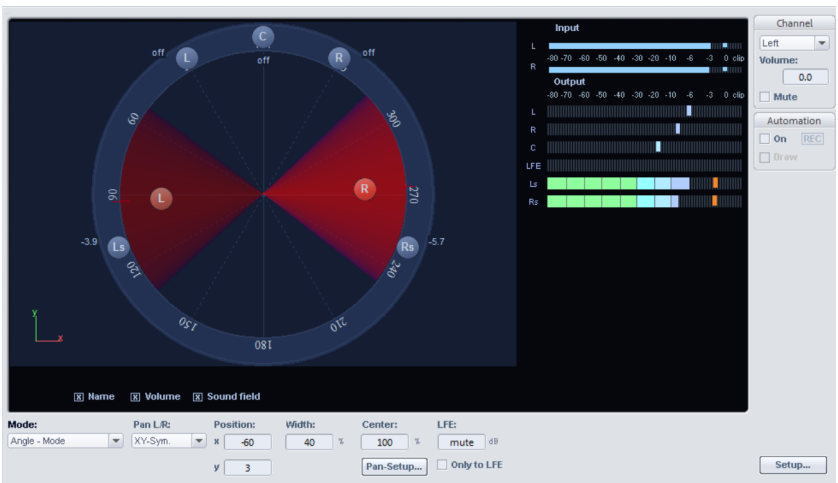
Angle Mode

Here a sound field is displayed that radiates from the middle point of the circle. The sound source is located on the middle axis of this sound field. The loudspeakers are arranged on a sphere. The level components of a sound source on the respective

channels are determined by the angle ratio between the sound source and the loudspeaker as well as the opening angle of the sound field. If the angle of the sound source and the channel perfectly match (i. e. the middle axis of the sound field is pointing directly at the loudspeaker: angle difference = 0), the level on this channel is the highest. The level on a channel drops as the angle difference increases.

Example of use: Good localization of direction of movements (e. g. jet flying overhead).

Angle Mode is not suitable for distance panning.



Width: This parameter determines the size of the sound field opening angle.

Pan Setup (view page 487) – Sound Field Characteristic: The sound field characteristic can be adjusted smoothly. "Invers. logar." produces a quick drop of the sound field and short fades between loudspeakers. "Logarithmic" provides a slow reduction of the sound field and longer transitions between speakers.

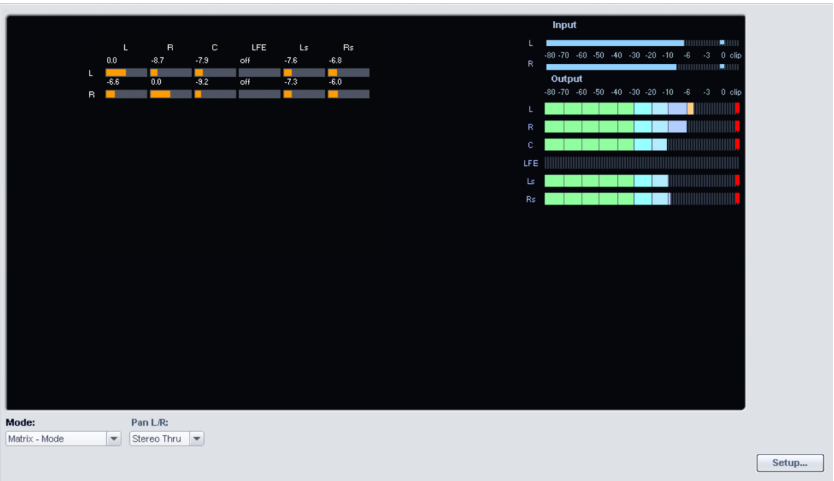
Pan Setup (view page 487) – Maximum sum output level constant: If this function is selected, the maximum level of all channel outputs of the Surround Editor will not exceed the set value. In Angle Mode, level drops in the sound field in connection with large opening angles can be avoided during movement.

Matrix Mode

In Matrix Mode you can directly enter the level that will go from the input signal to the individual Surround buses.

To enter the value manually, double-click on the numeric field. You can also set the level by dragging with the mouse over the level bar while holding down the mouse button. For very fine adjustments, hold down "Shift" key while doing this.

Examples of use: Analytical tasks like routing after track bouncing or simultaneous distribution of a signal over several Surround channels. Distribution over 3-dimensional alignments, e. g. 2+2+2 setup.



In contrast to the other modes, the settings from "Pan L/R" in Matrix Mode have the following meaning.

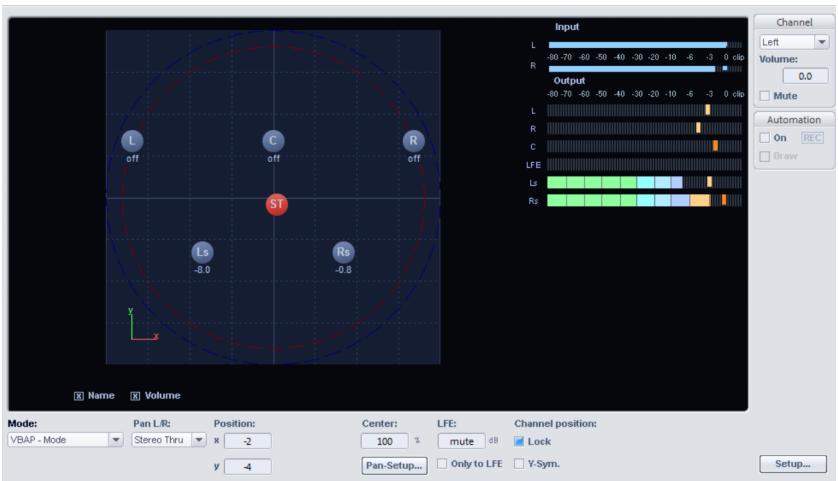
Pan L/R mono: The levels of the left and right input signals are applied equally.

Pan L/R stereo: Controls the levels of the left and right input signals individually.

The remaining "Pan L/R" settings do not apply. No reflective sound sources will result.

VBAP

The VBAP mode (Vector Based Amplitude Panning) is characterized by a higher precision when generating phantom sources. It can be used as an alternative for the Sound field mode or the 3D Sound field mode.

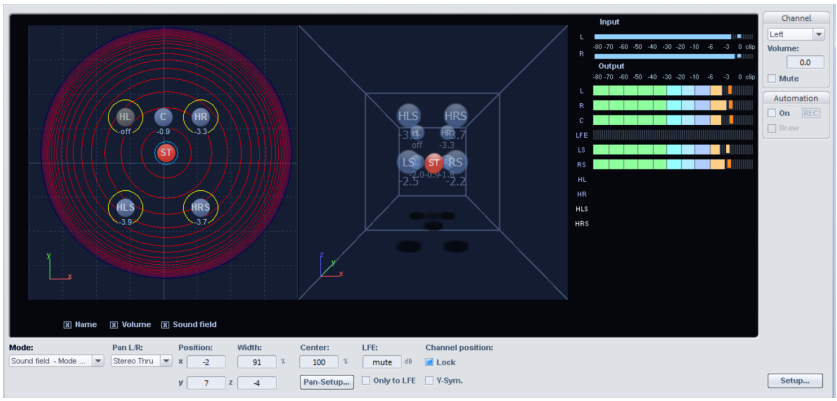


The operation differs from the sound field mode only in the point that no specification of the width of a loudspeaker sound field is required.

3D Surround (Sound field/VBAP)

The 3D Surround modes of the Surround Editor, which are available for the Sound Field Mode and VBAP modes, differ from these only in the addition of a 3rd coordinate z to indicate the height of the sound source.

To the right of the top view there is a second, spatial display for the visual adjustment of the height coordinate.

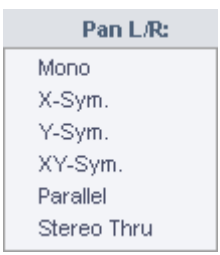


Surround presets

Save your own frequently used settings of the Surround Editor as presets here. In addition to the position of the sound source, the mode and the settings of Pan L/R are also saved.

With "Preset > Load" you can recall your presets. All surround presets are available also with a left click on the Surround pan display in the mixer. There are also a couple of pre-defined settings for frequently used pannings.

Pan L/R: Stereo and Mono Signal Processing in Surround Projects



Mono: For stereo signals a mono sum is created from the left and right channels. This is positioned as a single mono sound source in the panorama.

Mono signals are positioned as a single mono sound source in the panorama.

X-Sym: The following applies to stereo signals: The left and right channels are symmetrically aligned to the x-axis. For example, this enables you to pan a stereo signal to front L (Left) / LS (Left Surround).

The following applies to mono signals: A second (mono) mirrored sound source of this signal is positioned in addition to the original mono source.

The x-axis is the mirror axis.

Y-Sym: The following applies to stereo signals: The left and right channels are symmetrically aligned to the y-axis. For example, this allows you to pan a stereo signal to front L (Left) /R (Right).

The following applies to mono signals: A second (mono) mirrored sound source of this signal is positioned in addition to the original mono source. The y-axis is the mirror axis.

XY-Sym: The following applies to stereo signals: The left and right channels are symmetrically aligned to the x and y axes. For example, this enables you to pan a stereo signal to front L (Left) / RS (Right Surround).

The following applies to mono signals: A second (mono) mirrored sound source of this signal is positioned in addition to the original mono source. This mirrors the original source in relation to the X and Y axes.

Parallel: The following applies to stereo signals: The left and right channels are kept in constant distance from each other and are moved in parallel. The distance between the two sound sources can be changed when the Ctrl key is pressed.

The following applies to mono signals: The original and mirrored sound sources are kept in constant distance from each other and are moved in parallel. The distance between the two sound sources can be changed when the Ctrl key is pressed.

Stereo Thru: For stereo signals, depending on the position of the sound source, the same level ratios are reproduced on different channels. However, only the left signal is used for all left channels, and only the right signal for all right channels, and the mono share for the center and LFE channels.

The following applies to mono signals: No special function, identical with mono mode.

Working with multiple Masters.

In <Program name_short> it is possible to create further surround masters in addition to the master defined by the mixer setup when creating a new project (stereo or any surround configuration). This makes it possible to provide different multi-channel mix variants in the same project.

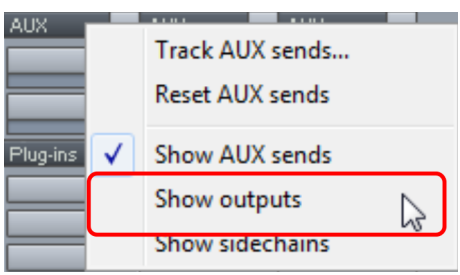
To create a new master, select the command "Menu Track" > "Insert new tracks" > "New Surround Master". The new master is placed in the mixer to the left of the existing one.

Each track / bus can be routed either to each master individually or to several masters.

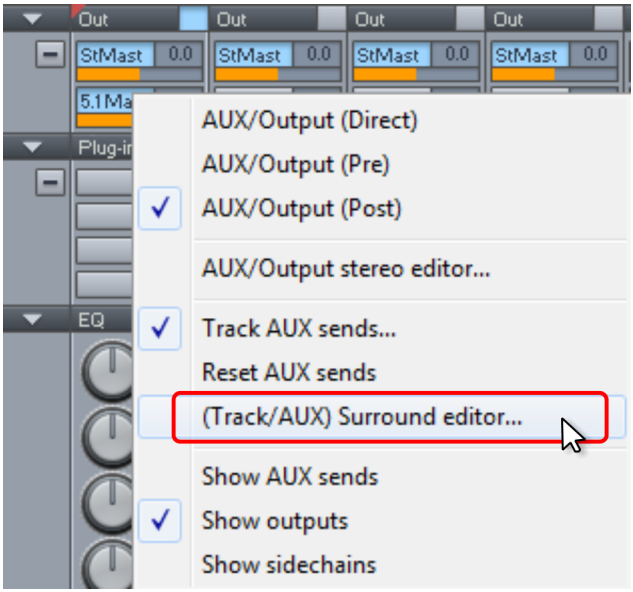
This means that several different mixing scenarios are now possible:

- Create a 7.1 surround mix, create an additional 5.1 surround master and create a downmix of this mix in 5.1 (see "Surround up or down mix" below).
- Create an additional surround master in the stereo project and route individual tracks additionally to the surround master. (In the Mixer chapter you can find more information about using multiple outputs (More information can be found in the section "Assign track to multiple outputs" on page 371).) Create a simple surround version of your stereo mix.
- For greater control over the additional surround mix, you can use Surround busses or Surround AUX busses.

In a combined routing to stereo and surround masters, the stereo Pan control is displayed on the mixer interface, although a surround editor is also active for the surround mix of this channel. To access the Surround Editor interface, set the AUX section of the mixer to "Show outputs".



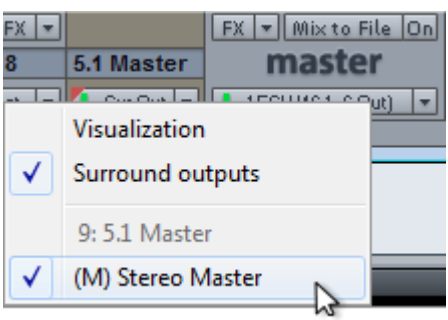
Now right-click on the slider of the corresponding bus/master and select "(Track/AUX) Surround Editor..." to open the Surround Editor.



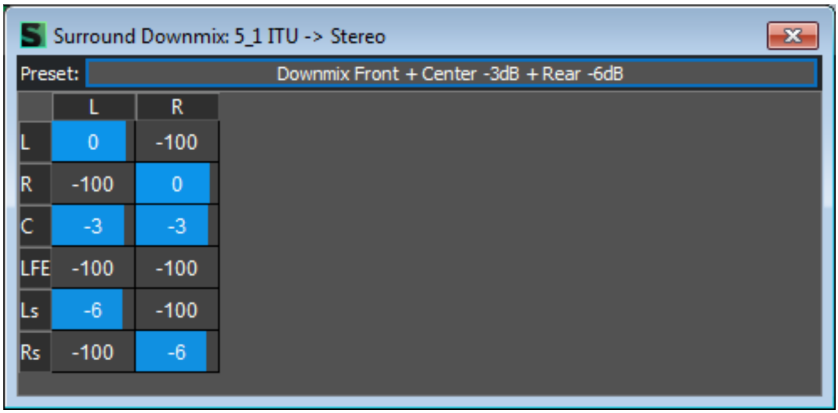
Surround up or downmix

A master bus can also be routed to any other surround or stereo master, while the required upmix or downmix is performed. For example, you can create a stereo mix on the stereo master from a project with a surround master or, in addition to a 7.1 surround mix, a 5.1 surround mix.

Open the output menu in the mixer of the corresponding Surround master and select the additional master while holding down the Ctrl key.



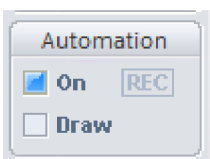
To open the downmix dialog where you can set the downmix coefficients, select the Master in the Output menu again.



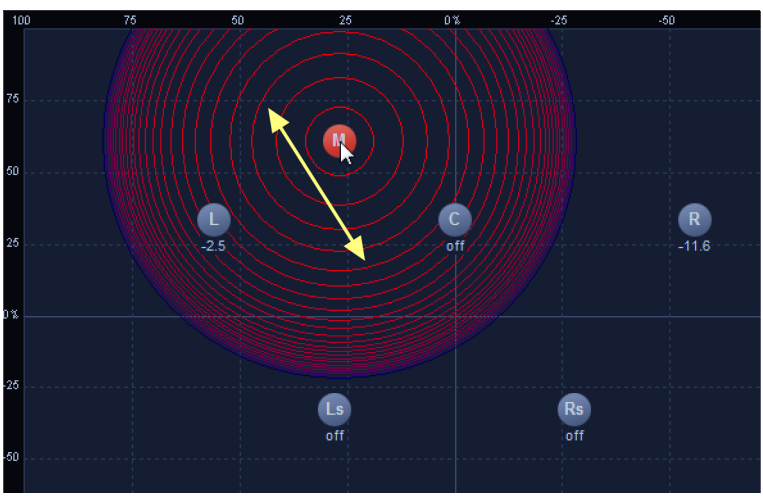
In the preset list above there are common presets for downmixes of 5.1 in stereo. Additionally, you can also set and save your own downmix coefficients. The downmix is accounted for during Stereo Master bouncing.

Track-based Automation in the Surround Editor

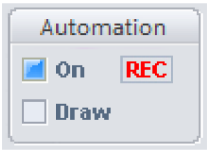
To automate panning movements, first activate the Automation option in the Surround Editor.



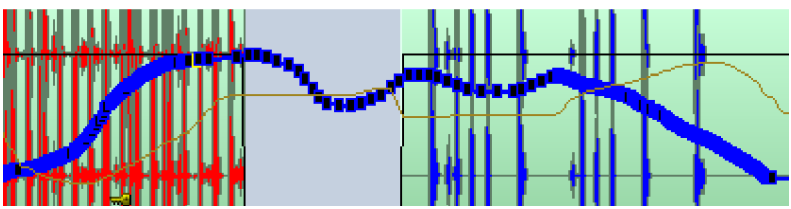
Now start playback. When you move the sound source with the mouse pointer during playback,



these movements will now be drawn



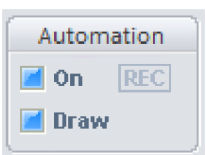
and displayed in the arranger as a curve.



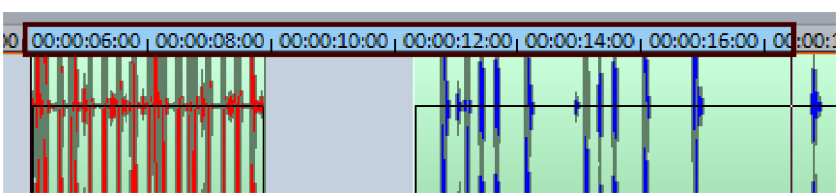
If the curve is not visible on the track, click on "Show" in the "Automation" section of the track editor to display it. You can edit the automation at any time by using the Automation Draw Mode in the arranger.

Drawing Surround Automation Curves

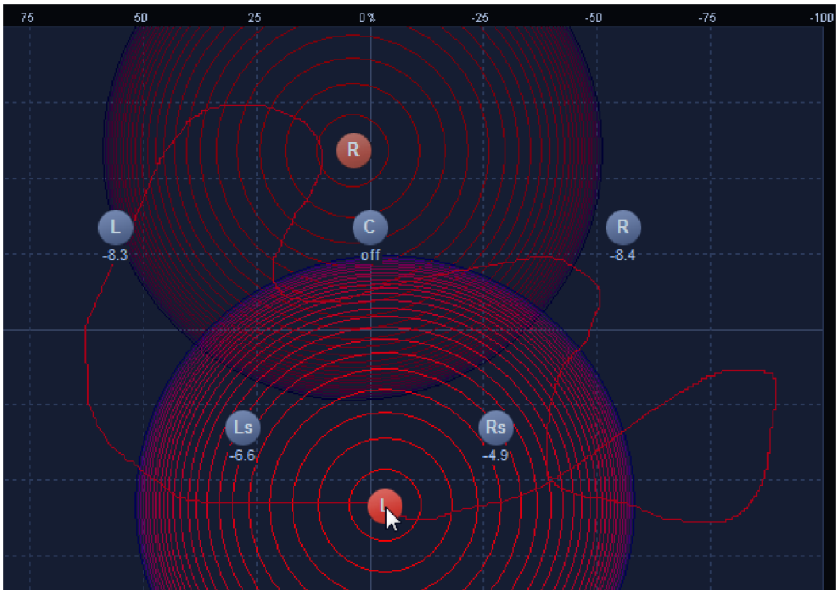
Activate the "Draw" option as well if you want to draw the automation directly in the Surround Editor.



First, in the arranger, select the track range where the automation will take place.



Now you can draw the movement progression of the sound source in the Surround Editor.



This progression will then be applied the next time the selected range is played back. Retroactive editing of the curve is possible here too with the help of the Automation Draw Mode in the arranger.

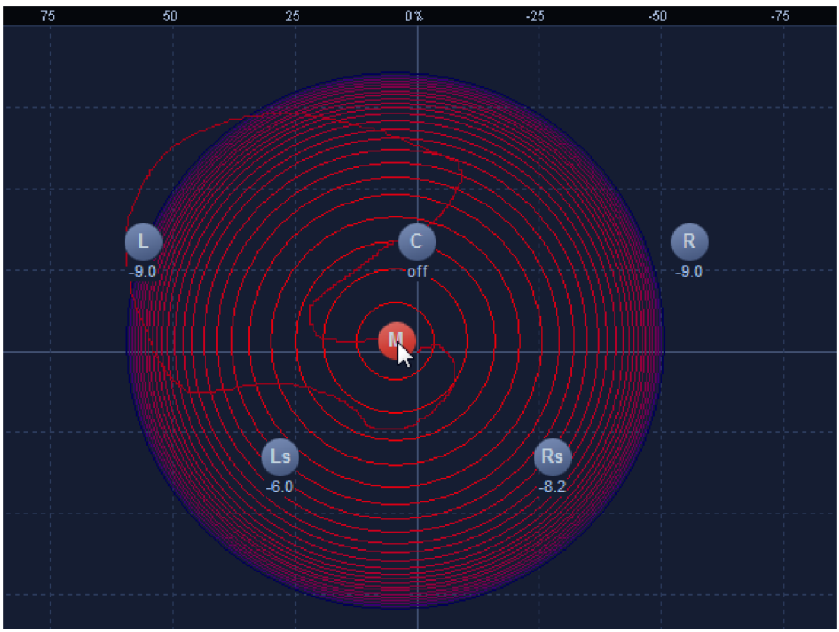
Object-based Automation in the Surround Editor

First open the Object Editor of the object to be automated. The setting "SurMaster" provides access to direct routing of the object to the Surround bus, i.e. to object-based Surround Panning.



Right-clicking in the Surround Panorama Field opens the corresponding Surround Editor.

Activate the "Draw" option as well if you want to draw the automation directly in the Surround EditorSurround Editorion of the sound source in the Surround Editor.



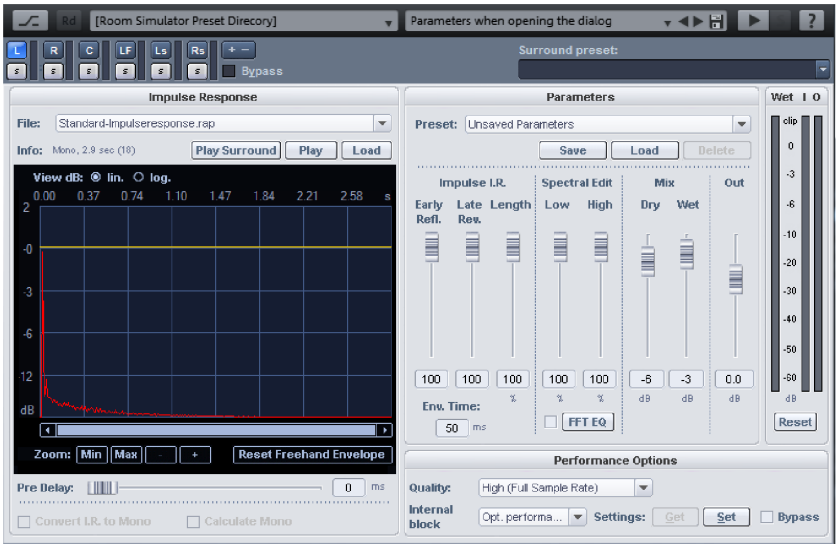
This progression will then be applied over the length of the object the next time it is played back. Retroactive editing of the curve is possible here too with the help of the Automation Draw Mode in the Object Editor.

Effects in Surround projects

Surround Effect Plug-ins

In Sequoia there are Surround effects available for all Surround buses which can be accessed through the "Plug-ins" section of the mixer. Here you can also add Surround-compatible VST effects from third parties. The surround channels can be grouped together to Surround Effect Groups.

- Dynamics
- Multiband Dynamics
- Advanced Dynamics
- sMax11
- EQ116
- FFT Filter
- Room Simulator
- DeHisser
- Vocoder
- Delay



In Surround processing, each effect can be applied on up to 12 channels.

The Surround effects sequence can be changed by

1. opening the Effect Routing Dialog with a right-click on the "Plug-ins" button in the mixer.
2. Here you can change the order of the effects.

In this case the plug-ins in the individual channels are always calculated before the Surround effects.

Note: This new functionality is not compatible with the characteristics of older program versions.

Surround Control Group

Surround control group The Surround channel buttons for each Surround mode are located in the Surround control group that appears with the Surround-capable effects above the effects dialog.



You can also switch the individual Surround channels to solo.

Grouping Mechanism for Surround Effects

Grouping and ungrouping:

To turn on Group Mode, click on the "+" button in the Surround Effect dialog. Now click on the buttons of the Surround buses that you want to group or ungroup. The corresponding Surround channels will now be marked with points of the same color.



If you click on "+" again to deactivate Group Mode, you can still open the created groups by clicking on the corresponding channels.

Adding Effect to Surround Buses: If you have grouped the Surround buses,



all the channels in the group that are open in the mixer will be calculated with an instance of the effect.



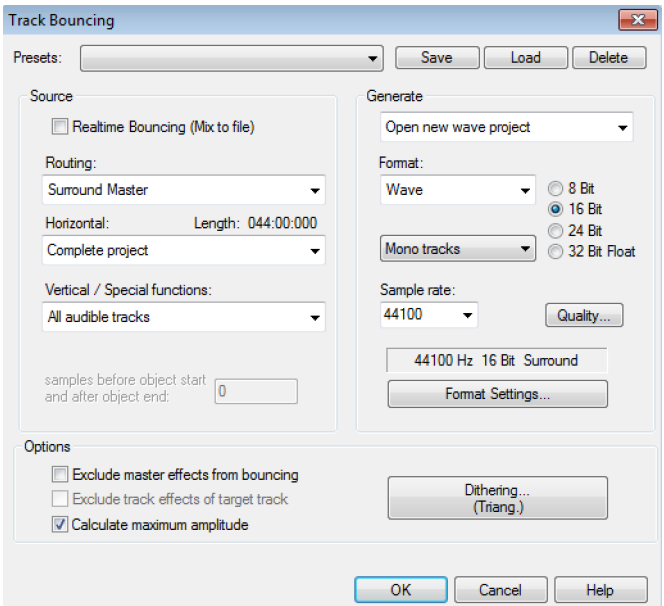
Effects of Grouping

- The parameters entered in the effect dialog are applied to all the channels in a group.
- For dynamic effects (Advanced and Multiband Dynamics), the control signals are compiled from all channels in the group, just as they are compiled from two stereo channels in the case of stereo.

Surround Bouncing

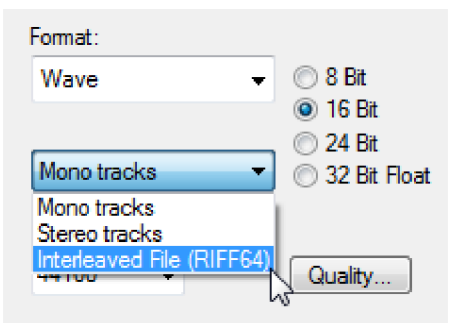
Surround bouncing enables you to compile a complex Surround mix. The outputs of the individual surround busses are used as a recording signal.

The "Track Bouncing" option can be accessed through the "File" menu.



Format: The following output formats are available for Surround bouncing:

- Wave files (Mono)
- Wave files (Stereo)
- Wave files (Interleaved RF64)



RIFF64 (also known as RF64) is a format for combining individual channels into a Surround sound production (e.g. for 5.1) as an interleaved file.

If you use the Stereo and Surround Master simultaneously in the mixer, you can select the setting "Surround + Stereo Master" in the "Routing" section of the track bouncing dialog to export RF64 files in the 5.1+2 format, for example.

The remaining settings can be selected during regular track bouncing (view page 643).

The abbreviations for individual Surround buses set in the "Surround Setup" (view page 477) will be added at the end of the selected names of files to be bounced.

Example: Select the name "5_1_Surround" when Surround bouncing a 5.1 Surround mix as 6 mono files. This will generate the following wave files: „5_1_Surround_L.wav“, „5_1_Surround_R.wav“, „5_1_Surround_Ls.wav“, „5_1_Surround_Rs.wav“, „5_1_Surround_C.wav“ „5_1_Surround_LFE.wav“. For stereo file output, the following files will be generated: „5_1_Surround_LR.wav“, „5_1_Surround_LsRs.wav“, „5_1_Surround_CLFE.wav“.

Synchronization

Synchronization is very important to an audio system, since it is necessary to align playback and recording speeds of different system components. Synchronization enables studio equipment such as tape machines, drum computers, video recorders, or sequencers to be linked with different formats like MIDI clock, MIDI time code, or SMPTE time code so that all of the devices run at precisely the same time.

Sequoia can be configured to as the master or slave within a synchronization system. As a master program, it generates the necessary timing information required for other system components; as slave. Sequoia receives the time code data to follow during playback or recording.

Clock on digital systems

In case digital signal flows are processed, e.g. via ADAT, SPDIF, or MADI, it is important to define a clock reference within the overall system. The clock signal is transferred via Blackburst, World Clock, or the digital input, and it may be received by the receiving or playback computer. External devices such as converters or mixers may also provide a clock reference. For these systems, there can be only one master but several slaves.

Note: Don't confuse (Word) clock with time code. The clock signal is only a digital pulse for comparison of the timing between the connected devices and to keep the bit rate constant. This enables transmission errors to be prevented. Professional digital audio devices normally feature a word clock input and are able to generate and receive a clock signal.

Time-related synchronization of projects

For more complicated configurations, it is usually necessary to connect various audio and video systems via absolute time references. This enables them to operate according to the behavior of the master, and the time code information (transport and progress) is implemented correspondingly.

Synchronization formats

The synchronization format features slave device information about the start position, start and stop signals and, in extended formats, the precise timing information, which are continuously transmitted.

Sequoia understands and transmits the synchronization formats MIDI clock, MIDI timecode (MTC), and SMPTE (Society of Motion Picture and Television Engineers).

MIDI Clock

The MIDI clock acts as a MIDI transmission clock consisting of system messages (F8H), and its time reference is not based on hours, minutes, and seconds, but rather on "ticks". One "tick" is the 24th part of a quarter note (24 ppqn [pulses per quarter note]) – MIDI clock signals are therefore transmitted 24 times per quarter note. Since the MIDI clock is based on the tempo as a reference, the number of ticks output depends on the tempo of the generating device. The MIDI clock also does not transport any information concerning where in the song the listener is located at any particular time.

Depending on the options the external MIDI devices provide for synchronization, start (FAH), stop (FCH), and continue (FBH) signals are also transmitted. These parameters ensure that the connected devices start via MIDI, stop at any point in the song, and that they may be restarted at the stop position.

Depending on the device specification, the MIDI clock signal may also transmit the song position pointer (SPP). This acts as a beat counter that provides the number of MIDI beats that have been played back since the start. SPP is also sent all 6 ticks as system exclusive data, which corresponds with a 1/16 note rhythm.

The song position pointer makes it possible to link the connected devices with a precision of 1/16 notes at any position in the song.

Note: Look at the specifications of your external MIDI devices to learn which synchronization parameters are supported. Sequoia is able to process MIDI clock (F8H), start (FAH), stop (FCH), continue (FBH), and song position pointer (SPP).

MIDI clock synchronization is especially useful for synchronizing devices with ticks as a time reference with Sequoia, e.g. external samplers or drum computer or if you want to link LFOs or arpeggiators of multiple external synthesizers.

SMPTE Timecode

SMPTE Timecode is a conventional synchronization standard for linking different audio and imaging devices via absolute time reference. Every SMPTE message sends exactly how much time has passed since the beginning of the projects.

The SMPTE Timecode refers to the units hours:minutes:seconds:frames. The concept of "frames" originates from the film and video industry, which the SMPTE code was originally developed for. The different frame rates specify the resolution in the individual images per second, which is common in the film and video industry.

MIDI Time Code (MTC)

MIDI Time Code (MTC) is an eight-digit format like SMPTE timecode that integrates an absolute timecode (hours, minutes, seconds, and frames) as a synchronization reference. MTC is transferred with other MIDI data via the MIDI interface and transmitted as system messages or as universal system exclusive messages.

MIDI Time Code is especially useful for synchronization of multiple sequencers or DAWs.

Note: To synchronize devices that output SMPTE Timecode with MTC, the SMPTE Code is converted into MIDI Time Code first via an SMPTE to MTC Converter.

MIDI Machine Control (MMC)

This is a signal for controlling external controllers that support the MMC format. Sequoia may also be remotely controlled via MMC commands from an external controller or mixer.

You can find out more about this is available under MIDI Machine Control (MMC) (view page 512).

APP (ASIO Positioning Protocol)

The ASIO Positioning Protocol enables transmission of sample-exact time information from a third device that is connected via a digital interface with the sound card. As a component of the ASIO driver architecture, digital signal connections (SPDIF, ADAT) or special sound card interfaces (LTC, Video Burst In) may also be used to transmit the ASIO Positioning Protocol. This enables synchronization with SMPTE or MTC-equivalent timecode, without requiring an additional MIDI connection. Sequoia is therefore able to follow the input signal of an external source exactly.

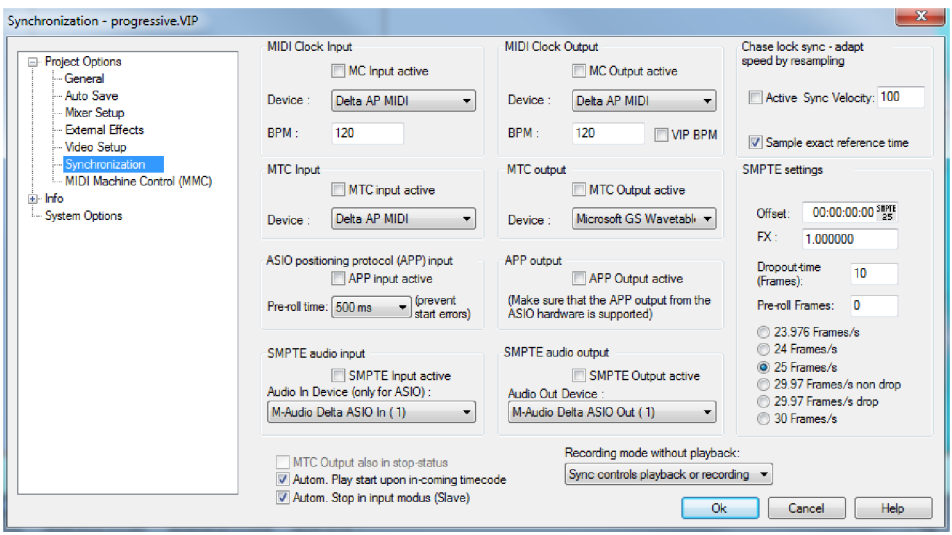
Note: Inform yourself regarding about whether your audio device and the ASIO driver support the APP protocol.

Synchronization Setup Dialog

The settings options for synchronization are available in the synchronization setup dialog.

Menu: „File“ > „Program Preferences“ > „Synchronization Setup...“

Keyboard shortcut: Shift + G



This window is for customizing the synchronization according to your requirements. The available formats are MIDI clock, MTC, APP, and SMPTE.

MIDI Clock

MIDI Clock Input

MC input active: Activate the input for the incoming MIDI clock signal by clicking this check box.

Device: Select the device that should create the MIDI clock signal for synchronization.

BPM: Define the tempo that the MIDI clock should receive.

MIDI Clock Output

MC output active: This check box activates the output for the MIDI clock signal generated by Sequoia.

Device: Select the device that should receive the MIDI clock signal for synchronization.

BPM: Define the tempo that the MIDI clock should receive.

VIP BPM: This check box defines the project speed set in Sequoia for the synchronization project.

MTC (MIDI Timecode)

MTC Input

MTC input active: This check box activates the input for the incoming MIDI Timecode signal.

Device: Select the device that should create the MIDI timecode signal for synchronization.

MTC output

MTC output active: This check box activates the output for the MIDI timecode signal generated by Sequoia.

Device: Select the device that the MIDI timecode signal should be sent to for synchronization.

ASIO positioning protocol (APP) input

APP Input

APP input active: This check box activates the input for the incoming APP input signal.

Preroll time: In order to hinder problems during starting, you can set the preroll time at up to 1500 milliseconds.

APP output

APP output active: This check box activates the output for the APP signal generated by Sequoia. Please make sure that the APP output from the ASIO hardware is supported.

SMPTE Audio

SMPTE audio input

SMPTE input active: This check box activates the input for the incoming SMPTE input signal.

Audio in device: Select the audio input of the device via which the external SMPTE timecode signal is sent for synchronization. This function is only available for ASIO.

SMPTE audio output

SMPTE output active: Activate the output for the sent SMPTE signal by clicking this check box.

Audio out device: Select the sound card's audio output device via which the SMPTE timecode signal should be sent to external devices for synchronization.

Information concerning the generation of SMPTE timecodes within Sequoia is presented in the menu item "Effects -> SMPTE generator (view page 1088)".

Automatic Stop in Input Mode (Slave)

If Sequoia is receiving MTC from an external device and the corresponding track is ready to record, Sequoia will begin recording as soon as the external signal is recorded. If sending of external synchronization signals stops, Sequoia will end recording and enter into playback mode automatically. If the timecode is received again, Sequoia will begin playback. If you want Sequoia to record again when receiving MTC, activate this option.

Automatic Playback with Incoming Timecode

If Sequoia is receiving MTC from an external device and the corresponding track is ready to record, Sequoia will begin recording as soon as the external signal is received.

MTC Output also in stop-status

If you select this option, Sequoia continuously sends the current time position as MTC – even in case playback has stopped.

Chase lock sync - adapt speed by resampling

Sequoia supports real chase lock synchronization, i. e. audio playback can be controlled by an incoming timecode signal. In this case not only the starting point of audio playback is controlled externally, but sync also controls playback speed if "Chase Lock" is activated. Sequoia is therefore capable of following analog tape devices or VCRs that always feature some slip over longer periods.

If fluctuations in speed occur, Sequoia can make adjustments in slave mode that ensure exact temporal synchronization. This function is called "chase lock". Use chase lock in case a device involved in the synchronization cannot be clocked centrally via

Blackburst, Wordclock, or the digital input and if Sequoia is the "slave". This is the case if the timecode is located on a track featuring a multitrack cassette player.

Active: Activate chase lock synchronization by clicking this check box.

Sync velocity: Entering the sync velocity enables you to influence the speed of the tempo adjustment. The greater the sync velocity value, the faster Sequoia will follow master device tempo changes; however, this also means that the pitch fluctuation in the audio material is greater. The preset value for the sync velocity is 100; experiment with values larger than 100 in case the synchronization does not run exactly.

Note: Please note that Sequoia will apply resampling to the recording in real time according to the timecode fluctuations in case chase lock is active. This causes increased CPU load and may result in unwanted changes to the audio material, if this is played later with other timecode references.

If the systems are coupled via a central digital audio clock, then chase lock should not be activated.

Use sample-accurate position: If this option is active, Sequoia uses the time position of the sound card as the clock reference (timer) and not its own internal timer. This ensures sample-exact synchronization between the recorded audio material and the sync signal.

SMPTE settings

SMPTE-Offset: Specify an offset to be deducted from the incoming SMPTE time before the time is used for synchronization. An offset of 01:00:00:00 (1 hour) therefore allows you to synchronize a tape with an SMPTE code that starts at 1 hour. The SMPTE offset behaves relative to the project start time (view page 607).

FX: This factor allows you to equalize possible irregularities with regard to positioning in long samples. Perfect synchronization at the beginning of the sample is nevertheless a prerequisite.

Drop out time (frames): Specify after how many sequentially passing frames synchronization should quit.

Preroll frames: Enter the number of frames that Sequoia should ignore before synchronization begins. This takes into account the fact that analog devices normally require a certain amount of time before they reach the correct speed. Sequoia will now synchronize at a valid time, since this startup time is able to be skipped according to the set preroll frames.

Type: Select a suitable frame rate, e. g. 24 for video/movies, 25 for PAL video and audio synchronization, 29.97 for NTSC video, and 30 for HDTV.

Synchronization - Sequoia as master

Sequoia supports MIDI clock, MIDI timecode, and APP timecode output as a master.

Sync output is directly linked to audio playback so that no delays occur between Sequoia and external devices, even with long tracks.

Note regarding linking Sequoia as the master:

- In this case, use a virtual MIDI router for linking the programs internally.
- Wherever possible, favor the MIDI timecode synchronization with MIDI clock, since this does not need to include tempo changes.
- If Sequoia is running as the master, then set the FX factor to 1.0.
- For optimal synchronization stability, deactivate the virtual memory (if possible).

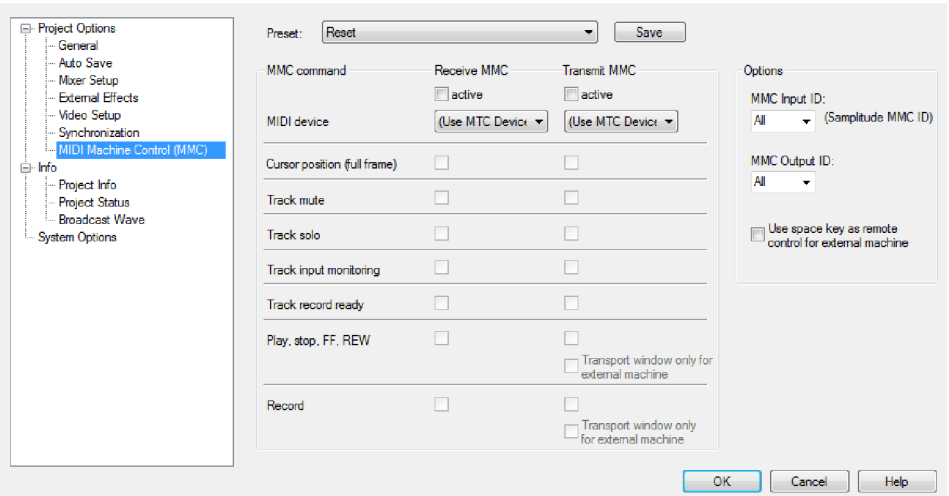
MIDI Machine Control (MMC)

MMC modes

In the project options, you will find a dialog for remote control via MIDI machine control. Sequoia supports synchronization of external devices via MMC. The following modes are available:

Receive MMC: If you have set up Sequoia to work as a slave, activate this option. Sequoia follows JOG shuttle operations, cursor position, track mute, track solo, track input monitoring, track record ready, play, stop, fast forward, rewind, and record commands from external devices.

Send MMC: Sequoia operates as master. When you use commands like cursor position, track mute, track solo, track input monitoring, track record ready, play, stop, fast forward, rewind, and record, the external device follows. If you also activate the "Receive MMC commands" option, Sequoia receives the current playback position of the device and displays it as an additional blue cursor on the timeline.



While playing a range in Sequoia, the MMC device stops once the end of the range is reached.

Transport window only controls external device

If you activate this option, the transport console no longer controls playback and recording directly within the program but instead controls the external MMC device. Play, stop, fast-forward, fast-backward and rewind no longer have a direct effect on Sequoia playback. The "Record" button starts recording in Sequoia in synchronization with the playback of the MMC device.

Use spacebar to control external devices: If this option is activated, the MMC device's start and stop commands are controlled with the spacebar.

Additional options

Input MIDI device: Set the MMC device which Sequoia receives.

Output MIDI device: Set the MMC device which Sequoia sends to.

MMC input ID: Enter the Sequoia MMC input ID here.

MMC output ID: Set the input ID of the MMC device you want to control remotely.

List of implemented MMC commands

Command	Receive	Send
Full frame	Yes	Yes
Stop	01	01
Play	02	02
Deferred play	03	-
FFWD	04	-
Rewind	05	-
Record ON	06	06
Record OFF	07	07
Record Pause	08	-
Pause	09	-
Locate	44	44
Variable Play	45	45
Shuttle	47	-
READ location	42 01	-
READ motion control tally	42 48	-
READ record status	42 4d	-
READ track record status	42 4e	-
Track Record Ready	42 4f	40 4f
Track Input Monitoring	42 53	40 53
Track Mute	42 62	40 62
Track Solo	42 77	40 77

MAGIX MMC extension

F0 7F <device_id> 06 6F <...> F7

Set Marker	6F 01	New auto numbered marker
Navigate Playback Marker to marker left/right	6F 02 n	n = 00 to left marker n = 01 to right marker
Set all tracks RecRdy on/off	6F 03 n	n = 00 off n = 01 on
Set Monitoring Mode	6F 04 n	n = 00 manual monitoring n = 01 tape monitoring
Set Input Monitoring on/off	6F 05 n	n = 00 off n = 01 on
Set Sync Mode on/off	6F 06 n	n = 00 off n = 01 on
Set Loop Mode on/off	6F 07 n	n = 00 off n = 01 on
Set Punch Mode on/off	6F 08 n	n = 00 off n = 01 on
Delete Punch Marker	6F 09	

Synchronization applications

Application 1. Synchronization using an external hardware sequencer. Sequoia is the master.

To synchronize Sequoia with an external hardware sequencer (drum computer, groove box, etc.), first select the type of synchronization. As these external devices usually use the song's tempo or rhythm-dependent information, they are often synchronized using MIDI clock. In this case, impulses are passed on from the master to the slave. These control impulses are transmitted for each quarter note. Their number per quarter is always identical. The tempo information is always clear, since the impulses do not occur in absolute time units but are rhythm-dependent. This secures the tempo synchronization of the two units that are to be aligned. Of course the same tempo (more or less) should be set to both units from the beginning. The MIDI song position pointer is another important MIDI clock synchronization tool which provides information on how may control impulses have already been transmitted. You can therefore start at any position of the sequencer. The same position of both synchronization units is thereby guaranteed.

Important: The value of the MIDI song position pointer is limited to 1024 bars (4/4 beat). After this, no further synchronization via MIDI clock is possible; connected slave devices simply stop. In this case, connected slave devices remain still.

1. Make sure that your external hardware sequencer supports synchronization via MIDI clock as slave.
2. Connect MIDI OUT of the master (MIDI OUT on the MIDI interface of the computer running Sequoia) to the MIDI IN of the external hardware sequencer.
3. Set the hardware sequencer to "Slave" mode. If you apply this setting, refer to the instruction manual for the respective device. You will now notice that the transport control and tempo setting of the hardware sequencer are deactivated. It will now receive data from the master, i.e. from Sequoia.
4. Open the synchronization window in Sequoia and activate the "MIDI clock output -> MC output active" option.
5. Select the MIDI port that connected to the slave device. Select the tempo you would like the external hardware sequencer to run at. This does not necessarily have to be the tempo of your Sequoia VIP project (VIP BPM), but in most cases it is sensible, since the bars in the Sequoia arranger window will correspond with the bars of the hardware sequencer.
6. If you press play in Sequoia, the external device runs in sync with the tempo of your VIP project.

Note: Please also observe that the "Transport window only controls external devices" option must be deactivated if the device that is to be controlled cannot process MIDI machine control data (MMC), since the transport keys neither control Sequoia nor the external sequencer.

Application 2: MIDI Clock synchronization using an external device. Sequoia is the slave.

In principle, it makes sense to choose a master within a group of devices with the most stable timing. In most cases, this would be Sequoia. Of course you can also set Sequoia as slave, for example, if the main parts of a production are prepared using an external hardware MIDI sequencer and Sequoia is only used as an additional audio device, or if the sequencer only permits MIDI Clock synchronization as a master.

The following should be observed: unlike a MIDI-only device, Sequoia can contain audio files which might need adjusting. This is only possible with the "Chase lock sync (time alignment by resampling) -> Active" function. If the timing of the master is inaccurate, audible pitch fluctuations may occur.

Note: This function should only be used if a common word clock is not available If a common word clock is available, synchronization variations will be created.

1. To synchronize via MIDI clock as slave, connect the MIDI OUT of the external device to the MIDI IN of the MIDI port selected in the synchronization setup.
2. Now select "MC Input active" in "Synchronization/MIDI clock input" and enter the tempo of the external sequencer. Sequoia is now operating as a slave. Unlike many other hardware sequencers, Sequoia does not deactivate the transport keys but “listens” to the incoming master commands.
3. Make sure that the external device is switched to "master synchronization" (many devices also call this "Internal sync mode" with "Sync Out" activated).
4. If you start the external device now, the VIP project in Sequoia starts, too.

If "Chase lock sync" is active, the fluctuations Sequoia has to compensate are displayed in the bottom right arranger window.

If your external sequencer is set to a different tempo than from Sequoia, this will attempt to adjust to the incoming tempo by changing the playback tempo relative to the initial BPM after starting the sequencer. However, this can take up to several bars and results in clearly audible pitch fluctuations. Make sure from the start that both synchronization units are set to the same tempo.

Application 3: MIDI time code (MTC) synchronization using a second software sequencer. Sequoia is master .

Sequoia can be synchronized with another software sequencer or hard disk recording system. MIDI time code synchronization is particularly useful for this.

The advantage of the synchronization method is that it is independent of a song's BPM entries. Rhythm and beat-dependent changes of tempo are therefore no longer a problem within a project, since communication now takes place via time values. Division into hours is possible according to the SMPTE code: minutes: seconds: frames (format: 00:00:00:00). Frames are the smallest SMPTE units. These originate from movie technology and state the number of images per second (e.g. 24 frames for film, PAL: 25 frames for video).

Note: If Sequoia should be synchronized with audio data and not film, then the frame rate can be set however you like. It's important that both of the systems to be synchronized feature the same sample rate.

Preparing for synchronization:

The second software sequencer can be running on the same or on a different machine as from Sequoia. If two sequencers are running parallel on one computer, make sure that the audio drivers being used permit simultaneous use of multiple sound card outputs. Alternatively, you can also use two different audio interfaces.

During synchronization via the MIDI interface, note that a MIDI port that is opened by the software cannot be used by other applications. To overcome this limitation, two

MIDI interfaces that have been connected may be used. It's more elegant to use a freeware help program like "Hubis Loopback", "Marblesound Maple", or "MIDI Yoke". These programs install virtual, open MIDI ports on a PC that may be "connected" freely with the corresponding applications.

MIDI time code (MTC) synchronization using a second software sequencer. Sequoia is master:

1. Start both programs first.
2. Now connect the MIDI output of the MIDI port used by Sequoia with the MIDI input of the system to be synchronized, or patch this virtually using an auxiliary program.
3. Select the corresponding MIDI port in Sequoia's "MTC output" window and activate the MTC output.
4. On the slave system select the corresponding “slave” or “receive MTC” option. This setting is independent of the second program being used. Now, activate the used MIDI in-Port.
5. Ensure that the same frame rate is set for both systems.
6. Now start playback in Sequoia. The slave system will follow and be in sync with the master.

Important: Always make sure that the sync start position of the slave system is identical with the start position of Sequoia. If SMPTE sync start cannot be set for the other sequencer, you can also enter the starting time of the other system in Sequoia in "Synchronization -> SMPTE settings -> Offset".

Application 4: MIDI time code (MTC) synchronization using a second software sequencer. Sequoia is slave.

This method is generally the same as in point 3. However, the master/slave setting has to be reversed, i.e. MIDI OUT from the other sequencer to Sequoia's MIDI IN.

Backup Recording with Two Programs

In order to make sure that everything is recorded properly during important studio sessions, using parallel systems is strongly recommended. In the past the parallel system was normally a hardware device that recorded the stereo master or the complete multitrack setup. These days this type of backup can be recorded simply by using a second DAW.

Requirements for Backup Recording

- Two DAW systems e. g. Sequoia/Sequoia, Sequoia/Samplitude Pro X5 or Samplitude Pro X5/Samplitude Pro X5
- MIDI MMC Connection

You can let the backup system run uncontrolled. In this case you don't need an MMC connection. This technique runs the backup system the whole time and does not generate objects (takes). A timecode synchronization of the systems using MTC/SMPTE is not necessary as long as the digital recording and the word clock is identical in both systems.

Creating the Backup Recording

The MMC settings include the presets "Record Backup (Master)" and "Record Backup (Slave)". Set suitable presets on the master and slave systems.

Now set up an identical project for the master and slave, activate the necessary tracks and deactivate the display of the recording dialog at the end of the recording ("System Options" > "Record" > "Show confirmation dialog (OK, Delete) after record"). In the slave system, the option "Set cursor to record end position - next recording starts at this position" should also be activated. This will add each new take to the end of the project even when an insert punch is applied in the master.

Parellel Recording

For parallel recording set the MMC presets "Parallel Systems (Master)" and "Parallel Systems (Slave)".

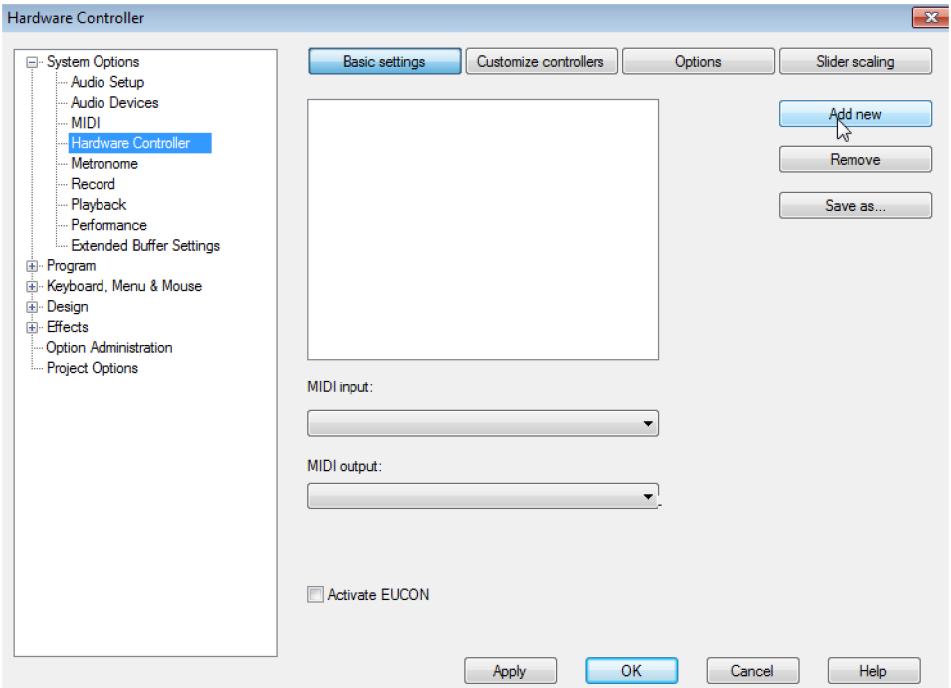
Now set up an identical project for the master and slave, activate the necessary tracks and deactivate the display of the recording dialog at the end of the recording ("System Options" > "Record" > "Show confirmation dialog (OK, Delete) after record").

In parallel recording the following information can be exchanged between the programs in addition to the transport functions: cursor position, mute, solo, input monitoring and record ready. You can adjust the settings to meet your own requirements.

In this way an almost identical project is created in the slave system whereby insert punch recordings are also added at the matching position. Only take names and markers are not taken into account.

Now if one of the systems fails during the session, you still have a complete project including the take structure and can continue working without having to add the takes again.

Hardware controller



Basic Settings

The following tasks are performed in the basic settings:

- Adding a hardware controller using a factory-preset (view page 549)
- Specifying MIDI ports for the hardware controller
- Determine the order and dependencies of the hardware controller:
  Hardware controllers can be operated independently or together as a unit.
- Adding a new hardware controller preset

Using EUCON Controllers

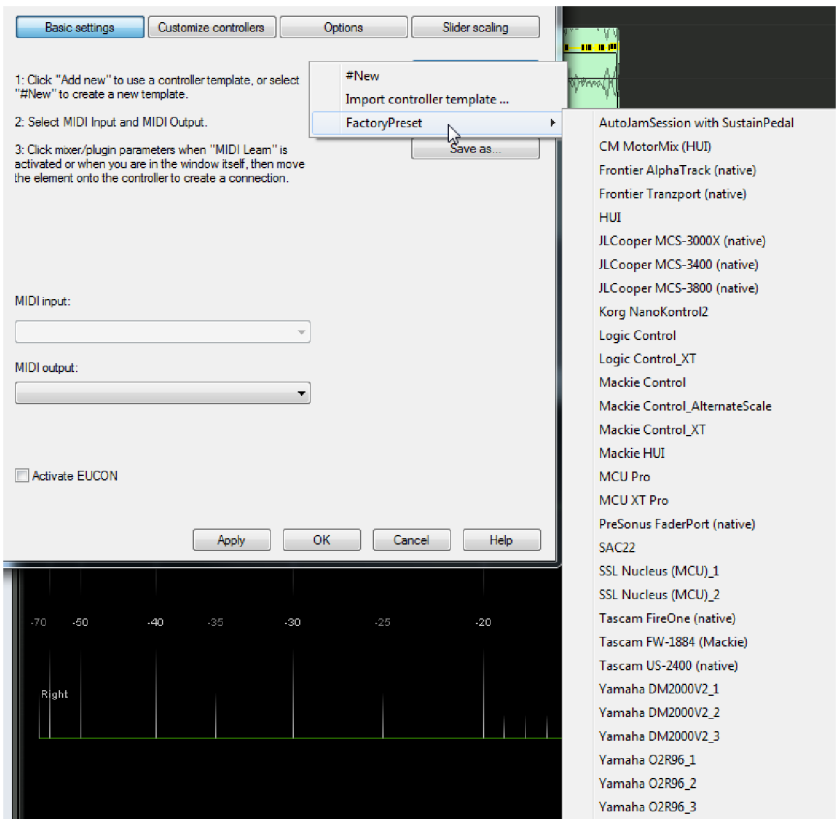
EUCON controllers are integrated through a network connection and contain an independent configuration environment. They are configured exclusively with the software included with the EUCON controller.

Activate EUCON: Activate this check box if you are working with an EUCON controller. The connection can be updated at any time using the corresponding button.

Adding a Hardware Controller using a Controller Preset

Tip: Many hardware controllers have a Mackie emulation mode. In this mode, most functions that are available in the Mackie Control are supported. If you are using this type of Mackie control emulation, simply select the preset "Mackie Control".

1. Click the "Add New" button in the controller settings.
   This opens the preset menu:



2. In the „Factory Preset“ list (view page 549) select your hardware controller or a compatible device.
   Result: The selected device appears in the list of hardware controllers.

Note: when you load a "Factory Preset", Sequoia saves this factory preset in a separate folder. Changes to this preset will then only be saved in this separate folder. This ensures that the factory presets are not changed and can be accessed again at any time.

Factory presets that have been loaded once appear additionally in the first level of the "Add New" menu. It is possible that adjustments have already been made to these presets.

3. Select the MIDI In and MIDI Out ports that your controller is connected to.
4. Click on the "Apply" button to confirm the changes.
5. If you want to edit the controller preset, open the "Customize Controller" view and edit the control elements as necessary (view page 534).
6. If you are using several controllers, repeat this procedure until all of the controllers have been edited.

Using Multiple Controllers, Extenders and Multibank Controllers

Note: When using several hardware controllers of the same type, add a separate preset to each controller. This ensures that the adjustments only have an effect on one of the controllers.

A maximum of four controllers can be positioned in the controller selection list. For example, if you select a Mackie Control and want to add an extension to it, add a "Mackie Control XT". This will be added automatically directly below the first one. In this case, the second controller is treated as the extension. Specify the correct MIDI ports for this additional controller.

1. Set up the controller as described above.
2. Use drag & drop to adjust the order of the controllers until they are actually in the right order.
   - Multiple controllers used in parallel are displayed on one level.
   - Expansion modules are displayed indented.
3. Activate or deactivate individual hardware controllers with a checkmark in front of the controller-entry. This can be useful for the following purposes:
   - Using Different Controllers for Various Applications / Environments
   - Functional Test of an Extender or Additional Controllers during Configuration
4. Click on the "Apply" button to confirm the changes.

Adding a New Empty Hardware Controller Preset

1. Select "#New" to create a new template.
2. Select the MIDI In and MIDI Out ports that your controller is connected to.
3. Click on the "Apply" button to confirm the changes.
4. Select the option "Save As..." and give the file a clear name.
5. Switch to the „Customize Controller“ (view page 523) view and map the control elements for your controller.

Removing a Controller from the List

Prerequisite: The list of used hardware controllers contains at least one entry.

Note: When a hardware controller is removed, the preset configuration remains. If you load the corresponding preset later, all of the settings for the hardware controller will be restored.

1. Simply click on the hardware controller that you want to remove.
2. Now click on the "Remove" button.
   Result: The selected hardware controller is removed from the list and is no longer used by Sequoia.

Customizing Controllers

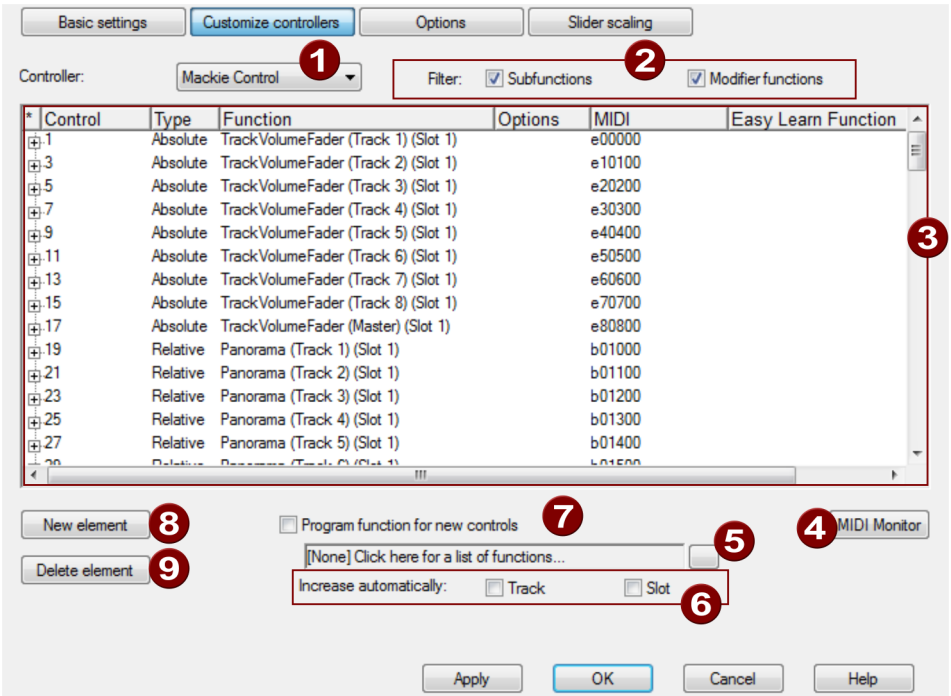
In the "Customize Controller" view you can make the following adjustments:

- Adjust already learned control elements
- Learn new control elements
- Apply or remove Easy Learn functions
- Remove already learned control elements

Note: when you load a "Factory Preset", Sequoia saves this factory preset in a separate folder. Changes to this preset will then only be saved in this separate folder. This ensures that the factory presets are not changed and can be accessed again at any time.

Factory presets that have been loaded once appear additionally in the first level of the "Add New" menu. It is possible that adjustments have already been made to these presets.

Controller Customization - Setup and Functioning



1 Controller

This is where you select the hardware controller that will be edited. When the "Customize Controller" view is opened, the controller that was last set in this view will be selected by default. Clicking on the dropdown menu opens a list of all currently available controllers.

2 Filters

Here you can specify whether sub-functions and modifier functions (view page 529) will also be displayed. Both of these options are activated by default.

Sub-functions: These are functions that belong to the control element but are accessed with a separate MIDI command.

- Push Function for Encoders
- Touch Function for Touch-Sensitive Faders

Modifier functions: Additional assignments to control elements. The modifier functions can be accessed by simultaneously pressing the assigned modification button and the corresponding control element key.

3 **List of Mapped Control Elements:** This list contains all of the control elements that have already been mapped.
The list is made up of several columns where the entries can be sorted by clicking on the column headers:

***:** If the control element has sub-functions or modifier functions, they will be displayed in this column.

> **Note:** If the list is sorted according to this column (*), the assigned or new control element will be displayed at top of the list when a MIDI command is received.

Element: Automatically numbered list of the control elements. Sub-functions and modifier functions are displayed here in color and with a corresponding short description:

Sub-functions: Touch/Push - Displayed in green

Modifier functions: Shift / Control / Option (also in combination) – Displayed in red.

Type: Displays the control element type (view page 526) that is assigned to an entry.

Function: Displays the function that is assigned to the control element.
Right-clicking opens the "Edit Function" dialog. The program functions that should be assigned to control element are set in this dialog.

Options: Displays whether various options for this control element are active.
Right-clicking opens the context menu where the available options can be selected.

MIDI: If the function is already assigned to a control element, the MIDI message that opens the assigned function is displayed here.
Right-clicking opens the "Select MIDI Data" dialog.

Easy Learn Function: If a control element is assigned an additional function with Easy Learn, the function will be displayed here.
Right-clicking opens the context menu where the function can be permanently assigned or deleted.

4 **MIDI Monitor** (view page 530)
This button opens the MIDI-input monitor.

When the Shift key is pressed during opening, the HWC MIDI-output monitor is also opened.

5 **Dropdown Menu Function**
The program functions that should automatically be assigned to the next newly added control element are set in this dialog. The "Program Function for New Elements" checkbox must be activated for this.

6 **Raise Automatically (Track/Slot)**
The track or slot assignment of new control elements to be mapped is automatically raised when the previous control element is mapped.

This option is suitable for mapping control elements with which the same function should be controlled on different tracks/slots.

7 **Program Function for New Elements**
When this checkbox is activated, the next new added element will be assigned to the set underlying function.

8 **New Element**
This button sets a new entry in the list. The entry appears at the very bottom of the list.

9 **Delete Element**
This removes the selected control element from the list.

Types of Control Element

Value Input

- **Absolute:** Control elements with absolute values.
  This type should be selected when using a fader or knob.
- **Relative:** Control elements with relative values.
  This type should be selected when using an endless encoder, the jog/shuttle or a similar control element.

Note: Control elements for the values are already assigned tomodifier functions (view page 529).

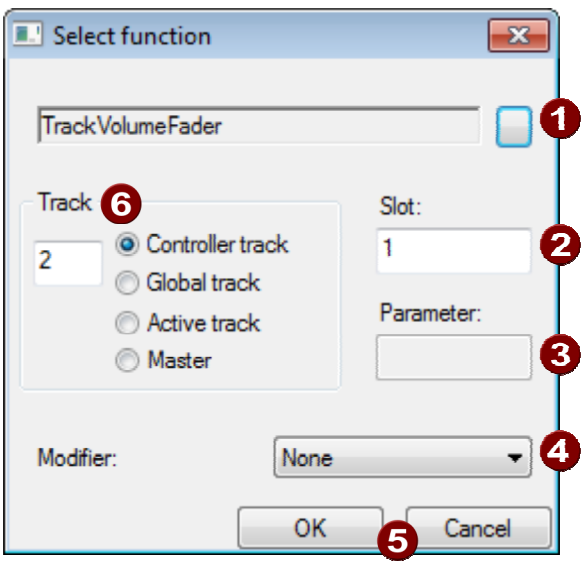
Keys

- **Button:** Simple keys
  Select this type of key when the keys on the hardware controller send a Note On command when pressed and a Note Off command when released.
- **Push Button:** Keys without a MIDI-command when released
  Select this type of key when the keys on the hardware controller ONLY send a Note On command when pressed.
- **State Button:** Key with a rest function
  Select this type of key when the keys on the hardware controller send a Note On command when pressed and a Note Off command when pressed a second time.

Other

- **LED:** Display function on the hardware controller
  LEDs or status lights are often integrated into other control elements or assigned to them. For this reason the "LED" status is most often changed using the same MIDI command that sends the corresponding superordinate control element to Sequoia.
  If a status display is not assigned to any control element, or is changed with a different MIDI command, please refer to the hardware controller manual to find out the correct MIDI command for the LED.

Select Dialog Function



1. **Function selection:** This is where the function assigned to the control element is displayed. The function that should be assigned to the control element is selected in the dropdown menu.

2. For certain functions there are several slots per track, e.g. for AUX sends, the equalizer (one slot per EQ band respectively) and the insert effects. **Slot:** Determines the number of the slot that the assigned control element affects. For functions that are only present once per track, this parameter does not cause any change.

3 **Parameter:** Additional dimensions for the selection of specific track parameters. If an effect has several parameters, the actual parameters are numerically ordered in this field. For example, the following scheme would be used for the track equalizer:

- **Slot:** Selection of frequency band (= 4 Slots)
- **Parameter:** Selection of parameter in the respective frequency band (Gain/Frequency/Q)

4 **Modifier:** The control element is only active when the set modifier key or the set combination is pressed.

The function opened by the modifier is called the modifier function (view page 529).

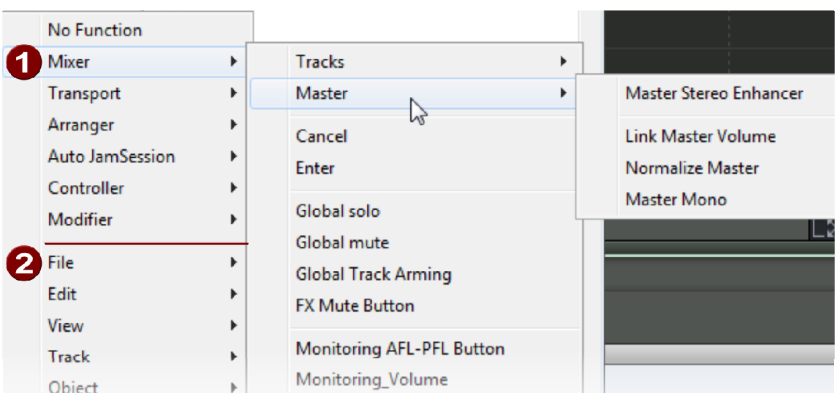
5 **OK:** The dialog is closed and the changes are applied.

Close: The dialog is closed without applying the changes.

6 **Track:** Track-based functions are assigned to the track through this parameter.

- **Controller Track:** The track allocation of the assigned function is controlled by the setting „Synchronize with“ (view page 540) in the „Options“.
- **Global Track:** The assigned function always affects the entered track.
- **Active Track:** The assigned function always affects the active track.
- **Master:** The assigned function always affects the master track.

Function Selection



1 The upper area contains the functions that are typically controlled with hardware controllers.

2 The lower area displays the complete Sequoia menu structure. You can use this to assign each menu command to a controller key.

Note: Please note that when working with menu commands, the key LEDs of the controllers remain inactive.

Modifier Functions

Modifier functions are multiple allocations of control elements that are only active while the modifier key is being pressed. These hardware controller modifier keys first have to be mapped (view page 537) in order to function. Any key on the hardware controller can be used as a modifier key.

The following modifier keys are available.

- Shift (alternatively locked)
- Option
- Control

Note: „Shift Lock“ has the same effect as „Shift“ but the corresponding key remains "snapped" when the key is released. Pressing the "Shift" key again turns the lock off. "Shift Lock" is particularly useful when combined with other modifier keys or when the status of the Shift key is displayed by the hardware controller.

As soon as one or more modifier keys is pressed, the respective additional allocation of the correspondingly mapped control element becomes active. After releasing the modifier key, the standard function becomes active again.

Combinations of Modifier Keys

The modifier keys can also be used in combination which increases the scope of remotely controlled commands even further.

The following combinations are available:

- Shift (Lock) + Option
- Option + Control
- Shift (Lock) + Control
- Shift (Lock) + Option + Control

Preset Modifier Functions

For Control Elements for Value Entry (view page 526) there are preset modifier functions:

- Shift (Lock) + FaderTouch (without moving the fader): The allocated parameter is reset to the default value (e.g. track volume: 0.0 dB).
- Shift (Lock) + Moving endless knob: A finer value gradation during a change to the value.

MIDI Monitor

The MIDI monitor shows either MIDI input data or MIDI ouput data.

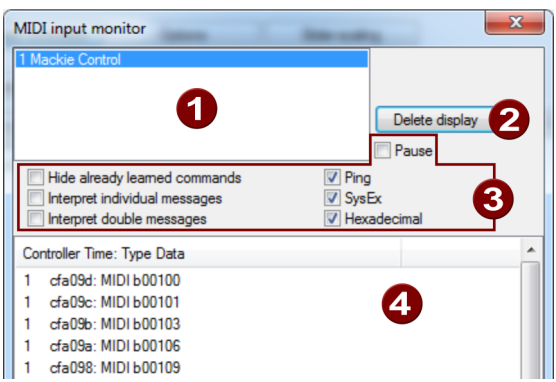
- MIDI Input Monitor: Shows all MIDI data being sent to Sequoia from the hardware controller.
  This dialog is modal, i. e. no other dialogs can be used when the MIDI monitor is open.
- HWC MIDI output monitor shows all MIDI data being sent from Sequoia to the selected the hardware controller.
  This dialog is not modal. This means that Sequoia is fully operational while the MIDI monitor is open.

The MIDI data that is displayed depends on the settings.

Open

- **MIDI Input Monitor:** Activate the „MIDI Monitor“ button.
- **HWC MIDI Output Monitor:** Activate the „MIDI Monitor“ button while holding down the Shift key.

Setup and Functioning



1 **Selection of Hardware Controller for MIDI Monitoring:** Clicking on the entries activates or deactivates the hardware controller for MIDI monitoring.

2 **Delete Display:** This deletes the values in the display field (4).

3 Options
- **Hide already learned commands:** MIDI commands that have already been mapped in Sequoia won't be shown in the list.
- **Interpret individual messages:** The MIDI commands are interpreted as 7 bit values.
- **Interpret double messages:** 2 MIDI commands are interpreted as 14 bit values.
- **Ping:** If this option is activated, the display of the MIDI Ping command entered under "Options" (view page 539) is disabled.
- **SysX:** If this option is activated, SysEx data is displayed.
- **Hexadecimal:** The MIDI commands are displayed in hexadecimal format. If this option is deactivated, the MIDI commands are displayed in decimal format.
- **Pause:** If this option is activated, no new data is displayed.

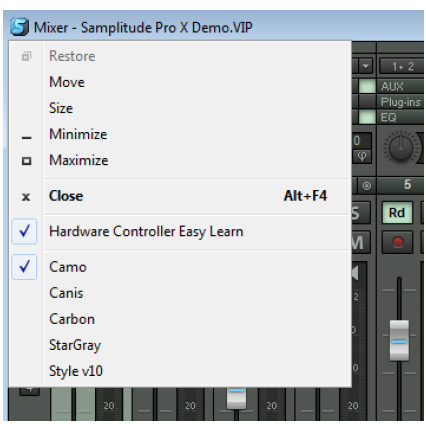
4 **Display range for MIDI data:** The MIDI data that is sent and received from the hardware controller is displayed here.

New MIDI data is displayed at the top of the list.

Controller Learning with Easy Learn (Plug-in and Mixer Elements)

The respective active mode can be adjusted for individual settings. The mixer system menu features the option "Hardware Controller Easy Learn" for this. Right-click on the title bar in the mixer or click on the icon to open this function.

Note: For assignment using Easy learn, only control elements that have been learned in the hardware controller dialog can be used.
Assignment using Easy Learn changes the active mode until the option "Restore Modes" is selected in the „Options“ (view page 539) menu.



1. Click on the command „Hardware Controller Easy Learn“.
2. Activate a mixer element with the mouse.
3. Now move the desired control element on the controller.
   Result: The controller control element now controls the newly assigned mixer element.

The plug-in menus also include the option "Hardware Controller Easy Learn". The parameters of a plug-in can also be mapped to a controller in the same way.



The changes remain available until

- the option „Restore Modes“ is selected in the hardware controller dialog or
- a change is made to the hardware controller dialog settings.

Here are the instructions for mapping a hardware controller:

1. Select the ports you used to connect the controller under MIDI input and MIDI output.
2. If there is already a template for your controller in the "Factory Presets", you can select this and your controller will be instantly ready for use.
   In this case you can skip the following steps.
3. Create a new controller entry by clicking on „Basic settings“ then „Add New" and selecting the option "#New". If a prompt for activation of the MIDI SysEx input then appears, answer this with "Yes".
4. If you haven't yet set up any connection between your controller and the PC, do that now. Most controllers are either connected by USB or MIDI.

5. In the "Options" tab, select "Generic MIDI" as the protocol and enter the number of control elements on your controller as the number of tracks (e. g. if your controller has 8 faders, enter "8").

   Note: If you are not sure, leave this value set to "0". You can also leave the other values in their default settings.

6. Now switch to the "Customize Controllers" tab.
7. Here you can first assign all of the control elements that you want to use. The following process is the same for every control element:
   - Open the mixer or plug-in you want to control and move the desired parameters with the mouse.
   - Now move the desired control element on the hardware controller. After 1 or 2 seconds it will appear in the Element List. The assigned parameter now appears in the "Easy Learn function" column of the control element.
   - Then check that the type has been correctly set and correct the setting if necessary (e. g. non-motorized faders and knobs should be set to the "Absolute" type).

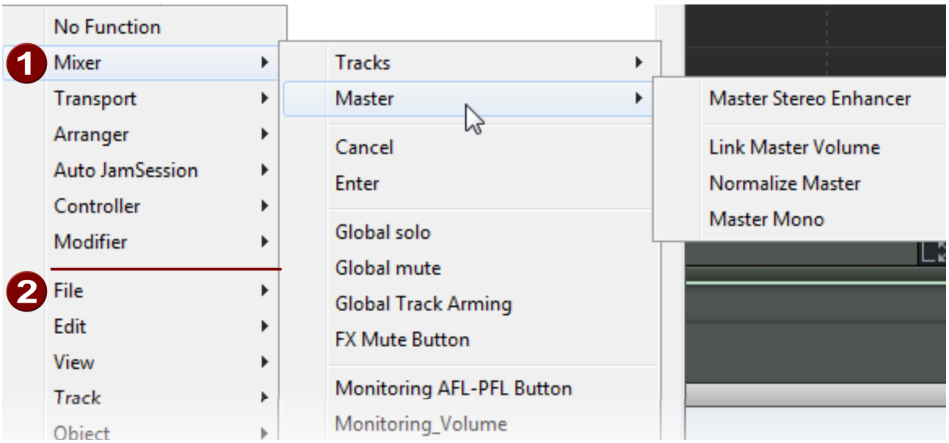
Notes and tips:

- As soon as you have opened the hardware controller settings, you are automatically in the Easy Learn Mode, i.e. every mouse movement of a mixer or plug-in parameter causes the assignment of this mixer or plug-in parameter to the last touched control element of your hardware controller.
- When you are finished assigning the controls be sure to shut "Hardware Controller Easy Learn" off to avoid making any unwanted changes to the settings.
- If it is not possible to assign a control element, first check in the "MIDI Monitor" ("Customize Controllers" tab) whether MIDI data is being received at all when moving the element. If this is not the case, your controller is either wrongly configured (wrong MIDI connection or incorrect internal mode of the controller) or the relevant control element is not sending data (other elements work). If the latter is the case, unfortunately you cannot use the element.
- In addition to the normally assigned "function", every control element can also have the "Easy Learn Function".
- The normal function is permanently assigned to the element whereas the "Easy-Learn Function" is a temporary function that has priority over the normal function if present.
- - You can also apply this after mapping an "Easy Learn Function". To do this, click with the right mouse button on the relevant element in the "Easy Learn Function" column. The "Easy Learn Function" is now transferred to the normal function.

Adjusting the assignment of a control element

Assigning a new function

1. Right-click on the desired entry in the "Function" column.
   The "Select Function" dialog will open.



2. Open the dropdown menu and select the desired function.
3. Click "OK" to close the dialog.
4. Click on the "Apply" button to confirm the changes.
5. Test the assignment by moving the control element on the hardware controller.

Mapping control elements

Mapping mixer or plug-in control elements

All of the control elements on the mixer or plug-ins that should be controlled by the hardware controller can be mapped using the Easy Learn function.

Requirement: The „Hardware Controller Setup“ dialog is opened in the "Adjust controller" view.

To do this, proceed as follows:

1. Select the mixer element that will be assigned to the control element.
2. Now move the control element on the hardware controller.
   Result: A new entry will be added to the list of control elements. The MIDI command that will be sent from the hardware controller is displayed in the "MIDI" column.

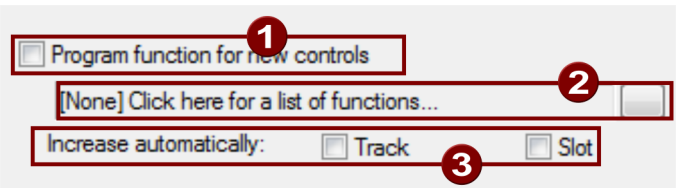
Note: If the control element on the hardware controller is touch-sensitive, a fader touch message will be sent when it is touched and when it is released. In this case 2 new elements are added.

3. Repeat this procedure for all of the control elements that should be mapped.
4. Select all of the entries in the list.
5. Move the mouse pointer over the entries in the "Easy Learn Function" column.
6. Right-click to open the context menu and select the option "Apply Easy Learn Function".
   The entries will be assigned automatically.
7. Click on the "Apply" button to confirm the changes.
8. When additional fader touch messages were added (button): Assign subfunction (view page 537).

Adding the same control element to multiple tracks/slots

Hardware controllers capable of simultaneously controlling multiple tracks/slots can be mapped quickly and easily.

To do this, proceed as follows:



1. Activate the option „Hardware Controller Easy Learn“ in the Mixer's menu.
2. Activate the option „Program Functions for New Controls“ 1.
3. Activate the option „Increase Automatically: Track/Slot“ 3.
4. Assign the desired function:
   1. Click on the field for the function selection 2.
      The "Select Function" dialog will open.
   2. Open the dropdown menu and select the desired function.
   3. If necessary, activate the desired modifier key in the dialog.
      The mapped function is then only activated when the set modifier key is also pressed.
      The modifier key must also be mapped (view page 537).
   4. Click „OK“ to close the dialog.

Note: Hardware controllers with touch-sensitive faders send a message whenever they are moved and released (FaderTouch). In this case 2 entries are automatically added, but the entry for the fader touch message is not assigned.

5. Move all of control elements of this type on the hardware controller.
   When the control elements are moved, a new entry is added and the track/slot number is increased by 1.
6. If additional fader touch entries are generated, assign these as subfunctions (view page 537).
7. Deactivate the option „Hardware Controller Easy Learn“ in the Mixer's menu.
8. Click on the "Apply" button to confirm the changes.

Mapping individual control elements

Control elements can be mapped individually, e.g. for special program functions. The modifier keys are also mapped individually.

To do this, proceed as follows:

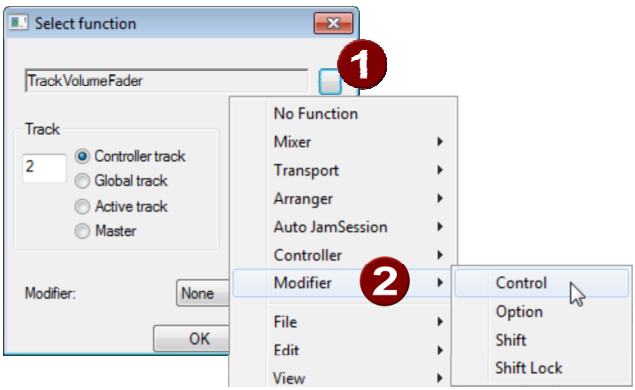
1. Manipulate the desired control element on the hardware controller.
   A new entry will be added to the list of control elements. The MIDI command that will be sent from the hardware controller is displayed in the "MIDI" column.

   Note: Hardware controllers with touch-sensitive faders send a message whenever they are moved and released (FaderTouch). In this case 2 new elements are added.

2. Assign the desired function:
   1. Right-click on the list entry in the "Function" column.
      The "Select Function" dialog will open.
   2. Open the dropdown menu and select the desired function.
   3. If necessary, activate the desired modifier key in the dialog.
      The mapped function is then only activated when the set modifier key is also pressed.
      The modifier key must also be mapped (view page 537).
   4. Click "OK" to close the dialog.
3. Repeat this process for all of the control elements that should be mapped.

Mapping modifier keys

Mapping modifier keys is exactly the same as mapping individual control elements (view page 536) of a hardware controller.

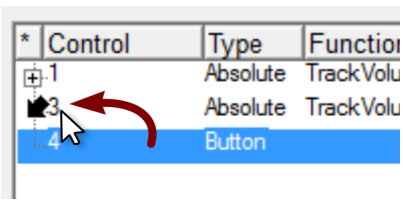


The modifier keys can be found in the "Select Function" dialog in the function selection menu 1 under "Modifier" 2.

Assigning subfunctions

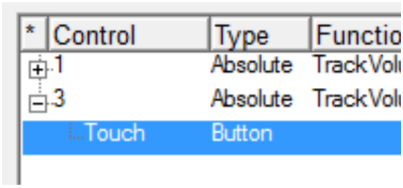
Tip: Assigning elements can be simplified if they are displayed in the order in which they were detected.
Select the "Elements" column to sort the detected control elements in this order.

1. Manipulate the control element on the hardware controller without moving it.
   The corresponding entry will be selected in the list.



2. Now drag & drop the entry onto the corresponding entry for the movement, e. g. „TrackVolumeFader (Global Track 1) (Slot 1)“.
   A popup menu will open.
3. Here you can specify the subfunction:
   - When dealing with a touch-sensitive control element such as a touch-sensitive fader:
     Select the entry „Insert as touch element“,

- When dealing with a control element with additional key function such as an encoder that can be additionally pressed:
  Select the entry „Insert as push element“,



Assigned touch function of a touch-sensitive fader

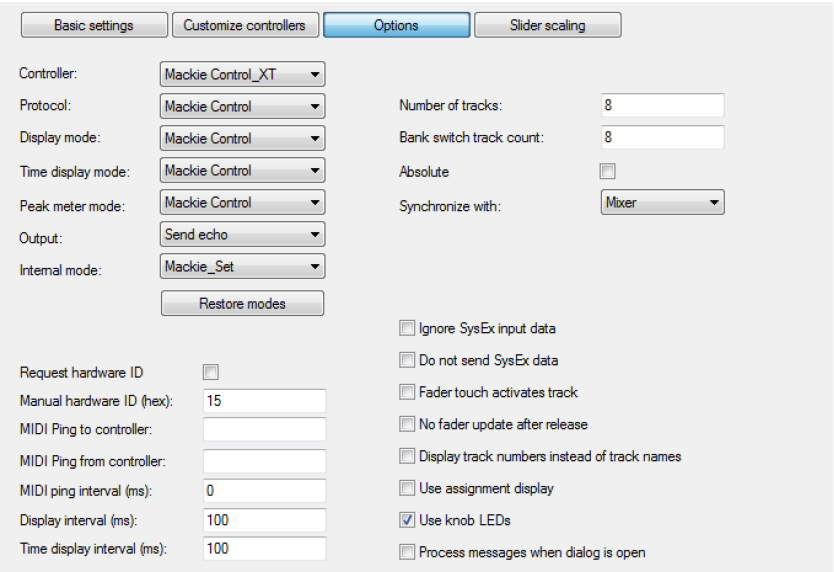
Correcting incorrectly recognized control elements

In rare cases some control elements might be improperly recognized during mapping, e.g. as "relative" instead of "absolute". This can be caused by various factors, e.g. large value jumps when moving the fader.

You can correct these incorrectly recognized control elements as follows:

1. Select the incorrectly recognized control elements:
   - with <Shift> for multiple control elements positioned one after the other
   - with <Ctrl> for multiple control elements
2. Position the mouse pointer over one of the entries in the "Type" column.
3. Right-click to open the context menu.
4. Select the appropriate type for the control element (view page 526).
5. Click on the "Apply" button to confirm the changes.

Options



Received MIDI messages are interpreted differently depending the the type of controller being used. In the "Options" view you can adjust the settings for the controller behavior to fit your personal workflow.

General Options

Controller

The hardware controller whose options will be edited is selected here. The list contains all hardware controllers that have been set in the "Basic Settings" view.

Protocol

The protocol determines how Sequoia communicates with the hardware controller. This determines the structure as well as the number and the order of the MIDI messages that make up a command.

The following protocols are available:

- Generic MIDI
- Mackie Control
- HUI
- JLCooper
- Tascam 2400
- FaderPort

Sequoia options

Number of tracks

Number of tracks: Specifies the number of tracks in a bank available to the controller. This is usually eight tracks. Mackie Control and HUI protocols can only communicate with eight channels per bank. However, some controllers have a native mode which lets you learn several channels (64 max.). For single track controllers, the value is "1".

Number of tracks for bank switch

A project in Sequoia can include more tracks than are available to the controller. In this case the controller can only display a section of the available tracks. To change this section you can perform a channel switch or bank switch. A channel switch changes the section by one track while a bank switch changes the section by several tracks. You can set different step sizes for each bank depending on the number of tracks on your controller and your preferred working method.

Synchronize with

The following options deal with the synchronization between the controller area, mixer area, and the active track. The controller area corresponds to the section of available tracks in the project that can be operated with the controller. The mixer area is the section of available tracks in the project that can be controlled with the Sequoia mixer.

Synchronize with Active Track: If the active track is changed, the controller area will be adjusted so that the active track is within the controller area.

Synchronize with Mixer: If a change is made to the controller area, the beginning of the mixer area will be adjusted according to the beginning of the controller area.

Note: Make sure that the synchronization of the view in Sequoia with the hardware controller is dependent on the control element settings (Track) (view page 527).

Fader touch activates track

This option activates the track that belongs to the fader in the application when a touch-sensitive fader is affected.

Use assignment display

Some controllers have an additional display field for showing the current operational mode (track, EQ, AUX, etc.). If a controller does not have this kind of display, you can

deactivate this option to prevent the control commands from incorrectly influencing other controller elements.

Process messages when dialog is open

Normally, when hardware controller dialogs are open, MIDI messages are no longer processed so that unwanted changes are avoided. When a controller is being customized or programmed it is sometimes necessary to send back messages to the controller. In this case, you can activate incoming messages while the dialog is open. Make sure to deactivate this feature when you are finished customizing the controller.

Options for controlling the hardware controller

Display mode

Many hardware controllers have LCD displays which differ in size and the number of lines and which need to be controlled differently. For hardware controllers without a display you can simply set the display control to "Off".

The following display modes are available:

- Off
- JLCooper
- Mackie Control
- HUI

Display interval (ms)

Display control occurs in certain time intervals and not continuously. This is meant to limit the amount of display information. A short interval accelerates the display on the controller but increases the amount of MIDI data that must be transferred.

Show track numbers instead of track names

If this option is active, track numbers will be shown instead of track names in the controller display.

Time display mode

Some hardware controllers have a special display for the current time position (Locator). The time display mode can be set for the following types of display:

- Off
- JLCooper
- Mackie Control
- HUI

Time display interval (ms)

Locator display control also takes place in time intervals. If the time position changes within an interval, only the last current value will be sent to the controller. If you shorten the interval, the locator display will react faster and more directly.

Peak Meter Mode

Some hardware controllers have a special peak meter display. The peak meter mode sets the controls for this display.
The following peak meter modes are available:

- Off
- Mackie Control
- HUI
- Tascam 2400

Output

- **Deactivate:** Hardware controller data is received and processed, Sequoia doesn't send any data back to the controller.
- **Normal Send:** The same as „Deactivate“, but changes in Sequoia are sent to the hardware controller.
- **Echo Send:** The same as "Normal Send" except that the received hardware controller data is immediately sent back and end at the controller, e.g. data for status LEDs.
- **Echo Send (without fader touch):** The same as "Echo Send", but no fader values are sent while the faders are being moved. The fader value is only sent back to the hardware controller when the fader is released.

Note: Some controllers (e. g. „Logic Control“) have problems receiving echo messages (termination). In this case select the setting "Normal Send".

Do not send SysEx data

Sequoia sends different messages via SysEx. Exactly what is sent depends on the selected display mode, locater mode etc.

If SysEx messages start to cause problems, you can prevent these kinds of messages from being sent to the controller.

No fader update after release (Motorized Fader)

When you move touch-sensitive faders, the fader messages to the controller are normally ignored by the controller. After release the hardware controller expects a position message as termination.

If this message is not sent, the fader will slide back to the position it was in before it was moved. Some controllers do not need this new position message. When they receive the new position they first jump to the old position and then to the new one.

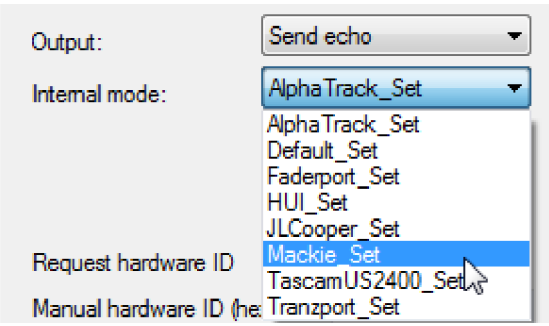
If the fader update after release of the fader should not be sent, you can deactivate this sending of the new position here.

Control LED knobs

Some controllers have LED displays that are assigned to the knobs. These can be used to visually display the current parameters of the knob. If you are using a controller with no knob LEDs, you can deactivate this option.

Options for the interpretation of hardware controller demands

Hardware Controller - Internal Mode ("Options" tab)



Note: The parameter „Internal Mode“ is only available for hardware controllers that have been added to the "Basic Settings" view on the first level and therefore act as the "Main Controllers".

The "Internal Mode" determines how Sequoia reacts to incoming remote commands from hardware controllers and enables adaptation to conceptually different hardware controllers.

Depending on the mode, different information is displayed on the controller and Sequoia interprets the controller messages correspondingly. The effect is that a particular physical control element controls different parameters in Sequoia, depending on the internal operating mode.

These data structures can be manipulated using the function „Hardware Controller Easy Learn“ in the mixer menu and the menus of the VST plug-ins (view page 531). In this case, the active operating mode can be changed so that it is able to function

differently than its preset. The changed data structures are saved accordingly and are immediately available the next time the program is launched. This ensures that you are able to continue working with the most recent settings.

Note: If you are using a hardware controller that is not listed in the hardware control setup, select one from the list that is most similar in concept and design.

The following internal modes (Mapping Modes) are available:

Frontier AlphaTrack_Set

Track

Press first time: Panorama mode

- Panorama control
- Navigate to markers
- Move selected track

Press second time: Active control mode

- Changes the value of the selected element in the mixer with the fader.

AUX

- Select other track
- Select other slot
- Change AUX send of the selected track/selected slot

EQ

Controlling the input volume, frequency, and quality bands of the selected EQs. The individual bands are selected with "Mixer track up/down".

Plug-in

- Select track
- Select plug-in slot in the track with "Mixer track up/down" key

Now you can control 3 parameters of the corresponding plug-ins with the vpots. The display page is toggled with the keys "Mixer track up/down".

Default Set

Presonus FaderPort_Set

Active control

The currently active mixer element (recognizable by red highlighting) is controlled by the first fader.

Mackie HUI_Set

Track

Track mode

Active control

The currently active mixer element (recognizable by red highlighting) is controlled by the first fader.

EQ (Phat Channel EQ mode)

Controls the EQ for the selected track via the knob controllers.

AUX (Phat Channel AUX mode)

Controls all AUX sends in the selected channel.

AUX [slots] (1 - 8) (Track AUX mode)

Controls AUX sends for the different tracks in the selected slot.

JLCooper_Set (MCS)

Track

Press first time: Track mode

Press second time: Active control mode

- Each activated mixer element (recognizable by the red marking) is controlled by the first fader

AUX

Controls all AUX sends in the selected channel.

EQ

Controls the EQ of the selected track via the knobs.

Mackie Multitrack

Track

Press first time: Track mode

Press second time: Active control mode

- Each activated mixer element (recognizable by the red marking) is controlled by the first fader

PAN

- Pan mode (like track mode, except for the display)

EQ

Press first time: EQ Phat Channel Bandwise mode

- Gain, frequency, and quality accessible via the knobs
- Use the bank switch keys to access the other bands

Note: The bank switch functions have to be learned, which is usually the case

Press second time: EQ Phat Channel Typewise mode

- Gain, frequency, and quality accessible via the knobs (differently grouped than in Bandwise mode)
- Using the bank switch keys, each of the other EQ functions can be accessed

AUX

Press first time: AUX track mode

- AUX 1 to 6 of the active track using the knobs

Press second time: AUX slot mode

- Set AUX for each track
- Select number of AUX tracks to be controlled using the Bank Switch keys

Plug-in

- The VST plug-in slots are listed for the active track (flip through via bank switch keys)
- Selection of the VST plug-in to be adjusted by pressing the corresponding knob
- Thereafter the parameters for the selected plug-ins are shown
- Changing the parameter using the knob
- Page through the parameters

Frontier Tranzport_Set

Functions: For transport only

Request hardware ID

Activate or deactivate the automatic hardware ID request. Deactivate this function if you aren't using the "Mackie Control" display mode. This helps to avoid incorrect interpretation of the request message.

Manual Hardware ID

For display in "Mackie Control" display mode, an additional parameter is necessary: The hardware ID. This can be automatically requested with new firmware versions or manually entered

MIDI Ping to controller

Most controllers expect a periodic incoming MIDI message (ping) to ensure that there is communication with the application. Different messages can also change the operating modes of a controller. Here you can enter the message that should be sent to the controller.

MIDI Ping from controller

In response to ping messages, controllers usually send a ping message back. To ensure that these are not falsely mapped as MIDI commands you can enter these explicitly so that they are ignored by the application.

MIDI Ping interval (ms)

Here you can specify the time period between the two ping messages in milliseconds.

Ignore SysEx input data

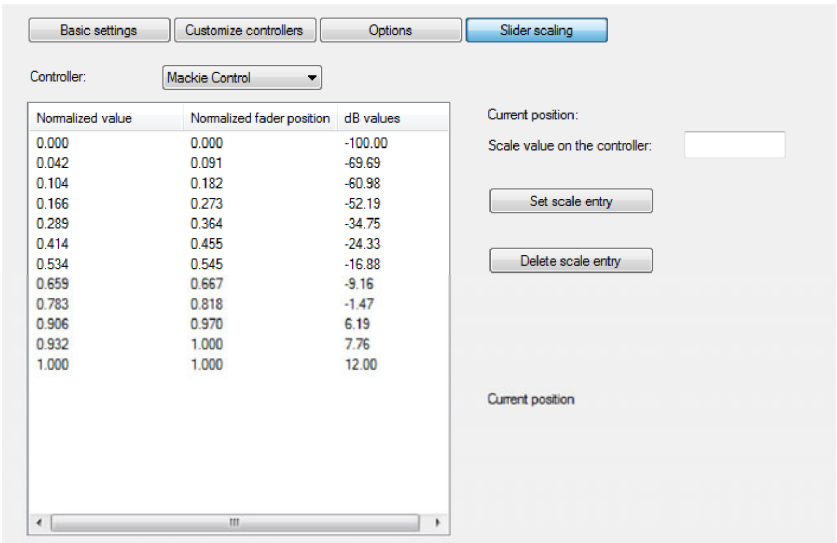
SysEx data from the hardware controller will not be processed by the application.

Fader Scaling

Fader scaling offers you the option of calibrating the scale on the hardware controller to the volume fader values in Sequoia.

Control with a hardware controller can take place in a range from +12 dB to -100dB. Normally at +12 dB the fader position is at the very top and at -100 dB at the very bottom.

Note: Calibration of the fader scale only makes sense when the fader scale on the controller actually includes numerical values.



Add scale values

1. In the „Controller“ section select the hardware controller whose printed scale should be calibrated in Sequoia.
2. Delete all previous entries. Click on the entry in the list and then press "Delete scale entry".
3. Enter the value "6" into the "Scale entries on controller" and move the fader all the way up (to the scale value +6dB). Next, click "Set scale entry" to set the new entry.
4. Move the fader all the way to the bottom and enter the lowest scale value (at least „-100“) in the field „Scale entry on controller“. Click "Set scale entry" again.
5. Enter further scale positions in the same way (e. g. -12, -24, -48...).

Test the mapped scale

1. Click "OK" to close the dialog.
2. Compare the entered fader position of the hardware controller with the fader position in the program mixer.

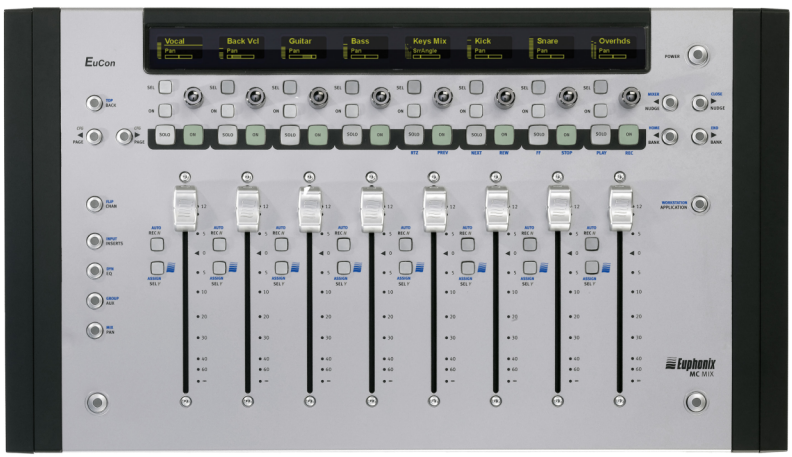
Changing entered scale values

1. Select the incorrectly mapped value
2. Click on the button "Delete scale entry"
3. Add the correct scale value.

Hardware Controller Presets

The following is a list of the most important presets for hardware controllers including their integration and configuration.

Artist Mix



1. Close your controller (according to manufacturer's instructions) and install the required driver software.
2. Go to System Options/Hardware Controller and select "Activate EUCON".
3. Close the dialog. Your controller is now ready to use.

Note

The modes INPUT, DYN, GROUP and MIX are not available in the software.

Track range

Control element	Function	Shift function
SEL 1~8	AUX: on/off, EQ: Switch between Freq and Q	
ON 1~8		
KNOB 1~8	Depending on mode (PAN, AUX, EQ, INSERTS)	
SOLO 1~8	Track solo	
ON 1~8	Track is on (inverted mute)	

Control element	Function	Shift function
RTZ		Cursor at project start
PREV		Previous marker
NEXT		Next marker
REW		Fast rewind
FF		Fast forward
STOP		Stop
PLAY		Playback
REC		Recording
FADER 1~8	Track volume	
AUTO REC N	Track record	Track automation
ASSIGN SEL Y	Track selection	Arrange tracks using EuControl

Navigation

Control element	Function	Shift function
TOP BACK	Return from a subpage	Return to upper most page
< PAGE	Parameter page to the left e.g. in channel editing mode, AUX or Inserts	
PAGE >	Parameter page to the right e.g. in channel editing mode, AUX or Inserts	
FLIP CHAN	Channel editing mode (Phat-Channel)	Flip mode KNOBS and FADERS swap functions
INPUT INSERTS	Plug-in Mode	
DYN EQ	EQ mode	
GROUP AUX	AUX mode	
MIX PAN	Pan mode	
SHIFT	Shift key	

Control element	Function	Shift function
MIXER < NUDGE	Controller track 1 to the left	Mixer
CLOSE NUDGE >	Controller track 1 to the right	Close application
HOME < BANK	Controller track 8 to the left	View goes to track 1
END BANK >	Controller track 8 to the right	View goes to last track
WORKSTATION APPLICATION	Switch the current Windows application	Workstation switch

Artist Control



1. Close your controller (according to manufacturer's instructions) and install the required driver software.
2. Depending on the program version, copy the files "Sam.xml" or "Sequoia.xml" from the program folder „/Customize/EUCON“ to „Programs\Euphonics\EuCon\SQRL“ or to the location of your user template.
3. Go to System Options/Hardware Controller and select "Activate EUCON".
4. Close the dialog. Your controller is now ready to use.

Note

The modes INPUT, DYN, GROUP and MIX are not available in the software.

Track range

Control element	Function	Shift function
SOLO 1~4	Track solo	
ON 1~4	Track is on (inverted mute)	

Control element	Function	Shift function
FADER 1~4	Track volume	
AUTO REC N	Track record	Track automation
ASSIGN SEL Y	Track selection	Arrange tracks using EuControl
HOME SOFT KEYS	SOFT KEYS upwards	SOFT KEYS to first page
END SOFT KEYS	SOFT KEYS downwards	SOFT KEYS to last page

Display area

Control element	Function	Shift function
KNOB 1~8	Function independent of view	
SOFT KEYS 1~12	Function independent of view	

Navigation

Control element	Function	Shift function
MONITOR CONTROL ROOM	Monitor volume	Solo-Bus volume
TOP BACK	Return from a subpage	Return to upper most page
WORKSTATION APPLICATION	Switch the current Windows application	Workstation switch
< PAGE	Parameter page to the left e.g. in channel editing mode, AUX or Inserts	
PAGE >	Parameter page to the right e.g. in channel editing mode, AUX or Inserts	
MIXER < NUDGE	Controller track 1 to the left	Mixer
CLOSE NUDGE >	Controller track 1 to the right	Close application
HOME < BANK	Controller track 8 to the left	Display switches to Track 1

Control element	Function	Shift function
END BANK >	Controller track 8 to the right	Display switches to last track
REC	Recording	
PLAY	Playback	
STOP	Stop	
FF	Fast forward	
REW	Fast rewind	
SHUTTLE JOG	Positioning	Scrubbing
VERTICAL ZOOM HORIZONTAL ZOOM		
SHIFT	Shift key	

Artist Transport

1. Close your controller (according to manufacturer's instructions) and install the required driver software.
2. If the files "Sam.xml" or "Sequoia.xml" don't exist in the driver software in Programs\Euphonics\EuCon\SQRL, copy and paste it here from the folder program folder "/Customize".
3. Go to System Options/Hardware Controller and select "Activate EUCON".
4. Close the dialog. Your controller is now ready to use.



Note

The modes INPUT, DYN, GROUP and MIX are not available in the software.

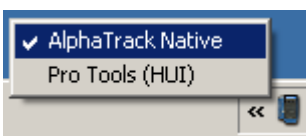
Controller configuration (Artist control)

Control element	Function	Shift function
SOFT KEY 1	Jog	
SOFT KEY 2	Scrub	
SOFT KEY 3	Loop mode	
SOFT KEY 4	Metronome	
SOFT KEY 5	Sync	
SOFT KEY 6	Position input Apply using ENTER Abort using *	
Numeric keys (0 ~ 9)	Number input	
-		
+		
.	Decimal place	
*	ESC/Cancel	
Enter	Enter	
TRANSPORT KEY 1	Cursor at project start	
TRANSPORT KEY 2	Fast rewind	
TRANSPORT KEY 3	Fast forward	
TRANSPORT KEY 4	Stop	
TRANSPORT KEY 5	Playback	
TRANSPORT KEY 6	Record	
TRANSPORT KEY 7	Loop mode	
WORKSTATION APPLICATION	Switch the current Windows application	Workstation switch
SHIFT	Shift key	

Frontier AlphaTrack



1. The "AlphaTrack" is supported in Native Mode. For this reason you should set the device to „AlphaTrack Native“ by right-clicking on the "AlphaTrack" button in the taskbar.



2. In Samplitude/Sequoia go to „System Settings/Hardware Controller“, click on the "New" button and select "AlphaTrack" from the list.
3. Now set the right side of the MIDI port to "AlphaTrack".
4. Close the dialog. Your controller is now ready to use.

Controller configuration

Control element	Function	Shift-Label	SHIFT function
Foot switch	Recording		
FADER	Track volume		
ANY SOLO (LED)	Indicates that at least one track in the arrangement is switched to SOLO		
AUTO WRITE (LED)			
AUTO READ (LED)			
REC	Record track		Monitor track
SOLO	Track solo		Global solo
MUTE	Mute track		Global mute
SHIFT	Shift		
PAN	Press first time: Pan mode Second pressing: Active Control Mode (see table 2)		
SEND	AUX mode (see table 2)		
EQ	EQ mode (see table 2)		
Plug-in	Plug-in Mode (see table 2)		
AUTO	Track automation		Copy automation mode to all tracks
F1	Touch automation	F5	Read automation
F2	Latch automation	F6	Automation off
F3	Overwrite automation	F7	Restore
F4	Trim automation	F8	Undo

Control element	Function	Shift-Label	SHIFT function
< TRACK	Previous track	IN	Punch Start Marker
TRACK >	Next track	OUT	Punch End Marker
LOOP	Loop Mode	PUNCH	Punch Marker Mode
FLIP	Mixer	WINDOW	Delete punch marker
<<	Fast rewind	RTZ	Playback marker to project start
>>	Fast forward	END	Playback marker to project end
STOP	Stop	Esc	Cancel
PLAY	Playback	Enter	Enter
RECORD	Recording	MODE	
TOUCHPAD	Positioning		

Special modes

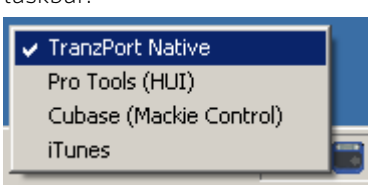
Mode	Control element	Turn	Press
Pan mode	KNOB 1	Track selection	
	KNOB 2	To the marker	Set marker
	KNOB 3	Pan	Reset pan
ACTIVE CONTROL MODE	FADER (controls the active element in the mixer, track editor, or plug-in that must be selected with the mouse beforehand)		
	KNOB 1	Pan	
	KNOB 2	Pan	
	KNOB 3	Pan	
AUX mode	KNOB 1	Track selection	
	KNOB 2	AUX level	
	KNOB 3	AUX selection	
EQ mode	KNOB 1	Intensity	

Mode	Control element	Turn	Press
	KNOB 2	Frequency	
	KNOB 3	Quality	
	< TRACK	Previous EQ band	
	TRACK >	Next EQ band	
Plug-in Mode Plug-in slot selection	KNOB 1	Track selection	
	KNOB 2		
	KNOB 3	Slot selection	
	< TRACK	Select plug-in	
	TRACK >	Select plug-in	
Plug-in parameter	KNOB 1	Change parameters	
	KNOB 2	Change parameters	
	KNOB 3	Change parameters	
	< TRACK	Previous parameters	
	TRACK >	Next parameters	

Frontier TranzPort



1. „TranzPort“ is supported in „Native Mode“. For this reason you should set the device to „TranzPort Native“ by right-clicking on the "TranzPort" button in the taskbar.



2. In Samplitude/Sequoia go to "System Settings/Hardware Controller, click the „Add New“ button and select „Frontier_TranzPort (Native)“ from the list.
3. Now,set the right side of the MIDI port to TranzPort.
4. Close the dialog. Your controller is now ready to use.

Controller configuration

Control element	Function	SHIFT function
< TRACK	Previous track	
> TRACK	Next track	
REC	Record track	Monitor track

Control element	Function	SHIFT function
MUTE	Mute track	Global mute
SOLO	Track solo	Global solo
ANY SOLO (LED)	Indicates that at least one track in the arrangement is switched to SOLO	
UNDO	Undo	Restore
IN	Punch Start Marker	
OUT	Punch End Marker	
PUNCH	Punch Marker Mode	Delete punch marker
LOOP	Loop Mode	
SHIFT	Shift	
PREV MARKER	Previous marker	Metronome
ADD MARKER	Insert marker	
NEXT MARKR	Next marker	Scrub mode
<<	Fast rewind	Playback marker to project start
>>	Fast forward	Playback marker to project end
STOP	Stop	Cancel
PLAY	Playback	Enter
RECORD	Recording	
DATA WHEEL	Positioning	Track volume
DATA WHEEL + STOP	Selected range	
Foot switch	Recording	

JLCooper MCS-3800



1. In Samplitude/Sequoia go to "System Settings/Hardware Controller, click the „Add New“ button and select „JLCooper MCS-3800 (Native)“ from the list.
2. Now set up the MIDI ports with the respective inputs and outputs of your MIDI interface on the right side.
3. Close the dialog. Your controller is now ready to use.

Notes

- In Shift Mode the button LEDs are not updated for the status of the commands mapped in "Shift Mode"
- LEDs for Bank3 (time information) and Bank 4 (video window) are inactive.
- Plug-in control is inactive.

Track range

Control element	Function	Modifier	Modifier function
SELECT 1~8	Select track		
AUX 1~8	Record track	SHIFT	Track monitoring 1~8 (ASIO only)
SOLO 1~8	Track solo		
MUTE 1~8	Mute track	SHIFT	Track automation
FADER 1~8 (with fadertouch)	Track volume		

Navigation

Control element	Function
Cursor left	Mixer track 1 to the left
Cursor right	Mixer track 1 to the right
Cursor up	Mixer track 8 to the right
Cursor down	Mixer track 8 to the left

Area to view

Control element	Function
PAGE 1	**Track mode** (see special modes)
PAGE 2	**AUX mode** (see special modes)
PAGE 3	**EQ mode** (see special modes)
PAGE 4	**Active control mode** (see special modes)
PAGE 5	
PAGE 6	
PAGE 7	
PAGE 8	
BANK 1	Mixer
BANK 2	Transport console
BANK 3	Time display
BANK 4	Video window
KNOB 1~5	Change parameters

Functions window

Control element	Function
F1	Punch Marker Mode
F2	Loop
F3	Metronome
F4	Synchronization
F5	
F6	

Control element	Function
F7	
F8	
SHIFT	Shift

Mode range

Control element	Function	Modifier	Modifier function
M1	Read automation	Shift	Automation off
M2	Touch automation		
M3	Latch automation		
M4	Overwrite automation		
M5	Trim automation		

Shuttle range

Control element	Function	Modifier	Modifier function
Shuttle wheel	Positioning	W1	Scrubbing
Rotation ring			
W1	Scrub mode		
W2	Punch Start Marker		
W3	Punch End Marker		
W4	Delete punch marker		
W5			
W6			
W7			

Locator range

Control element	Function
LOCATOR numbers (0 ~ 9)	Entry position: Apply using ENTER or PLAY Abort using STOP
Enter	Apply position
CLEAR / CANCEL	Erase single digits

Control element	Function
+/-	
LAST	
SET LOCATE	
MODE LOCATE	

Transport range

Control element	Function
REWIND	Fast rewind
FAST FORWARD	Fast forward
STOP	Stop
PLAY	Playback
RECORD	Recording

Special modes

Mode	Control element	Function
Track mode	KNOB 1	
	KNOB 2	
	KNOB 3	
	KNOB 4	Track pan
	KNOB 5	To the marker
AUX mode	KNOB 1	AUX level
	KNOB 2	AUX level
	KNOB 3	AUX level
	KNOB 4	AUX level
	KNOB 5	Selection AUX range
EQ mode	KNOB 1	Level EQ band 1
	KNOB 2	Frequency EQ band
	KNOB 3	Quality EQ band
	KNOB 4	

Mode	Control element	Function
	KNOB 5	Band selection
Active control mode	FADER TRACK 1 (controls the active element in the mixer, track editor, or plug-in that must be selected with the mouse beforehand)	

Logic Control

1. In Sequoia go to „System Settings/Hardware Controller“, click on the "Add New" button and select "Logic Control" from the list.
2. Now set up the MIDI ports with the respective inputs and outputs of your MIDI interface.
3. Close the dialog. Your controller is now ready to use.

TRACKS

Control element	Function	Modifier	Modifier function
KNOB 1~8	Pan; Eq; Aux (see **special modes**)		
REC/RDY 1~8	Activate track	SHIFT	Monitor track
SOLO 1~8	Track solo		
MUTE 1~8	Mute track		
SELECT 1~8	Select track		
FADER 1~8 (with fadertouch)	Track volume		
MASTER FADER	Master volume		

ASSIGNMENT

Control element	Function
1	Press first time: Track mode Press second time: Active Control mode (see special modes)
2	Press first time: AUX mode Press second time: AUX mode (phat channel) (see special modes)

PAN/SURROUND	Pan mode (see special modes)
Plug-in	Plug-in Mode (see special modes)
EQ	EQ mode (see special modes)
INSTRUMENT	

FADER BANKS

Control element	Function
< BANK	Mixer track 8 to the left
BANK >	Mixer track 8 to the right
< CHANNEL	Mixer track 1 to the left
CHANNEL >	Mixer track 1 to the right
FLIP	Flip mode: Exchange functions of knobs and faders
GLOBAL VIEW	Mixer

DISPLAY

Control element	Function
NAME/VALUE	
SMTP/BEATS	Timecode switch

FUNCTION BUTTONS

Control element	Function	Modifier	Modifier function
F1-F8	Set / jump to marker 1-8	SHIFT CNTRL	Move markers 1-8 Delete markers 1-8

GLOBAL VIEW

Control element	Function
MIDI TRACKS	
INPUTS	
AUDIO TRACKS	
AUDIO INSTRUMENT	

Control element	Function
AUX	
BUSSES	Select previous object.
OUTPUTS	Select next object
USER	Crossfade Editor

MODIFIERS

Control element	Function
SHIFT	Control
OPTION	Option
CONTROL	
# / ALT	Shift

AUTOMATION

Control element	Function	Modifier	Modifier function
READ/OFF	Read automation	SHIFT	Automation off
WRITE	Overwrite automation		
TRIM	Trim automation		
TOUCH	Touch automation		
LATCH	Latch automation		
GROUP	Copy automation mode to all tracks		

UTILITIES

Control element	Function	Modifier	Modifier function
SAVE	Save		
UNDO	Undo	SHIFT	Repeat
CANCEL	Cancel		
Enter	Enter		

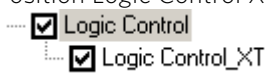
Special modes

Mode	Control element	Turn	Press
Track mode	KNOB	Pan	Reset
ACTIVE CONTROL MODE	FADER (controls the active element in the mixer, track editor, or plug-in that must be selected with the mouse beforehand)		
Pan mode	KNOB	Pan	Reset
EQ mode	KNOB 1	Level EQ band 1	
	KNOB 2	Level EQ band 2	
	KNOB 3	Level EQ band 3	
	KNOB 4	Level EQ band 4	
	KNOB 5	Frequency EQ band 1	
	KNOB 6	Frequency EQ band 2	
	KNOB 7	Frequency EQ band 3	
	KNOB 8	Frequency EQ band 4	
AUX mode AUX 1~8 selectable using BANK	KNOB 1~8	AUX level	
AUX mode (phat channel) depending on selected track	KNOB 1	AUX 1 level	
	KNOB 2	AUX 2 level	
	KNOB 3	AUX 3 level	
	KNOB 4	AUX 4 level	
	KNOB 5	AUX 5 level	
	KNOB 6	AUX 6 level	
	KNOB 7		
	KNOB 8		
Plug-in Mode	KNOB 1~8		Select plug-in

Mode	Control element	Turn	Press
Plug-in parameter	KNOB 1~8	Change parameters	
	< BANK	Previous parameters	
	BANK >	Next parameters	
Marker mode	REWIND		Previous marker
	FAST FWD		Next marker
Object mode	REWIND		Previous Object
	FAST FWD		Next object

Logic Control XT

1. In Samplitude/Sequoia go to „System Settings/Hardware Controller“, click on the "Add New" button and select "Logic Control XT" from the list.
2. Position Logic Control XT under Logic Control

   If you have set up Logic Control XT to the left of Logic Control, the entries in the list will look like this:

3. Now set up the MIDI ports with the respective inputs and outputs of your MIDI interface on the right side.
4. Close the dialog. Your controller is now ready to use.

The functions correspond with the Logic Control.

Note

The number of tracks to be jumped when a bank switch is pressed can be changed in the hardware controller setup options. These settings are made in "Logic Control".

Mackie Control

1. In Samplitude/Sequoia go to „System Settings/Hardware Controller“, click on the "Add New" button and select "Mackie Control" from the list.
2. Now set up the MIDI ports with the respective inputs and outputs of your MIDI interface on the right side.
3. Close the dialog. Your controller is now ready to use.

Note

- All entries are based on MCU Firmware 2.1.2.

TRACKS

Control element	Function	Modifier	Modifier function
KNOB 1~8	Pan; Eq; Aux (see **special modes**)		
REC/RDY 1~8	Activate track	SHIFT	Monitor track
SOLO 1~8	Track solo		
MUTE 1~8	Mute track		
SELECT 1~8	Select track		
FADER 1~8 (with fadertouch)	Track volume		
MASTER FADER	Master volume		

ASSIGNMENT

Control element	Function
1	Press first time: Track mode Press second time: Active Control mode (see special modes)
2	Press first time: AUX mode Press second time: AUX mode (phat channel) (see special modes)
PAN	Pan mode (see special modes)
Plug-ins	Plug-in Mode (see special modes)
EQ	EQ mode (see special modes)
DYN	

FADER BANKS

Control element	Function
< BANK	Mixer track 8 to the left
BANK >	Mixer track 8 to the right
< CHANNEL	Mixer track 1 to the left
CHANNEL >	Mixer track 1 to the right
FLIP	Flip mode: Exchange functions of knobs and faders
EDIT	Mixer

DISPLAY

Control element	Function
NAME/VALUE	
SMTP/BEATS	Timecode switch

FUNCTION BUTTONS

Control element	Function	Modifier	Modifier function
F1-F8	Set / jump to marker 1-8	SHIFT CNTRL	Move markers 1-8 Delete markers 1-8
F9-F13			
F14	Select previous object.		
F15	Select next object		
F16	Crossfade Editor		

MODIFIERS

Control element	Function
CNTRL	Control
OPT	Option
SNAPSHOT	
SHIFT	Shift

AUTOMATION

Control element	Function	Modifier	Modifier function
READ/OFF	Read automation	SHIFT	Automation off
WRITE	Overwrite automation	SHIFT	Copy automation mode to all tracks
UNDO	Undo	SHIFT	Repeat
SAVE	Latch automation		
TOUCH	Touch automation		
REDO	Trim automation		

UTILITIES

Control element	Function
FDR GRP	Cancel
CLR SOLO	Enter
MRKR	
MIXR	Save

TRANSPORT

Control element	Function	Modifier	Modifier function
< FRM	Marker mode (see table 2)		
FRM >	Object mode (see table 2)		
END	Loop mode		
PI	Punch Marker Mode		
PO	Synchronization		
LOOP	Metronome		
HOME	Activate LCD meter by holding "F1", "F2", "F3", or "F4"		
REWIND	Fast rewind	SHIFT	Playback marker to project start

Control element	Function	Modifier	Modifier function
FAST FWD	Fast forward	SHIFT	Playback marker to project end
STOP	Stop		
PLAY	Playback		
RECORD	Recording		
Cursor left	One section left	ZOOM	Horizontal project zoom (-)
Cursor right	One section right	ZOOM	Horizontal project zoom (+)
Cursor up	Next highest track	ZOOM	Vertical project zoom (+)
Cursor down	Next lowest track	ZOOM	Vertical project zoom (-)
ZOOM	Zoom mode		
SCRUB	Scrub mode		
Shuttle wheel	Positioning	SCRUB	Scrubbing

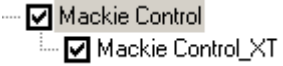
Special modes

Mode	Control element	Turn	Press
Track mode	KNOB	Pan	Reset
ACTIVE CONTROL MODE	FADER (controls the active element in the mixer, track editor, or plug-in that must be selected with the mouse beforehand)		
Pan mode	KNOB	Pan	Reset
EQ mode	KNOB 1	Level EQ band 1	
	KNOB 2	Level EQ band 2	
	KNOB 3	Level EQ band 3	
	KNOB 4	Level EQ band 4	
	KNOB 5	Frequency EQ band 1	

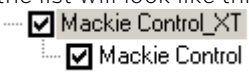
Mode	Control element	Turn	Press
	KNOB 6	Frequency EQ band 2	
	KNOB 7	Frequency EQ band 3	
	KNOB 8	Frequency EQ band 4	
AUX mode AUX 1~8 selectable using BANK	KNOB 1~8	AUX level	
AUX mode (phat channel) depending on selected track	KNOB 1	AUX 1 level	
	KNOB 2	AUX 2 level	
	KNOB 3	AUX 3 level	
	KNOB 4	AUX 4 level	
	KNOB 5	AUX 5 level	
	KNOB 6	AUX 6 level	
	KNOB 7		
	KNOB 8		
Plug-in Mode Plug-in parameter	KNOB 1~8		Select plug-in
	KNOB 1~8	Change parameters	
	< BANK	Previous parameters	
	BANK >	Next parameters	
Marker mode	REWIND		Previous marker
	FAST FWD		Next marker
Object mode	REWIND		Previous Object
	FAST FWD		Next object

Mackie Control XT

1. In Samplitude/Sequoia go to „System Settings/Hardware Controller“, click on the "Add New" button and select "Mackie Control XT" from the list.
2. Position Mackie Control XT under Mackie Control.

   If you have set up Mackie Control XT to the left of Mackie Control, the entries in the list will look like this:

3. Now set up the MIDI ports with the respective inputs and outputs of your MIDI interface on the right side.
4. Close the dialog. Your controller is now ready to use.

The functions correspond with the Mackie Control.

Note

The number of tracks to be jumped when a bank switch is pressed can be changed in the hardware controller setup options. These settings are available in Mackie Control.

PreSonus FaderPort

1. In Samplitude/Sequoia go to "System Settings/Hardware Controller, click the „Add New“ button and select „PreSonus FaderPort (Native)“ from the list.
2. Now set the MIDI ports on the right side to USB audio device.
3. Close the dialog. Your controller is now ready to use.



Notes

- The foot switch is functional starting with the hardware controller firmware 1.2.
- To map control elements for the "FaderPort", enter the response values to the controller for learning so that the button LEDs also function. To do this copy and paste the value from the last column of the assignment table (view page 576) of the corresponding control element into the field „Send MIDI Data“.

Controller configuration

Control element (SHIFT)	Function	SHIFT function	MIDI Command	Return value for LED
FADER	Track volume			
PAN	Track pan			
MUTE	Mute track	Global mute	A01200	A01501
SOLO	Track solo	Global solo	A01100	A01601

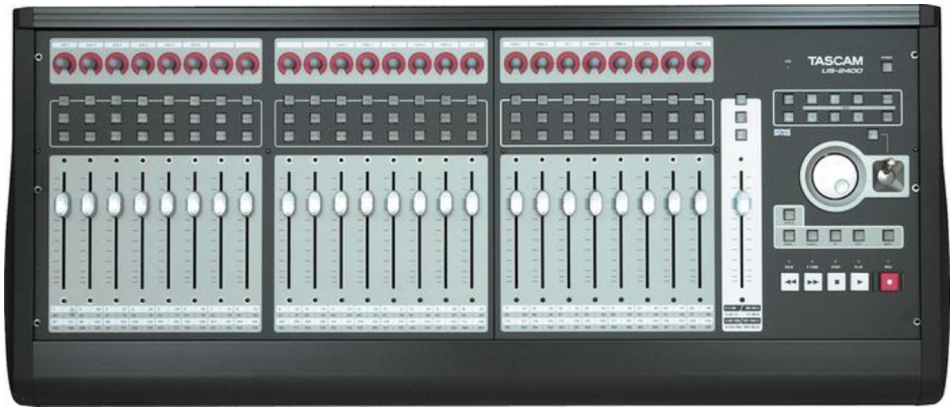
Control element (SHIFT)	Function	SHIFT function	MIDI Command	Return value for LED
REC	Record track	Monitor track (ASIO only)	A01000	A01701
<	Previous Track/Bank		A01300	A01401
BANK	"<" and ">" switch bank		A01400	A01301
>	Next Track/Bank		A01500	A01201
OUTPUT	ACTIVE CONTROL MODE (see special modes (view page 578))		A01600	A01101
READ	Read automation		A00A00	A00D01
WRITE	Overwrite automation	Latch automation	A00900	A00E01
TOUCH	Touch automation	Trim automation	A00800	A00F01
OFF	Automation off	Copy automation mode to all tracks	A01700	A01001
MIX	Mixer		A00B00	A00C01
PROJ	Focus on arranger		A00C00	A00B01
TRNS	Transport console		A00D00	A00A01
UNDO (REDO)	Undo	Restore	A00E00	A00901
SHIFT	Shift		A00200	A00501
PUNCH (PREV)	Punch Marker Mode	Previous marker	A00100	A00601
USER (NEXT)	Metronome	Next marker	A00000	A00701

Control element (SHIFT)	Function	SHIFT function	MIDI Command	Return value for LED
LOOP (MARK)	Loop Mode	Insert marker	A00f00	A00801
REWIND (START)	Fast rewind	Playback marker to project start	A00300	A00401
FAST FORWARD (END)	Fast forward	Playback marker to project end	A00400	A00301
STOP	Stop		A00500	A00201
PLAY	Playback		A00600	A00101
RECORD	Recording		A00700	A00001
FOOT SWITCH	Recording		A07e00	

Special modes

Mode	Control element
ACTIVE CONTROL MODE	FADER (controls the active element in the mixer, track editor, or plug-in that needs to be selected with the mouse beforehand)

Tascam US-2400



1. The "US-2400" is supported in "Native Mode". Use the most current firmware. Start the device with the "Power" button while holding down the "SEL" (above the master fader) and "CHAN" buttons. This puts the device in "Native Mode".
2. In Samplitude/Sequoia go to "System Settings/Hardware Controller, click the „Add New“ button and select „Tascam US-2400 (Native)“ from the list.
3. Now set the MIDI ports on the right side to USB audio device.
4. Close the dialog. Your controller is now ready to use.

Notes

- The number of tracks to be jumped when a bank switch is pressed can be changed in the hardware controller setup options in the program.
- When using the TASCAM US-2400 as an exclusive recording panel, it's recommended that you relearn the SEL buttons as RecordReady.

Track range

Control element	Function	Shift function
KNOB 1~24	Pan, EQ, AUX (see table 2)	
SEL 1~24	Select track	Track automation
SOLO 1~24	Track solo	Record track
MUTE 1~24	Mute track	Monitor track (ASIO only)
FADER 1~24 (with fadertouch)	Track volume	

Control element	Function	Shift function
SEL (master)		Copy automation mode to all tracks
CLR SOLO	Global solo	Global record
FLIP	Global mute	Flip mode: Exchange functions of knobs and faders
MASTER FADER	Master volume	

Navigation

Control element	Function	Shift function
CHAN	**EQ mode** (see special modes)	
PAN	**Pan mode** (see special modes)	
AUX 1	KNOB 1~24 to AUX 1	Read automation
AUX 2	KNOB 1~24 to AUX 2	Touch automation
AUX 3	KNOB 1~24 to AUX 3	Latch automation
AUX 4	KNOB 1~24 to AUX 4	Overwrite automation
AUX 5	KNOB 1~24 to AUX 5	Trim automation
AUX 6	KNOB 1~24 to AUX 6	Automation off
METER	Activate level meter in knob display	
F-KEY	Active control mode (see special modes)	
NULL		
Shuttle wheel	Positioning	
Shuttle wheel + STOP	Select range	
SCRUB	Scrub mode	

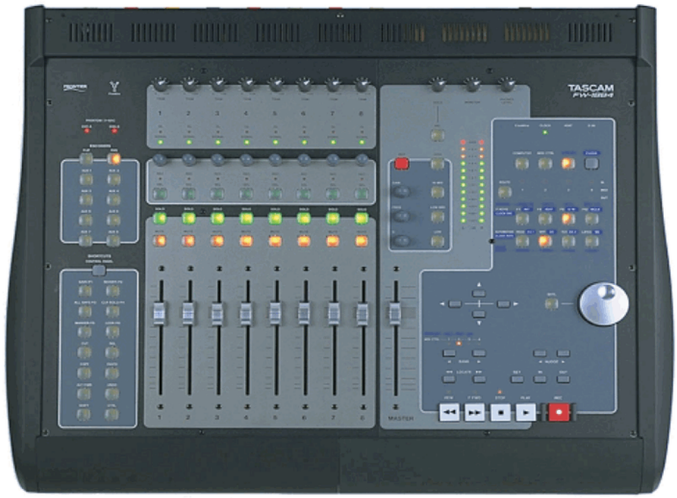
Control element	Function	Shift function
BANK	Mixer track 24 to the left	Mixer track 1 to the left
BANK+	Mixer track 24 to the right	Mixer track 1 to the right
IN	Loop Mode	
OUT	Metronome	
SHIFT	Shift	
REW	Fast rewind	Playback marker to project start
FFW	Fast forward	Playback marker to project end
STOP	Stop	Cancel
PLAY	Playback	Enter
REC	Recording	
Foot switch	Recording	

Special modes

Mode	Control element	Function
Pan mode	KNOB 1 ~ 24	Panorama
EQ mode (1 press)	KNOB 1	Level EQ band 1
	KNOB 2	Frequency EQ band 1
	KNOB 3	Quality EQ band 1
	KNOB 4	
	KNOB 5	Level EQ band 2
	KNOB 6	Frequency EQ band 2
	KNOB 7	Quality EQ band 2
	KNOB 8	
	KNOB 9	Level EQ band 3
	KNOB 10	Frequency EQ band 3
	KNOB 11	Quality EQ band 3
	KNOB 12	

Mode	Control element	Function
	KNOB 13	Level EQ band 4
	KNOB 14	Frequency EQ band 4
	KNOB 15	Quality EQ band 4
	KNOB 16	
EQ mode (2 press)	KNOB 1	Level EQ band 1
	KNOB 2	Level EQ band 2
	KNOB 3	Level EQ band 3
	KNOB 4	Level EQ band 4
	KNOB 5	Frequency EQ band 1
	KNOB 6	Frequency EQ band 2
	KNOB 7	Frequency EQ band 3
	KNOB 8	Frequency EQ band 4
	KNOB 9	Quality EQ band 1
	KNOB 10	Quality EQ band 2
	KNOB 11	Quality EQ band 3
	KNOB 12	Quality EQ band 4
ACTIVE CONTROL mode	FADER TRACK 1 (controls the active element in the mixer, track editor, or plug-in that must be selected with the mouse beforehand)	

Tascam FW-1884



1. Open the FW-1884 Control Panel (see TASCAM documentation) and set the "Control Protocol" to "Mackie Control Emulation".
2. In Samplitude/Sequoia go to "System Settings/Hardware Controller, click the „Add New“ button and select „Tascam FW-1884 (Mackie)“ from the list.
3. Now set the MIDI ports on the right side to "FW 1884 Control".
4. Close the dialog. Your controller is now ready to use.

Notes

The following elements do not send MIDI values and cannot be learned:

- PFL, HIGH, HI-MID, LOW-MID, LOW, GAIN, FREQ, Q, COMPUTER, MIDI CTRL, MON, CLOCK, ROUTE
- All elements with CTRL configuration can be mapped in the control panel without providing a modifier because the CTRL configuration already takes place in the controller and sends separate IDs.
- All elements with a CTRL configuration can also be set up with SHIFT-CTRL. In the control panel, set the modifier to "Shift" (since the CTRL configuration takes place in the controller already), next press CTRL + the element and learn.
- LED for AUX-elements is not functioning properly.
- FLIP LED is not functioning.
- The SoftLCD application that was installed with the FW-1884 drivers can be used to display track names and other typical Mackie Control information

Encoder

Control element	Function	Shift function
FLIP	Flip mode: KNOBS and FADERS swap functions	
PAN	Pan mode	
AUX 1~6	Aux select	
AUX 7, 8		

Shortcuts, control panel

Control element	Function	Modifier	Modifier function
SAVE/F1	Lock project		
REVERT/F2	Global mute	SHIFT	Global recording
ALLSAFE/F3			
CLR SOLO/F4	Global solo		
MARKER/F5	Metronome		
LOOP/F6	Loop Mode		
CUT	Split object	SHIFT	Split Objects with Alternate (Linear) Fade
DEL	Delete	SHIFT	Ripple delete
COPY	Copy	CTRL	Duplicating and moving
PASTE	Insert		
ALT/CMD			
UNDO	Undo	SHIFT	Repeat
SHIFT	Shift		
CTRL	Ctrl		

Track range

Control element	Function	Modifier	Modifier function
ENCODERS 1-8	Send pan/aux		
SEL 1-8	Select track	CTRL	Monitor track
SOLO 1-8	Track solo		
MUTE 1-8	Mute track		

Control element	Function	Modifier	Modifier function
FADER 1-8	Track volume		
FADER MASTER	Master volume		
REC	Arm track with SEL 1-8		

Navigation

Control element	Function	Modifier	Modifier function
F7	Automation OFF		
F8	Transfer automation mode to all tracks		
F9	Trim automation		
F10	Mixer		
READ	Read automation		
WRIT	Overwrite automation		
TCH	Touch automation		
LATCH	Latch automation		
SIITL	Scrub mode	SHIFT	Zoom mode
WHEEL	Jog/Scrub		
CURSOR UP	Previous track	SHIFT	Zoom project vert.(+)
CURSOR DOWN	Next track	SHIFT	Zoom project vert.(-)
CURSOR LEFT	Timeline positioning	SHIFT	Zoom project horiz.(-)
CURSOR RIGHT	Timeline positioning	SHIFT	Zoom project horiz.(+)
BANK LEFT	Previous bank	SHIFT	Mixer channel 1 left
BANK RIGHT	Next bank	SHIFT	Mixer channel 1 right
NUDGE LEFT	Move object left		
NUDGE RIGHT	Move object right		
LOCATE LEFT	Previous marker	SHIFT	Previous Object
LOCATE RIGHT	Next marker	SHIFT	Next object
SET	Create marker (auto #)	SHIFT	Delete marker
IN	Set punch in marker	SHIFT	Delete punch marker
OUT	Set punch-out marker		

Control element	Function	Modifier	Modifier function
REW	Fast rewind	SHIFT	Playback marker to project start (Return to Zero)
FFW	Fast forward	SHIFT	Playback marker to project end (Go to End)
STOP	Stop	SHIFT	Cancel
PLAY	Playback (second press: stop at position)	SHIFT	Enter
REC	Recording	CTRL	Punch mode on/off

SSL Nucleus

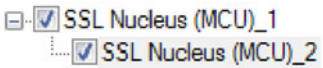


Set up SSL Nucleus

1. In the "Nucleus Remote" configuration tool of the SSL Nucleus set the "Samplitude Default" profile.
   - If this profile is not displayed, you can add it through "Edit Profiles" > "Restore". The file can be found in the Samplitude/Sequoia program folder under "Customize\SSL Nucleus\".
2. The "SSL Nucleus" is connected with the computer through ethernet:
   - SSL provides a help program for this („ipMIDI Ethernet Midi Ports“) that supplies the operating system with virtual MIDI ports.
   - Configure this program so that there are at least 2 ports available.
   - You should also deactivate the "Loop Back" option.
   - Up to three DAWs can be controlled with the Nucleus. For this the virtual MIDI ports are assigned as follows:
     DAW 1: ipMIDI ports 1 and 2

DAW 2: ipMIDI ports 3 and 4
DAW 3: ipMIDI ports 5 and 6

3. Go to „System Settings/Hardware Controller“ in Sequoia and click on „Add New“ to select „SSL Nucleus (MCU)_1“ from the list.
4. You can also add „SSL Nucleus (MCU)_2“ in exactly the same way.
5. Position „SSL Nucleus (MCU)_2“ indented under „SSL Nucleus (MCU)_1“.



6. Now set MIDI ports for both entries:
   - „SSL Nucleus (MCU)_1": „1. Ethernet MIDI"
   - „SSL Nucleus (MCU)_2": „2. Ethernet MIDI" (if Samplitude/Sequoia is DAW 1).
7. Close the dialog. Your controller is now ready to use.

Various Modes

Control element	Function
INSTRUMENT	
Plug-in	Plug-in Mode (see special modes (view page 591))
EQ	1. Press: EQ Mode 1 (Band) 2. Press: EQ Mode 2 (Type) (see special modes (view page 591))
PAN	
SEND	1. Press: AUX mode (phat channel) 2. Press: AUX Mode (Track Mode) (see special modes (view page 591))
TRACK	1. Press: Track mode 2. Press: Active control mode (see special modes (view page 591))

Control element	Function
USER1	User1 Mode (view page 590)
USER2	
PARAM	Mixer Window
EQ	
REC	Record Mode (view page 589)
FLIP	Flip Mode: Knobs and faders exchange their functions.
DYN	
AUTO	Automation mode (view page 590)

General Control Elements

Control element	Function	Modifier	Modifier function
PARAM	Mixer Window		
< BANK	Bank to the left		
BANK >	Bank to the right		
< CHANNEL	One channel to the left		
CHANNEL >	One channel to the right		
FLIP	Flip mode: Exchange functions of knobs and faders		
SHIFT	Modifier Shift		
OPT	Modifier Option		
CNTL	Modifier Control		
ALT			
DAW 2	Control of second DAW (Assignment in Nucleus Remote)		
Enter	Enter		
SAVE	Save		
F1			
DAW 3	Control of third DAW (Assignment in Nucleus Remote)		

Control element	Function	Modifier	Modifier function
Esc	Cancel		
UNDO	Undo	SHIFT	Redo
F2			

Record Mode

Control element	Function	Modifier	Modifier function
SEL 1 ~ 16	Activate track	SHIFT	Track monitoring

Tracks

Control element	Function	Modifier	Modifier function
KNOB 1~16	Pan; Eq; Aux; Plug-in parameters (see Special Modes (view page 591))		
CUT 1~16	Mute track		
SOLO 1~16	Track solo		
SEL 1~16	Select track		
FADER 1~16 (with fader touch)	Track volume	SHIFT	Touch fader: Reset to 0 dB

Navigation

Control element	Function	Function in Zoom-Mode
MODE	Zoom mode	
Arrow up	Select previous track	Zoom in arranger view vertically
Arrow down	Select previous track	Zoom out arranger view vertically
Arrow left	Move arranger view forward on the timeline	Zoom in arranger view horizontally (along the timeline)
Arrow right	Move arranger view backward on the timeline	Zoom out arranger view horizontally (along the timeline)

Transport

Control element	Function	Modifier	Modifier function
PREVIOUS	Playback marker to previous object	SHIFT	To project start
NEXT	Playback marker to next object	SHIFT	To project end
CYCLE	Loop Mode		
REWIND <<	Fast rewind		
FFWD >>	Fast forward		
STOP	Stop		
PLAY	Playback		
RECORD	Recording		
Shuttle wheel	Positioning	SCRUB	Scrubbing
Wheel Button	Scrub mode		

Automation

Control element	Function	Modifier	Modifier function
Rd/Off	Read automation	SHIFT	Automation off
Write	Overwrite automation	SHIFT	Copies automation mode to all tracks
Touch	Touch automation		
Latch	Latch automation		

User1 Mode

Control element	Function	Modifier	Modifier function
Mixer	Mixer window		
Transp	Transport window		
Markers	Previous/Next marker with FFWD/REWIND		
Nudge	Previous/Next object with FFWD/REWIND		
SMPTE	Timecode switch		
Click	Metronome on/off		

Control element	Function	Modifier	Modifier function
1	Set/Jump to Marker 1	SHIFT CNTL	Move Marker 1 Delete Marker 1
2	Set/Jump to Marker 2	SHIFT CNTL	Move Marker 2 Delete Marker 2
3	Set/Jump to Marker 3	SHIFT CNTL	Move Marker 3 Delete Marker 3
4	Set/Jump to Marker 4	SHIFT CNTL	Move Marker 4 Delete Marker 4
5	Set/Jump to Marker 5	SHIFT CNTL	Move Marker 5 Delete Marker 5
6	Set/Jump to Marker 6	SHIFT CNTL	Move Marker 6 Delete Marker 6

Special modes

Mode	Control element	Turn	Press
Plug-in Mode Plug-in Parameter	KNOB 1~6	–	Select plug-in
	KNOB 1~16	Change parameters	–
	< BANK	Previous parameters	–
	BANK >	Next parameters	–
EQ Mode 1 Banks <,> are not available in this mode	KNOB 1	Level EQ band 1	–
	KNOB 2	Frequency EQ band 1	–
	KNOB 3	Quality EQ band 1	–
	KNOB 4	Level EQ band 2	–
	KNOB 5	Frequency EQ band 2	–
	KNOB 6	Quality EQ band 2	–
	KNOB 7	Level EQ band 3	–
	KNOB 8	Frequency EQ band 3	–
	KNOB 9	Quality EQ band 3	–
	KNOB 10	Level EQ band 4	–

Mode	Control element	Turn	Press
	KNOB 11	Frequency EQ band 4	–
	KNOB 12	Quality EQ band 4	–
EQ Mode 2 Banks <,> are not available in this mode	KNOB 1	Level EQ band 1	–
	KNOB 2	Level EQ band 2	–
	KNOB 3	Level EQ band 3	–
	KNOB 4	Level EQ band 4	–
	KNOB 5	Frequency EQ band 1	–
	KNOB 6	Frequency EQ band 2	–
	KNOB 7	Frequency EQ band 3	–
	KNOB 8	Frequency EQ band 4	–
	KNOB 9	Quality EQ band 1	–
	KNOB 10	Quality EQ band 2	–
	KNOB 11	Quality EQ band 3	–
	KNOB 12	Quality EQ band 4	–
AUX Mode (Phat Channel) depending on active track	KNOB 1	AUX 1 Level	–
	KNOB 2	AUX 2 Level	–
	KNOB 3	AUX 3 Level	–
	KNOB 4	AUX 4 Level	–
	KNOB 5	AUX 5 Level	–
	KNOB 6	AUX 6 Level	–
	KNOB 7	–	–
	KNOB 8	–	–

Mode	Control element	Turn	Press
AUX Mode (Track Mode) For example, on all knobs there is AUX 1, selectable through Ban k< and/or Bank >	KNOB 1~16	AUX level	–
Track mode	KNOB	Pan	Reset
ACTIVE CONTROL MODE	FADER 1 (controls the active element in the mixer, track editor, or plug-in that was selected with the mouse beforehand)	–	–
Marker mode	REWIND <<	–	Previous marker
	FFWD >>	–	Next marker
Object mode	REWIND <<	–	Previous Object
	FFWD >>	–	Next object

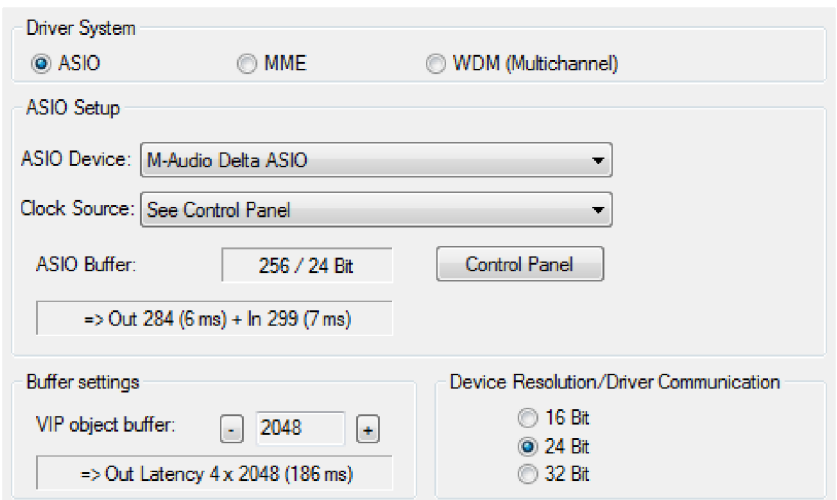
System settings

You can configure settings for your system and sound card in the system settings. Press the "Y" key on your keyboard or open the global system settings via "File" > "Program preferences" > "System/Options" (view page 670).

This allows you to access information about any connected playback and recording devices as well as MIDI, metronome, or program settings. You can also select the target folders for projects and VST plug-ins and the view options and coloring of screen elements. The most important dialogs are explained briefly in the following.

Audio Settings

Enter all essential settings for the driver system, buffers, driver communications, and monitoring here.



Driver system: A so-called "driver system" is used for the necessary communication between Sequoia and your sound card. In order to take full advantage of the program, we recommend that you use ASIO.

MME is the standard Windows multimedia driver system with the best compatibility. It also supports 16-bit playback. If you are recording 24/32-bit audio material, you can use MME/WDM. This driver system is suitable for multitrack recordings of up to 64 tracks that don't place severe requirements on the monitoring processes. For performance critical recordings, this provides greater security compared to ASIO drivers but with many sound cards, multitrack recordings will not be synchronous.

ASIO: Use a sound card model fitted with ASIO drivers if possible. This offers a number of decisive advantages over the MME/WDM driver system:

- Lower latency (input/output delay) in terms of the driver system. Resulting response times during real-time editing are clearly reduced. This makes it possible to use software monitoring for inputs and VST instruments.
- ASIO is intended for editing multitrack recordings featuring several sound cards that are all using the same ASIO drivers. The sound cards are synchronized by the ASIO driver.
- Advanced hardware monitoring options are also available by using ASIO direct monitoring.

ASIO settings

ASIO device: 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. Sequoia also displays the relevant output and input latencies.

Buffer settings: The VIP object buffer indicates the buffer size for internal processing of object effects and economy tracks in hybrid mode. In every other monitoring mode, the buffer size also determines the editing of track effects. With a smaller VIP object buffer size, the amount of playback delay also decreases (latency). In these cases, the processor may become overloaded, which will result in dropouts during playback. Large buffer sizes actually increase stability, but they also increase the latency of the system. The field below displays the resulting latency in relation to the buffer settings.

Tip: The VIP object buffer size should normally be bigger than the ASIO buffer size and be set between 1024 and 8096 samples.

Device Resolution/Driver Communication: This option enables you to select the bit resolution for communication with audio devices. The preset value correlates with that of the sound card installed on your system.

The ASIO driver system specifies the bit resolution of the ASIO drivers. Sequoia always assumes the bit resolution set for the ASIO drivers. The settings options displayed for device resolution/driver communication only specifies the bit depth applied by dithering (view page 703). For 32-bit data transfer, dithering is 24-bit or 16-bit. The settings for 32-bit are recommended especially for Sequoia

- if no dithering is intended.
- if the input is supplied directly by a DSP card instead of from the converter unit of the sound card.

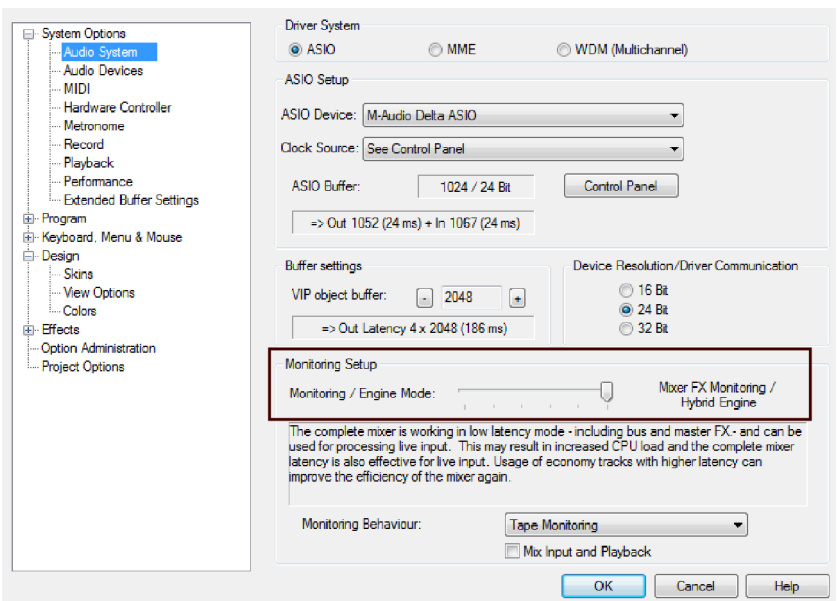
If MME driver system is selected, Sequoia opens the drivers with the bit depth set under Device resolution/Driver communication. If the output device cannot process the set bit resolution, the corresponding lower resolution will be produced and this will be transmitted to the MME driver.

To specify the plug-in buffer and the path to your VST plug-ins, switch to the dialog "System options (Y) > Effects > VST/ReWire (view page 390)".

Monitoring settings

Monitoring is essentially the process of making the input signals audible, e.g. so musicians who are playing on the recording can hear what they are doing. The routing of monitoring signals is an important and complex application that has various requirements.

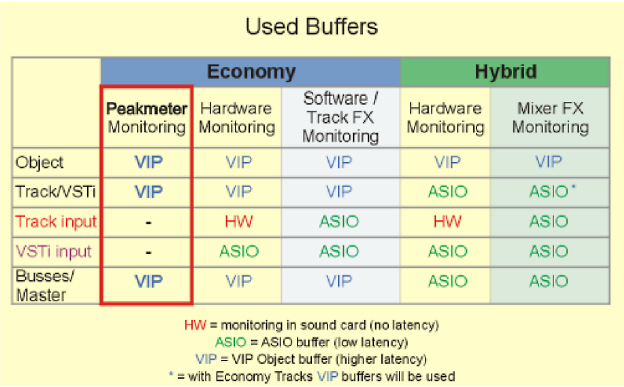
The dialog can be found in the System Options (keyboard shortcut: Y) under "Audio Setup > Monitoring Setup".



For each selected monitoring setting a matrix display also opens which shows which buffer sizes are used and what effect the currently selected engine mode has on the latencies.

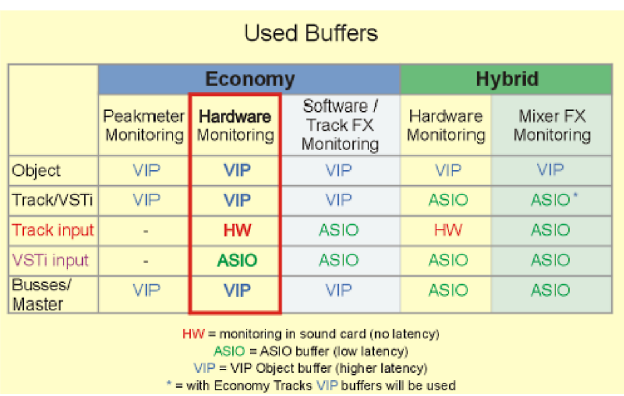
Overview of monitoring and engine modes

No Audio Monitoring (peak meter only): Displays the input level, but cannot be heard. This monitoring option is useful if you are using an external mixer for audio monitoring.



With this matrix display of Peakmeter Monitoring, input signals on tracks or VST instruments cannot be heard in Sequoia. Already existing objects, (VSTi) tracks, as well as all busses and the stereo master output (Master) will still be played back through the monitoring by means of a VIP object buffer with high latency.

Hardware Monitoring: Audio monitoring through the sound card. This is the only possible monitoring type for MME drivers; For ASIO, many sound cards are able to process functions such as mute/solo, volume, and pan directly. For large ASIO or VIP buffers, minimal latency can be achieved with just a few samples. Effects cannot be applied to the input signal.



With this matrix display for hardware monitoring, the input signals on tracks are played back directly through the sound card and not in Sequoia which results in almost latency-free monitoring. VSTi input signals are played back within Sequoia with low latency by means of an ASIO buffer. Already existing objects, (VSTi) tracks as well as all busses and the stereo master output (Master) can be monitored by means of a VIP object buffer with high latency.

Software Monitoring/Economy Engine: Audio monitoring that takes into account the recording track levels and the playing of software instruments. This monitoring option is only available when using ASIO drivers. No effects are applied to the input signals which means that latency and CPU load remain the same even in complex projects.

Used Buffers

	Economy			Hybrid	
	Peakmeter Monitoring	Hardware Monitoring	Software / Track FX Monitoring	Hardware Monitoring	Mixer FX Monitoring
Object	VIP	VIP	VIP	VIP	VIP
Track/VSTi	VIP	VIP	VIP	ASIO	ASIO *
Track input	-	HW	ASIO	HW	ASIO
VSTi input	-	ASIO	ASIO	ASIO	ASIO
Busses/ Master	VIP	VIP	VIP	ASIO	ASIO

HW = monitoring in sound card (no latency)
ASIO = ASIO buffer (low latency)
VIP = VIP Object buffer (higher latency)
* = with Economy Tracks VIP buffers will be used

With this matrix display of Peakmeter Monitoring, input signals on tracks or VST instruments can be played back in Sequoia with low latency by means of an ASIO buffer. Already existing objects, (VSTi) tracks as well as all busses and the stereo master output (Master) can be monitored by means of a VIP object buffer with high latency.

Track FX Monitoring: Audio monitoring including the track effects of the recording track. This monitoring option is only available when using ASIO drivers. Effects applied directly to the recording track are included in the monitoring signal, but bus and master effects are not.

Used Buffers

	Economy			Hybrid	
	Peakmeter Monitoring	Hardware Monitoring	Software / **Track FX** Monitoring	Hardware Monitoring	Mixer FX Monitoring
Object	VIP	VIP	**VIP**	VIP	VIP
Track/VSTi	VIP	VIP	**VIP**	ASIO	ASIO *
Track input	-	HW	**ASIO**	HW	ASIO
VSTi input	-	ASIO	**ASIO**	ASIO	ASIO
Busses/ Master	VIP	VIP	**VIP**	ASIO	ASIO

HW = monitoring in sound card (no latency)
ASIO = ASIO buffer (low latency)
VIP = VIP Object buffer (higher latency)
* = with Economy Tracks VIP buffers will be used

With this matrix display of Track FX Monitoring, input signals on tracks or VST instruments can be played back in Sequoia with low latency by means of an ASIO buffer. Already existing objects, (VSTi) tracks as well as all buses and the stereo master output (Master) can be monitored by means of a VIP object buffer with high latency.

Hardware Monitoring/Hybrid Engine: In this case, the complete mixer works in low latency mode. This keeps playback latency to a minimum. The input signals are monitored via the sound card in use.

Used Buffers

	Economy			Hybrid	
	Peakmeter Monitoring	Hardware Monitoring	Software / Track FX Monitoring	**Hardware** Monitoring	Mixer FX Monitoring
Object	VIP	VIP	VIP	**VIP**	VIP
Track/VSTi	VIP	VIP	VIP	**ASIO**	ASIO *
Track input	-	HW	ASIO	**HW**	ASIO
VSTi input	-	ASIO	ASIO	**ASIO**	ASIO
Busses/ Master	VIP	VIP	VIP	**ASIO**	ASIO

HW = monitoring in sound card (no latency)
ASIO = ASIO buffer (low latency)
VIP = VIP Object buffer (higher latency)
* = with Economy Tracks VIP buffers will be used

With this matrix display for Hardware Monitoring through the Hybrid Engine, the input signals on tracks are played back directly through the sound card and not in Sequoia which results in almost latency-free monitoring. VSTi input signals are played back within Sequoia with low latency by means of an ASIO buffer. (VSTi) tracks as well as all busdes and the stereo master output (Master) can be monitored by means of an ASIO buffer with low latency. Already existing objects will still be played back through the monitoring by means of a VIP object buffer with high latency.

Mixer FX monitoring/Hybrid Engine: The Hybrid Audio Engine enables audio monitoring throughout the entire mixer and, in doing so, also calculates playback tracks in the mixer with short latency. This way you can also mix data from the hard drive with the lowest possible playback delay. We recommend this mode for input signals as well as for live mixing with hardware controllers because access to all bus and master effects is guaranteed.

Used Buffers

	Economy			Hybrid	
	Peakmeter Monitoring	Hardware Monitoring	Software / Track FX Monitoring	Hardware Monitoring	**Mixer FX** Monitoring
Object	VIP	VIP	VIP	VIP	**VIP**
Track/VSTi	VIP	VIP	VIP	ASIO	**ASIO** *
Track input	-	HW	ASIO	HW	**ASIO**
VSTi input	-	ASIO	ASIO	ASIO	**ASIO**
Busses/ Master	VIP	VIP	VIP	ASIO	**ASIO**

HW = monitoring in sound card (no latency)
ASIO = ASIO buffer (low latency)
VIP = VIP Object buffer (higher latency)
* = with Economy Tracks VIP buffers will be used

With this matrix display of Track FX Monitoring through the Hybrid Engine, input signals on tracks or VST instruments can be played back in Sequoia with low latency by means of an ASIO buffer. (VSTi) tracks as well as all busses and the stereo master output (Master) can be monitored by means of an ASIO buffer with low latency. Already existing objects will still be played back through the monitoring by means of a VIP object buffer with high latency.

The following applies to all monitoring modes: The set track effects on the recording track are not included in the recording which means that the track actually recorded does not have any of the effects that have been applied to the track. Depending on the monitoring mode used during the recording process – and during playback of the recording, of course – you will hear the recorded signal with the effects applied to it. For example, it is not possible to record the vocal signal with the Sequoia reverb VariVerb II directly. However, you can save the recorded track afterwards along with all track effects using the "Track Bouncing (view page 643)" function.

Tip: In most cases we recommend using the "Mixer FX Monitoring/Hybrid Engine". This monitoring mode enables audio monitoring throughout the entire mixer and also calculates playback tracks in the mixer with short latency. This also gives you access to all the bus and master effects.

If the CPU load becomes too high when using "Mixer FX Monitoring/Hybrid Engine", switch to the Monitoring/Engine Mode that best meets your requirements and can be handled by your system.

For example, if you want to do without effect monitoring of input signals, "Software Monitoring/Economy Engine" might be the right mode for you.

If you want to include the monitoring of track effects of the input signals and to do without the bus and master effects in the monitoring path, we recommend the "Track FX Monitoring" mode. If you prefer to monitor to the track input signals through the sound card, you can choose either "Hardware Monitoring" or "Hardware Monitoring/Hybrid Engine" depending on the performance capability of your system.

Note: If you use the "TotalMix" software from RME in combination with a hardware monitoring mode, set the panning law in "TotalMix" to -6 dB. This will ensure that the recording level in Sequoia corresponds to the monitoring level in "TotalMix".

Hybrid Audio Engine



In general, "Hybrid" refers to a system in which two separate technologies are combined with one another. Sequoia's Hybrid Audio Engine is a special combination of a Low Latency Engine and the classic Sequoia Playback Engine with higher latency. The Low Latency Engine calculates live input signals and the output of the playback engine. This provides short reaction times for calculating track effects and also provides monitoring with low latency. The classic playback engine saves resources and enables the integration of high-performance effects.

The Hybrid Engine provides a combination of Low Latency Engine and a classic engine for calculating the track and effects. For example, processor-intensive VST instruments can be played on the so-called "Economy" tracks (see below), while only the VST instrument that is being recorded is calculated by the Low Latency Engine.

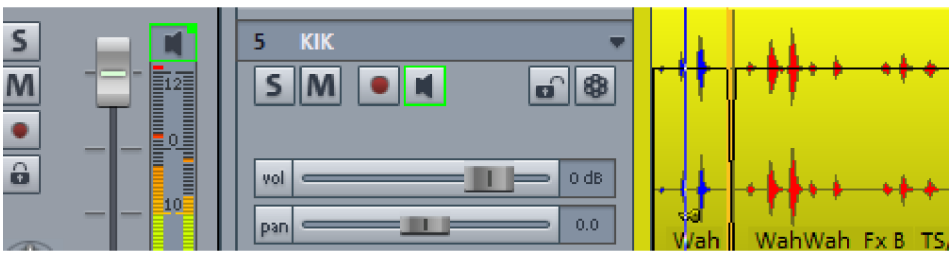
Go to System Options (shortcut "Y") ->"Audio System" -> "Driver System" and select "ASIO". "Monitoring Setup" features the mode "Mixer FX monitoring/Hybrid Engine".

Mixer FX monitoring provides audio monitoring including all insert and AUX effects and also calculates the playback tracks with the least amount of latency. This means that the entire mixer can operate in "Low latency" mode, including the bus and master effects, and it can also be used for editing input signals.

Economy Track

If you have selected the ASIO Hybrid Engine as the driver system, you can take individual tracks from the Low Latency Engine and calculate the track effects with a larger VIP buffer size for VIP objects during playback. This prevents the system processor from being overloaded but can lead to an increased delay time during playback.

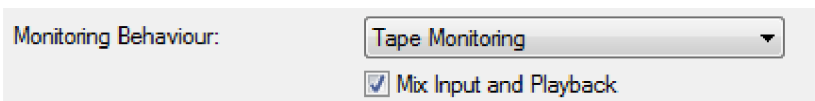
To switch a track to "Economy", select the option "Track -> Track type -> Economy Track".



The "Volume" button for the economy track is framed in green.

Note: The Hybrid Engine compensates for latency of the track effects in economy tracks so that the total latency of the mixer does not increase for other tracks.

Monitoring switch behavior



Tape Monitoring is the preset monitoring behavior and the monitoring behavior familiar from earlier Sequoia versions which works like a tape device: In stop status the input signal is played back; in play status the track content is played back. During a recording, the input signal is played back, except for punch-in recordings; here, outside the punch marker, the track content is output and inside the punch marker, the input signal is output.

Manual Monitoring: For each track you can manually specify whether the monitoring is active for the track using the loudspeaker button in the track header, track editor or mixer. This switch behavior is only available when the ASIO driver system is used.

Mix Input and Playback: If you set the check here, the input signal can also be heard when playback is running if track monitoring is active.

The monitoring settings and switch behavior can be set in "System Options > Audio System" and can also be accessed directly by right-clicking on the "**Mon**" button. This displays two more MIDI recording options:

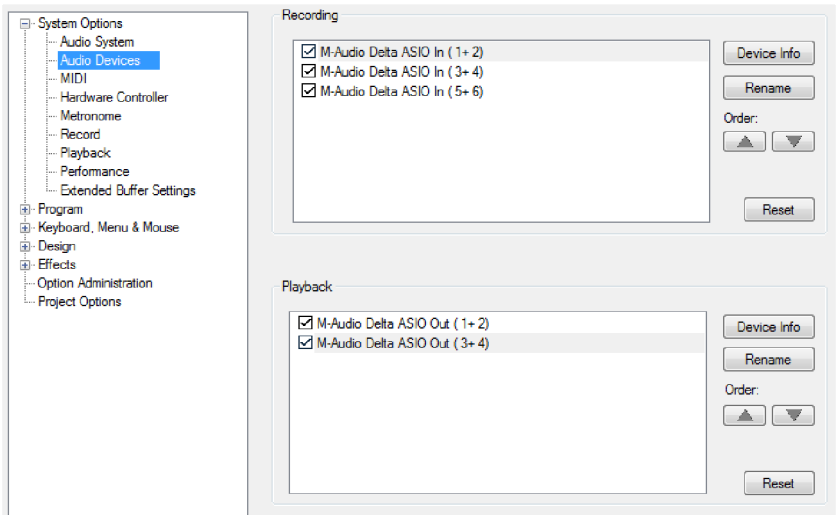


Automatic MIDI record switch on current track: This option ensures that MIDI tracks are always ready for recording as soon as they are selected. This is indicated by the pink recording buttons.

Automatic MIDI monitoring (thru): If this option is selected, every MIDI track that you activate for recording will be switched on automatically, e.g. if you are playing a MIDI keyboard that is being recorded on this track, you will always immediately hear the output signal of the software instrument for this track.

Audio Devices

This dialog is where you select the inputs and outputs for the installed sound cards that will be used in Sequoia.

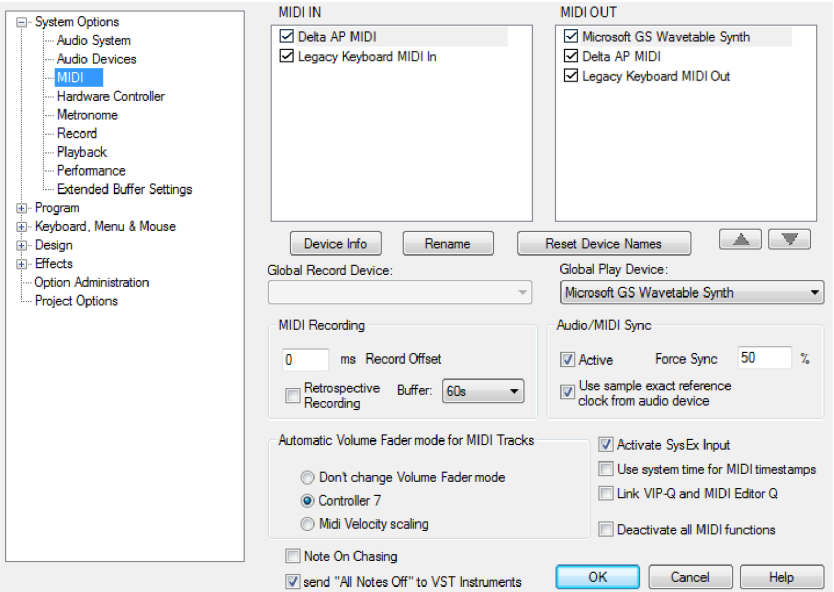


You can activate or deactivate a device by adding or removing a check mark in the box next to it. You can set the sequence with the arrow keys. The "Reset" button activates all of the devices displayed. With ASIO, only the first 4 stereo channels are activated in the default setting. However, if you hold down the Shift key, all of the inputs and outputs will be activated. Clicking on the "Info" button opens the control panel for your sound card. With the "Rename" button you can customize each device name individually.

MIDI Settings

Tip: For smooth playback of VST instruments, please always use the monitoring settings "Track FX monitoring", "Hardware monitoring/Hybrid Engine", or "Mixer FX monitoring/Hybrid Engine".

Global MIDI Devices



Open the System Options – MIDI through "File" > "Program Preferences > MIDI Options" and set the standard input and standard output. The set devices are used to play back MIDI files directly, and they are also used as the standard settings for new tracks. You can rename the devices at any time.

Record offset

Enter a recording offset value in milliseconds to determine the time difference which passes between a MIDI command and placement in the arrangement.

Sequoia places played notes in the MIDI object on the right according to the set time, i.e. MIDI notes are delayed and displayed later in the timeline.

Retrospective MIDI recording

Activate the "Retrospective recording" option here for the same function available under "Play/Rec -> Retrospective MIDI recording (view page 849)". Sequoia creates a MIDI object on the MIDI track that can be adjusted in terms of buffer length.

Audio/MIDI synchronization

Force sync specifies how precisely Sequoia synchronizes MIDI tracks with audio tracks. On faster systems, this setting should be at 100% in order to achieve the closest sync between the MIDI and the audio. If your system experiences difficulties synchronizing the MIDI tracks with the audio tracks, select a lower sync reciprocation

value. Normally you will also want to use sample-exact reference times from your audio device (e.g. sound card) for audio/MIDI synchronization.

Automatic volume fader mode for MIDI tracks

The behavior of the MIDI track's volume fader is preset to controller 7 (MIDI volume).

Alternatively, you can also select that "Volume fader" mode should not change for MIDI tracks or that the volume fader for MIDI tracks should match the MIDI velocity scaling. New MIDI tracks will assume the currently set Volume fader mode (view page 366).

Activate SysEx input

Selecting this option causes Sequoia to receive SysEx data from external devices.

Use system time for MIDI timestamps

If this option is activated, the MIDI device driver's timestamp will be ignored. This is helpful if drivers provide a timestamp that is not synchronized with the audio or is completely incorrect.

This function detects invalid driver timestamps and reverts back to the system time automatically. This helps fix MIDI recording problems with MIDI devices.

Link VIP Q and MIDI editor Q

This option links the grid or quantization settings between the VIP and MIDI editor, i.e. changing the grid/quantization settings in the VIP will automatically be assumed for the grid/quantization settings in the MIDI editor and vice versa.

Deactivate all MIDI functions

You can shut off all the MIDI functions in MIDI or VIP projects. If you do this, the MIDI functions in the arranger, track editor and track settings dialog will no longer be available.

Note On Chasing

"Note On Chasing" ensures that held MIDI notes are played back even if they are positioned before the current playback start position.

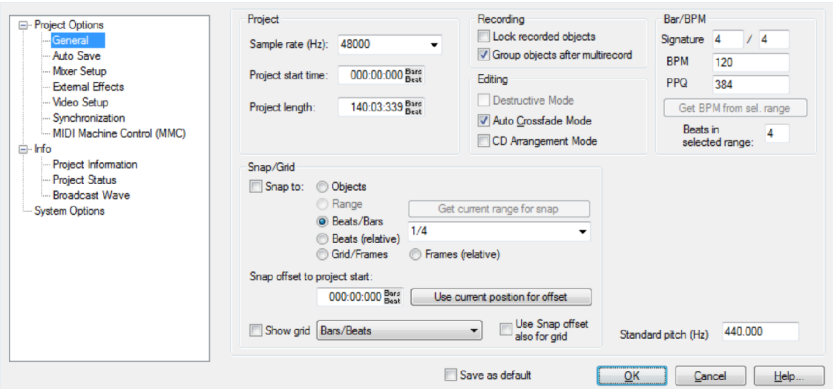
Send "All Notes Off" to VST instruments

As an alternative to the "All Notes Off" mechanism, an individual "Note Off" mechanism is available for VST instruments, since "All Notes Off" is ignored by some VST instruments. You can completely shut off "All Notes Off for VSTis in the MIDI settings.

Project Options - General

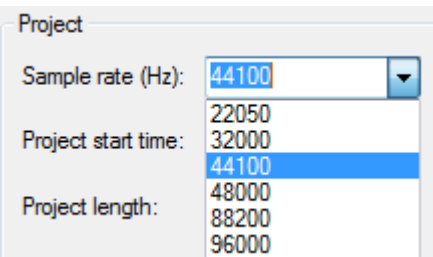
Shortcut: I

Here you can see the most important information about the current project at a glance. This includes Bar/BPM, Recording, Editing, and Snap/Grid settings.

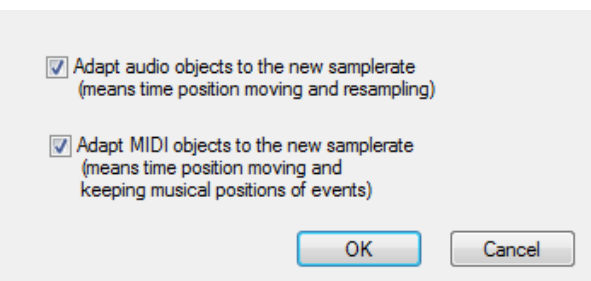


Project Options - General

Sample rate (Hz): This indicates the project's sample rate. You can also change the sample rate for the project



and adjust audio and MIDI objects to the modified sample rate.



Project start time: Specify the project's starting time here.

Project length: The project length is displayed here in bars and beats.

General project options – recording/editing

Here you'll find various presets for recording and editing:

Lock recorded objects: Protects against unintentional moving of recorded objects.

Group objects after multirecord: Objects that belong together will be grouped in a multitrack recording.

Destructive editing mode (only for audio files): Detailed information about destructive offline audio editing can be found in the chapter "Basic Terms > Audio Editing in Sequoia" (view page 62).

Auto Crossfade Mode: Use this function to activate a mode which adds a crossfade to all newly recorded objects, objects created from cuts or audio files that you imported to your project. Every object can be assigned a standard fade-in which can be edited in the "Object editor fade" menu under "Get/set global crossfade". If two objects overlap in this mode, a realtime crossfade will occur at the intersection.

CD Arrangement Mode: If this menu item is activated, Sequoia arranges newly added objects to insert a "Red Book Standard" pause between the objects. "Red Book" is a standard for audio CDs that sets the technical specifications for all CD and CD ROM formats.

Project options - Bar/BPM

Bar/BPM/PPQ: Enter the time signature (counter/signature), the tempo in beats per minute (BPM), and the timer resolution in peaks per quarter (PPQ/clicks per quarter note) here.

Get BPM from selected range: If you enter the number of beats into the field beside "Selected range has beats:", Sequoia will calculate the BPM based on the selected range when the button is pressed.

General Project Options – Snap / Grid

Snap to: Switches the global snap on/off.

Objects: This option activates the object snap. This lets objects snap exactly to the edges of other objects.

Range: Activates the range snap and enables the current range to be used as the basis for snapping.

Beats/Bars: Activates a grid with bars as the basis for snapping.

Beats (relative): Also this option activates a grid which uses beat intervals as its basis. For example, here you can move a selected object while maintaining its relative distance to the closest marker according to the set bar grid.

Grid/Frames: This option activates a frame-based grid.

Frames (relative): This option also activates a frame-based grid where the object maintains its relative distance to the closest grid marker when moved.

Snap offset to project start: Sets the snap offset relative to the beginning of the project. "Use current position for offset" specifies the current position of the playback marker as the grid's zero position.

Show grid: If a check is placed here, the grid will be displayed for the project according to the snap unit set in the selection box beside it.

Use snap offset also for grid: The snap offset is used as a reference size for the grid.

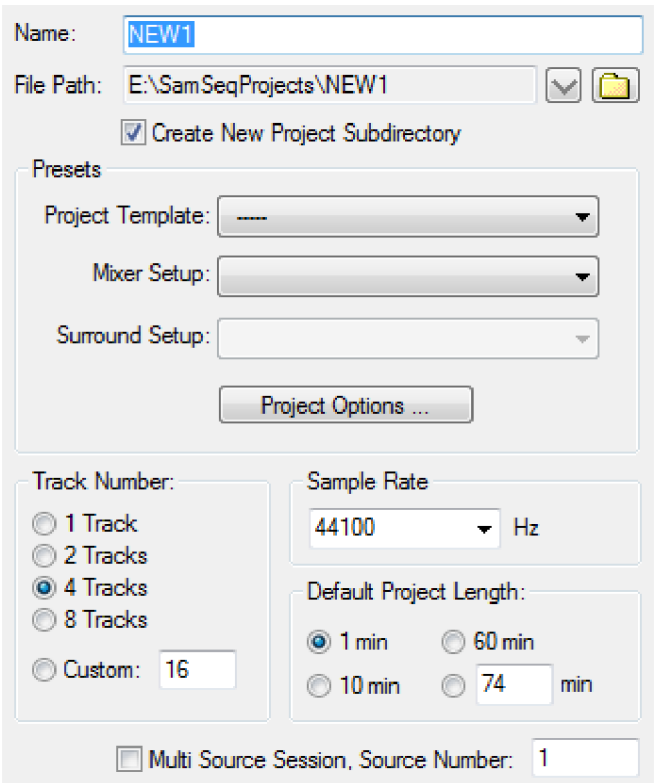
General Project Options - Standard pitch for tuner

This field indicates a standard pitch of A at 440 Hz. You can change this setting if you want to use an alternative tuning for the Sequoia internal tuner (view page 1143).

File Menu

This allows you to apply comprehensive settings for creating, managing, loading, and saving projects.

New Virtual Project (VIP)



Shortcut: E

A new virtual project is created in the arranger with the specified number of tracks:

Name: Enter the name for the new project.

File path: Select the folder for saving the project.

Create new project subdirectory: A subdirectory featuring the name of the project is automatically created in the directory selected in "File path".

Project template: Choose between previously selected project templates in this list field. These presets contain project settings such as track number, device assignments, etc. Create project templates via "File -> Save project as template (view page 621)".

Mixer setup: Select a preset mixer setup here. Mixer setups define the number and kind of tracks/buses as well as the routing of inputs/outputs and effects.

To create a Surround project, define the Surround setup first. The list field located below, i.e. "**Surround setup**", is activated for selecting a number of Surround formats.

Project options: All settings adjustments made here are valid for every new virtual project. They will be saved in a separate VIP (templates\template.VIP). The template.VIP file may also be edited directly to define additional settings for all projects:

- Record arming setting of the first track
- Some mixer setup settings
- Playback mode
- Effects sequence in tracks and mixer
- Grid offsets

Please also read "System settings -> Project options -> General (view page 607)" for more details about project options.

Track number: The start track count of the virtual project is defined here. Use "Track -> Insert new tracks... (view page 759)" to add more tracks at any time.

Sample rate: Defines the sample rate of the virtual project.

Preset project length: The initial project length may be set as a preset here. This setting will be automatically adjusted when loading or recording and changes during zoom operations.

Note: VIP projects may only be up to 168 hours in length.

There are no more length restrictions on WAV files with NTSF. Recordings are written to a large RIFF64 file. If under 2 GB, it will be compatible with normal wave files. Using FAT 32, files will be a split into W01, W02, etc.

Multi-source session, number of sources: Enter the number of desired source projects for a multi-source session. A destination VIP and the desired number of source VIPs will be created by Sequoia.

Open

You can open various file types here and load them into Sequoia. To load audio files, use the command "Import -> Load Audio File".

Virtual Project (*.VIP)

Use this menu command to load a multitrack project as an arrangement.

Shortcut: O

RAM Wave (*.RAP)

RAM Wave Projects contain audio data in an audio format unique to Sequoia that is loaded into the RAM memory. Additional information for display of the audio data, marker positions, etc. is contained by this data.

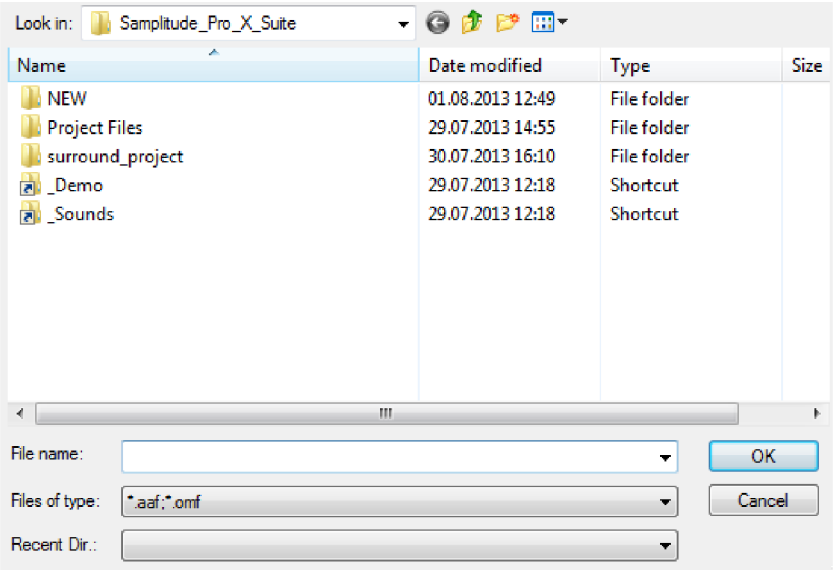
HD Wave (*.HDP)

HD Wave Projects contain audio files that are loaded directly from the hard disk. Additional information for display of the audio data, marker positions, etc. is contained in them. The files use WAV audio format.

Shortcut: Shift + L

Note: If a VIP is active, the loaded wave projects are loaded immediately into the VIP as objects. The range of the active track displays the position. "CD arrangement" mode is the exception. In this case, the objects are displayed below one another independent of the range. The distance between the objects corresponds to the pause time (CD/DVD menu -> Set start pause time (view page 1117)).

Import AAF/OMF



The**AAF import** transfers the following content:

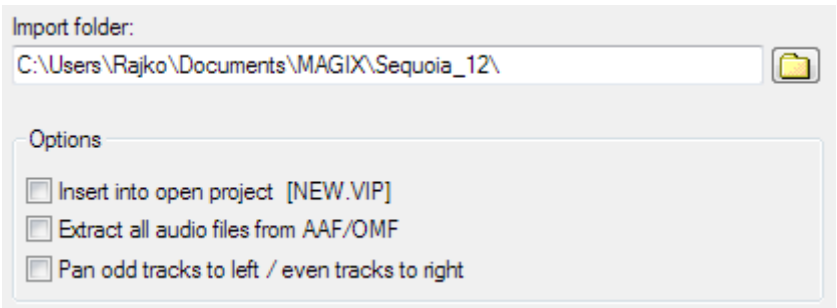
- Object position/wave offset - sample exact
- Object fade in/out - linear only
- Object crossfade - linear only
- Object volume
- Object pan
- Object volume automation
- Object pan automation
- Track names
- Track Volume
- Track Pan
- Track Volume Automation
- Track Pan Automation
- VIP markers
- Timecode offset
- Timecode format (24, 25, 30 fps)
- Processing timestamps in audio files

The**OMF import** transfers the following content:

- Object position/wave offset - sample exact
- Object fade in/out - linear only
- Object crossfade - linear only
- Object volume
- Track names

Note: When importing AAF/OMF files video files will also be imported. These can only be played when the program 'understands' the format and when the corresponding codecs are available on the system.

Find the AAF/OMF file you want to import using the import dialog. In a further dialog you can choose the track panorama when importing, so that the uneven track numbers in the panorama are placed on the left and the even on the right.



Object (*.OBJ)

An object contain playback instructions for the underlying wave project. The object contains information about the time position as well as the editing steps involved. Objects are used in virtual projects.

Edit List (*.EDL)

The EDL helps exchange projects with other sequencer programs. A virtual project that resembles the EDL is created. The list is a text file that contains information about the wave files used, video files, object borders, object volumes, markers, and volume & panorama automation.

Table of Content (*.TOC)

With the table of contents (toc) of an audio CD the information about the contents of a CD is saved.

DPP Import

To check a DDP image created in Sequoia using a different application, load it here. All files belonging to the DDP should be placed into one folder on the hard disk.

Session (*.SAM)

This command loads a Sequoia session. This opens all projects and arranges all windows how they were distributed on the screen at the time the session was saved.

This is especially useful to open multi source sessions featuring all of their individual projects at one time.

Import

Load audio file

Here you can open a dialog for loading of audio files.

The following formats are supported and read directly by Sequoia:

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

Note: Open all other formats with the import audio function "Import audio" to convert these to wave files for local storage on the hard drive.

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".

If a virtual project is opened, new objects are created simultaneously in the VIP which refer to loaded audio files. If an area has been selected in the VIP, the files are positioned at the beginning of the range or with a separating pause of 2 seconds after the last object. The length of the pause can be adjusted in the "CD/DVD" menu via "Set pause index".

If you want to load the audio file as a separate wave project, click the corresponding check box.

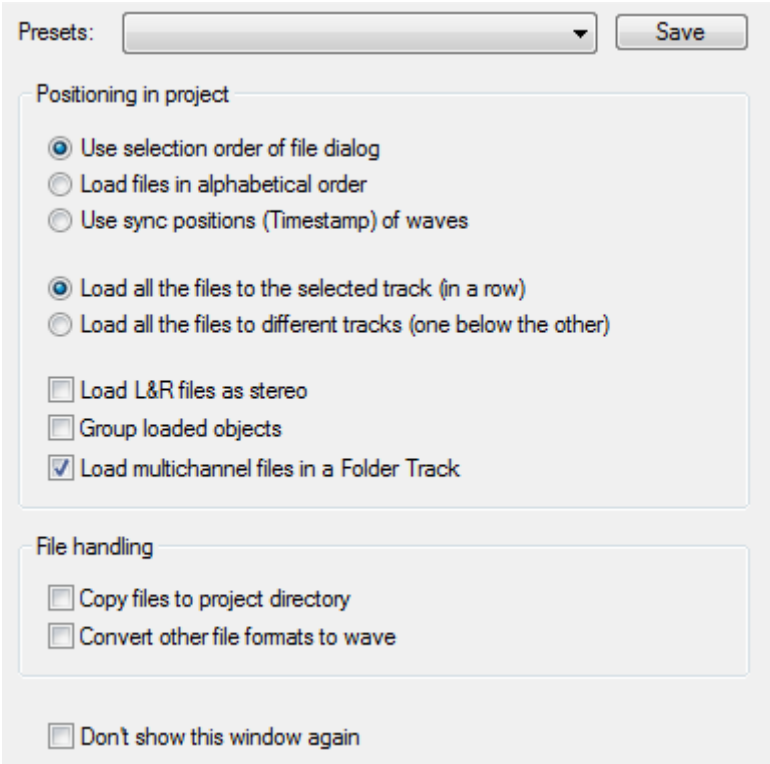
Clicking the payback button enables each audio file to be listened to in advance.

Attention: The monitoring function uses the Windows Multimedia MME standard output device. Many audio cards shut down the Windows MME system when ASIO drivers are used, which means problems may occur during monitoring. For this reason, the preview function is deactivated by default when ASIO drivers are used.

The "load audio file" option activates the monitoring function again at any time.

The "options" button opens the options for loading an audio file (view page 616).

Options for loading an audio file



Use selection order of file dialog: If this option is activated, Sequoia remembers the sequence in which the files were selected and then sorts them accordingly.

Load files in alphabetical order: If this option is activated, Sequoia sorts the selected files alphabetically in the VIP.

Load files to sync positions (timestamp in wave): Broadcast wave files containing time stamps are positioned precisely at this position in the VIP.

Load all files to selected track: The selected files are loaded successively in one of the selected tracks.

Load files to different tracks: The files are now sorted in vertical order from the selected track to the next one. If necessary, an additional track is added.

Load L&R files as stereo: Here you can load files separated by left and right stereo channels as stereo files.

Group loaded objects: All loaded files are grouped. They can be ungrouped anytime.

Load multichannel files into a folder track: This option is activated by default. When loading multi-channel files (i.e. files with more than two channels), it is assumed that these are Surround files in Interleaved format. A folder (view page 132) track with the required number of tracks is created and routed to the surround master (that is also created if necessary). The tracks in the folder track get the surround panning according to the determined surround format. If this option is not active, the individual channels are loaded onto the track and the tracks below it.

Other (only present for "Load audio file")

File preview with Windows Media Player Control: Files may be previewed via the Windows Media Player.

File Management

Copy audio files to project directory: Copies the file to the corresponding project folder.

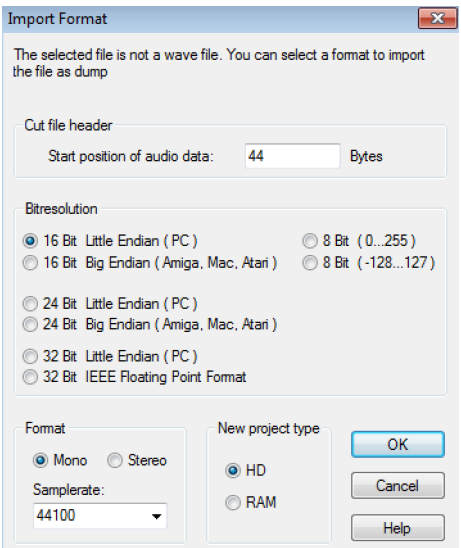
Convert compressed files to wave format: Compressed audio formats such as MP3 can also be loaded and played directly in Sequoia. This requires increased CPU calculation. Activate this option to convert files like this into an uncompressed wave format.

Sound Designer II

Use this function to load Sound Designer II files.

Import raw data (dump)

With the "**Import raw data (dump)**" you can try to import damaged files that sound like white noise on playback. The header is usually damaged or missing. In this case you can enter "0 bytes" as the header length in the following dialog.



Wave or MP3 projects will be imported as raw audio data (PCM) in RAW format (Little Endian for PC, Big Endian for Amiga, Mac, Atari).

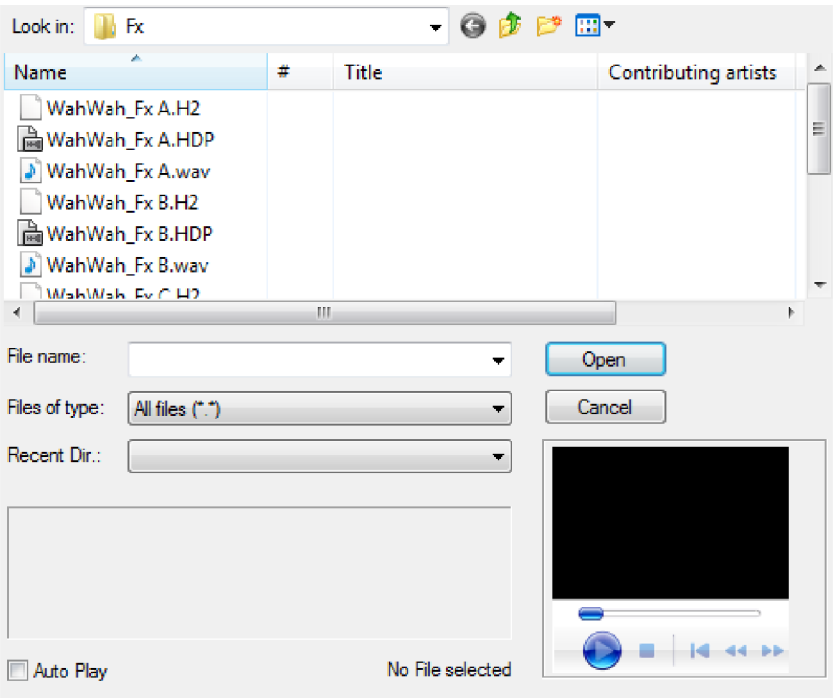
Load MIDI File

This function loads a MIDI file. Detailed information about this topic is provided in "MIDI in Sequoia -> MIDI - record, import, edit -> Importing MIDI files (view page 292)".

Keyboard shortcut: Shift + M

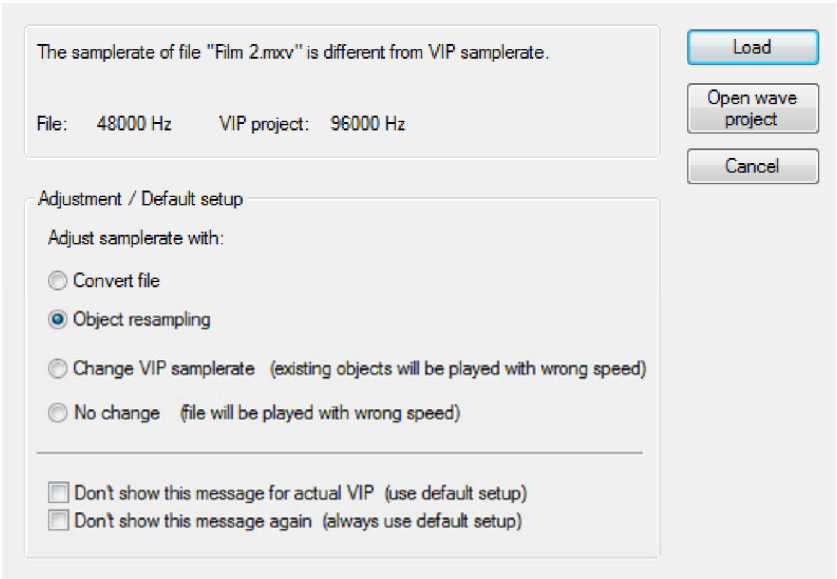
Load Video File

This dialog helps load a video into the current project. After selecting a video, preview it or inspect its properties. If a check is placed next to "**auto play**", the selected video file will play automatically in the preview window.



Before loading the video, you will be asked if you want to load the video's audio track into the project. This is positioned in the track underneath the video track. If the sample rate of the audio file deviates from the VIP's sample rate, then the sample rate may be adjusted. The following options are available for adjusting the sample rate:

- Convert file: Sequoia creates a converted audio file in the VIP's sample rate. The file name will now also indicate the new sample rate.
- Object resampling: If this option is active, Sequoia will resample the object. The quality of object resampling may be set under "**System option -> Effects -> Resampling/bouncing** (view page 681)".
- Adjust VIP sample rate (existing objects are played back at incorrect speed)
- No change (file plays back at incorrect speed)



Detailed information about displaying video objects is available in "Video setup (view page 659)".

Detailed information about recording a video track is provided in "Playback -> Video recording (view page 847)".

Import Audio CD Track(s)

This command imports the audio contents from a CD into Sequoia (view page 1089).

Import Audio DVD

This command imports the audio contents of a DVD into Sequoia (view page 1092).

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

Save Project

The current disc project is saved with the name displayed in the project window. If you have not yet specified a name for your project, Sequoia will ask you to do so.

Shortcut: Ctrl + S

Save Project as...

A standard dialog will open, where you can specify the path and name of the project to be saved. If you would like to save wave projects and virtual projects under a new name and in a different location, the original file will remain untouched. Then you can continue working in the new project (automatic).

Shortcut: Shift + S

Save Project Copy...

A dialog open to create a copy of the project, which may be saved under a different name. In this case, the date and time are added automatically to the project name. Of course, you can name the copy any way you like. Once saved you will be able to continue working on your original project.

Save Project as Template...

This option saves the current project as a project template. The template will save all project-based settings like grid form, track number, etc., but not the used objects or wave files. Templates may be loaded when creating new multitrack projects ("File" > "New Virtual Project").

Save project as EDL...

The active virtual project will be saved as a list in the EDL text format. This edit list is a text file containing information about used wave files, video files, object borders, object volumes, markers, volumes & panorama curves.

This option effectively lets you save Sequoia projects into an exchange format that allows a project to be incorporated into a different sequencer or video editing program. EDLs may be converted together with the audio files into other formats using a conversion program like EDL ConvertPro.

Save Object...

This function allows you to save a single object. This file contains all the data of the object editor like fades, effects, object names, etc.), but not the actual audio data. The wave project referenced by the object also has to be available if you want to reload the object.

Save Session...

A Sequoia session is a meta project that includes the file paths and window positions of all loaded projects upon saving. This is useful for starting to work again at the same position where you ended, without having to load various projects.

If you save a session with the name "startup.sam" and place it into the directory with the program folder, Sequoia will load that session automatically the next time you start the program.

Export

This menu option allows you to export projects in various formats.

The export dialogs for the respective formats are almost identical.

File name and path selection: 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. With "Recent Dir." 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. Format settings: Here you'll find the settings and options for the corresponding export format. (see below)

Export settings: Under Export Settings you will find a menu where you can make further settings for the export. Selecting a menu item enables or disables the corresponding option.

In the upper part of the menu you can specify which part of the project should be exported:

- **Marked Range only:** Here the export will be conducted only over the length of the range selected in the arranger.
- **Complete project (up to last object):** The time selection for export goes from the beginning of the project to the end of the last object plus the reverberation time. This is the default option.
- **Complete CD:** If you select this option, the project will be used until a CD end index is created, without such an index the whole project will be used until the end of the project.

The next section of the menu determines whether the export is to a single file or to several separate files whose beginning is determined by track markers or other markers:

- **Create one file:** A single file is created.
- **Each CD track in a file:** Individual files are created, ranging from one CD title index to the next.

- **Each CD track in a file (from track to pause marker):** Individual files are created, ranging from one CD title index to the next CD pause index. The sections between this pause index and the next CD track index are not exported.
- **Any marker:** Individual files are created, each of which extends from one marker to the next.

Note: When exporting to single files, the name of the corresponding marker is used as file name instead of the specified file name.

More options:

- **Use track number as prefix:** When exporting to individual files, the file name (marker name) is preceded by a consecutive number.
- **Export CD markers only:** If supported by the export format, all markers present in the project are saved as audio markers in the exported audio file. With this option you can limit this to the CD track markers.
- **Write cue list:** When exporting to a single file, it is possible to save contained markers in a cue list of the same name as a .cue file.
- **Write playlist file:** When exporting to multiple files, a playlist of the files is saved as a *.m3u file.

WAV

Here you can export Wave files.

Format settings: A dialog window featuring a list of all compression codecs appears here. Select the codec and compression rate you require. If you click the "**Dithering**" button, the dithering options dialog will open. Detailed information about this is available in "File > Program Preferences > Dithering Options (view page 703)".

MP3

This exports the project as an MPEG Layer 3 with the included encoder. Clicking on format settings opens a window for setting encoder options. Here you can set the export format, encoder quality, stereo coding, VBR (variable bit rate) options and labeling using the MP3 ID editor.

MPEG...

Exports the project as an MPEG file. These formats may also be set in the advanced settings dialog, which is opened via the button "Format settings". The options "stereo", "joint stereo", and "mono" are featured.

Windows Media...

Exports the project in Window Media Format.

This format is an optimized audio/video format for the Internet from Microsoft. This format also enables streaming playback of audio files over the Internet. Select a profile for the Windows Media file in the "format settings" dialog. Enter the track and artist names, copyright details, and descriptions.

AAC

Exports the project as an AAC file (Low Complexity or High Efficiency). The encoder can be set up in the advanced dialog that is opened through "Format Settings". You can also set the quality and transport format here. AAC export uses the file ending *.m4a.

Note: During AAC export the sample rate varies depending on the bit rate.

For example, if you load a 44.1 kHz wave file into a 44.1 kHz VIP and then export it as an AAC with 128 kBit/s, you will get a 44.1 kHz file.

However, if you load a 44.1 kHz wave file into a 44.1 kHz VIP and export it as an AAC with 256 kBit/s, you will get a 48 kHz file.

FLAC...

The FLAC format allows you to compress 16-bit or 24-bit files during export. The properties of the FLAC encoder may be accessed via the "Format settings" button. Set the compression degree, bit resolution, format, sample rate, and dithering here.

OGG Vorbis...

Exports the project in OGG Vorbis format.

The "Format settings" button opens an additional dialog to select the desired bit rate. The scale of compression ranges from 46 kBit/s to 500 kBit/s (for MP3 from 32 kBit/s to 320 kBit/s). In the stereo options, choose between "Stereo" and "Mono" and also select the "Variable bit rate" mode.

AIFF...

The project will be exported in AIFF format. In the advanced dialog, select the bit resolution, stereo format, sample rate, and quality level of sample rate adjustments.

Mono / Stereo Convert

Stereo Wave -> 2 Mono

This function saves both the left and right channels of a stereo wave file (file.wav) separately as a mono file (file_left.wav, file_right.wav).

"Stereo to 2 Mono" may also be applied to objects in the VIP. Stereo objects to be split are replaced by two mono objects that are arranged one below the other and which feature panorama settings at the object level to the left or right.

2 Mono -> Stereo Wave

This combines two mono wave projects to one stereo wave project.

After executing this command, a dialog opens displaying all loaded mono files. You can load additional files with "Load file". From the opened wave projects, select the right and left files with the "^" keys. With "<->" keys you can switch channels. After clicking "Link files", specify the name of the new file to be created in the project folder.

Please note that only mono wave projects of the same word width and sample rate can be linked. The length of both projects are adapted to one another, the longer mono file sets the total length.

LR Wave -> 2 Mono

LR wave projects may be split into two independent mono projects using this function.

LR Wave -> 1 Mono

The active LR wave project is set to mono mode. In this case, the two channels are mixed together by adding the corresponding samples each at 100% and then splitting the sum in two in order to avoid overmodulation. This corresponds to sinking the level by 6 dB.

2 Mono -> LR Wave

This creates an LR wave project with two channels from two mono wave projects, in which case one mono wave project in panorama view is positioned to the left, and the other one to the right. Select both mono files with consecutive mouse clicks. In the following dialog you can decide to "Link". "Convert to stereo" creates a stereo wave file.

1 Mono -> LR Wave

In this case, LR wave project with two identical channels is created from a mono wave project. The original sample is duplicated here.

Save in format

Use this function to save wave projects in various formats.

This can be useful when, for example, RAM projects are to be converted into HD Wave Projects or LR Wave Projects (two mono samples linked to one another) are to be converted into Stereo Wave Projects.

In "**Area settings**", select whether only the **selected area** or the **entire wave project** should be saved in the new format.

Under "**General options**", select whether the **maximum amplitude** should be displayed or not and if the **master effects** should also be copied when the file is saved.

The **Dithering Options** (view page 703) can also be adjusted here.

In the "**Format**" section you can determine in what format the file should be saved (**Wave**,**MP3**, **AIFF**, **Ogg Vorbis** or**FLAC**).

In the case of Wave, AIFF and FLAC projects you **also** have the **options "Stereo"**, **"left & right"**, **"only left channel"**, **"only right channel"**, **"mono mixdown"** and **"Mono"** and the **selection/quality of the sample rate** options open to you.

By clicking on the **format settings** for **MP3 format** a new window opens, where **encoder options** can be defined. Here you can set the **Output format**, **Encoder quality**, **Padding-mode**, **Content options**, **VBR** (variable bit rate)**options** and tagging using the **MP3 ID Editor**.

In the dialog for **AIFF** you can specify the **bit resolution**, **stereo format**, **sample rate**, and **quality level of sample rate adjustments**.

Change Bit Resolution

Select the desired word width/bit resolution of wave projects here.

Working with the 32 Bit Float Format

If you convert fixed comma wave projects into the floating comma format, the signal will remain a 16 or 24-bit signal and the quality will not be improved. This will only yield a result if you are applying destructive changes to the audio material as precise calculation is practically impossible after editing.

You use overmodulation resistance and level-independent dynamics. The quantization hiss will not increase, even if you set very low levels.

The downside is that twice the amount of hard disk space is required and only half the tracks are now playable simultaneously (depending on the CPU).

If you convert from 32-bit float to 16 or 24 bits, dithering occurs to reduce subjective quality loss.

Working with 8 Bit Wave Projects

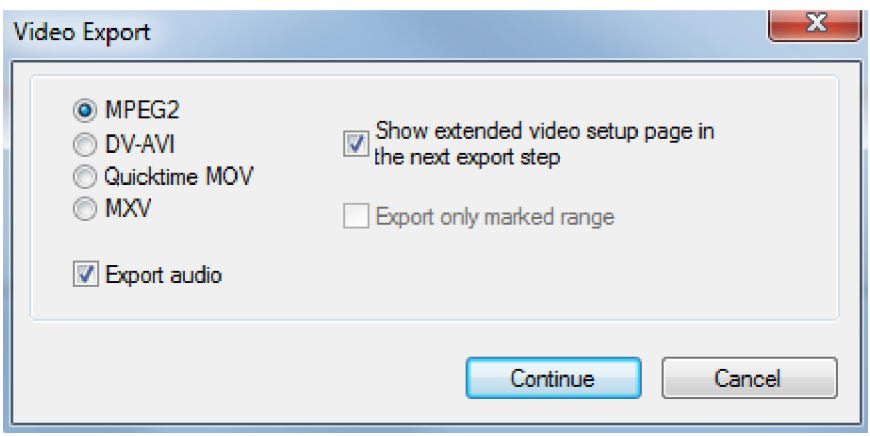
A smaller word width may be useful when editing audio for multimedia applications, since these applications may use a resolution of 8 bits to save storage space.

The downside of such a low bit resolution is that as the quality of the so-called SNR (signal-to-noise ration) drops, and the quantization noise increases. This quantization noise is not consistent, but is modulated by the signal and may be unpleasant.

The word width of the project is always indicated in the title bar of wave projects. To perform manifold destructive editing on an 8-bit project, you should convert it into 16 or 32-bit float before editing. The created calculation precision will then occur in the 16-bit range, this being considerably smaller. After editing, convert the sample back to 8 bits.

Export video

A dialog to export videos will appear if you go to "File > Export > Export video".



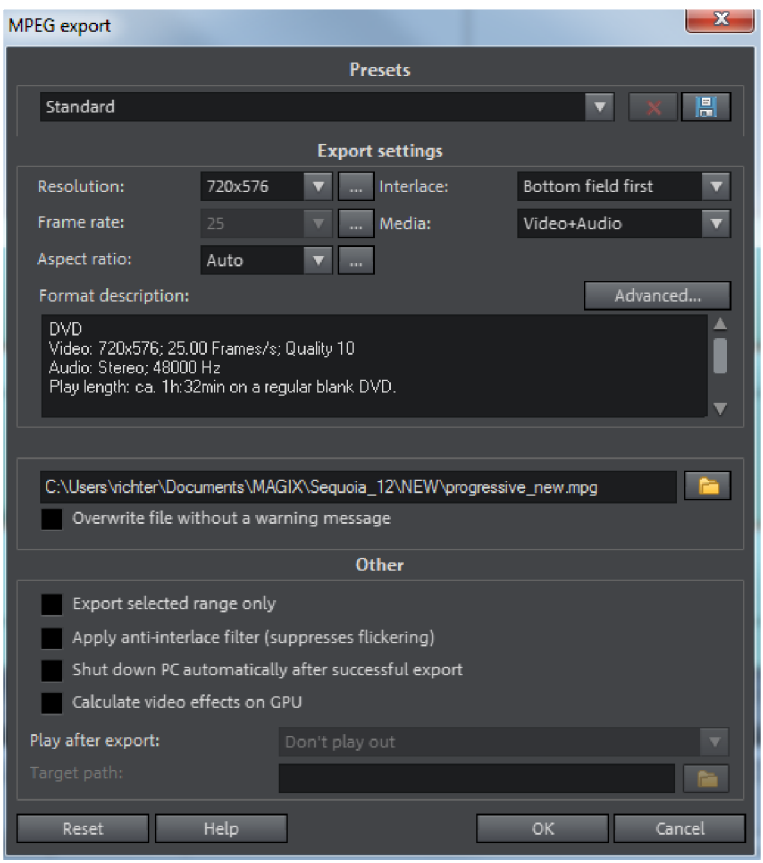
The **MPEG-2**, **DV AVI**, **QuickTime MOV**, and **MXV** formats are available for exporting. If you want to export the associated audio material as well, place a check next to "**Export audio**". Optionally, the **video codec settings can also be displayed in the next step**. The options available vary according to the selected format.

If you **only want to export a previously marked area** tick the appropriate checkbox.

Then click on "**Continue**". In the next dialog window enter the file name, file type and destination folder, then click "OK".

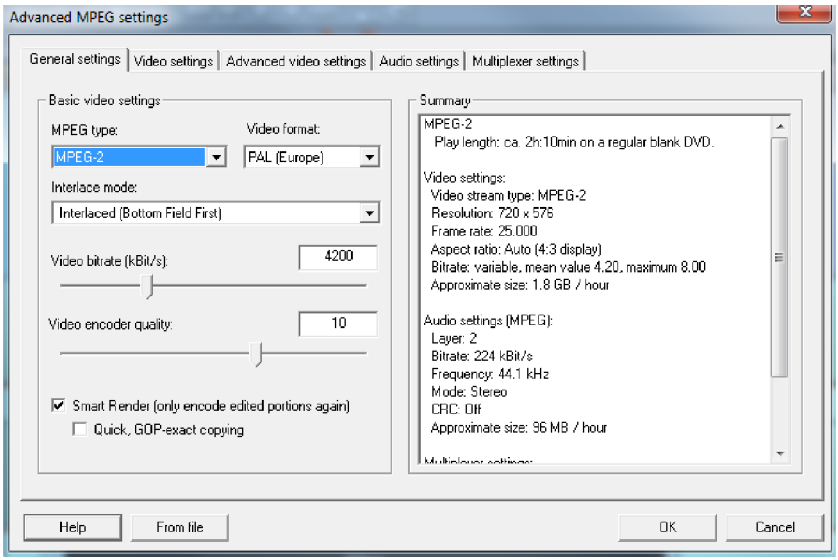
Video export - Export settings

Here, you can specify general export parameters like **resolution**, **aspect ratio** and **frame rate**. Select the most-used values from the list field, to set your own values click on the "..." button. For individual settings click on the "..." keys. The "**Advanced**" button opens the specific settings for the selected video format. "**File**" enables you to export your file to any folder. The option "**Overwrite file without confirmation**" option allows you to execute multiple exports in the same file.



MPEG-1/2 encoder settings

General settings: Here you can see an overview of all the encoder's settings.



MPEG type: Here you can set the output type of the created MPEG file. Available formats: MPEG-1, VCD, MPEG-2, SVCD, DVD, DVD-VR, HDV HD1, HDV HD2, Blu-ray.

Video format: Automatically specifies the video format, aspect ratio and frame rate. You can choose between PAL or NTSC.

Interlace mode: The movie is encoded interlaced (i.e. in two half-screens, so-called fields). This is essential for later playback on TV screens. If you want to view the exported movie on your PC only and like a "cinemascope" view, you can also encode by frame (progressively). Some video projectors also support playback of progressively encoded image material.

Video bit rate: The bit rate determines the memory requirement of the completed video. The amount of data available can be used differently for different display modes. 20 MB can be 4 seconds of DVD Video or 5 minutes Internet streaming at the thumbnail size. The quality of an MPEG video is measured by the "width" of the created "data stream". The bit-rate is the amount of the transmitted data per time unit; it is stated in kBit/s or bit per second. This is the amount of the transmitted data per time unit; it is stated in kBit/s or bit per second.

Video encoder quality: Determines the quality of the encoding process, to be more precise, the quality of the motion estimation. The higher the quality, the better the

finished video will look; however, encoding will take considerably longer. The preset value of 10 is a good compromise between speed and quality.

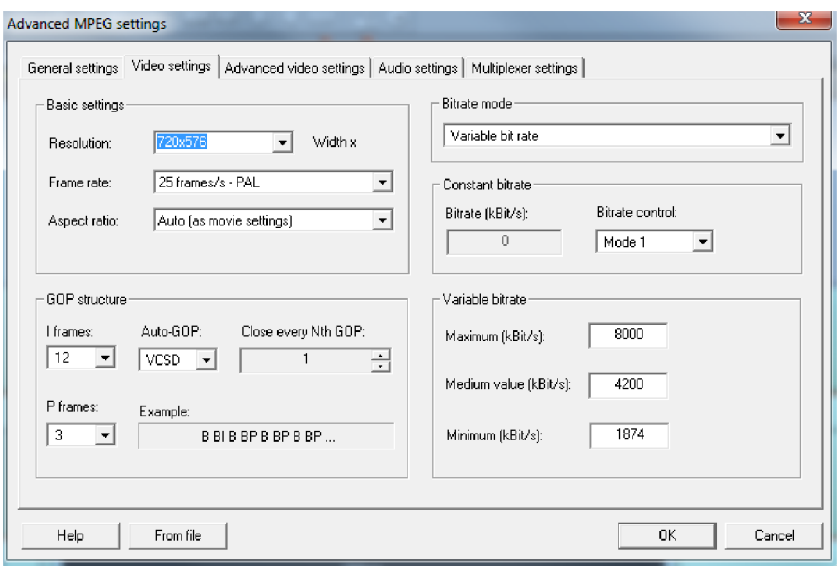
Smart rendering: With Smart Rendering you can considerably reduce the encoding processing strain for MPEG files. With the production of MPEG files, only those parts of the movie that were changed in the program (e.g. by video cleaning or effects) are re-encoded. Please note: The MPEG files contained in the movie must have the same format, i. e. the bit rates (variable or constant), audio formats, image resolutions and video formats must match.

Quick, GOP-precise copying: This special mode of Smart Rendering enables MPEG material to be transferred without having to encode it for the target medium, thereby increasing the encoding speed manifold. The video material cannot appear to have been altered in any way, only hard cuts (without fades) are permitted. These won't be executed precisely to the frame, but rather take place at the next GOP borders. For this reason, you should set cuts more generously.

Note: To burn DVDs in original 5.1 Surround Sound (Dolby Digital Audio) without having to re-encode, this option has to be activated.

From file: With this button you can rip the necessary settings from an existing MPEG file. This can be useful, for instance, if you want to merge MPEG files together without re-rendering via Smart Rendering.

Video settings



Resolution: Width and height of the video corresponds with the settings in the export dialog. This setting corresponds to the setting in the export dialog.

Frame rate: PAL requires 25 frames/sec, NTSC 29.97 frames/sec.

Please note: Encoding NTSC material as PAL or vice versa may be possible; however, it will result in jittery images.

Aspect ratio: Here you can set the aspect ration. With MPEG 2 the image format is displayed as image ratio; correspondingly there is 1:1 (square screen), 4:3 (regular video), 16:9 (Widescreen) and 2,21:1 (cinemascope).

GOP structure: Here you can change the settings of the GOP sequence, i.e. the number of P frames per I frame within a GOP. Please note that the total length of the GOP must not exceed 15 frames, which corresponds to the default setting (1 I frame + 4 P frames + 5*2 B frames = 15 frames).

However, you can reduce the length of the GOP sequence (while compromising the image quality at the same bit rate). This speeds up the encoding process considerably as the motion estimation while encoding P and B frames requires high CPU performance. The speed of decoding, i.e. playback, is also increased.

If I frame is set to 1, each frame is an I frame. If the value is set larger, the setting describes the total length of the GOP.

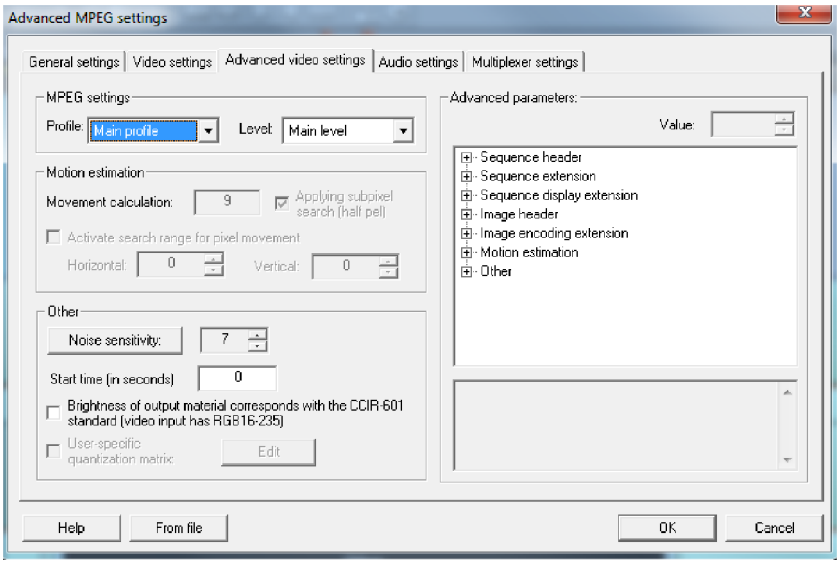
Auto GOP: Closed GOPs do not contain relations to frames from subsequent GOPs. MPEGs where all GOPs are closed, make MPEG editing easier as the I frames only can be edited alone if the GOP is closed. Auto GOP closes GOPs at scene changes, whereby two different and complex procedures ("fast" and "VSCD") are used.

Bit rate mode

In "**constant bit rate**" mode a bit rate that constantly remains the same is used. This option should be used for Video CDs, as it is only required here. With a constant bit rate the full power of MPEG compression can not be used properly as bandwidth for non-moving scenes is lost.

"**Variable bit rate**" mode attempts to adjust the available storage space to the requirements of the video you want to encode. The actual bit rate fluctuates around a **mean value (kBit/s)**. During calm sections of the video it may drop to a **minimum value kBit/s)**, if there are movements in the scene it may rise to the **maximum value(kBits/s)**. There are two different regulation processes (mode 1 and mode 128).

Advanced video settings



MPEG profile and level: The MPEG-2 standard defines so-called "**profiles**" and "**levels**".

For creating SVCDs and DVDs you can use the "**Main profile**" and "**Main level**".

The "**high profile**" adds additional properties to the data stream, such as the option of displaying an image at a reduced resolution for restricted transmission quality (SNR scalable profile) or locally scaled, for instance an HDTV data stream on a standard TV set).

The "**4:2:2 profile**" is used if the image data is to be encoded for alternative chroma scanning. However, these profiles are only supported by very few encoders, mainly for professional use.

The different levels define the restrictions to the image resolution and the maximum data rate. "**Low level**" can only reach a reduced resolution (352x2888 = CIF), "**high level**" or "**High -1440** enables encoding in HDTV format.

Estimate movement: These parameters are controlled via the quality controller (view page 629) (see General Settings).

Other

Noise sensitivity: This factor defines how sensitive the encoder will react to noise in the source material. If your source material only contains little noise (digital recordings, computer animations or already de-noised by video cleaning), you don't

have to change the default value 4, or you can even reduce it in order to increase the quality further.

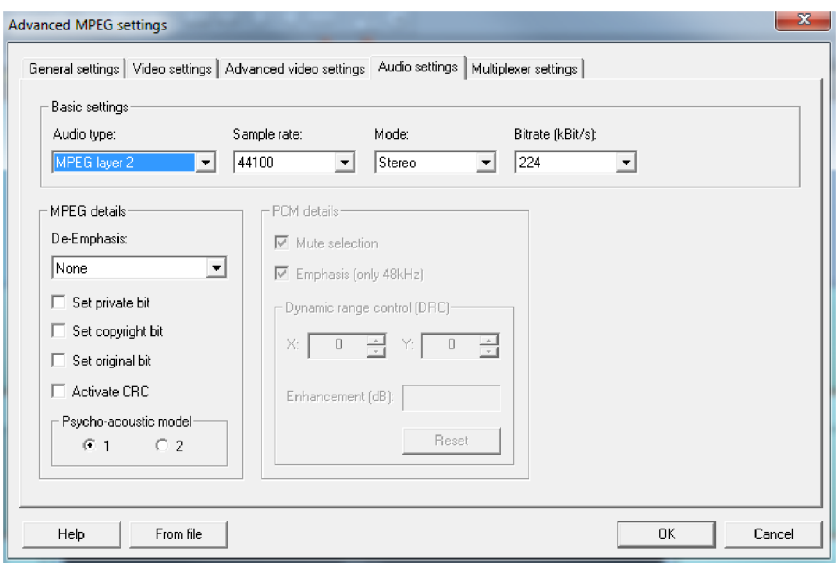
However, if you want to encode noisy material, a too low factor will considerably increase the encoding time without producing higher quality. For an unedited analog video you can increase the factor to 8-14.

By clicking on the **Noise reduction** button (click on noise sensitivity): A noise filter is used which has adjustable settings from 1-31.

Advanced parameters

Additional expert settings are available in the tree to the right of the window. This should only be changed if you are an experienced user. These settings have been optimized for general applications to such an extent that alteration is only necessary in exceptional cases.

Audio settings



Audio type: You can use MPEG Layer 2 audio and PCM (WAV). Select "No audio" in the export dialog.

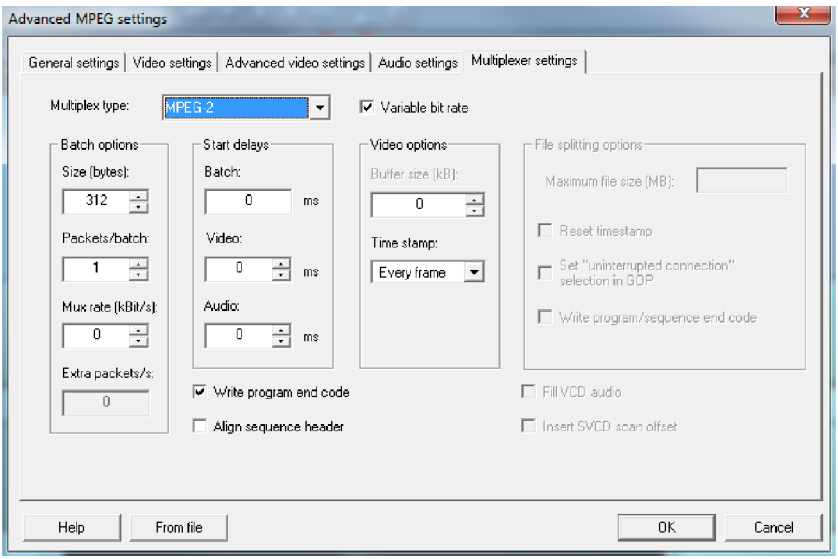
Sample rate: You can set a sample rate of 32, 44.1 or 48 kHz for the audio track. VCDs and SVCDs require 44.1 kHz, DVDs require 48 kHz. To reduce the size of audio data it is recommended to lower the bit rate instead of the sample rate.

Mode: You can use Mono, Stereo, Joint Stereo or Dual Channel.

- Dual Channel enables encoding of two mono tracks (e.g. different language sound tracks) that can be switched during playback.
- Joint Stereo is an optimized stereo encoder which takes advantage of the fact that the signal of both stereo channels is largely identical. Use Joint Stereo if you can only use small audio bit rates but still require a stereo signal.

Bit rate: Here you can set the bitrate of the audio signal. The higher the bitrate, the better the playback quality. VCD requires 224 kBit/s, and for SVCDs and DVDs select a value between 384 kBit/s and 448 kBit/s.

Multiplexer settings



The multiplexer is a component of the encoder which combines audio and video streams. With these expert settings you can manually control the data streams.

MPEG glossary

Motion estimation

Motion estimation is a further element for reducing data used in MPEG encoding.

Motion estimation also occurs in the B and P frames. The image difference that still exist after prediction are examined. Complex algorithms are used to search for an original occurrence of the macro block in the reference frame of each macro block of the P or B frame (these are units of 2x2 blocks specially combined for this purpose), which have only been moved, either by movement or a camera pan. They can then be left out in the P and B frame. Only the information by how far and to where the macro block has been moved is saved instead. This vector is called motion detector.

In the General encoder settings you can specify the "video encoder quality" of the final MPEG video. This factor also influences the time required for encoding. The longer it takes, the better.

Block

For almost all image file editing techniques the image is subdivided into 8 x 8 pixel blocks (image points). This should be noted if you would like to used user-defined image resolutions (width/height) – these should always be a multiple of 8.

Chroma format

The color value of each image point consists of the color values for the primary colors red, green and blue (RGB), and for traditional and technical reasons is transformed into one brightness value (Y = 0.299*R = 0.587*G + 0.114*B) and two color difference values (U = R - Y, V = G -Y).

The Y value alone produces the black and white picture. These signal components let brightness and color information to be handled separately. The first data reduction occurs already when single rows comprising a picture are read. Because the human eye has a lower color resolution than a brightness resolution, the color components are recorded only for every other point of a row (4:2:2) for each four pixels grouped (4:1:0) (under-reading of the color signal).

4:2:2 this corresponds to the established TV standard. One piece of color information is transmitted per row for two pixels, which corresponds to a 2/3 compression of the output data.

4:1:0 this is the color coding used for DVDs and most other consumer video applications. For each 4 pixels grouped together on two rows, one unit of color information is saved. This corresponds to a output data compression of 1/2.

Field

A half-image, two of which combine to produce a frame.

Frame

A frame is a single image from a video sequence, also called a full image. PAL video, for example, contains 25 frames per second, NTSC 29.97 frames.

Video recordings, with the exception of computer animations and still frames, don't contain full images. Instead, they have double numbers of half-images (fields), which are transmitted in an "interlaced" state. However, we still talk about frames, because

many predecessors of MPEG compression we based on such frames. In video editing, frames are also talked about.

GOP

Group of Pictures, this what the sequence of I frames and the P and B frames that belong to them is called.

Example: **I B B P B B P B B I ...**

(This GOP has a length of 9, with 2 P frames and 2 B frames)

I frames contain the entire image information of a frame, while P and B are reduced in information. So-called prediction and movement approximation are methods used for reduction.

The combination P B B is called a subgroup.

I frames must appear in regular intervals in the data stream for image and sound to be synchronized. Between the I frames only a limited count of P and B frames is allowed. This explains a few things: Since P and B frames contain only differential information, these differences will be larger with time, since more and more changes takes place from frame to frame. Because of this, a large count does not make much sense, GOP has a maximum length of 15 (4P, 2B) in PAL and 18 (5P, 2B) in NTSC. (More than 2 B frames between P frames is not allowed).

In a closed GOP, B frames of the last subgroup may contain only backward predictions, or references to the preceding P frame, but no references to the following I frame, since it belongs to the next GOP.

I frames

Intra Frames. In these pictures, the entire image information of a frame is saved, and only information from this frame is used to do so ("intra-frame encoded"). In contrast to the I frame, P and B frames, which save only the differences between the current frame and preceding and /or following frame are also found in MPEG video (P frame = "predicted frame", B frame = "bidirectional frame).

Interlace

For historical reasons, pictures in a movie are always recorded and transmitted in the form of two fields, first the lines with even numbers and then those with odd numbers. These fields are alternatively displayed with a double-frame rate. The (lazy) eye of the viewer or the processing of the TV tube puts the two frames together to form one.

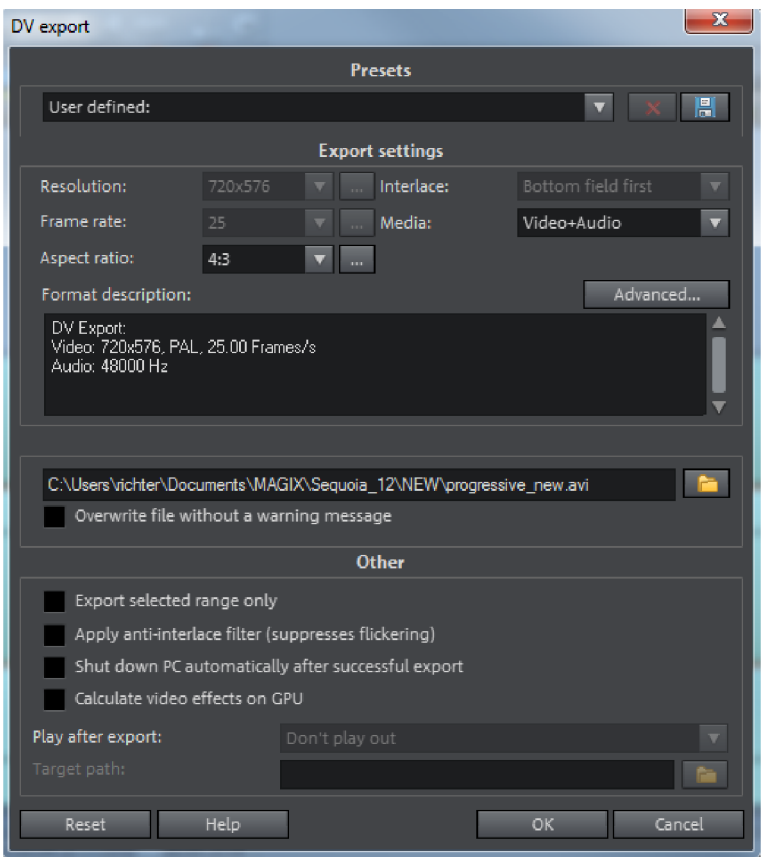
You normally don't have to worry about field processing. The video material goes through the entire processing chain as fields and is exported again as fields or burned onto DVD or shown on TV when played back on a DVD as a full picture. Only in certain rare conditions is it necessary to delve deeper into this process.

P frames and B frames

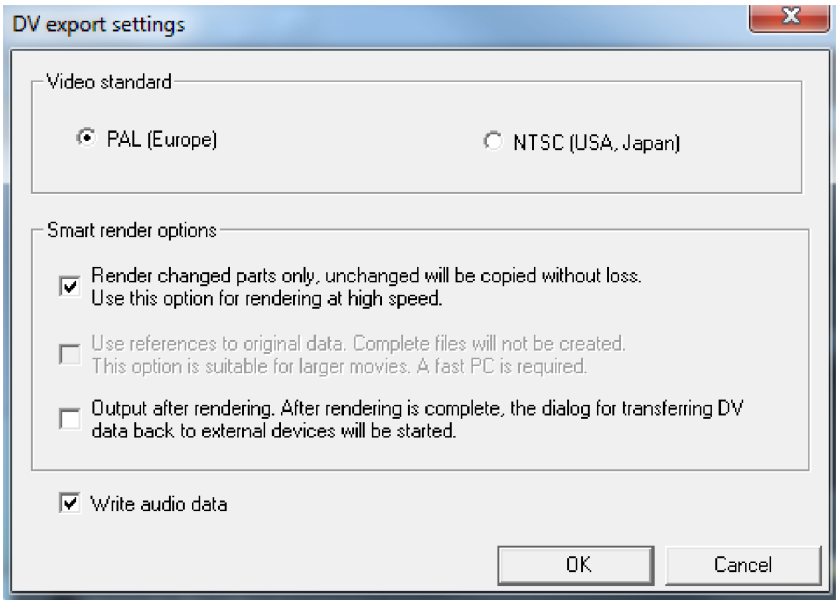
P frames save only the difference between the current picture and the preceding I frame. The "P" comes from the term "prediction", which describes this process.

B frames save the differences between the current picture and the I or P frame preceding and following - therefore the information which is neither in the previous frame nor in the following frame remains the same. Both directions are thus analyzed, pointing to the "B" in the name, which means "bidirectional-predicted".

AVI export



When exporting to AVI video, you can set and configure the size, frame rate, aspect ratio, interlace mode and other options of the AVI video using the "Advanced" button, as well as define the video standard (PAL or NTSC) and set and configure smart render options.



Here you can also select whether audio should also be written during export.

General notes on AVI videos

The AVI format (Audio Video Interleaved) isn't actually a "proper" video format. Rather, it is a so called "Container", where the conventions for transferring audio and video files to the program are only loosely defined. The codec (coder/decoder) actually defines what storage format is used. A codec compresses audio/video data into its own unique format which can only be read by the codec itself and is decoded when the film is played.

In concrete terms, a computer-generated AVI file can only be loaded by and played on a different computer if the same codec is installed on it.

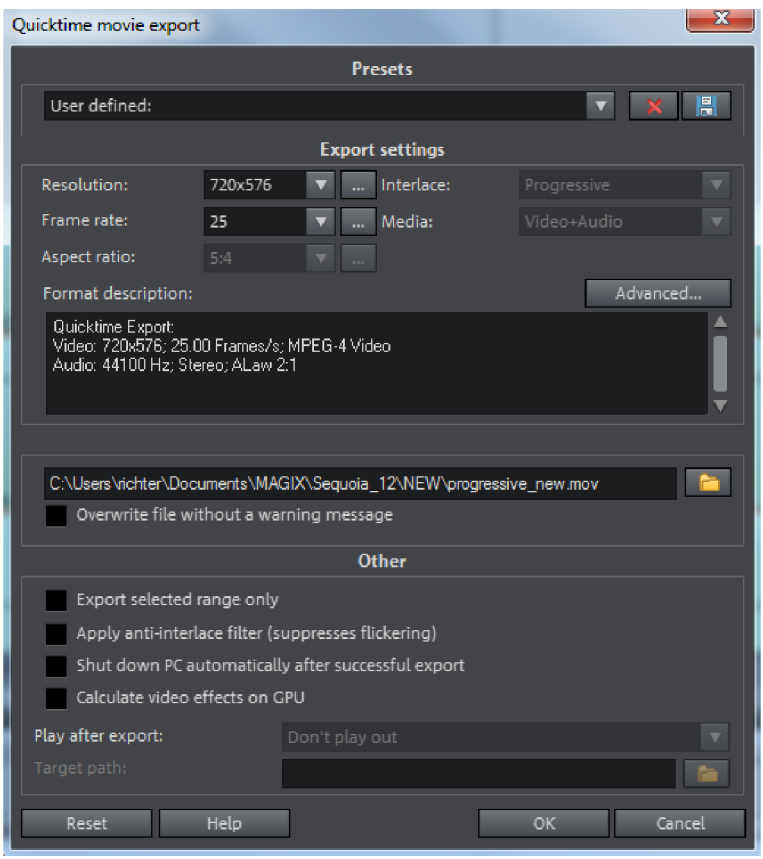
Many codecs (e.g. Intel Indeo Video) have now become standard components of the Windows installation. Others like the popular DivX codec are not standard. If you are generating an AVI file for future play on another computer using one of these codecs, you should first install this codec on the other PC. The best method available is to copy the codec installer to your export directory and burn it every time you create a video disc (slideshow disc) for play on computers.

You may encounter some problems when using older video editing cards with codecs which only function with the card's hardware. This type of AVIs can only be used on the PC on which they were created. Avoid using this kind of codec if possible.

Quicktime Movie Export

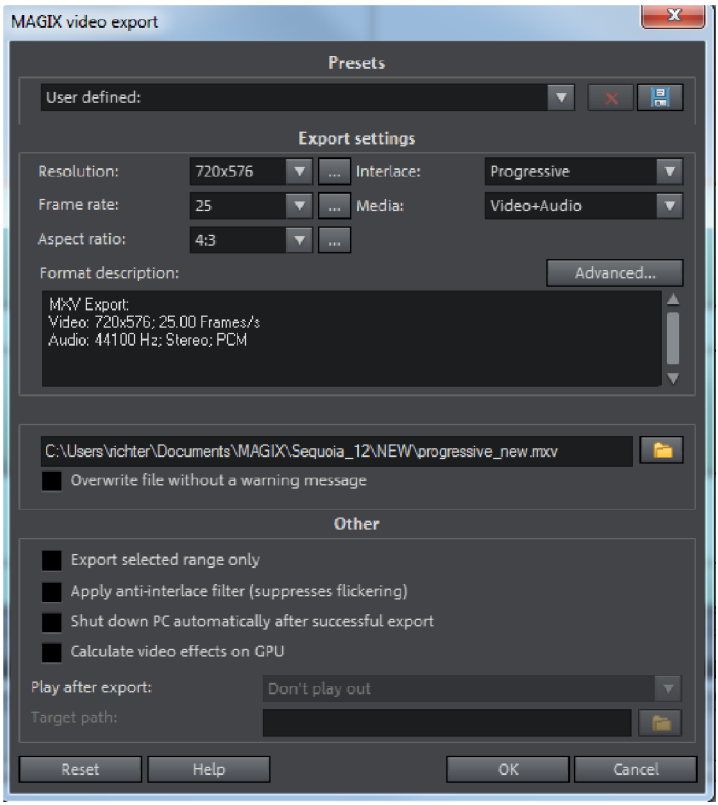
Exports the project in QuickTime Movie format. This format enables streaming playback of audio or video files via the Internet.

As with Real Media Export, you can adjust the corresponding settings for video size, frame rate, codec, compression and audio.

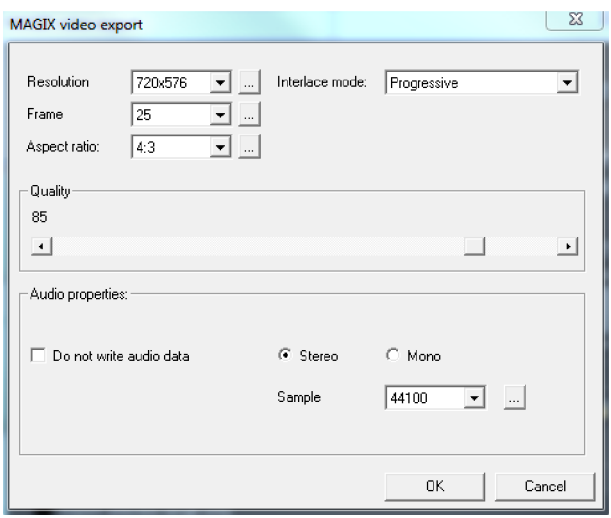


Note: For Quicktime files (*.mov), the Quicktime library must be installed.

MAGIX Movie Export



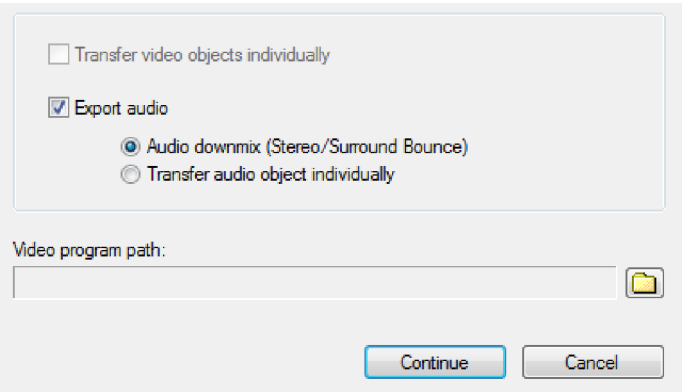
With this dialog you can export the project in MAGIX Movie format.



This format is used for video recording with MAGIX Video Pro X or MAGIX Movie Edit Pro software and is optimized for digitally editing high quality video material.

Export to "MAGIX Video Pro X"...

The following trackbouncing dialog opens for exporting to the external MAGIX program "Video Pro X".



Select whether only the video objects should be transferred, whether an **Audio downmix (stereo/Surround bounce)** should be made, or if the **Audio objects should be transferred individually**. If the "Continue" button is pressed, a save dialog will appear.

Export Video Sound...

After successfully editing the video sound in Sequoia, you can write the sound back into the video file. You may replace the original sound of the video or create a new video file.

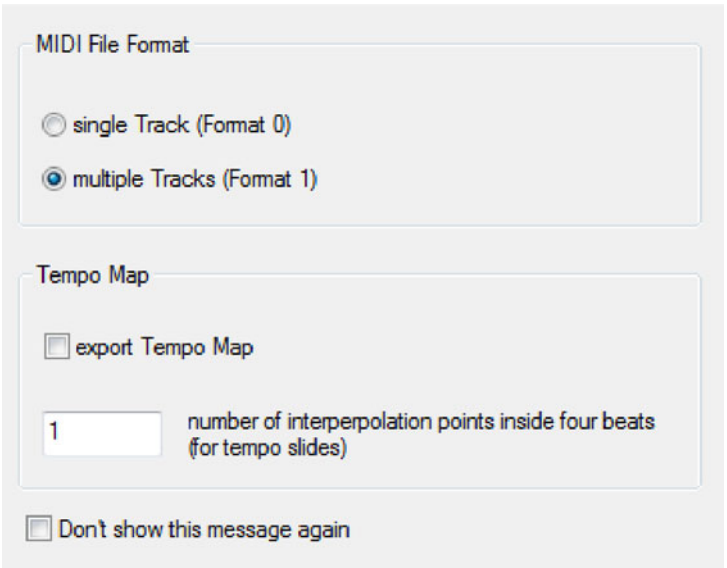
Set the source file under "Video source".

Replace audio in the original AVI file: This option carries out trackbouncing on the current VIP. This creates a temporary wave file which can then be integrated into the selected AVI file. It's necessary that the AVI contains an audio track of the same format (sample rate and bit resolution).

Save a new AVI file to: This option applies trackbouncing to the current VIP. This creates a temporary wave file which may be copied with the selected AVI file into a new AVI file. This process contains the original AVI file.

Note: If the lengths of the audio and video files differ from one another, a warning dialog opens. After the export, the longer component is cut off; an attempt to automatically synchronize the data is not carried out. If you receive this warning, try to resample the soundtrack to the correct length.

Export MIDI File...



The particular file format may also be specified when exporting a MIDI file. One track **corresponds to format 0**, **multiple tracks to format 1**. In addition, the tempo map may also be exported, in which case the support point count per beat for tempo runs can be selected freely.

The standard MIDI file (SMF) export always takes place in the current VIP-PPQ resolution. The markers are also exported.

Exporting MIDI Objects as Individual MIDI Files

This command exports all selected MIDI objects as separate MIDI files.

List Export

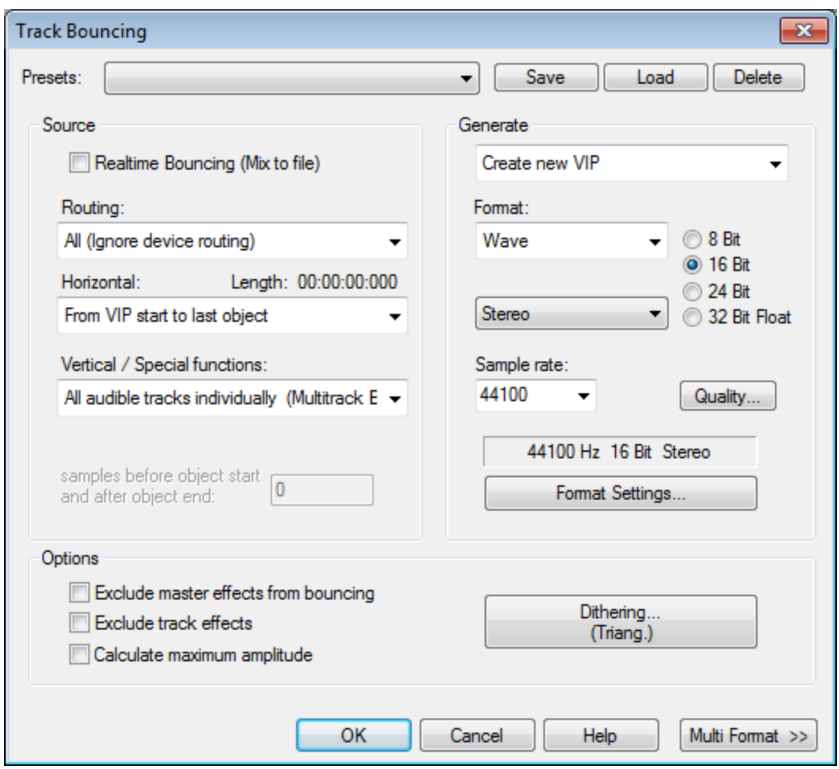
A current **marker list**, **object list**, or **track list** for your project can be written as a text/CSV file and stored in the project folder. You can select the parameters that should be included in the list in the export options.

Advanced export & Trackbouncing

For exports that go beyond simply rendering the project into an audio file, use the Track Bouncing dialog. Here you can create one or more audio files by combining the project or a section of the project, selected objects and tracks.

There are other differences to the simple export:

- Note: All files created as a result of trackbouncing are opened in Sequoia. This means that you cannot bounce into the same audio file twice in a row, but have to close it first. If you bounce often and do not require any special mastering options, you can work more effectively by using the simpler "File > Export" (view page 622) command instead of the track bouncing dialog.
- All formats that cannot be directly opened by Sequoia cannot be created in the track bouncing dialog. For bouncing into AAC, MPEG or WMA formats, use the "File -> Export (view page 622)" command.
- The export settings related to CD track markers, e.g. to create single files at CD track markers when exporting, are not available in the track bouncing dialog.



All settings made in the dialog can be saved as presets.

Source

Under **Source** you define which audio material from the project should be used for track bouncing.

realtime bouncing (Mix to File) The "Mix to File" (view page 376) function allows you to quickly perform a mixdown in real time and change your mix during playback.

Routing: This drop-down menu is used to define the outputs whose signal output should be written to the file(s).

- **Master:** The mixer's master output is written to a file. This is the default setting for exports.

If a surround master is present, there are the additional options

- **Surround master** (view page 502): This bounces the surround mix.
- **Surround + Stereo Master:** The surround mix is bounced together with the stereo mix.

The remaining two options are only relevant if bouncing of individual tracks is selected under "Vertical/Special functions".

- **All (Ignore Device Routing):** A mono or stereo file is created for each unmuted track (see Track Bouncing Settings: Format), regardless of its routing to the outputs.
- **All Devices (Use device routing):** All unmuted tracks are combined according to their routing to the individual outputs. For each physical output, individual mono files, stereo files or an Interleaved RIFF Wave (Multichannel Wave) with the corresponding number of channels are generated, depending on the settings under **Format**.

Horizontal: In this drop-down menu, select the time range of the project for bouncing.

- **Marked Range only:** Here the export will be conducted only over the length of the range selected in the arranger.
- **From VIP start to last object:** The time selection for bouncing goes from the beginning of the project to the end of the last object plus the reverberation time. This is the default option.
- **Complete project:** If you select this option, the complete virtual project will be bounced, including the silence after the last object until the project end.

Vertical/Special functions: Use this drop-down menu to specify whether you want to combine the output of all tracks or only selected tracks into one file, create separate files per track, or bounce individual objects.

- **All audible tracks:** All unmuted tracks are used for bouncing.**This is the default option.**
- **Only selected tracks**: All selected tracks are used for bouncing.

The following two options are only useful if **All (Ignore Device Routing)** is selected under Routing.

- **All audible tracks individually (Multitrack bounce):** All tracks that have not been muted are individually calculated into new files.
- **All selected tracks individually (Multitrack bounce):** All selected tracks that have not been muted are individually calculated into new files.

For the last two options, there is an additional setting **Samples before object start and after object end**, with which you can include additional samples before and after the object boundaries in the file to be calculated.

- **Bounce all selected objects individually:** All selected objects will be bounced individually, including the object effects, into a new file. This enables a large number of single objects to be exported quickly to separate files (e.g. for creating sample archives).
- **Glue selected objects together**: Per track, all selected objects are processed and written into a new file including object effects. In this process, all objects between the frontmost and the last selected object in each affected track are glued together.

Notes:

- If a **Surround Master** or **All (Use Device Routing)** is selected under **Routing** (see above), track bouncing of individual tracks or objects is not available.
- Note: If a selected track is a submix bus, all tracks that are routed into this bus are also bounced. This is done recursively, i.e. if these tracks are busses again, their sources are calculated as well. This permits fast mixing of individual groups.

Generate

In this drop-down menu you can define what should happen to the generated audio files after track bouncing.

Create new audio file: The audio file or audio files are simply created.

Open new wave project: The audio file or audio files are created and opened as wave projects in new windows.

Create new object in VIP: The audio file or audio files are created and inserted as new objects in a new track in the existing VIP. This selection automatically activates the option "Exclude master effects from bouncing" to prevent master effects from being applied twice.

Replace object(s): The objects used in trackbouncing are removed and the result is inserted into the selected track. This selection automatically activates the options

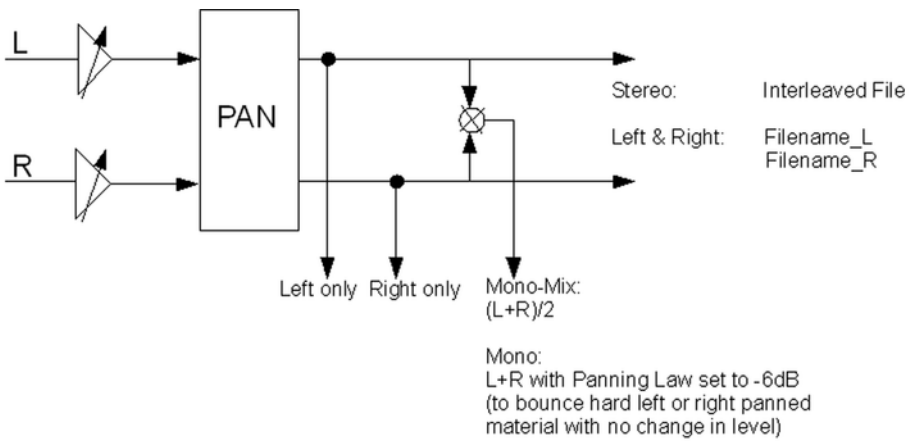
(see next section) "Exclude master effects from bouncing" and "Exclude track effects" to prevent these effects from being applied twice.

Create new VIP: A new virtual project (VIP) will open. The bounced audio material is inserted into the first track of the created project.

Format

The target format may be set flexibly. **Wave** and **AIFF** files in 8/16/24/32-bit mono/stereo at variable sample rates and four different quality levels are supported, as are **MP3**, **Ogg Vorbis**, and **FLAC** files at **variable sample and bit rates**. The desired codec can also be selected by clicking "**Format settings...**"

The following diagram displays the signal flow for the target format's setting.



- The format setting "**Stereo**" creates a file that contains the stereo information as an "**Interleaved file**".
- The format setting "**Left & right**" can be used to create two files, i.e. "**Filename_L**" and "**Filename_R**", which provide separate information about the left and right channels.
- The format setting "**Left channel only**" monitors and outputs the **left channel only**.
- The format setting "**Right channel only**" monitors and outputs the **right channel only**.
- The format setting "**Mono mixdown**" calculates the left and right channels together according to the formula "**(L+R)/2**" and then outputs these.
- The format setting "**Mono**" is especially useful for hard left or right tracks, since it bounces these at the same level. Mono is formed according to the formula "**L+R**". If **panorama** is set in the middle in the format settings, the trackbouncing process will automatically result in a **level reduction of 6 dB** per channel (panning law -6 dB). This will ensure that even centered mono signals are not bounced at a too high level, but at the original level.

If you set **Surround Bouncing** (view page 502) in "Source -> Routing", the formats "Mono", "Stereo", and "Interleaved file (RIFF 64)" will be available for selection.

Options

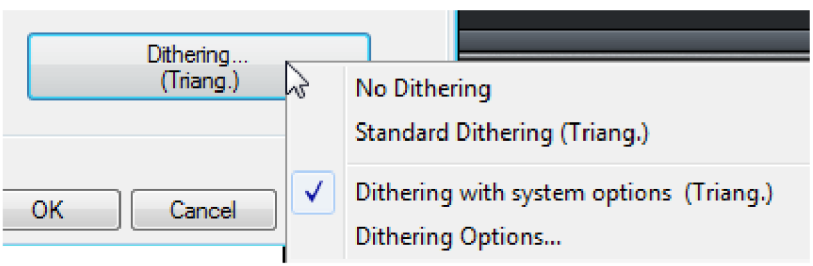
Exclude master effects from bouncing: Master effects are not included during bouncing.

Exclude track effects of target track: Track effects of the target track are not included if they are identical with the source track.

These options are important in combination with the "Generate" functions "Create new object in VIP" and "Replace object(s) in VIP". Note: these are set automatically so that the track effects are not applied twice.

Calculate maximum amplitude: Displays the maximum volume in dB after bouncing in order to correctly set equipment for additional editing or to correct the master level. Once bouncing has been completed, the corresponding information window will be displayed.

Dithering: Every track bouncing process can have its own dithering process, independent of the global settings. This enables dithering to be bypassed or to apply the standard dithering (dithering featuring triangular noise).



Use this dialog to apply dithering according to the system options or to access the dithering options in the system settings. The button value in brackets (e. g. **Triang.** or **POW-r 1**) indicates the currently set dithering algorithm.

Detailed information on this can be found in the menu reference under "File" > "Program Preferences" > "Dithering Options (view page 703)".

Notes on the trackbouncing options

The numerous options in the track bouncing dialog are designed to give you maximum flexibility for rendering your projects or parts of them. However, this means that not all combinations of options can be usefully applied.

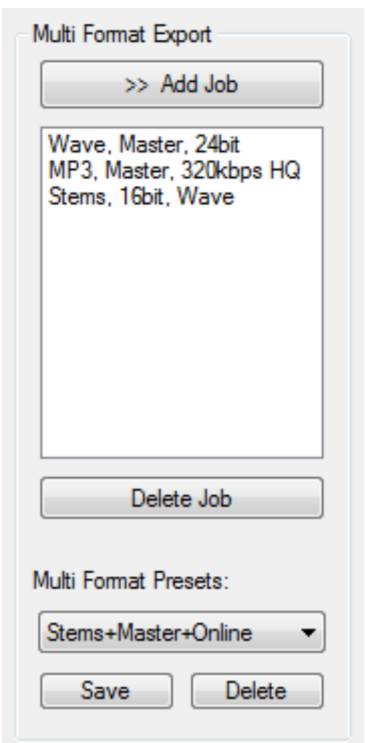
For example, rendering single tracks (Source - Vertical/Special functions - All audible tracks individually) makes no sense if the Master is set under Routing instead of All outputs. These operating errors are caught as far as possible - for this example, single tracks are still created - but it is still recommended to think through and test the settings before bouncing.

Use the possibility to store trackbouncing options in presets for specific, recurring tasks!

Multi-format export

With the Multi Format Export it is possible to perform several track bouncing operations with different settings one after the other. For example, you can render a mix both as MP3 and as WAV as well as single tracks.

Click "Multi Format" to display the controls for the multi-format export.



This is how you proceed for a multi-format export:

1. Set the desired export options in the corresponding areas of the Track bouncing dialog.
2. Click "Add Job" to add an export job with these settings to the job list. You will be prompted to enter a name for the job.
3. Repeat these steps with the desired other export jobs.
4. If you select a job in the list by clicking on it, the corresponding settings are always displayed in the dialog on the right.
5. Click on "Multi Format Export" to close the dialog and execute the exports one after the other.

Note: You cannot change the settings of a job directly. To change a job, select it and the dialog takes over the settings for this job. Now make the desired changes and save the job again. The old job can be removed from the list with "Delete Job".

The entire job list can be saved as a Multi Format Preset and can be reloaded later by selecting it from the list box. The last used job list is always available as the preset "last".

Tip: Also save all your regular trackbouncing settings as jobs in a multi format preset! By this you can quickly select them with one click from a handy list and then return to the normal track bouncing dialog with another click on "Multi-Format" and then use the selected settings.

Range Bouncing (Internal Mixdown)

Use this function to convert the objects within a selected range to a new wave file. The objects are then replaced in the arranger. This is useful if you want to combine multiple objects into a single object for easy editing later.

To create performance room you can render real-time object effects using the range trackbouncing feature.

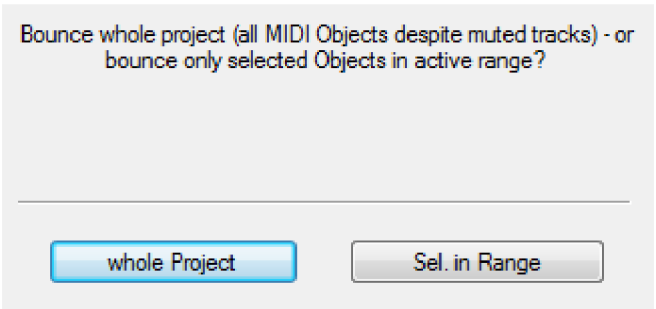
If the rate of your objects is higher than 16-bits, a dialog box will ask you whether the new file will be created as a 32-bit float or 16-bit file:

32-bit (float): The resulting file features a 32-bit (float) rate. This setting is practical if 24-bit objects or float objects are used in the bouncing and their high rates should be retained.

16-bit (integer): The resulting file is 16-bit rate. This setting is practical when the recording needs to be burned to CD and when there are no 24-bit objects in use. If 24-bit objects are being used during bouncing, these may be reduced to 16-bit using dithering options.

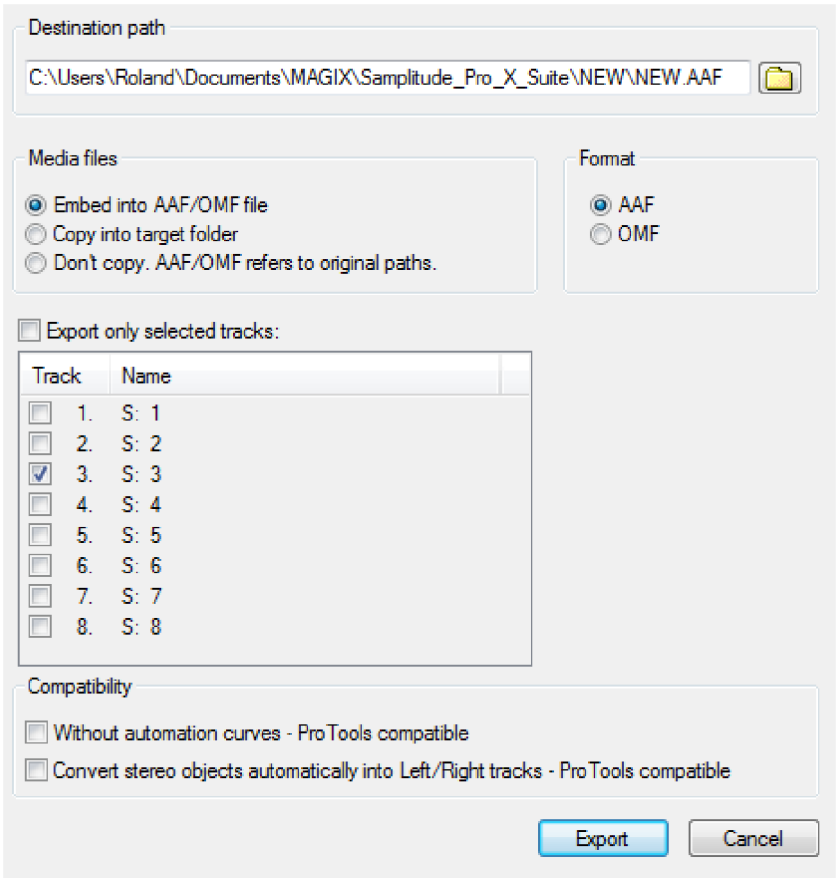
MIDI bouncing

In another prompt you can choose whether you want to mix the entire project with all MIDI objects or just the selected MIDI objects within the current area into one single MIDI object. In this case, the MIDI track and object effects like timestretching, track MIDI transpose, program change, etc. are bounced into the resulting MIDI object.



Note: Don't confuse MIDI bouncing with the "MIDI object freeze (Object -> Object freeze) (view page 804)" function, which replaces the MIDI objects with audio objects in the software instrument's audio return signal.

Export project as AAF/OMF



The**AAF Export** transfers the following content:

- Object position/wave offset - sample exact
- Object fade in/out - linear only
- Object crossfade - linear only
- Object volume
- Object pan
- Object volume automation
- Object pan automation
- Track names
- Track Volume
- Track Pan
- Track Volume Automation
- Track Pan Automation
- VIP markers

- Timecode offset
- Timecode format (24, 25, 30 fps)
- Processing timestamps in audio files

OMF Export transfers the following content:

- Object position/wave offset - sample exact
- Object fade in/out - linear only
- Object crossfade - linear only
- Object volume
- Track names

Note: During AAF/OMF export, media files will always be converted to WAV files. You can't export video files. Corresponding objects will not be transferred; the video track therefore remains empty.

Destination path

Enter here where you would like to export the AAF/OMF file.

Media files

- **Embed in AAF/OMF file**: If you select this option when exporting then the whole project will be saved as a single file, which contains both media content and metadata.
- **Copy into target folder**: If you select this option, media files will be placed next to the AAF/OMF into the target folder.
- **Don't copy. AAF/OMF refers to original paths**: If you select this option, no media files will be exported – the AAF/OMF file will only refer to the original media file paths.

Note: Select the "Embed in AAF/OMF file" or "Copy to destination folder" if you want to have the entire project together with the corresponding Audio/Video files as AAF/OMF files in another studio.

Format

Select whether you want to export the file in AAF format or in OMF format.

Compatibility

Without automation curves (Pro Tools-compatible): If you select this option, AAF/OMF export will take place in a Pro Tools-compatible fashion. Track volume and panorama as well as automation curves will not be transferred.

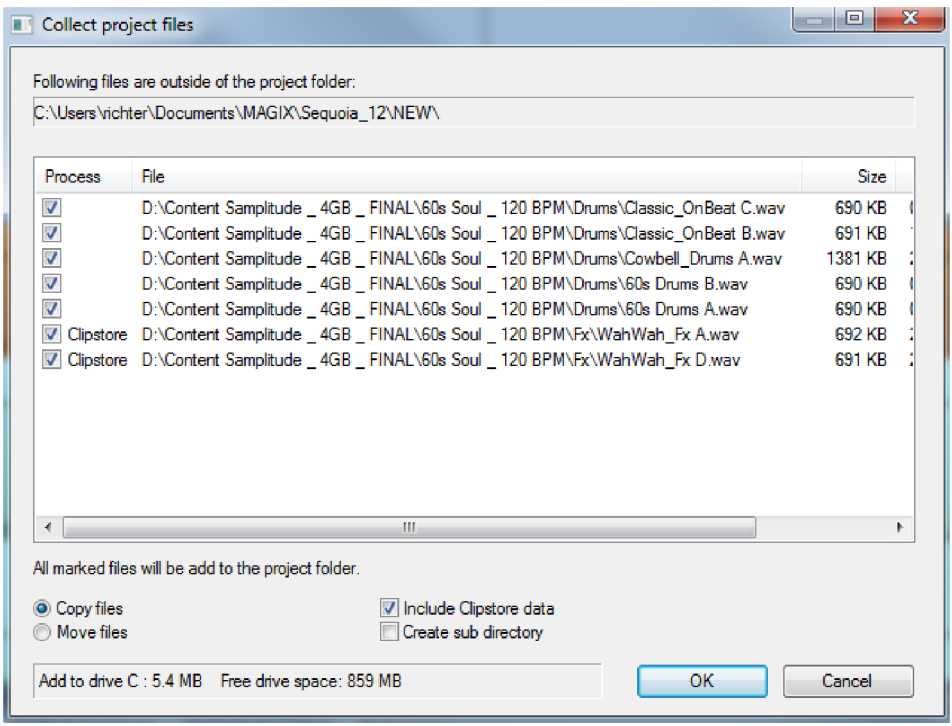
Convert stereo objects into left/right tracks automatically - Pro Tools compatible: Stereo objects are not supported by Pro Tools. Convert the corresponding objects into Pro Tools-compatible objects prior to AAF/OMF export as required.

Note: Timestamps saved with ProTools in Frames for improved compatibility.

Clean up

Collect Project Files

This dialog displays all files located outside of your project folder. All files marked in the "Edit" column can be compiled in the project folder. You can ether copy or move the files outside of the project folder into it or create a new subfolder. Sequoia displays the amount of space required and how much is currently available.



ClipStore data can also be optionally included in the calculation.

After processing, it's a good idea to save the project!

Delete Virtual Project (VIP)

Use this menu command to easily delete wave projects from the hard disk. The integrated wave files may be individually blocked from deletion by removing check marks under "Delete" next to each file. The remaining files belonging to a virtual project will be deleted.

Warning: Once the dialog has been confirmed, the data will be deleted without a repeat query.

Save Complete VIP in...

The complete virtual project and all associated wave files will be saved in the specified folder. This function is useful for making backups on a different drive, etc.

Keep project subdirectories: If you have already created subfolders in your project directory, e.g. to sort samples, then these will also be copied into the new directory.

Copy unfreeze data: Copies all unfreeze data in the VIP into the new directory.

Copy Revolver track data: Copies all Revolver track data included in the VIP.

Copy ClipStore data (Sequoia only): Copies all of the ClipStore data used in the VIP.

Copy video data: Copies associated video files.

Copy effect used files: Copy the files used by effects into the new directory, e.g. Room Simulator or the vocoder.

Copy only samples used in VIP: Only the sections of wave projects that are actually used by objects in the VIP are copied.

This function is practical for saving disk space; however, the objects in the newly saved project may be only shortened, and not extended. This is because all of the audio data outside the object edges is not copied with it, and therefore, cannot this cannot be used.

A security area in may be defined in samples for this purpose. The **number of samples** will be left as **additional space** in front and at the end of the corresponding object limits to leave a reserve, in case the object may be altered with a fade in/out. A value of 22,050 samples is preset (this corresponds to 500 ms at a rate of a 44.1 kHz).

Burn Project Backup on CD/DVD

Use this menu command to burn an entire project backup onto multiple CDs or DVDs. Sequoia's burner program, MXCDR, will open for this.

Delete Wave Project(s)...

Use this menu command to easily delete wave projects from the hard disk. All files belonging to the HD wave project (the wave file containing the audio files, files with the graphics data, and the HDP file with the project information) will be deleted together.

Note: An HD wave project may only be deleted once it's closed. As long as parts of the project are still active as objects in the VIP, deletion is impossible. Once the dialog has been confirmed, you will be able to delete the data.

Delete Unused Samples

This function lets you edit all wave projects belonging to the current virtual project in such a way that all ranges that aren't used are deleted. It basically deals with the part of the audio data to which the objects in the arranger do not refer and don't get played at any point. The objects in the virtual project are customized automatically so that nothing changes in the arranger view.

This function deletes physical data and does not feature an "Undo" option. You should therefore apply this command carefully.

If multiple virtual projects refer to the same wave files, all of these projects should be open as well. This is the only way to monitor and prevent data loss.

Using the "Remove unused samples" function saves storage space, but corrections to the objects' lengths are limited, since all audio data outside the object borders are removed. For this reason, specify backup reserves with "**Save additional samples for each object**". This number of samples is left in front and behind the object borders in the audio material. The default value is 22,050 samples, which corresponds to 500 ms at a 44,100 kHz sample rate.

The dialog lists all of the wave projects used in the virtual project. You can see the total memory space used by the wave projects in the "Size" column here, and beside this the free space in the "Unused" column. In the "Edit" column, you'll find a check box for each file that may be used to add files to the process. Only those files that contain unused sections of audio data are selected. To keep these, remove the check.

Note: The list may also contain files not actually referenced by the VIP, but referenced by the VIP's undo chain. These files are 100% unused samples and will be deleted completely when checked.

For example, if you have discarded an entire recording session, you can delete the unused files as well. However, if you had opened audio material from other sessions or your private sample library and didn't use them any more after that, you should uncheck these files so that they don't get deleted.

We recommend deleting the "Undo" chain and closing all unused wave projects before using the function.

Tip: If the relevant audio files are required later on for additional production work, the following procedure is recommended for archiving the finished production:

Save your entire project in a new folder ("File -> Save complete VIP in..."). In the save dialog, select the command "**Copy only samples used in VIP**". Now only those audio files/samples that are actually required by your archiving project will be found in the few folder. You can now save the content of this folder to a backup medium (e.g. CD-ROM or DVD).

Delete Freeze Data

Use this function to delete unused data that was created when freezing but is no longer needed after "unfreezing".

Close Project

This menu point closes the active project.

Project Properties

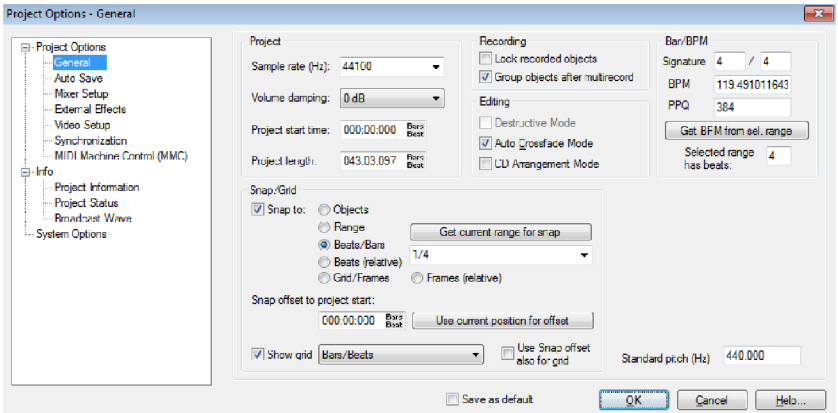
Snap and Grid Setup

In the general project options dialog you can choose the kind of grid and its activation.

Shortcut: I

Virtual projects have an object grid, area grid, beat grid, and frame grid. In the object grid, the objects may only be moved to the start, end or the hotspot (view page 802) ("Object -> Set hotspot/delete hotspot (view page 802)"). The reference point is normally the front edge of the object that is to be moved. As soon as a hotspot has been defined for the object, this is what will be used. If several objects are selected, alignment takes place at the forward edge of the hotspot of the object that was

selected last, and is currently located under the mouse cursor. The intervals between the selected projects stay constant.



Grid: Checking the tick box turns snap on globally.

Object grid: This option activates the object grid. This lets objects snap exactly to the edges of other objects.

Snap to range: Activates the range grid with the option to use the current range as the basis for snapping.

Grid/bars: Activates a grid with bars as the basis. The objects snap to the the next beat grid.

Grid/bars (Relative): Activates a grid with bars as the basis. This means that the relative distances between the objects adhere to the beat grid when you move them about.

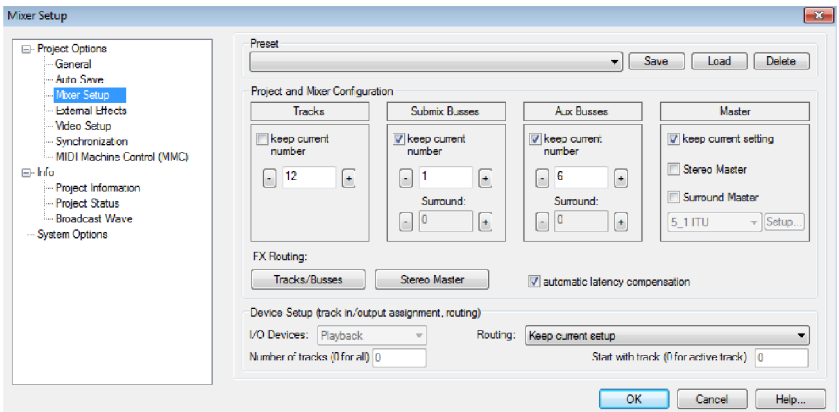
Grid/frames: Activates the frame-based grid.

Snap offset to project start: Here the snap offset can be set relative to the beginning of the project. "Use current position for offset" specifies the current position as the grid's zero position.

Use snap offset also for grid: The snap offset is used as a reference size for the grid.

Mixer Settings

In this dialog you can easily configure, save and load project and mixer settings.



Shortcut: Ctrl + Shift + M

Project and Mixer Configuration

Here you can set the number of tracks, AUX buses and submix buses. You can also determine the effects and signal routing for the tracks and the master section.

Automatic Latency Compensation: Activate this option for automatic latency compensation of VST plug-ins. This is active by default.

Master Routing

- Keep Current Setting: Each track is assigned to an audio device (or a submix bus). The final mix then takes place in external equipment or on the sound card.
- Stereo Master: Normal mixing in the Sequoia Stereo Master. If the sound card has several outputs, select one in the mixer (in the "out" bar below the volume faders) or in the playback parameters (shortcut: "P").
- Surround Master: If this setting is active, master channels are provided for surround mastering.

Detailed information about the Surround setup is available in "Surround Sound" -> "Surround Setup (view page 477)".

Keep Current Number (Tracks, AUX and Submix Buses)

Check this box to prevent the current settings from being overwritten when loading a mixer preset. The current status of this checkbox is also saved along with a preset. If you quickly want to add a few tracks, (e.g. 4 AUX buses and 4 submix buses), you only have to load the "4 Busses 4 AUXes" preset.

When loading a preset, the respective tracks and buses will be added where the box is not checked. In other words, place a check for settings that shouldn't be included in the preset before saving.

Device Settings

In this section you can conveniently select the input and output device assignment for several tracks simultaneously.

I/O Devices: Here you can define the routing for playback, recording or both.

Number of tracks (0 for all) / **Start with track (0 for active track)**: This option permits a change to I/O routing for a certain number of tracks. If you set the number of tracks to 4 starting at track 8, the selected routing setup is applied to tracks 8-11.

Note: To be able to switch these commands to active, you must first replace the "Keep current setup" option in the neighboring "Routing" field with another option from the selection menu.

Routing

Keep current setup: No routing assignment takes place if a preset is loaded.

Assign all tracks to stereo master: All tracks (incl. AUX and submix buses) are routed to the master. **L/R panning** sets the panorama settings for the tracks to alternate between left and right. Use this option if stereo sources are recorded as mono pairs.

Assign tracks to available stereo/mono devices: All tracks are assigned to the different output devices. In the system options window (shortcut: "Y") you can define which devices are available in Sequoia and which are not. When routing to mono devices, the tracks are also placed right/left in panorama alternatively.

Assign tracks to surround channels: The corresponding tracks are created here after the Surround setup has been selected .

Use setup from preset: The settings saved in the preset are used.

External effects

Detailed information on this topic can be found in "Effects -> Breakdown and functionality -> External hardware effect integration (view page 237)".

The integrated video engine allows you to add video objects to the first track. Video objects can also exist in other tracks, but they will not be played back, and will appear

shaded. Various video files like AVI, DV, MOV, MPEG, MXV can also be loaded. WMV or image files such as BMP or JPEG can be loaded into the arranger window using drag & drop or through "File -> Load/import -> Load video file" (view page 619).

Please ensure that the video files included in a project are identical in terms of dimension, aspect ratio, and frame rate. If possible, use only videos that work exclusively with keyframes and without interframes (DiVX or MPEG are not very suitable). If necessary, you can convert files into other formats and resolutions using the video editing program "Movie Edit Pro". We recommend the DV and MXV formats.

The video soundtrack can be loaded into the project. This process converts MPEG-2 audio into WAV format.

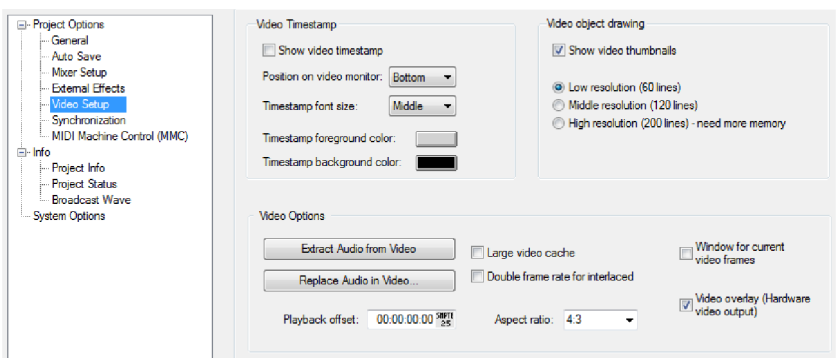
You can move or cut video objects just like all other objects, but no fades will be used. It is possible to edit a video using Source Destination Cut commands.

When working with many single bitmap files, keep in mind that these will be loaded completely into RAM and not successively read from the hard disk.

You can **right-click the video object** to

- open the "Video settings" window
- show/hide the video window
- replace the video file
- move/shift the video object by entering values

Video Setup - Dialog Options



Show video time stamp: This option opens the time stamp display in the video monitor. You can display it in three sizes either at the top, in the center or at the bottom of the project window.

Video object drawing: The option "**Show video thumbnails**" displays thumbnails in the video track in different resolutions. You can improve system performance by unchecking the box to shut off the video object display completely. The video file will continue to be played in the video monitor. VIP draw mode 2 ("Tab" key) can be used to display only the object start and end frames.

Extract audio from video: This button extracts and converts audio files from an AVI file into a wave file for insertion into the current project.

Replace audio in video: This option can be opened using the "Export video sound" function to replace the existing audio track in an AVI video.

Playback offset: Set a positive or negative playback offset in this field.

Large video cache: This option allows you to allocate more cache for the video clipboard.

Double frame rate for interlaced: This enables you to double the image repeat frequency when interlacing so that videos can be viewed without any flickering.

Aspect ratio: Select an aspect ratio of either 4:3 or 16:9.

Window for current video frames window: The beginning and the end are displayed precisely according to the frame for range and object markers. This window is divided into two sections and shows two frames at a time. If you select an object or a range, the first and the last frames of the object are displayed. Object borders and fade changes can also be added to video material in this way.

Video overlay (hardware video output): "Video overlay" indicates that the graphics card calculates the video display and the video picture is positioned on top of the actual Windows screen (overlay).

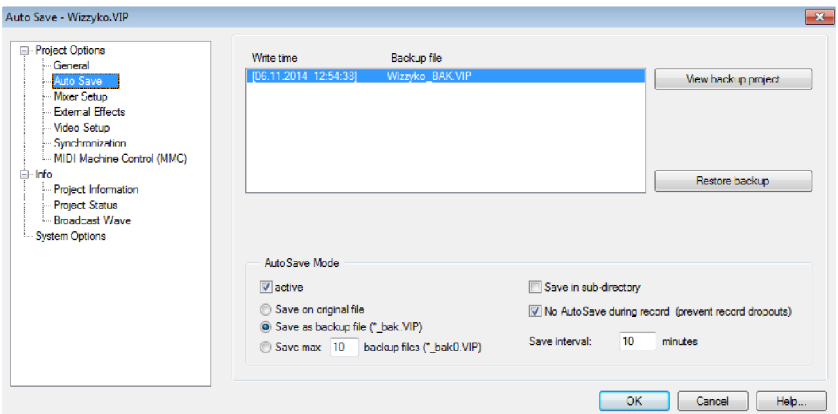
Playback Options

More information is available in "Playback -> Playback parameters (view page 830)".

Shortcut: P

Save automatically

The **Autosave** dialog is available via the project options.



Here you can adjust the settings for creating backup projects. The button **View backup project** opens the selected backup in the arranger.

The button **Restore backup status** ensures restores the project to the status of the backup. If the current project was changed beforehand and not yet saved, you can save the latest status with the addition of _OLD.vip.

The "Autosave" can be activated by clicking this checkbox. Select between the following options:

Save to original: Overwrites the original file with the current status at specific intervals. If the **Save in sub-folder** option is selected, the previous status, i.e. the last but one status, will also be updated in the **Backup** sub-folder.

Add backup file (*bak.VIP): Adds a backup file and updates this at specific intervals. If the **Save in sub-folder** option is selected, the backup file will be updated in the **Backup** sub-folder each time it is saved.

Max. 10 backup files (*bak0.VIP to *bak9.VIP): Specify save intervals for up to 10 backup files and number them. If the **Save in sub-folder** option is selected, the backup files will be updated in the **Backup** sub-folder each time they are saved. After 10 backup steps, the last backup file is given the extension *bak9.VIP: in contrast, the oldest backup file is deleted whereas the others move down in the numbering.

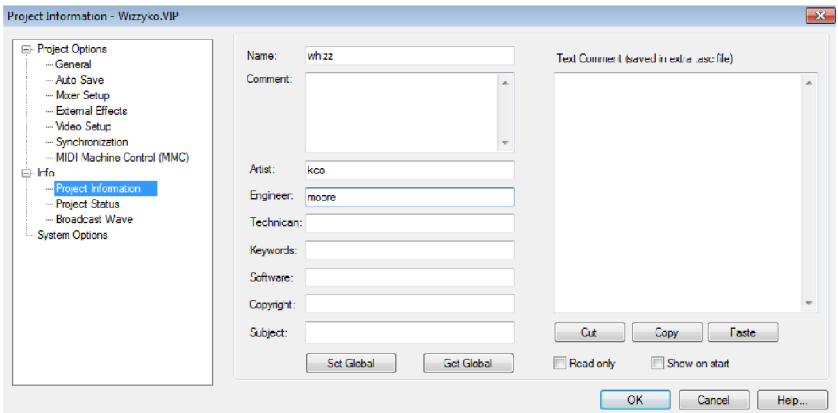
No automatic save during recording (avoid recording errors): Automatic saving during recording is set to inactive to avoid recording errors. Deactivate this option in order to save automatically during recording.

Save automatically after finished recording: When this option is activated, the project is automatically saved directly after the recording has been completed.

Project Information...

Enter information about the project here.

There is also a text field for comments. These may be set to show each time the project is opened.



Project Status...

This indicates all project status information like the name, path, number of ranges, markers and objects, wave files used in the current project, or the date of creation. You can also create a text file which includes all of this information.

Broadcast Wave Manager

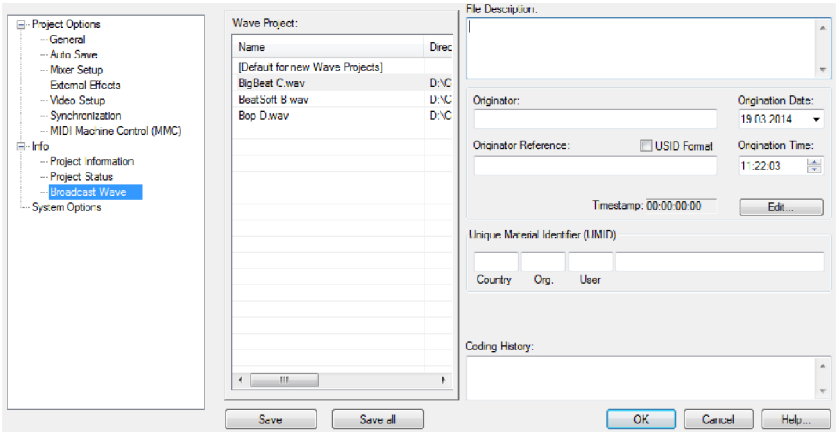
The Broadcast Wave Extension makes it possible to save information about the audio files as metadata in the extension of the Broadcast Wave Format (BWF) called the "Chunk". This metadata can basically be applied in a proprietary manner, but observing the respective EBU and SMPTE guidelines is recommended. The file ending of the Broadcast Wave format remains ".wav".

Wave Project

Here you'll see a list of all the WAV files in the project. To add or read specific information, simply click on the corresponding file. Its current BWF information will be displayed in a dialog on the right.

For newly recorded audio material you can define the metadata that should be written to the Broadcast Wave Extension.

This metadata can late be read or extracted and used for other purposes such as the management of audio files in a database.



Broadcast Wave Manager - List of individual fields

All of the settings values are saved according to the project and used for new audio files.

File description: You can use this field any way you like. You can enter a text of up to 256 ASCI characters.

Originator: This field contains details on the origin of the file, e.g. the description of the producer. Maximum of 32 characters.

Originator reference: This field is determined by the originator. This can be an internal reference number, for example. In the area of EBU, the EBU Recommendations R 99-1999 define how this field should be completed. To assign the entry of this property, activate the "USID Format" option. Afterwards you can format the entry according to the EBU recommendations. Maximum of 32 characters.

Date: The date of the file's creation is displayed and can be edited, e. g. if audio material was saved for the first time as a file, even though the recording is somewhat older and the date needs to be valid as a reference. If the entry is processed in the BW Manager, a value will result that is independent of the file properties.

Time: The time at which the file was initially created is shown here. As with the date, this is automatically created from the file properties but can be edited retroactively.

Only Sequoia: ISRC: Enter the ISRC (view page 665) of the file here. This is a 12-digit ID number that provides specific information like the originating country of the label, the label's company number, the year, and a sequential title number. ISRCs are used by the music industry for the identification and accounting of music titles.

Timestamp: The timestamp saved in the BWF extension is displayed here. This is the timecode for recording the file, which is identical with the timecode of the first recording if the recording was synchronous. For other applications, this timestamp can provide information on the time on the day of recording. Click on the "Edit" button to adjust the timestamp or to apply it from the object position. Here all audio files are assigned the timestamp from the object positions or the timestamp assignment is undone.

Unique Material Identifier (UMID): The specifications for UMID are regulated by the SMPTE (Society of Motion Picture and Television Engineers). The corresponding documentation has the code number SMPTE 300M-2000. We recommend observing these guidelines and agreements regarding the use of the UMID before using this feature, especially the sections that specifically apply to how you want to use it. Use of UMID is not absolutely necessary for a valid BWF.

Only Sequoia: Loudness: Here you can calculate the loudness metadata as BWF files or update it automatically.

Coding history: In addition to information about the file format (A: encoding, e. g. PCM, F: sample rate, W: bit width, these values are only used with non-transparently encoded material like MPEG or MP3, M: number of channels), each entry in this field contains a T value. This is a comma-free text string for entering the serial numbers of analog tape recorders, codecs, dither types, AD converters, or special signal editing applied to the file (such as de-noising).

An entry is made when a file is recorded in Sequoia. If this file is processed again (e. g. by bit width reduction or MPEG encoding), an additional entry is added.

Use of the coding history is regulated in the EBU Recommendation R98-1999.

Broadcast Wave Manager - Advanced Options

Update Loudness Metadata Automatically: Here you can automatically update the loudness metadata in BWF files.

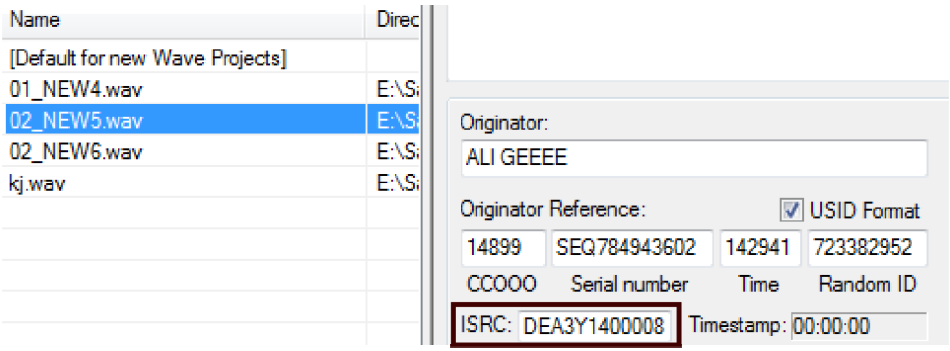
Save BWF Graphic Data in Wave Files: If this option is activated, Sequoia creates a so-called "Peak Chunk" that saves graphic information as metadata in the file. In Sequoia graphic data is normally saved separately in .ho files.

Read BWF Graphic Data and Use Data in Sequoia: If this option is activated, the included "peak chunk" is used instead of the .ho file.

ISRCs in the Broadcast Wave Manager

In BWF format you can save the ISRC (International Standard Recording Code) along with the metadata in the audio file. This info is then available when the file is

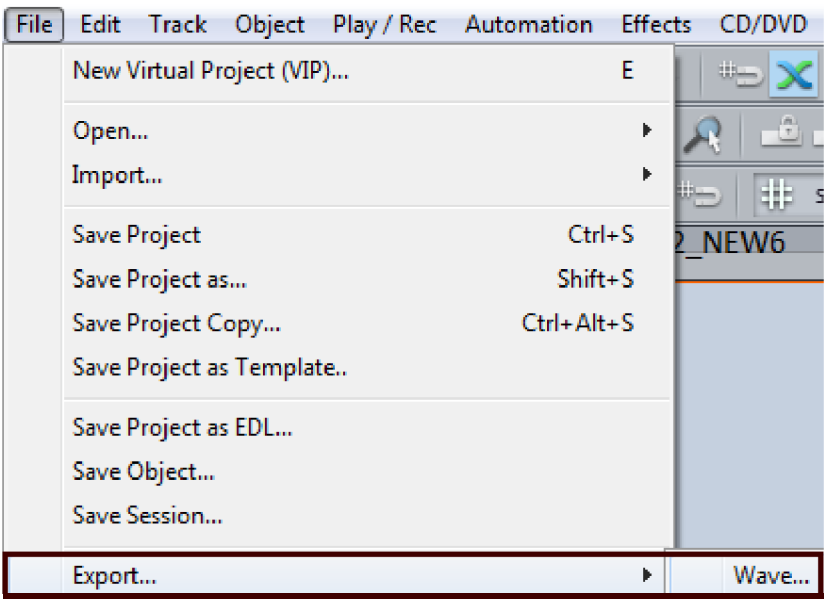
exchanged during the production process. If a WAV file already has an ISRC, it will be displayed in the Broadcast Wave Manager. Here you can edit the ISRC or enter it for a newly created music file.



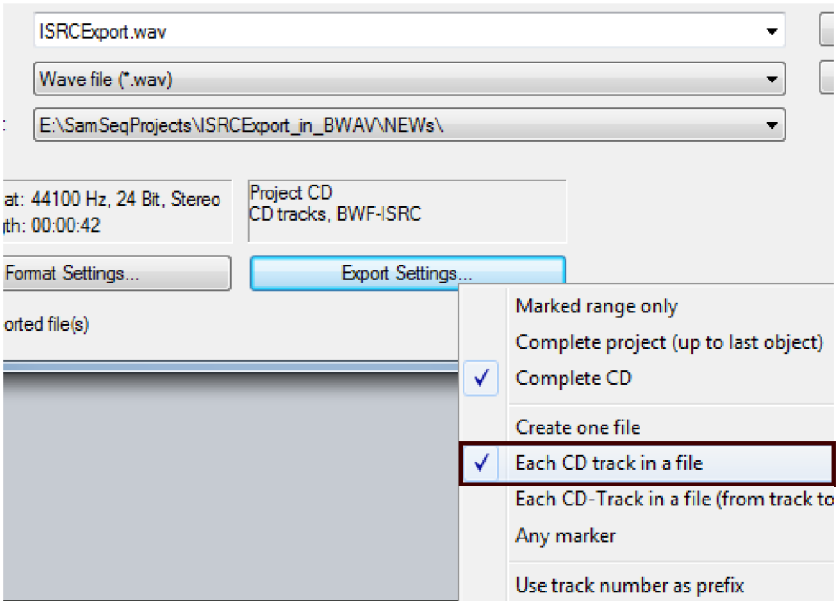
To transfer the entry to the WAV file, click on "Save" or "Save All".

ISRC Export in BWF Files

To export ISRCs that have been set in track indices to audio files, select WAV export



with the setting "Each CD track in a file". Because each track requires its own ISRC, it is not possible to enter only one ISRC for all included files. The label "BWF-ISRC" shows in the "Export Settings" dialog that ISRCs are written to the BWF files.

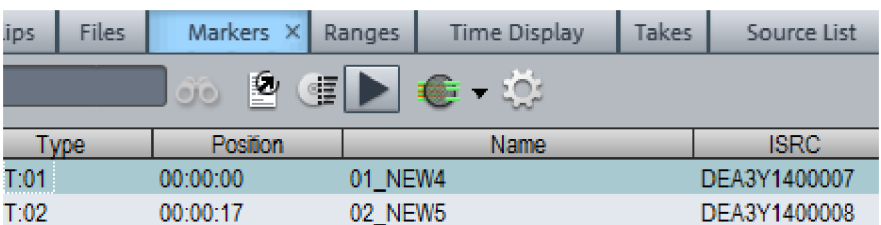


ISRC Import When Loading WAV Files

1. Load the WAV files sequentially onto the first track. To do this go to "File" > "Import" > "Load Audio File" > "Options" > "Load all files to selected track".

2. Now you can set track indices that contain the respective ISRCs by going to "CD/DVD" menu and selecting the function "Indices" > "Set Track Indices on Object Edges". You can use these when burning CDs or for DDP export.

3. The ISRC of the imported file is displayed in the "ISRC" field of the Marker Manager.

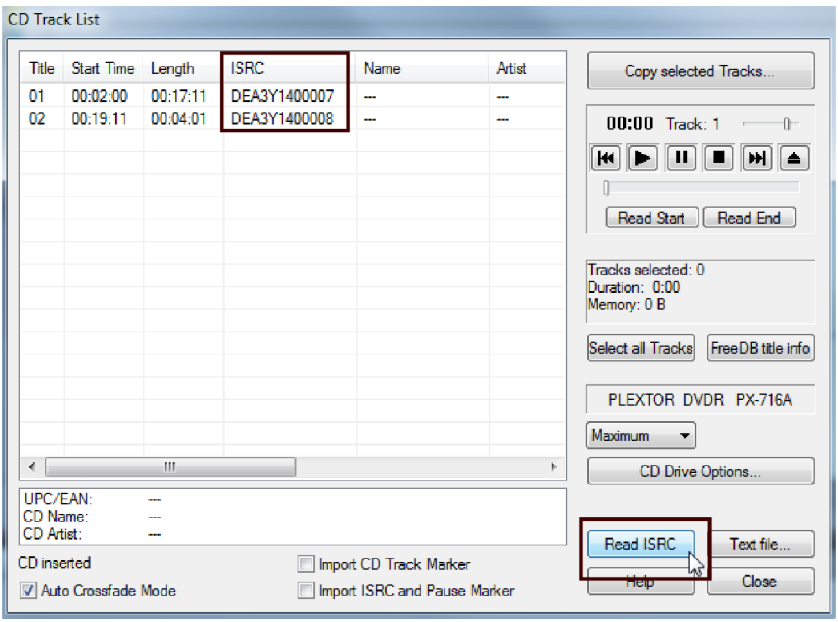


ISRC Import from CD Tracks

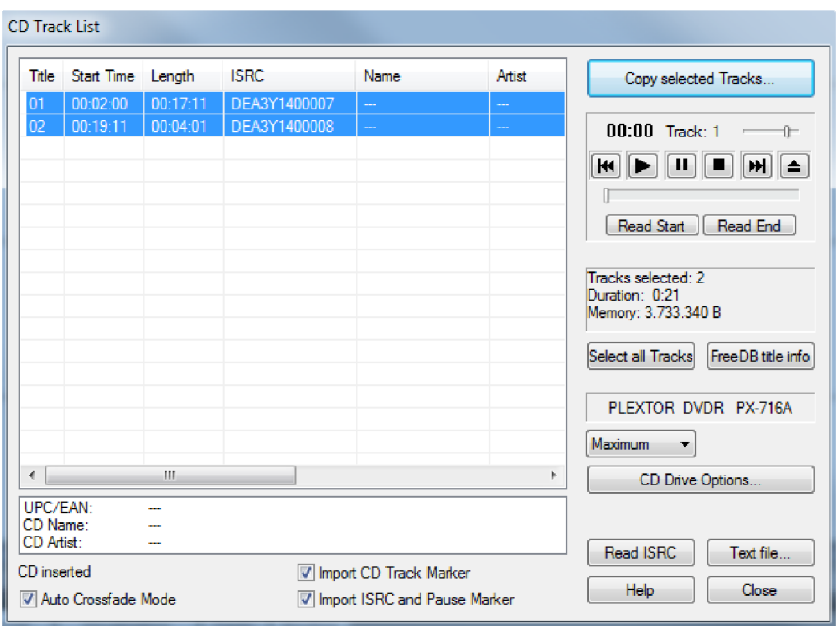
When importing single CD tracks each ISRC entry is transferred with the imported WAV file.

1. Open the dialog in the menu „CD/DVD“ > „Import Audio CD Track(s)...“ (view page 1090)

2. Activate the option "Apply ISRC and Pause Indices". To view the ISRC entries in the import dialog, click on the "Read ISRC" button.



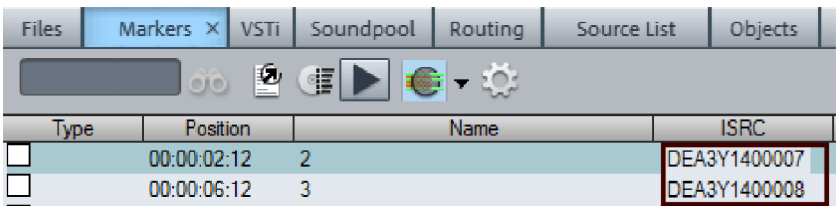
3. Select the tracks to be imported and click on the "Copy Selected Tracks..." button.



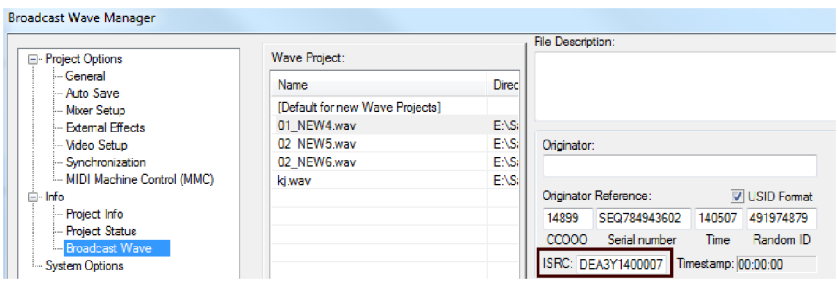
In the "Audio Import" dialog that appears select the option "Each track in its own file with names". This imports the selected tracks and names them automatically according to the following scheme:

(Track number) (Track name)
(File name_(Track number)
(Track number)_(File name)
(Track name)

4. The ISRC of the imported track is displayed in the CD track markers of the Marker Manager



and in the Broadcast Wave Manager of the imported object.



CD Arrange Mode

If this is activated, Sequoia arranges recently added objects to insert a Red Book Standard-compatible pause between the objects.

Detailed information about "CD arrange" mode is provided in "CD arrange mode (view page 1117)".

Offline Audio Editing (Destructive Wave Editing Mode)

This menu item activates or deactivates offline audio editing in the wave project window. If this feature is deactivated, Sequoia reverts to "Virtual Wave Editing" mode.

Detailed information about destructive editing is available in the chapter "Working in the Project Window > Audio Editing in Sequoia (view page 62)".

Program Preferences

System Options (System/Audio)...

Specify all of the important settings for Sequoia in this dialog.

These options include:

- **System options**: Audio system, audio devices, MIDI, hardware controller, metronome, record, play, performance, extended buffer settings
- **Program settings:** General, undo, object lock definitions, 4 point cut, crossfade editor
- **Keyboard, menu & mouse**: Keyboard/menu, MIDI editor, special keys, mouse, mouse wheel.
- **Design**: Skins, view options, colors
- **Effects**: Dithering, resampling/bouncing, VST/ReWire, destructive effect calculation
- **Option administration**: Reset program settings to default values
- **Project options**: Switches the tree structure of the project options.

Explanations and references regarding system settings are located directly in the "System options" dialog.

Audio Setup

Driver System

Please also read "System settings -> Audio setup (view page 594)" for more information.

Buffer Settings

Please also read "System settings -> Audio setup (view page 594)" for more information.

Device/Driver Communication

Detailed information about this topic can be found in the chapter "System Options" > "Audio Setup" (view page 594).

Monitoring settings

There are different options for monitoring available here. These settings are used as presets for new projects.

For more information about monitoring, please see "System settings -> Monitoring settings (view page 596)" for more details.

Audio Devices

Detailed information about this topic can be found in the chapter "System Options" > "Audio (view page 604) Devices".

MIDI

For more details about MIDI settings, see "System settings -> MIDI settings (view page 604)".

Hardware Controller

Detailed information about this is available in "Hardware controller (view page 520)".

Metronome options

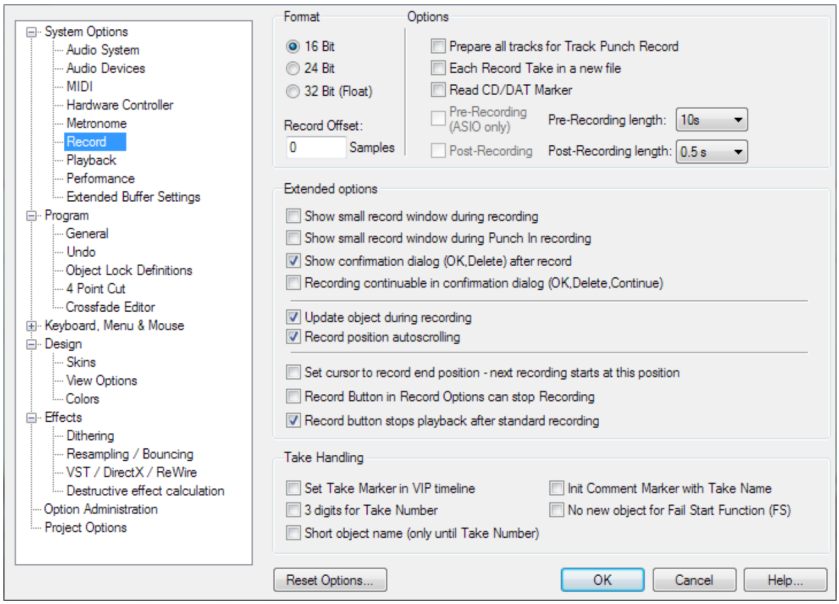
Detailed information about this is available in "File -> Program Preferences -> Metronome Options (view page 95)".

Record

This dialog enables the selection of various dialog configurations and playback marker behaviors when recording.

Format: Choose from 16 bit, 24 bit or 32 bit (float).

Enter the **Recording offset in samples** to ensure consistent movement of your audio recordings with regard to the existing audio material in the arrangement.



The following **options** are available:

- **Prepare all tracks for track punch record**

If this option is activated, all set recording devices will open during playback so that you can start a recording on all tracks using the track record button.

- **Save each take in a new file**

This option results in each take being saved to a new file. Loop recordings will be processed as one take and saved to one file.

- **Read CD/DAT Marker**

DAT devices and several professional CD players output digital marker information via an SPDIF output e. g. CD track markers or DAT markers). With this recording option all marker information is read from the soundcard's SPDIF input, provided the selected audio device supports it.

Pre-recording (ASIO only): If recording is activated from the stop state or during playback, this function inserts audio material that you have added at the beginning of the recording to the beginning of the current recording. You can set pre-recording lengths of 2, 5, 10, 30, 60 and 120.

If you drag out a recorded object to the left, you'll see the set pre-recording time before the start of the actual recording.

Post-Recording can also be activated in the recording options in order to record up to 2 seconds of audio material in the background after the actual recording is ended. If you drag out the object end of the recording to the right, you'll see the display of the part that was recorded after it was stopped. The preset for post-recording equals 0.5 seconds.

Advanced options

Show small record window during direct recording: Displays a small non-modal recording window with the most important recording controls while recording.

Show small record window during direct recording: Displays a small non-modal recording window with the most important recording controls while recording.

Show confirmation dialog (OK, Delete) after record: If this option is checked, a dialog will appear after the recording process to choose whether to keep or discard the recording.

Note: If you select the option "Display confirmation window after pressing the record button", the recording will always be saved straight after it has completed. On the one hand, the best security against data loss as a result of inadvertent deletion is guaranteed, and on the other hand, obviously unnecessary takes are saved to the hard drive and take up space there until they are manually deleted.

Recording continuable in confirmation dialog (OK, Delete, Continue): With this option, when you click on the record button during a running take, a dialog will appear where you can choose to delete, stop or continue the recording.

Update object during recording: This option increases the recorded object constantly along with continuous recording.

Record position autoscrolling: If this option is activated, the window will scroll along with the recording.

Set playback marker at the end of recording - the next recording will start at this position: If this option is activated, the cursor will be set at the end of the recording, and playback or recording will start at this position.

Recording Button In Recording Options Can Stop Recording: Specify that the **"Record" button** in the **Recording Options** (shortcut: **Shift + R**) is able to start or stop the recording process.

Record button stops playback after standard recording: If this option is activated, the running recording and playback can be ended by clicking the "**Record**" button in the transport console.

Take Handling

Set take marker in VIP time bar: This option places markers at new takes in the project.

3-figure take number: The take numbers are created with 3 figures starting with 001 (instead of 01). This prevents complications when there are more than 99 takes.

Short object names: Object names will be set only to take number.

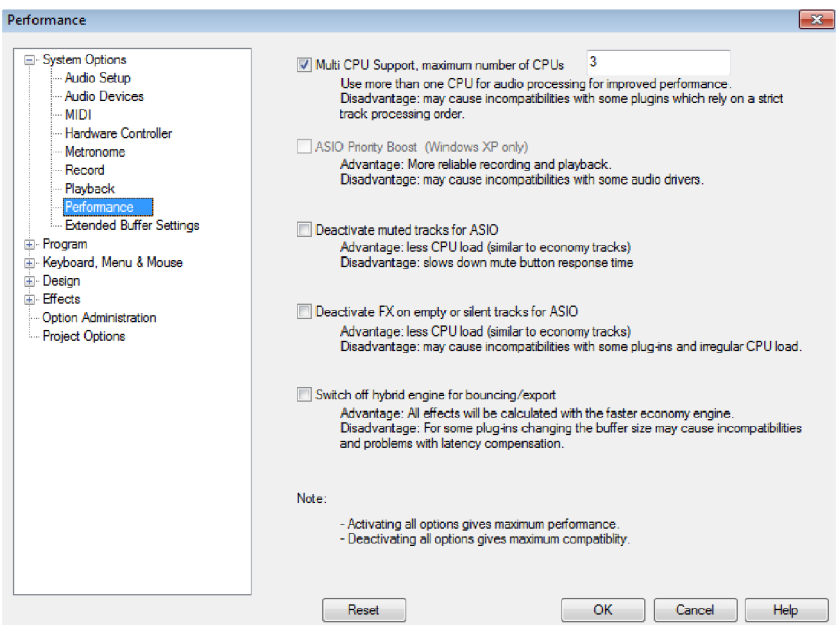
Initialize comment with take names: Comments are initialized with the take names.

No new object with the false start key (FS): After pressing the "False start" (FS) key, a take is recorded again with the same name and in the same object.

Playback

More information about playback settings is available in "Playback -> Playback parameters (view page 830)".

Performance



The options featured under "Performance" are provided to set up your system's performance optimally. Among other things, "**ASIO silence economy**" is available for switching off empty or silent tracks with ASIO.

Multi-CPU support: Sequoia supports use of multi-CPU/multi-core/HT workstations that distribute the load of various tasks across several processors. If you select the driver system "MME" or "WDM", the audio thread editing will run mainly on the first CPU. Disk I/O operations, graphics, and video integration are processed by additional CPUs.

"ASIO" activates this driver system and dual CPU support; the mixer tracks and their effects are distributed between the initial CPUs. Objects (including their effects), integrated video material, and the graphics engine are distributed between additional available CPUs. The program is currently optimized for work with up to eight cores; working with 3 cores is preset.

If you own a system with multiple CPUs, you will be able to get higher performance from your system for processing audio, provided that you activate this option. However, using multi-CPU support can lead to incompatibilities with plug-ins that are specified for a set track processing series.

ASIO priority boost: Among all threads of the application, special priority is granted to the ASIO thread. This option is a preset and does not need to be modified. This increases the reliability of your audio recordings as well as that of the playback.

Note: We recommend deactivating this option for LYNX and TASCAM devices.

Deactivate muted tracks for ASIO: Use this option to reduce the CPU strain on your system. When muted, however, there may be some delays.

Deactivate FX on empty or silent tracks for ASIO: Use this option to reduce the CPU strain on your system even more. This function can, however, cause fluctuating CPU efficiency.

Switch off Hybrid Engine for bouncing/export: This option calculates all effects during the bouncing/exporting process using the faster "Economy Engine". This may cause irregularities when some plug-ins attempt to compensate for latency.

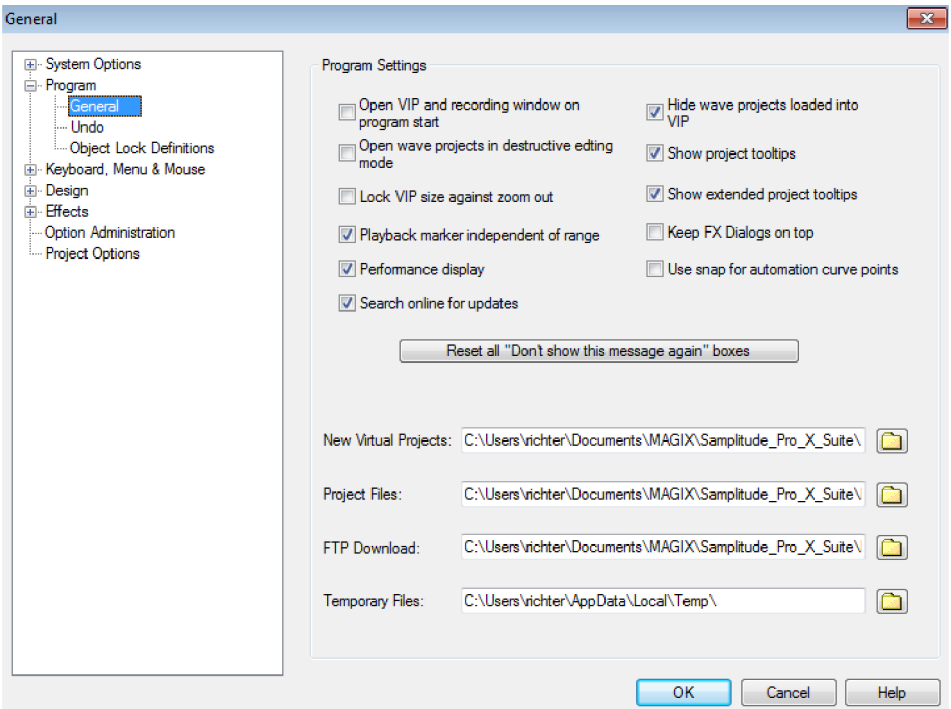
Note: If every option is activated, maximum system performance will be achieved. If each option is deactivated, maximum compatibility will be achieved.

Advanced buffer settings

Detailed information about this is available in "File -> Program Preferences -> Advanced Buffer Settings (view page 706)".

Program

General options



Open VIP and record window on program start: If this option is active, Sequoia will automatically open an empty virtual project (VIP) and the "Recording parameters" dialog when the program is started.

Open wave projects in destructive editing mode: If this option is activated, audio files are opened in offline editing mode.

Lock VIP size against zoom out: If you zoom the virtual project out beyond the section size, this option will automatically stop it from being extended.

Playback marker independent of range: This option allows the playback marker to be set independently in "Loop" mode and playback to be started outside of the selected loop range.

Performance display: This option activates the DSP performance display in the lower left corner. Your PC is overloaded if the displayed rate is 100% or higher. If this is the case, try one of the following steps:

- Reduce the number of realtime effects in the mixer or in the object editor.

- Reduce the number of tracks by muting tracks that are not needed.
- Increase the VIP buffer size (keyboard shortcut: Y) to 16000 or 32000 samples.

Please note the comments displayed in the status bar if an overload occurs during realtime previewing of complex effects such as the DeNoiser or FFT analysis filter.

Search for online updates: When this option is activated, Sequoia will automatically search for updates each time it is started.

Create fades for new objects with Auto Crossfade mode active: When loading audio files into the VIP, the objects will be created without automatic fades even if Auto Crossfade mode is active, as it is assumed that such samples have already been cut properly. If you still want fades in this case, enable this option.

Show project tooltips: If this option is active, additional information is displayed when the mouse pointer hovers over an interactive element.

Show extended project tooltips: If this option is activated, the extended tooltips are displayed if the mouse pointer is held over an interactive element.

Keep FX dialogs on top: This option keeps all effect dialog windows "on top" of all other windows.

Use snap for automation curve points: Select this option to set automation curve points while moving set values on the grid.

Reset all "Don't show this message again" boxes: Click this button to reactivate all tips dialogs that have been deactivated.

Path settings

New Virtual Projects: All new virtual projects as well as recorded and imported wave projects will be saved through this path.

Project Files: All other wave files which are to be saved on the hard drive and cannot be allocated to a specific project are saved through the specified path.

FTP Download: All files downloaded via the integrated FTP client will be saved through the specified path.

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.

Undo

Detailed information about this is available in "File > Program Preferences > Undo Definitions (view page 702)".

Locking objects

Detailed information about this is available in "File -> Program preferences -> Lock Definitions (view page 708)".

4 Point Cut

A dialog window for the cut operations and the source/destination cut (view page 410) options opens.

Crossfade Editor

This menu entry opens the dialog window for the crossfade editor's settings (view page 439).

Detailed information about the crossfade editor is provided in "Crossfade editor (view page 425)".

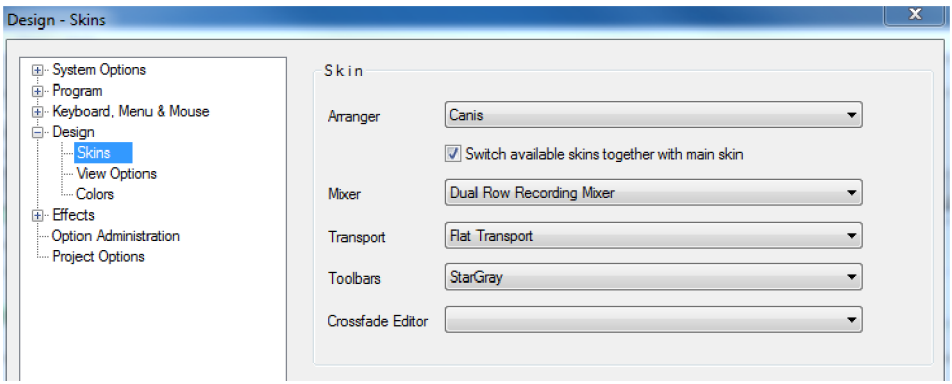
Keyboard, Menu & Mouse

Detailed information can be found via "Menu File -> Program Preferences -> Edit Keyboard Shortcuts and Menu (view page 688)".

Designs

Interface (Skins)

In this dialog, select different skins for the project, the mixer, and the transport console. Use the various skins to adjust the color and graphical display of the Sequoia interface.



There are specific skins adapted for working with certain processes like the "Multitrack Mixer", "Dual Row Recorder" or "Recording Mixer" which can help keep the overview clear in certain working situations.



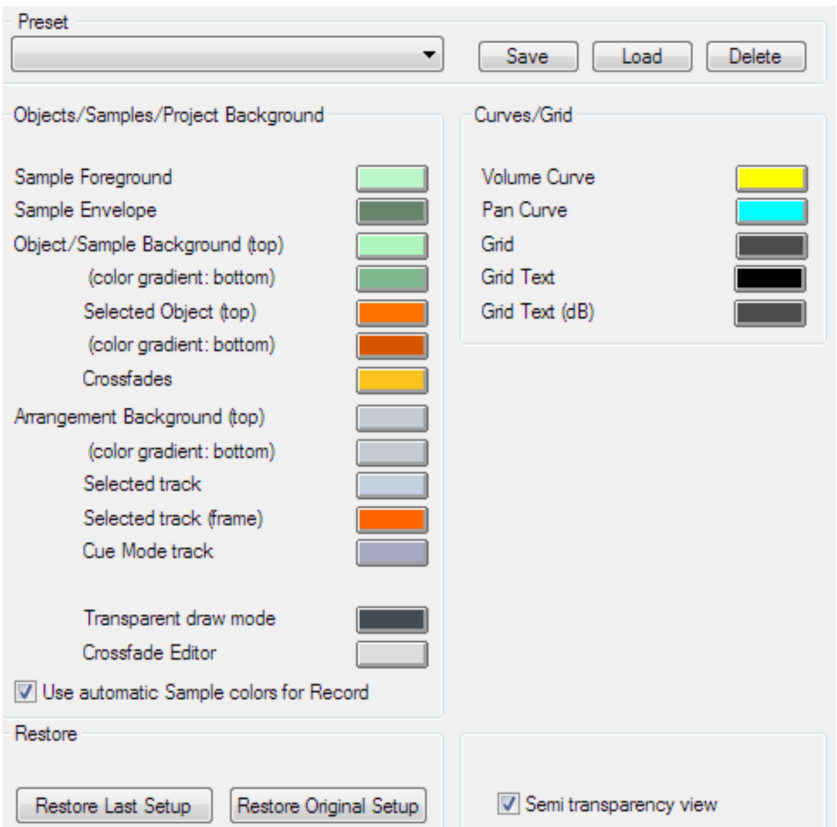
Skins can be selected by clicking on the icon at the top left of the mixer title bar (view page 363) or on the menu bar of the arranger.

Project Display

Detailed information on display options can be found in "File > Program Properties > Display Options (view page 694)".

Colors

All colors used in the program can be set here. You can also load and save these colors as presets.



Use automatic sample colors for record: A separate random color is chosen for each recorded object per track.

Redo

Restore last state: The last status of the color settings prior to opening the dialog was opened is restored.

Reset...: 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.

Reset object colors: This option resets all object colors. This may also be necessary when importing a project from a different PC with different color settings, or if the objects have colors that deviate from the standard object colors as a result of automatic assignment during recording.

Reset track colors: Resets the track colors.

Reset object colors to track colors: Sets the object foreground colors of the waveforms to the track colors.

Object background to track colors: Sets the object background colors to the track colors.

Semi transparency view: Some effect dialogs and editors feature transparent displays. These may be deactivated here to increase performance.

Effects

Dithering

You can find out more about this by reading "Dithering options..." (view page 703).

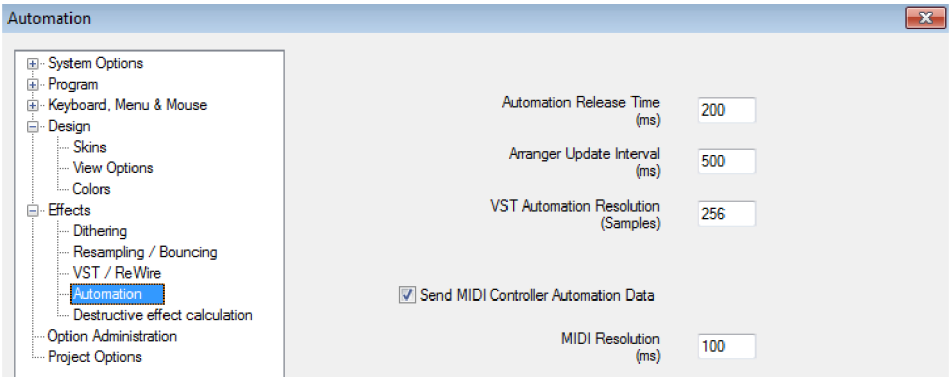
Resampling / Bouncing

Detailed information about this can be found in the section „Resampling / Freeze Options“ (view page 707).

VST/ReWire

Detailed information is available in the section "Mixer -> Effects routing/plug-ins dialog -> VST/ReWire setup plug-in buffer (view page 390)".

Automation



Automation Release Time (ms): This sets the fader return time in ms for the write automation modes.

Time for entries by users: As soon as a controller curve is updated (i. e. new controller data for the curve is received). If no further new data is received, Sequoia transfers the original data again after this time span has elapsed.

Arranger Update Interval (ms): When mapping controllers you can set the frequency of the display update in milliseconds in this field.

VST Automation Resolution (samples): Time constant (in milliseconds) for sending current controller values: When starting playback, the controller data is transferred after this time has lapsed if its value changes (within the frame of 7 bit MIDI resolution). During playback, the VST controller values are exported at this rate and sent to the corresponding VST plug-in. During playback the VST controller values are retrieved within this time gap and are sent to the respective VST plug-in.

Note: VST2 does not support sample-exact automation, only parameter changes at buffer limits. To increase the automation resolution enter a low sample value here. Lower values can lead to a higher load on the CPU for automated effects.

Send MIDI Controller Automation Data: Here you can globally activate or deactivate the sending of MIDI controller data. Program-internal automation is not affected by this.

MIDI Resolution (ms): Time constant (in milliseconds) for sending current MIDI controller data. The controller data is sent during playback after this time period if the value changes (within the frame of the 7-bit MIDI resolution). During playback the MIDI controller data is retrieved and sent to the respective MIDI device.

Destructive Effect Calculation...

Choose in the advanced destructive effects calculation options (view page 858) if you want to add the effect to the original file when editing destructively, write it into the effect file, or create a new effect file for each calculation.

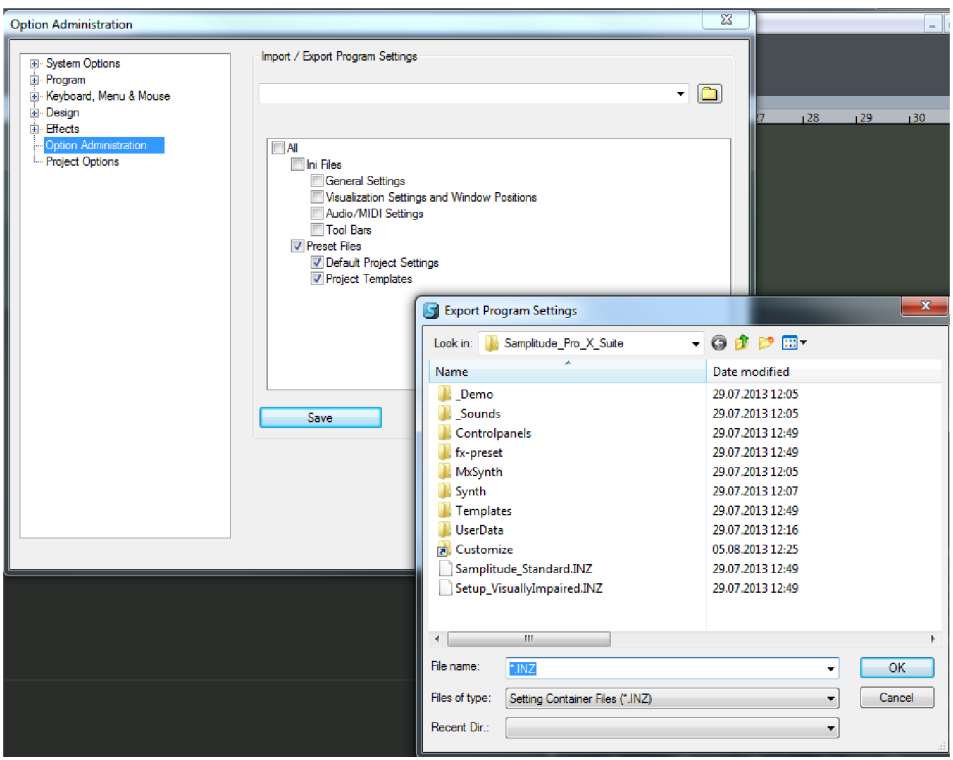
Loudness

In this dialog window you can specify the basic settings for loudness normalization (view page 862) compliant with EBU R128/ ITU-R BS.1771.

Options managerment

The Sequoia folder includes so-called "Ini" files and preset files, which contain program settings.

The following dialog enables the listed settings to be saved and loaded conveniently. This makes it possible to save the complete program settings onto a mobile storage device for use on other Sequoia workstations.

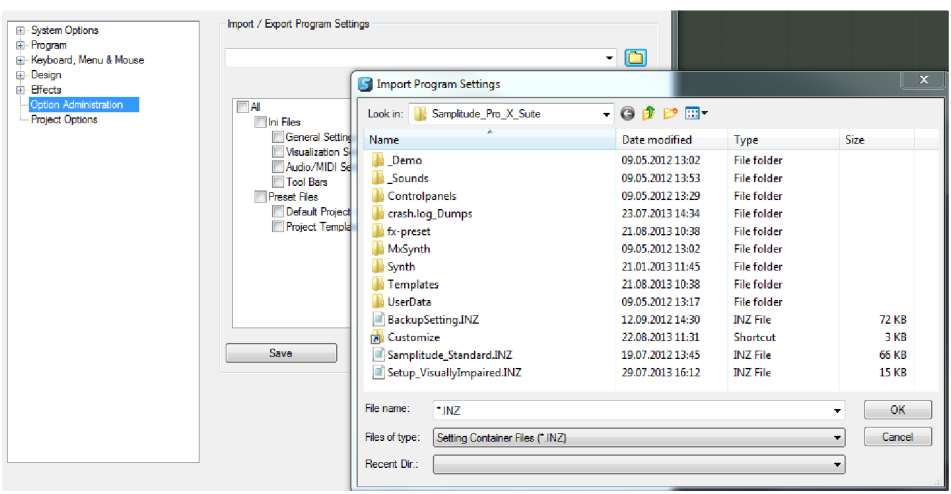


The program settings are saved as "**Settings Container Files (*.INZ)**".

Save: Press the "Save" button to save the current project and program settings in a "Settings Container". All of the program settings will be saved, including those that are not checked. The state of the individual check marks during saving only represents a pre-selection for loading. The saved file will now appear as a preset at the top of the list. Sequoia settings are provided for the following dialogs:

- General settings (view page 676)
- Settings displays and window positions (view page 694)
- Audio/MIDI Options
- Toolbars (view page 72)
- User administration (affects all users and user settings) (view page 711) (Sequoia only)
- Standard project settings (view page 607)
- Standard Crossfade-templates (view page 425) (Sequoia only)
- Project templates (view page 621)

Load: Select a program setting for Sequoia from the list above or import the settings via the folder button.



Specify via the tree structure which "Ini" files or "Preset Files" for the selected container file should be used. Now press "**Load and restart**" to activate the settings. Sequoia will restart and a backup of the last settings will be created.

Hint: Load your settings containers comfortably from the list box in the Start Wizard (view page 1155)!

The program **subfolder "Customize"** also features a series of "**INI patches**" that do not contain complete files, but rather only activate or deactivate individual settings. These may also be loaded just like a "Settings Container" via the folder button. After

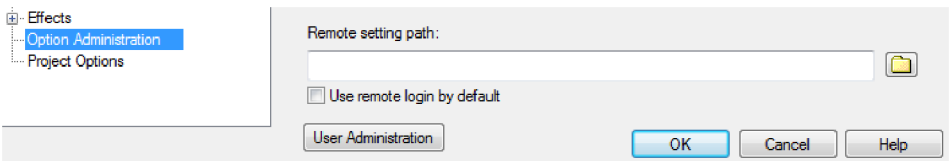
selection is confirmed, a separate dialog window featuring explanations about the respective "INI patch" will appear.

Note: During multi-user operation (view page 711) of Sequoia, loading of "settings containers" is not possible for users with the property "Fixed settings (view page 712)" or for remote users.

Network Path for Settings

Sequoia offers administrators (view page 711) the option to add path settings for remote setups via a network.

Click the folder icon to specify the "Network path to settings" for your network.



User administration: This enables the user to log in via the selected remote path (view page 713). Place a check next to "**Use network path**" in order to switch from a local login scheme to the network login.

Project options

General:

Please also read "System settings -> Project Options -> General (view page 607)" for more information.

Auto save:

Detailed information about this is provided in "File -> Project Properties -> Project Auto Save (view page 662)...".

Mixer setup:

Detailed information is provided in "File -> Project Properties -> Mixer Setup (view page 658)...".

External effects:

Detailed information on this topic can be found in "Effects -> Breakdown and functionality -> External hardware effect integration (view page 237)".

Video setup:

Detailed information is provided in "File -> Project Properties -> Media Link...".

Synchronization:

Detailed information is provided in "Synchronization (view page 505)"

MIDI Machine Control (MMC):

Detailed information is provided in "Synchronization (view page 505)"

Project information:

Detailed information is provided in "File -> Project Properties -> Project Information (view page 663)...".

Project status:

Further information is provided in "File -> Project Properties -> Project Status (view page 663)...".

Broadcast wave:

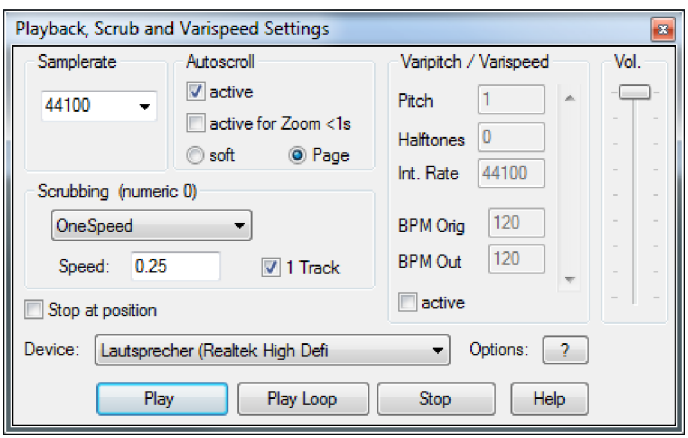
Detailed information is provided in "File -> Project Properties -> Broadcast Wave Manager (view page 663)...".

Playback Options

More information is available in "Playback -> Playback parameters (view page 830)".

Shortcut: P

Varispeed / Scrub Settings



In this part of the dialog you can make changes to the playback tempo for virtual projects. Right-click the scrub control button in the transport controls.

Sample rate: The preset project sample rate is displayed here. If you want to change this value, a dialog window appears where you can optionally adjust all audio objects to the new sample rate. The customizing is conducted, if necessary, through moving

and resampling. MIDI objects can also be adapted to the new sample rate. These are also moved if necessary without changing the musical position.

Autoscroll / Scrubbing: See "Playback Options (view page 830)".

With **Stop at current position**, the playback marker stays at the current position when the "Stop" button is pressed and doesn't jump back to the original position.

With "**Device**" you can specify the driver of the sound card that you want to use for playback.

Varipitch / Varispeed Mode

Vertical slider: This controller defines the playback from -200% to +200%, in which case negative values cause the project to play in reverse.

Pitch: Double-clicking on this field enables the speed factor to be input directly, e.g. 0.5 for half speed, 2.0 for double speed.

Halftones: This field specifies pitch changes according to semitones; a value of -12 causes the project to play an octave lower at half speed, and a value of +12 plays back the project one octave higher at double speed.

Internal rate: Here you can specify the sample rate for the Varispeed calculation. At a sample rate of 44100 kHz, a value of 22050 plays the project an octave lower at half the speed.

BPM orig: Displays the original speed in BPM.

BPM out: Enter the target speed in BPM. Sequoia calculates the Varispeed pitch factor from the quotient from the target to the original tempo.

Active: Checking this box activates Varipitch/Varispeed.

MIDI Options...

More details about MIDI settings are provided in "System settings -> MIDI settings (view page 604)".

Synchronization Active

This command activates synchronization.

Shortcut: G

Synchronization Setup...

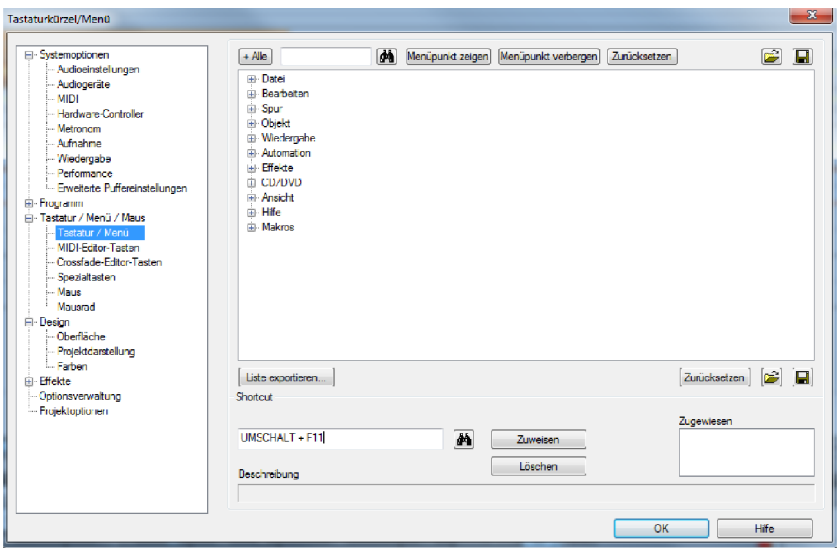
Shortcut: Shift + G

Read "Synchronization (view page 505)" for more information about synchronization settings.

MMC Setup...

Read "Synchronization >MIDI Machine Control (view page 512)" for more information about MMC.

Editing Keyboard Shortcuts and Menus



This dialog allows you to specify keyboard shortcuts for every menu function in Sequoia. 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 Sequoia the shortcut and menu settings will be saved automatically, so you can continue using them when you next open the program.

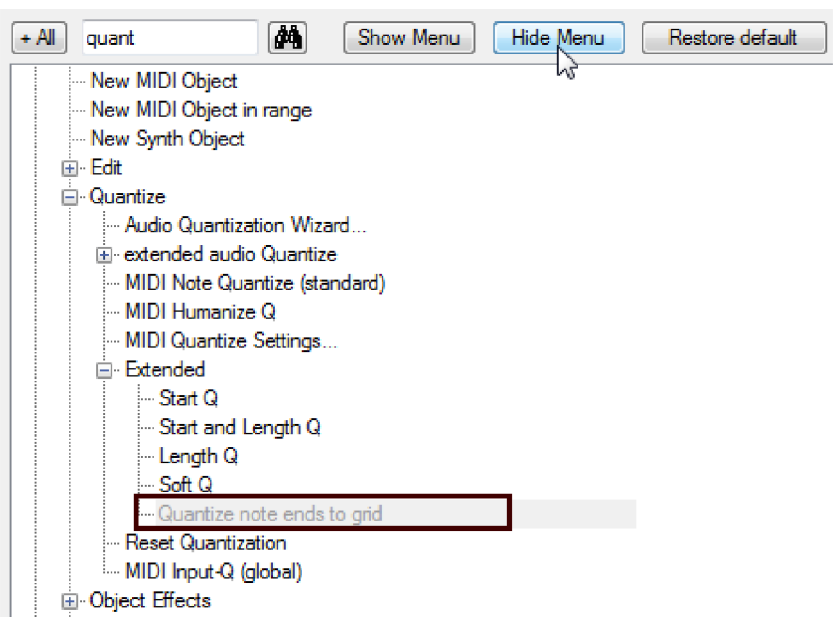
The main part of the dialog is the display of the complete Sequoia menu. Here you can select for which menu item you want to create a new shortcut or if the menu item should appear in the main menu of Sequoia.

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.



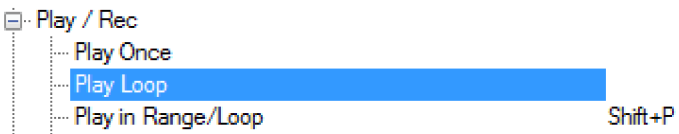
Show/hide menu item: Select a menu item which you would like to hide. "**Hide menu item**" removes the menu item from the menu It will be still shown in gray in the dialog's tree diagram.



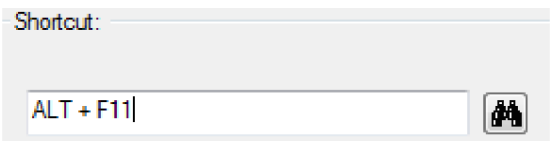
Please note that the menu item will no longer be selectable using the previously defined keyboard commands. "**Show menu item**" activates the hidden menu item again. "**Restore default**" resets the menu item to its default setting and makes the command visible again.

Creating shortcuts

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



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

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

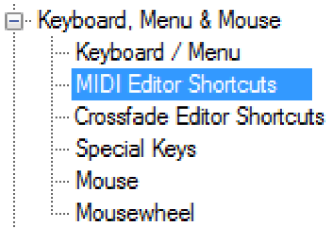
Deleting a shortcut

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.

MIDI Editor/Special Keys/Mouse Wheel/Mouse



Under "**MIDI Editor Keys**" you can find or assign new keyboard shortcuts for editing MIDI events.

Sequoia only: With the **"Crossfade Editor keys"** you have numerous keyboard shortcuts available that you can use in the Sequoia Crossfade Editor.

"**Shortcuts**" redefines the keys for temporarily changing mouse and object modes. Use these functions to quickly switch from "Universal" mouse mode to a different mouse mode and to temporarily activate "Link objects" mode or to temporarily switch to "Write automation" mode, for example.

- Temporary switch for "Link one track (view page 90)": Activates the function "Link one track" as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch for "Link all tracks (view page 90)": Activates the function "Link all tracks" as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch for object contents: Enables object contents to be moved (view page 155) with the mouse as long as the assigned shortcut combination/key combination is held down.
- Temporary switch for "Snap (view page 74)" mode: Activates the "Snap" mode as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch for "Object" mode: Activates "Object" mode from "Range" mode as long as the assigned key is held down.
- Temporary switch for "Curve (view page 82)" mode: Activates "Draw automation" mode from "Universal/range" mode as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch 1 for Scrub Mode (view page 831): Activates Scrub Mode as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch 2 for Scrub Mode (view page 831): Activates Scrub Mode as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch for "Zoom Mode: Activates Zoom Mode as long as the assigned keyboard shortcut/key combination is held down.
- Switch for "Scrub Mode (view page 118): Switches to Scrub Mode indefinitely until the next time play stops.

- Temporary switch for "Cut Mode (view page 85): Activates Cut Mode as long as the assigned keyboard shortcut/key combination is held down.
- Multi-tap sequence key: Pressing this key multiple times provides special functions.
- Temporary switch for "Automation write (view page 468)" mode: Activates "Automation write" mode as long as the assigned keyboard shortcut/key combination is held down.

More information about the different mouse and object modes is available in "Screen elements -> Program interface - overview (view page 72)".

"**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**
- **Downward-compatible use of "Shift"...**: This allows you to change the keyboard shortcut for "Switch to exclusive" for solo, mute, and record. Click "Solo/Mute/Record" and the selected shortcut "Shift + Alt" or "Shift" switches the individual channels temporarily to "Exclusive" mode.
- **Knob characteristics same as faders**: If this option is active, knobs can be pulled up and down like faders by dragging with the mouse.
- **2nd click required to move object**: This option requires a second click to moving objects. The first click selects objects and prevents unintentional moving.
- **Move thresh**: When an object is selected by clicking it, it could be moved unintentionally. If "Movement delay" is activated, Sequoia will wait a set time before executing the movement.

"**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.

Note: To help Sequoia beginners switching to Sequoia we changed the mouse wheel behavior to match the standard behavior in many other DAW. With version Pro X5 the mouse wheel scrolls in VIPs vertically (tracks) by default instead of horizontally (time position). With the „Reset“ button you can reset Sequoia to either the old or the new standard behavior.

Note: Pressing the key combination "Alt + Shift" might cause Windows to switch keyboard languages. In order to avoid this simply deactivate the keyboard shortcut used to switch input settings by going to the Control Panel and clicking on the "Clock, Language, and Region" icon. Navigate to the "Change keyboards or other input methods" tab and then click on the "Change keyboards" button to access the settings. In the "Advanced Key Settings" tab click on "Change Key Sequence" . Set the "Switch Input Language" and "Switch Keyboard Layout" options to "Not Assigned". This will prevent you from accidentally changing the input settings.

Record macro

The macro function gives you the option of creating and playing command orders. You can find detailed information in the "Working in the Project Window" > "Macro functionality" (view page 136).

Edit toolbars

see Customize Toolbar (view page 91)

Reset Toolbars

Resets the toolbars to default settings. Select whether individual toolbars should be reset or if all of them should be reset via "**Reset all toolbars**". If the file "Sequoia_TB.ini" is deleted from the Sequoia folder, then the default settings will be reset.

Font Selection...

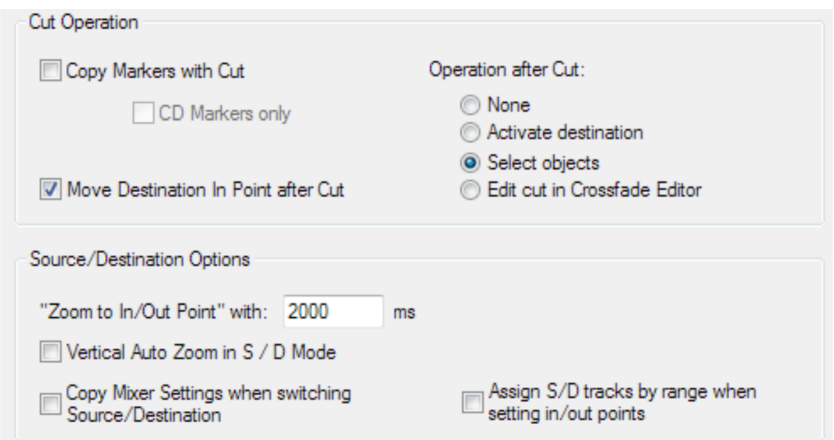
Select a font for text display in Sequoia.

Font for Manager...

Choose a font for the managers.

Source destination cut settings

Here you can open the dialog window for cut operations and source destination cut settings.



In "**Cut operations**", choose whether the **marker should be copied when cutting** or whether you **only want to copy the CD marker**. You can also **move the "Destination in-point"** after cutting.

Using "**Actions after the cut**" you can **activate the destination** after the cut, **select objects when cutting** or **open the cut in the Crossfade Editor**.

In the **Source Destination Settings** you can set the **zoom factor for the In/Out point**. Furthermore you have the chance to set the **vertical Auto zoom to S/D mode, to copy mixer settings during source destination swap** and **to use the S/D track assignment in a range when setting an In/Out point**.

Detailed information about source destination cuts is available in the"Source Destination Cut" chapter.

Crossfade Editor Preferences...

This opens the dialog window for the crossfade editor's settings.

Detailed information about the crossfade editor is provided in "Crossfade editor -> Crossfade editor settings (view page 439)".

Metronome Options...

Detailed information about metronome settings is provided in the chapter "Sequoia quick start -> Workshop: recording -> Metronome settings (view page 95)".

Project Display

Definition...

This dialog can be used to change between the two viewing options "**Draw mode 1**", "**Draw mode 2**", „**3**“ and „**4**“. Wave form display, object, VIP, and wave form color categories can be edited.

Keyboard shortcut: Shift + Tab

Mode 1/Mode 2/Switch Modes

Select a view mode here, or switchback and forth between the two viewing modes.

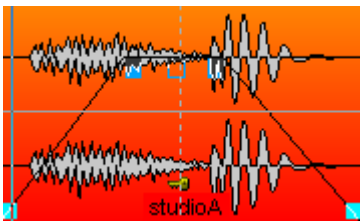
Shortcut: Tab

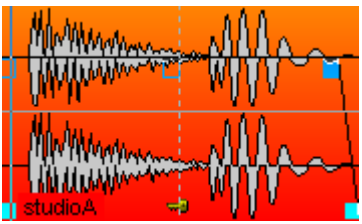
Waveform Display

Draw wave form: Turn the wave form display on and off here. When deactivated, you can see volume and panorama curves more clearly.

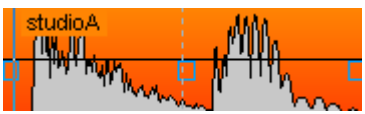
> **Note**: In "Drawing mode 2", the standard default setting for the wave form display is switched off. This may be changed at any time.

Scale with fades/curves: In this default display mode, the wave form overview is scaled according to the settings for the fade in/out or volume curves. This makes the decay of the sound during a fade out or for volume curve changes more visible. This mode also offers good visibility when crossfading audio material.

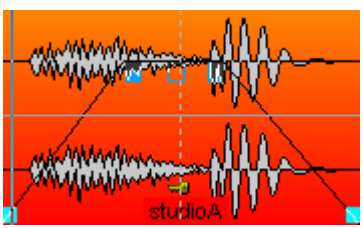




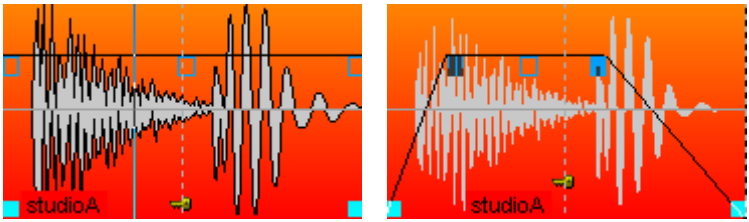
Halved waveform: Switches the graphical representation of the samples to half the waveform display size.



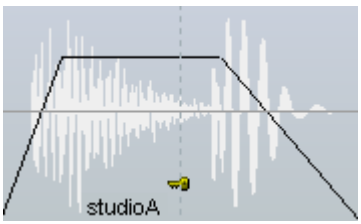
Separate stereo: If this option is active, stereo tracks are displayed separately one above the other.



Draw envelope: If this option is active, an outline is drawn around the envelope of the waveform.

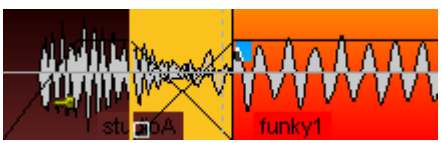


Gray out muted objects/tracks: Muted objects and tracks will be shown gray.

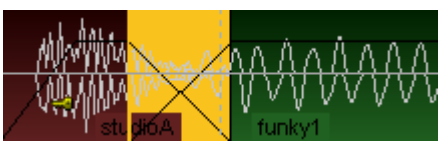


The second column is especially useful for a more clear crossfade display.

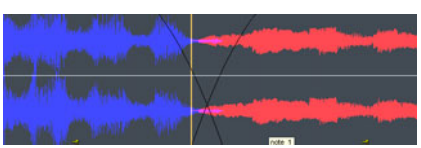
Standard: The waveform of the second object is drawn over the waveform of the first object:



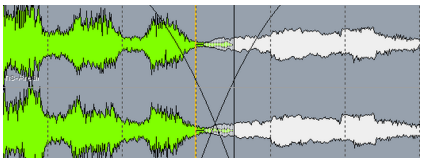
Envelope only: When this radio button is selected, only the waveform outline (envelope) is drawn. Both objects are well visible in crossfades:



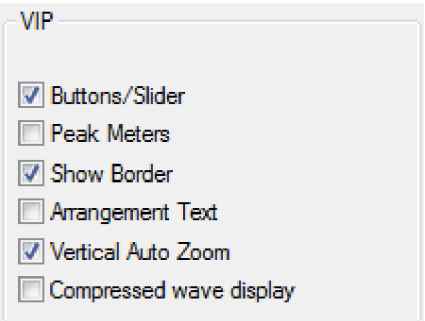
Transparent: If this button is selected, the colors of both crossfade waveforms are overlapped. The waveforms appear against a dark background. We recommend combining this and the "Waveform colors -> Red/blue alternation" to optimize the display of crossfade areas:



Interleaved: When this option is selected, a sample of the left object and a sample of the right object will be drawn. You may then visually assess the fade area, a task made especially easy if the adjacent objects have different colors:



VIP



Various VIP components (buttons/sliders, peak meters, show border, arrangement text, vertical auto zoom, compressed waveform, and CPU peak meter) can be activated/deactivated here.

Buttons/slider displays the track header with all mixer controls, i.e. the channel faders, solo buttons, etc. If you always have the track editor open, you won't have to use these functionally identical control elements here. However, if the track header is left open, then only the track header's **peak meter** will be able to be hidden.

If you **hide the borders**, the edge of the selected track will disappear.

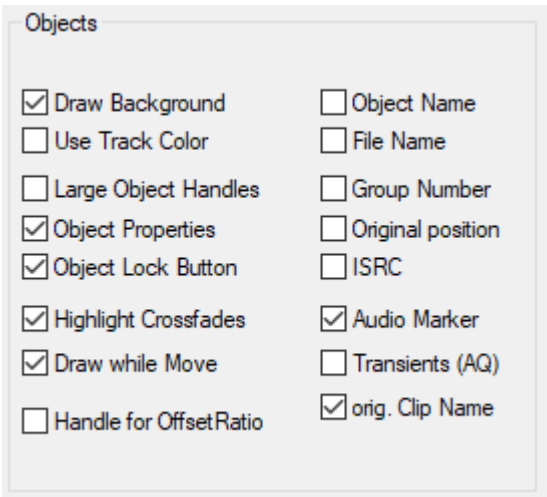
Arrangement text identifies the tracks for the two editing ranges in "Universal" mouse mode. If you set a check mark here, you will see "Playback Marker and Segment Manipulation Area" in the top track half, and "Object Manipulation Area" in the bottom half.

Vertical Auto Zoom: If you have activated this option, the waveform of a track is zoomed vertically so that the sample with the highest value fills the track height exactly. This also disables vertical zooming into the waveform via keyboard shortcuts "Ctrl + Arrow up" or "Shift + Mouse Wheel".

Note: Active vertical auto zoom also deactivates the buttons for "Zoom wave in" and "Zoom wave out".

Compressed wave display: This display option causes the waveform to be distorted according to the zoom level. The transition from silence (less than –90 dB) to signals with lower levels (e. g. noise) is also shown more clearly.

Objects



Draw background: This activates the background color view of the selected objects. Every object can be assigned its own color (see "Object .> Object Color/Name (view page 803)" or in the Object Editor under "Color").

Use track color: The track color will be used as the background color.

Large object handles: The object handles are displayed larger.

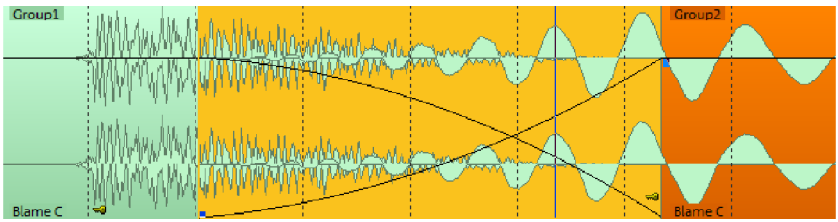
Object properties: This option enables object settings like EQ, dynamics, panorama, effects or plug-ins to be displayed with abbreviations in the object.

Object lock symbol: If a checkmark is set, a key will appear on the bottom of an object, which can be used to lock and unlock objects (view page 786).

Highlight crossfades: With this setting, the crossfades between the objects are set in contrasting colors.

Draw while moving: The user interface is updated while moving the mouse. This option can be deactivated for computers with low processing capabilities.

Handle for fade overlap: Use this handle to set the offset ratio for crossfades without having to open the crossfade editor or object editor.



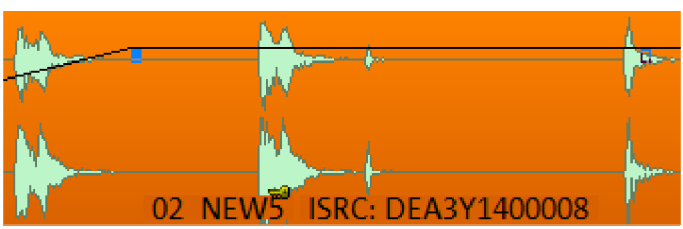
Object name: Shows the object name.

File name: The file names will be displayed.

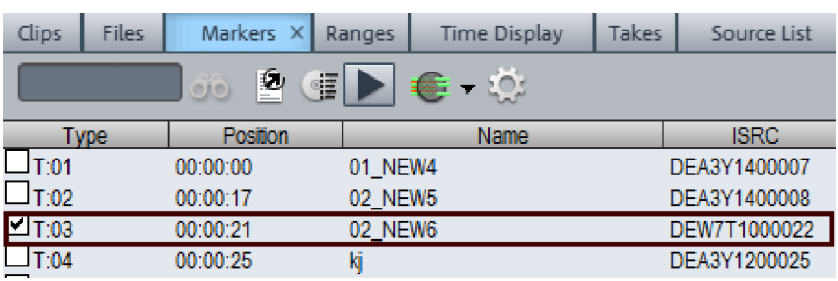
Group number: Objects in a VIP can be grouped. The groups are numbered. When displaying these group numbers a quick overview appears showing which objects belong to which groups.

Original position: This shows the original recording position at the beginning of the object in the set time code format. This way, a comparison of values in the object with the position on the timeline or on the time code of the picture helps clear up possible deviations in the synchronization.

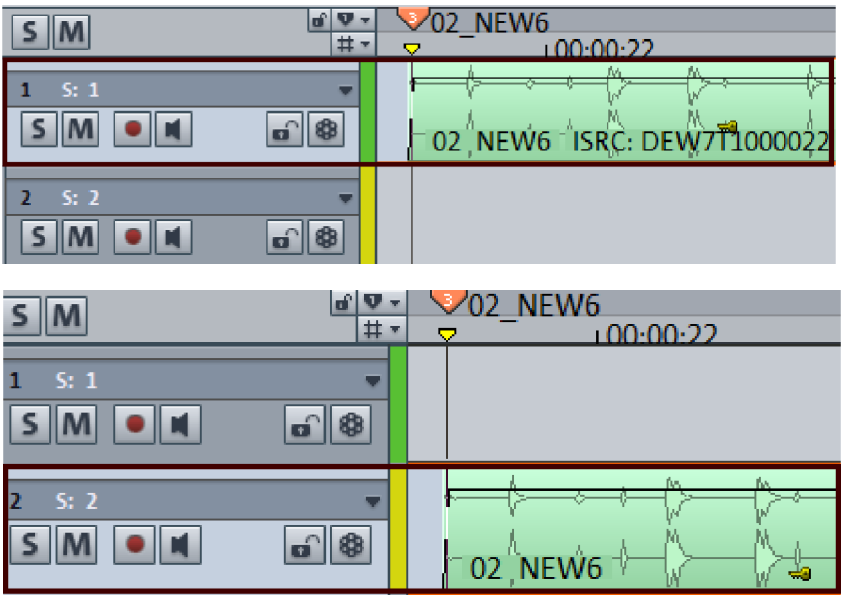
ISRC: This option displays the ISRCs in the objects.



When the ISRC for an object is not set as a Broadcast Wave File (BWF) entry (view page 665), the ISRC of the track marker will be displayed if it exists.



For this the object must be on the first track.

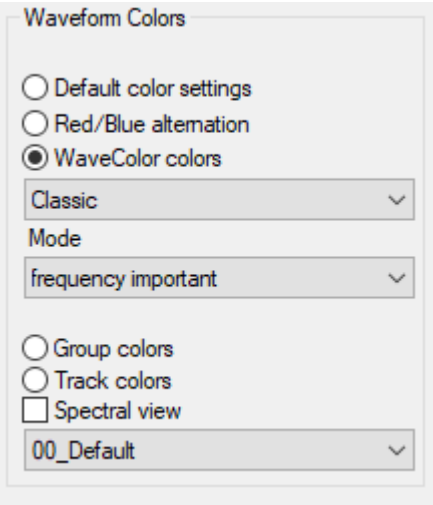


Audio Markers: Use this option to display markers that are present in the audio files in the corresponding objects.

Transients AQ: Use this option to display the transients in objects that were created beforehand with audio quantization.

Orig. clip name: This option displays the clip name on objects created by loading a clip (More information can be found in the section "Clip Manager ("Clipstore")" on page 178).

Waveform color



Default color setup: The waveform contains the default color set in the color setup.

Details about the color setup can be found in the menu reference under "File > Program Preferences > System/Audio -> Design -> Colors (view page 680)".

Red/Blue alternation: This display mode colors the objects beside the waveform alternately red and blue so that crossfades in the "Transparent" and "Interlaced" draw modes are optimally visible. This way, crossfades can be optimally viewed in the "Transparent" and "Stacked" drawing modes.

WaveColor color gradient: With WaveColor, two additional properties of the audio material are visualized in the waveform display using color. The pitch is represented by the hue: Low tones are red, mid tones are green, high tones are blue. The sound characteristics of the audio material are represented by the color saturation: The more tonal and harmonic the signal, the more saturated the color, the more noise there is, the less saturated the color will be.

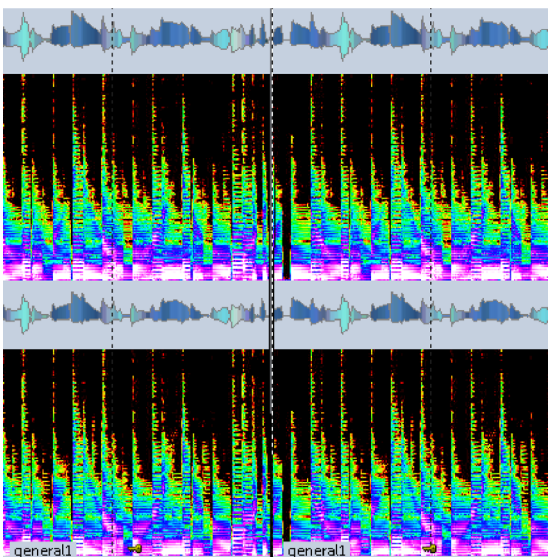
Various settings can be found in the drop-down menu: The "default" setting applies intense rainbow colors to the audio signal. The "Classic" presetting is similar to the colors used by Comparisonics. "Black-and-white pitch" and "Black-and-white noise factor" show only the pitch level or the noise components.

On the basis of the WaveColor colors, a completely new audio search function is available under "Object > WaveColor Audio Search (view page 813)" which can be used to easily find identical or similar audio material in your project.

Group colors: Object groups receive a randomly created individual group color when being made. This way, various groups can be displayed and differentiated very clearly from one another.

Track colors: The tracks will appear in colors selected under track options. The track options can be opened by right-clicking on a track name.

Spectral display: Here you can select various spectral color palettes. The graphical data that has been created will be saved in an extra *.hs file.



The graphical representation of the music is displayed in a spectrogram. This displays the frequency proportions in a time curve. The volume of frequencies is visualized via a color code or via its brightness.

Color Settings

Details about the color setup can be found in the menu reference under "File > Program Preferences > System/Audio > Definition > Color Setup... (view page 680)".

Undo Definitions...

Use this editor to switch the undo function on and off for virtual projects, wave projects, and plug-in settings. The amount of undo steps may also be entered. A value of 20 means that the last 20 changes to a project may be undone.

If you check the corresponding box, temporary undo files be created as HD wave projects for RAM wave projects.

Dithering Options...

How does dithering work?

An audio signal is quantized at a higher or lower resolution with every A/D conversion. During quantization, the signal receives an echelon form via limiting of the possible amplitude values. An 8-bit signal, for example, possesses only 256 amplitude values. This echelon formation causes distortion and thereby deforms the signal, especially impairing the sound of the lower levels.

Dithering can be understood as "mixing in" of low-level noise which seriously reduces the human ear's detection of sound defects.

When to dither?

Sequoia always dithers a signal in integer format when it is saved or exported.

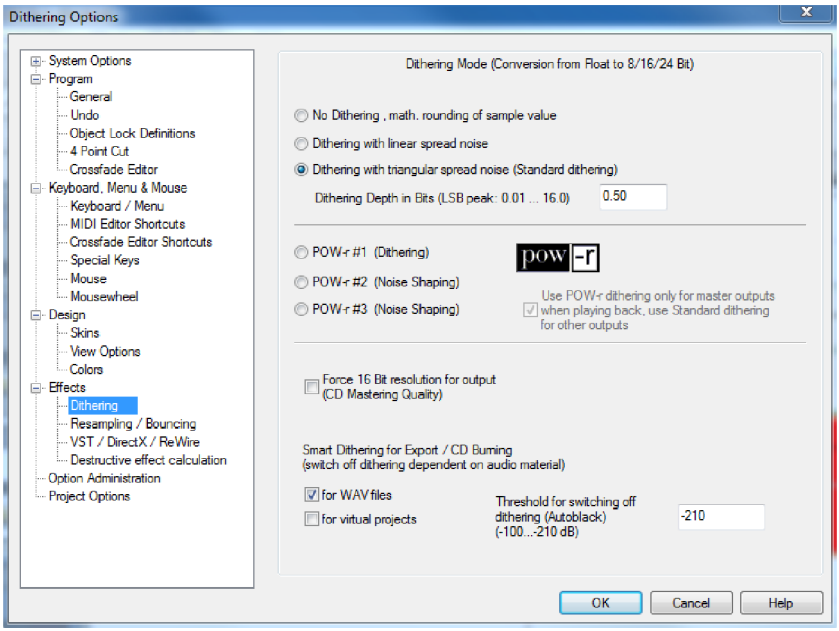
This is the case for the following situations:

- During 16-bit playback. All device addressing takes place in fixed comma resolution.
- When burning audio CDs in real time using an internal precision setting of 32-bit float.
- When using track-bouncing for virtual projects into 16-bit wave files, provided the internal precision is set to 32-bit float.
- When converting any 32/24-bit wave project into a 16-bit wave project.

Note: Sequoia does not execute dithering during recording.

The individual settings for dithering can be set during each trackbouncing process. Detailed information is provided in "Tools -> Trackbouncing (internal mixdown) -> Trackbouncing settings: options".

Dithering Options



No dithering, math. rounding of sample value: In this mode, conversion of the signals of 32-bit float takes place 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: In this mode, audio data at 32-bit float is converted via dithering with noise featuring amplitude values that occur at regular intervals. The noise level can be set with 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. This means that values are more regular in the medium range and less often at the maximum or minimum values. This type of dithering usually creates better results than linear dithering. The noise is not modulated through the signal, resulting in a fading signal being enveloped by one constant noise signal.

Dithering depth in bits: This sets the strength of the noise used in dithering. Input is in bits. Use this option to specify how many bits of the resulting 16-bits should be affected by dithering. In most cases, the values between 0.5 and 2 will produce good results. Increase the value until distortion effects are no longer audible. Provided you cannot perceive any distortion effects, values below 0.5 are usually sufficient. If you would like to add more severe noise to your signal, try entering values between 8 and 12.

POW-r Dithering/Smart Dithering

POW-r #1 (dithering): This function uses a special dithering curve to minimize quantization noise.

POW-r #2 (noise shaping): This function uses additional noise shaping across a wide frequency range to extend the dynamic range by 5-10 dB.

POW-r #3 (noise shaping): This function uses additional, optimized noise shaping to extend the dynamic range by up to 20 dB between 2 kHz and 4 kHz. The human ear is most sensitive to this frequency range.

Noise shaping minimizes side effects caused by bit reduction by spectrally displacing the quantization noise above 10 kHz, which is the range the human ear is least sensitive to.

Which dithering mode sounds the best depends mainly on the audio signal.

POW-r dithering during playback for master outputs only, use standard dithering for other outputs: If this option is selected, then only the master outputs will be dithered with the POW-r dithering algorithm. The individual outputs will use standard dithering, i.e. **dithering with triangular spread noise**.

Force 16-bit resolution for output (monitor CD mastering quality): In this case, audio playback dithering is also executed even when audio output supports higher bit rates. This enables a preview of how the CD will sound at the end of the project.

Smart dithering during export/CD burning (switch of audio material-dependent dithering)

Smart dithering for wave files: This option is set as the default. When burning CDs and exporting 16-bit files, dithering will only take place when the file is not currently 16-bit. Dithering is not calculated during silence.

For virtual projects: This option is switched off by default. In this case as well, dithering only occurs if the bit depth does not equal 16 bits.

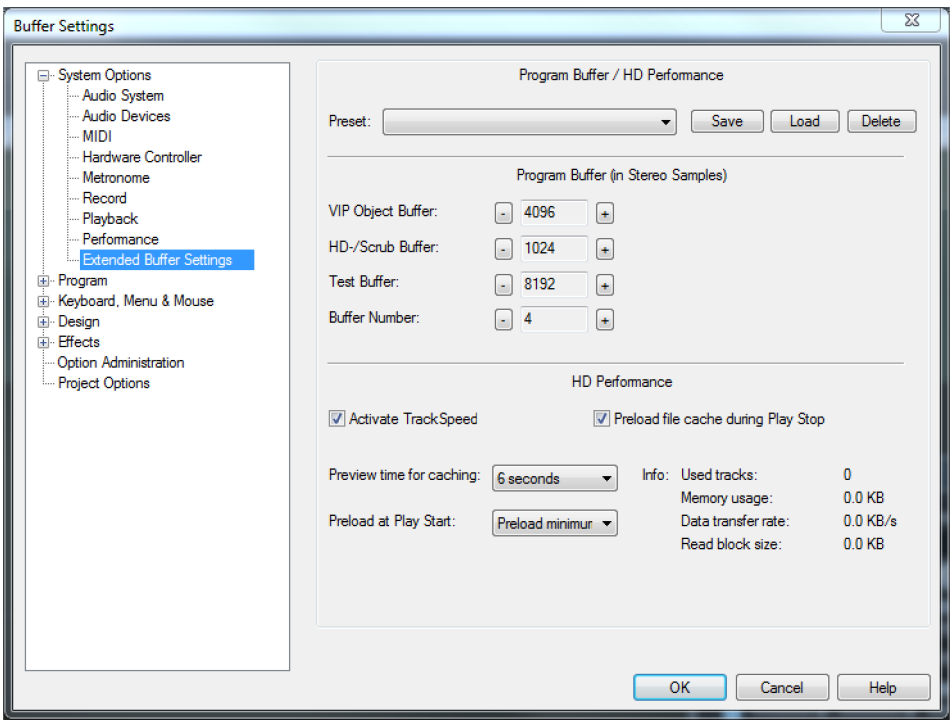
Threshold for dithering shutoff (autoblack): Specify a threshold value below which any dithering noise that is created will be muted. These values range from between -100 dB and -210 dB.

Advanced Buffer Settings

In this dialog you can optimize buffer settings for your virtual project, for hard disk caching and for plug-in manipulation. If you are not having any problems with audio dropouts or pops and clicks, you do not need to make any adjustments here.

Program Buffer/ Hard Disk Performance

For special applications you can set and save your own buffer settings. In addition to this, special presets are available, e. g. for scrubbing and ASIO applications.



Program Buffer (in stereo samples)

VIP Object Buffer: Because error-free playback is usually more important than fast reaction times, this value should be increased when playing lots of tracks. This is the only setting relevant for playback and editing in the VIP.

HD/Scrub Buffer: This buffer is put into use during direct playback of HD Wave Projects. Test smaller values here for even faster response times.

Test Buffer: This buffer is only used for realtime previewing of effects from the effects menu.

Buffer Number: Specify how many of the buffers described above should be used here. More buffers increase reliability, but also increase memory requirements. This results is longer response times. You can check the current buffer usage levels during playback in the status bar at the bottom right. We recommend using between 4 and 6 buffers.

Hard Disk Performance/TrackSpeed

With the help of the TrackSpeed technology you can increase the number of hard disk tracks simultaneously playable in Sequoia. TrackSpeed uses your PC's RAM to intelligently and effectively preload the necessary audio data.

By using TrackSpeed, the internal audio engine of Sequoia can even work using multiple tracks with small audio buffers (e. g. 8000 samples or less), without hurting performance. This makes short reaction times between user interactions possible, e. g. with the mixer and various object handles.

There are settings available which can be used to customize TrackSpeed to your individual requirements:

Activate TrackSpeed: Set a check mark here to activate TrackSpeed.

Preload file cache in "Stop" state

If this option is activated, the file cache will be preloaded each time the playback marker is repositioned. You can recognize this by the "preloaded cache" notification that appears in the bottom right status bar. The next play start happens automatically because the necessary audio files are already present in the RAM.

Preview time for caching: Enter how many seconds of the VIP can be preloaded in this field.

Preload at play start: This value determines how much of the cache will be loaded at play start (in percent). Larger values delay the start process but increase the playback stability when using a large number of tracks.

The dialog also displays the following project information:

- Used tracks:
- Memory usage:
- Data transfer rate:
- Read block size:

Resampling / Freeze Options

Use these options to set the following resampling/freeze calculations.

Resampling options

- Record resampling (chase lock sync, conversion to 44.1 kHz during recording)
- Playback resampling (scrubbing, chase lock sync, playback at altered sampling rate)
- Object resampling for new objects (load objects with sample rate different from project)
- Resampling during bouncing (prior to burning CD)

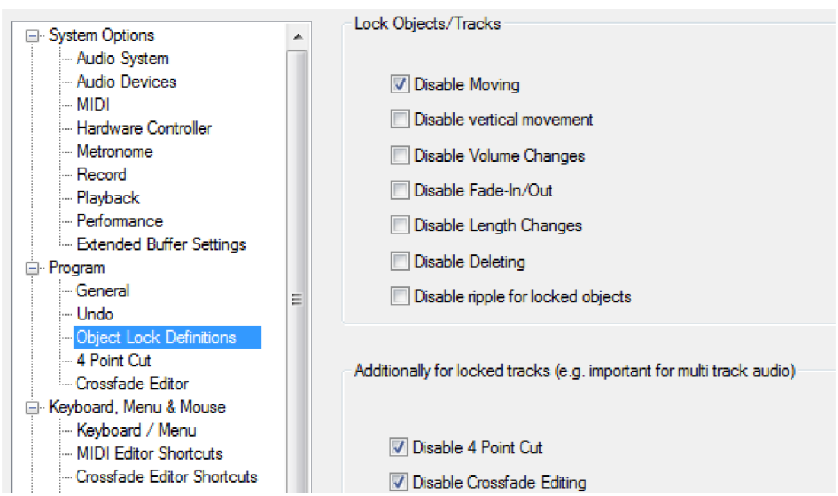
Time Stretching/Pitch shifting options

Here you can set the standard algorithm (view page 912) for objects.

Freeze options

- Keep mono, if possible
- Use additional samples for object freeze
- Object freeze without object volume
- Format (16-bit, 24-bit, 32-bit)

Locking options



Select which functions should be disabled by locking objects or tracks here. The following options are available:

Locking objects

Disable moving: Objects can no longer be moved horizontally. This avoids unintentional offsets between individual tracks for multitrack recordings.

Disable vertical movement: Objects can no longer be moved vertically.

Disable volume changes: This deactivates the object's volume handles.

Disable fade-in/out: Deactivates object's fade handles.

Disable length changes: Deactivates the object's length handles.

Disable deleting: This option prevents the deletion of locked objects.

Disable ripple for locked objects: Locked objects are not affected by functions like "Edit -> More -> Ripple delete" or "Edit -> More -> Ripple cut".

Locking objects/tracks

Tracks are locked in the track box by activating the lock symbol.

Disable source destination cut: This deactivates the source destination cut function for set tracks. The cut may no longer be changed.

Disable crossfade editing: This deactivates the crossfade function for set tracks. The crossfade can no longer be changed.

Set Preroll Time...

This option sets the pre-roll time for the cut simulation functions. The cut simulation function is started via the corresponding buttons in the punch/playback bar. The pre-roll time specifies how long is played until the selected range is reached.

Destructive Effect Calculation...

Choose in the advanced destructive effects calculation options (view page 858) if you want to add the effect to the original file when editing destructively, write it into the effect file, or create a new effect file for each calculation.

Loudness

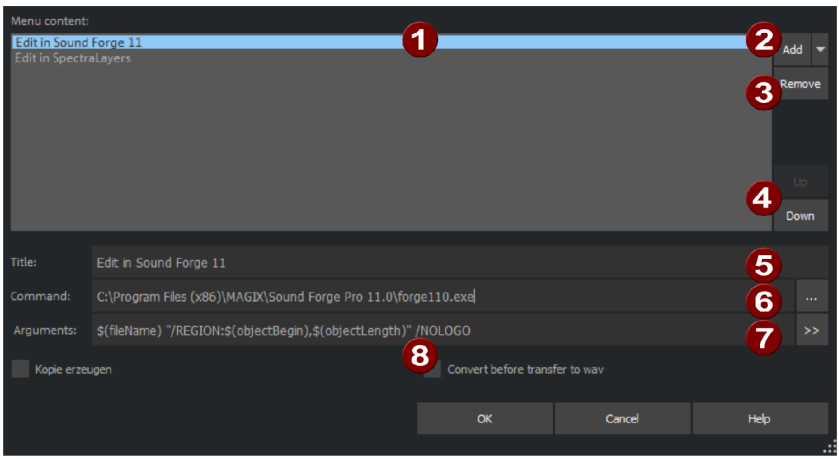
In this dialog window you can specify the basic settings for loudness normalization (view page 862) compliant with EBU R128/ ITU-R BS.1771.

Option Administration

See „System Options“ > „Option Administration“ (view page 683).

External Tools

This dialog is where you can configure objects for editing other audio programs.



1. **Menu content**: All entries in this list create entries in the context menu for an object (view page 163). The entries can be transferred to an external editor for editing.
2. **Add**: Click here to add a new tool. To do so, search for the executable file you need in your program folder. If Sound Forge or SpectraLayers is installed, the tool may be entered automatically with all its parameters via the dropdown menu.
3. **Delete:** Deletes the selected entry. If you delete an entry and close the dialog that then appears with "OK", the entry will no longer be displayed in the context menu for objects.
4. **Up/Down**: Use to change the sequence of menu entries.
5. **Title:** Enter text for the menu entry here.
6. **File selection dialog:** Search for an executable file on your hard drive using the file section dialog.
7. Predefined placeholders: In this menu you'll find the predefined placeholders to set in the commando line in order to transfer object parameters e.g. object start and end to the external program, if it supports them.
8. **Convert to wave before transfer:**Since some editors can't open MP3, FLAC or video files, these types of files are converted by the external editor and saved under another name. However, this will prevent you from transferring the file back to the virtual project. We recommend that you activate this option when working with these editors. This means the content of an object will always be converted to wave before transferring to the editor and the object will be adjusted for editing.

Administration

The administration function enables you to set up different users for Sequoia; every user or every user group can work with their own settings in this case.

Every user has their own independent set of settings that is loaded after logging on:

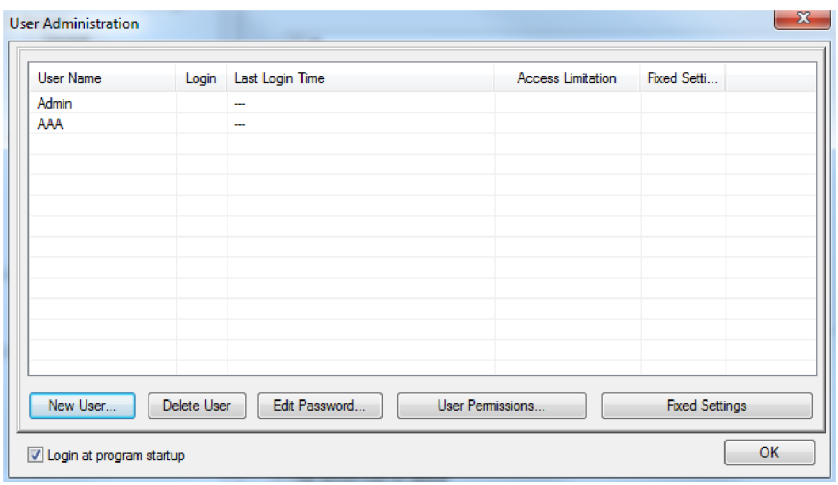
- Visible and hidden menu entries
- Keyboard shortcuts
- Color settings, skins, standard folder paths
- Toolbars and icons
- VIP display modes
- Options for settings the crossfade editor properties, undo settings, dithering, etc.
- Crossfade presets

Note: Users access a common folder for all effects presets (Reverb, Room Simulator, FFT Filter), mixer presets, and color schemes – in short, everything that is saved in .\fx-presets. If you want to protect your personal user presets then you should save these in another folder. This will cause them to no longer be shown in the preset selection list.

Set up multi-user operation

In order to set up Sequoia for multi-user operation, select the menu item "File > Program settings > Administration" or access the system options and then select "Option administration" and then the "User administration" option. The settings dialog for local multi-user operation is only available to Sequoia's administrator.

In case there aren't any entries present already, add an "Admin" user first and a password for this user. Next, the dialog for user management will open.



This dialog allows you to add or delete users and set passwords for them. Sequoia displays the time each user was logged in, whether they have access limitations to paths or whether the settings are able to be changed the user independently.

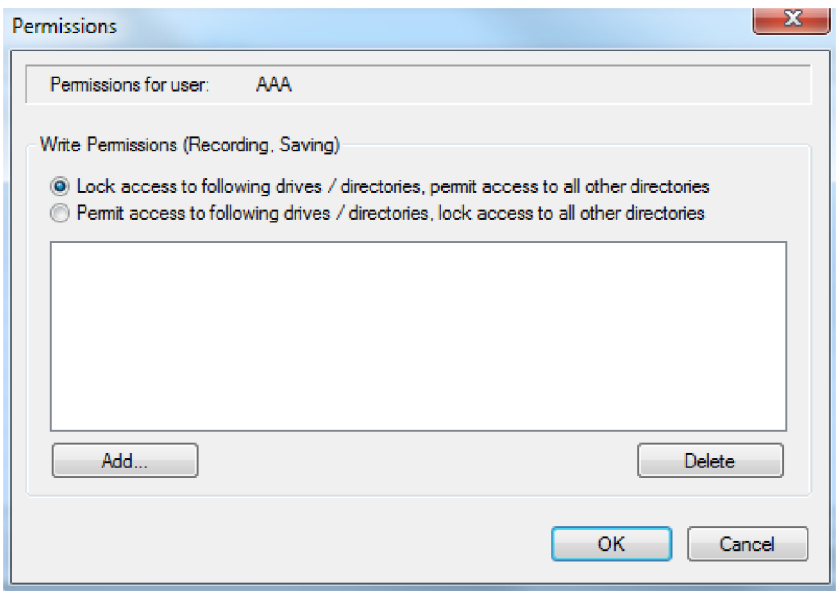
When multi-user mode is active, this option will only be available to the administrator, i. e. only the administrator may add or delete users and activate/deactivate multi-user operation. This option is grayed out for the regular user, and also in the network operation (view page 685).

Create New User

To set up a new user, click the button "**New user**" and enter the username and password. If you want to apply the settings from an existing user, select the user first and then click "New user". The option "**Copy all settings from...**" is now active. Confirm with "**OK**" to copy all settings from an available user to the new user.

User Permissions

This function sets the permissions for the respective user. You can also authorize or lock write permissions for drives or folders.



Fixed Settings

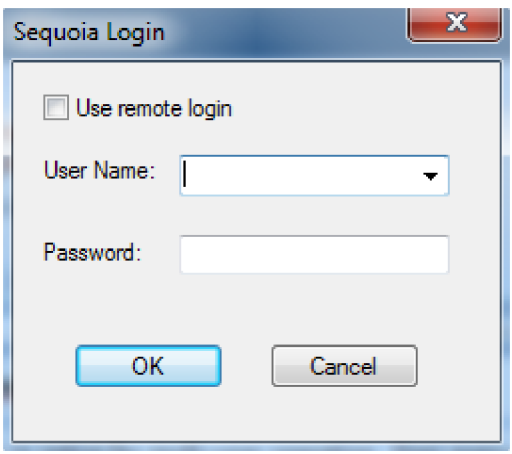
As **administrator**, you may fix the program settings for specific users via the option "**Fixed settings**".

For the affected **users**, this means that they may change their user settings while the program is running, but that when the program is restarted, they will begin again with the settings set by the administrator.

Activate/Deactivate Multi-User Mode

Login at program startup: If Sequoia is setup for multi-user operation, then every user must log in with a valid username and password each time Sequoia is launched.

The option "**Use network login**" allows the respective user to log in via the network.



Multi-user mode may be deactivated with the option "**Login at program startup**". In this case, the administrator's settings will be used.

If a valid network path for user administration (view page 685) is set and a check is set to activate its use, then the login screen will be displayed when Sequoia is launched. This makes it possible for remote users to log in with a specific profile or locally without a special username.

End Multi-User Mode

If the user "Admin" is deleted, Sequoia will be permanently restored to single user mode.

Attention: This will delete all users will be deleted!

A user cannot change their identity while the program is running.

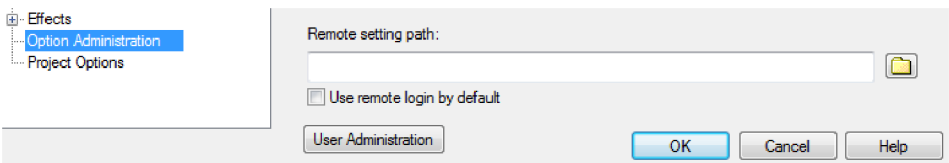
Multi-user mode on network drive

Sequoia provides the option of **central management and administration of user profiles on a network drive**. The network path must feature the settings for the user and the login data in the file "**Sequoia_Admin.ini**". There are two options for providing this data on the network.

1. Complete an **installation of Sequoia on the network path** that will be started via the network and managed via "User administration", where all desired users are located.

2. Save **a settings container for each user** and name it according to the user (e. g. User1.INZ, User2.INZ, etc.). These **containers must then be placed in the network path**. Now copy the contents of the file "**RemoteSetupTools.zip**" from the program folder to the network path. To create or update the login data in "Sequoia_Admin.ini", start the tool "**AdminCreator.exe** (view page 715)" in the program folder. For new users, an empty password will be created that may be changed via the menu item "Options > Change user password". The tool "AdminCreator.exe" may also be used to assign passwords to the users.

Each Sequoia work station that is intended to use this remote user administration must be set via "**Network path for settings**" in the options. Place a check mark for "**User network path**" in order to use the network login when Sequoia is launched.



If a user logs in remotely, then all of their settings will be copied from the network path to a temporary local folder.

Note: Copying settings from the server takes place via the batch file "**remote_setup_copy_from_server.bat**" that may be changed individually by the administrator (e.g. additional comparison of effects presets and template folder). The batch file is always searched for on the server first, and then locally.

Note: In order to display administrator information for all users in an additional text field in the login dialog, create a file named "AdminInfo.rtf" in the network path. This rtf file should be created with Wordpad in order to ensure correct display.

Note: If you work with "Remote Logins", please extract the contents from "RemoteSetupTools.zip" in your remote directory, because changes were made to "AdminCreator.exe" as well as "remote_setup_copy_from_server.bat".

- User-specific templates are now stored in "fx-preset/user_name/templates".

- If user-specific templates are available, they will only be shown to the user when creating a new project. If these templates don't exist then the templates of the local admin will be displayed.
- The auto jam session continues working with the templates in the template directory of the local admin, not with the user-specific path.
- Templates are now also saved in the INZ container. If there are no user specific templates available, the templates of the local admin will be saved.
- When importing INZ containers, previously available user-specific templates are deleted.
- "Recent Files list" and work paths for remote users are saved separately for each remote user in the "audio.ini" of the admin (in other sections)
- Current "Recent files list" is kept if you import INZ containers. You will only encounter troubles if the "audio.ini" e. g. is overwritten by an admin when loading INZ containers.

AdminCreator tool

Next unzip the contents of the "**RemoteSetupTools.zip**" file in the program folder and then start the AdminCreator Tool using "AdminCreator.exe".

AdminCreator.exe /T:"c:\ini" /U:"UserName" /P:"UserPassword" /N /R /I

/T:"" = path entry for Sequoia_admin.ini (remote), in case AdminCreator is not started on the network drive.

/U:"" = username

/P:"" = password

Once the username and password are indicated, only the password for the user specified will be changed; no user will be added or removed.

/N = no user will be added to Sequoia_admin.ini, even in case a new settings container is present. (this makes sense in context with option /R)

/R = users that are contained in Sequoia_admin.ini but and for whom no settings container exists will be removed from Sequoia_admin.ini

/I = All users in the Sequoia_admin.ini will be deleted. (if /N has not been set, then all available settings containers will be added again as users).

Change User Password

This option is only available if multi-user mode (view page 713) is activated. Registered users can change their passwords here.

More

Internet Connection

This menu item allows you to terminate an existing Internet connection directly from within Sequoia.

Append Project

With this function, a project can be appended to another one. The objects or wave files of a selected project will be immediately copied to the end of an existing object.

To execute this command, first click the project you would like to attach the material to. Next, select the "Append project" command, followed by a mouse click the project to add to. Finally, confirm via the appearing dialog by clicking on the "Add" button.

Add VIP to a VIP

Virtual projects may be inserted into other virtual projects and added to the end of already existing virtual projects.

All objects of the appended VIP remain separated and are inserted at the end of the existing project from the first track downwards.

Note: The mixer settings for the inserted project will not be applied with this command.

Note the convenient source/destination editing with multiple projects (view page 407) option in the manual chapter entitled "4-point cut (view page 737)".

Add Wave Project to a Wave Project

"Append Project" allows you to copy a wave project to the end of an already existing wave project.

Exit

The program Sequoia is being shut down. All projects changes will be lost if "Save" is not selected.

Recent File

In the last section of this menu the 10 projects opened last will be displayed.

Edit Menu

This menu features functions that are applicable only to wave projects (e. g. „Copy “ > „Copy as“), and others that are only for VIP projects (e. g. „Insert Silence“).

Edit functions like "Delete > Extract" require a selected range; other functions like "Copy" are available only for active objects. The start and end points of the selected range determine the start and end points of a cut, and the vertical position indicates the tracks that are affected by the edit.

Activate the auto crossfade function and all the newly created fade edges will automatically be made soft or disappear.

Mouse mode

The following options allow you to select from a variety of mouse modes like in the Mouse mode (view page 77) selector on the upper tool bar.

Universal Mode

This is the preset mouse mode for Sequoia. All necessary functions are available by left-clicking. Right-clicking opens a context menu.

Detailed information about Universal Mode is available in "Screen Elements > Toolbar - Overview > Mouse Mode > Universal Mode".

Range Mode

In this mode, only ranges and the playback marker can be manipulated.

Detailed information about Range Mode is available in "Screen Elements > Toolbar - Overview > Mouse Mode > Range Mode".

Curve Mode

This mode allows panorama and volume curves to be drawn, edited and deleted with the left mouse button.

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Curve Mode (view page 82)".

Object Mode

In this mode objects can be moved and curves can be edited with the left mouse button.

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Object Mode (view page 83)".

Object/Curve Mode

In this mode, you can move objects and edit curves using the left mouse button.

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Object/Curve Mode (view page 83)".

Left/Right Mode

Use this function to switch to the Left/Right Mode. The right mouse button then controls object functions, the left one controls manipulations of the range.

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Left/Right Mode (view page 84)".

Wave (HDP/RAP) Mouse Mode

Range

Use the left mouse button to set the playback marker and select ranges.

Draw Wave

Draw the sample material at a high zoom level, which can be helpful if you want to remove clicks and clipping.

Volume Draw Mode

In this mode, you can directly manipulate the volume using the mouse.

Scrub Mode

This mouse mode enables you to preview and control the playback speed. The project is played forwards or backwards, depending on the scrub direction.

Zoom Mode

Click with the left mouse button to zoom into the wave display; zoom out with the right mouse button.

Cut Mode

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Cut Mode (view page 85)".

Pitch Shift / Time Stretch Mode

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Pitch Shift / Time Stretch Mode (view page 85)".

Volume draw mode

This mode allows Volume Curves to be drawn with the left mouse button. You must first activate the volume curve on the corresponding track with the "Vol" button to the left in the track header. Then, click on the curve to create curve points. These can then be dragged to the desired position.

Left mouse button: Freehand drawing function for volume curve.

Right mouse button: Context menu

Automation Draw Mode

This mode allows automation curves to be drawn by using the left mouse button. Activate the automated parameters for the embedded plug-in beforehand by pressing "**Ctrl + Alt + Moving the corresponding plug-in control element**". Next, click with the right mouse button on the "Track automation" field in the track editor, and place a check mark for the desired parameters. Now you can draw in the curve of the selected parameter in the track in Automation Draw mode.

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Automation Draw Mode (view page 86, view page 468)".

Spectral Mode

In "Spectral Mode (view page 86)" you can remove noise from a track object without the wanted signal being audibly influenced. Editing can be performed directly in the arranger window.

After switching to "Spectral Mode", select the noise by clicking and dragging out a rectangle over it. The handle points can be used to make fine adjustments to the selected area. After making the correction, you can see the results in the Wave/Spectral display in the arranger window.

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Spectral Mode". (view page 86)

Preview (Scrubbing Mode)

This mouse mode enables you to preview and control the playback speed. The project is played forwards or backwards, depending on the scrub direction.

Zoom Mode

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Zoom Mode". (view page 88)

Color Mode

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Color Mode". (view page 88)

Object mode

Detailed information about object modes can be found under Symbol bars > Object modes" (view page 89).

Undo

Sequoia offers a secure way of tracking your changes in Virtual and Wave Projects, which makes it easier to undo work steps. Up to 100 changes can be kept in the memory using the undo function in "Menu > File > Program Preferences > Undo Definitions (view page 702)".

When editing audio material with destructive effects ("Offline Effects menu"), the Undo option only works when the option "Create copy" is activated in each dialog.

Shortcut: Ctrl + Z

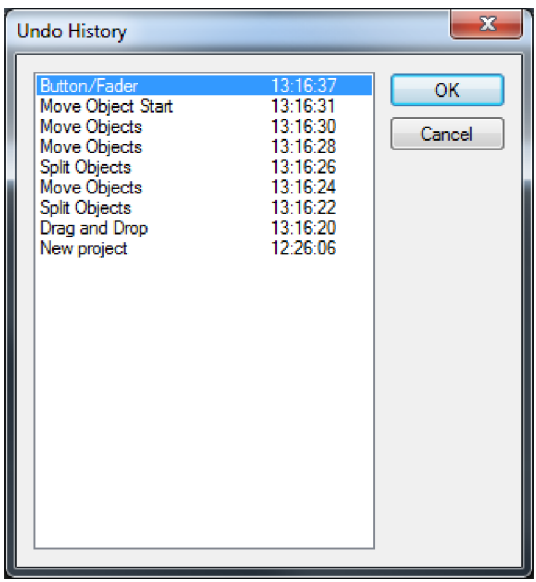
Restore

"Redo" (undoes) the most recent Undo command.

Shortcut: Ctrl + Y

Undo History

A list featuring the most recent executed commands appears.



In this dialog, you can return to an earlier work stage. This list can be removed in the "Edit -> Delete undo levels (view page 721)".

Delete Undo History

Deletes the undo levels of a project.

Copy

Copy

The current range is copied into the clipboard. Please note that any previous clipboard content will be deleted.

Shortcut: C, Ctrl + C or Ctrl + Ins

Copy as... (Wave-Project)

This function copies the selected range from a wave project into a new file.

Shortcut: Shift + C

Paste with Time/Ripple

Copies the selected range and saves it as an entry in the ClipStore. Read more about the ClipStore in the chapter "Manager (view page 178)".

Shortcut: Ctrl + Shift + C

Copy + Clear

Copies the current range to the clipboard (clip), removes it from the project, and replaces it with silence.

Shortcut: Ctrl + Alt + C

Cut

Cut

Virtual Projects (VIP)

In the case of virtual projects, the selected area will be copied into the virtual clip (VirtClip) and removed from the active project. The VirtClip, like the virtual projects, does not contain the actual audio files, just the links to them. The VirtClip spans over the same number of tracks as the selected area.

Keyboard Shortcut: X

Wave projects

The data of the selected range is copied into the clip (view page Fehler! Textmarke nicht definiert.) and removed from the current project. Data behind that range takes its place. The Wave project is reduced in size by the length of the removed segment.

If you cut a piece out of a mono wave project, the clip will be a mono wave project, and if you cut out a piece from a stereo wave project, the clip will also be a stereo wave project. Similarly, the bit resolution and the sample rate from the project are applied. The old content of the clip is overwritten.

After the function has been carried out, the playback marker will be at the cut edge of the removed segment. You can now place the clip back into the wave project with the command "Paste from clip".

Cut with Time/Ripple

This command deletes the marked range and copies it into a clip. The material behind the removed range will be moved forward by the length of the range (ripple, shuffle).

Shortcut: Ctrl + Alt + X

Cut to ClipStore

Copies the selected range from the project and saves it as an entry in the ClipStore.

Shortcut: Ctrl + Shift + X

Split

Split Objects

Use this option to split all selected objects at the cursor position and separate them into individual objects.

If the range is selected, the cut will be made at each border. Ensure that the object that is to be edited is contained within the active range. If necessary, all objects may be activated by double clicking on the range selected in the range bar. In this case, it is not necessary to select the objects beforehand.

Shortcut: T

Split Objects with Alternate (Linear) Fade

Use this option to split all selected objects at the cursor position, thereby separating them into independent objects. The curve defined as Default_Crossfade_ALT in the crossfade editor will be applied as the crossfade.

Shortcut: Shift + T

Detailed information about Sequoia's crossfade editor is provided in "Crossfade editor -> Crossfade presets (view page 436)".

Split Objects at Project Markers

The object is split into separate objects at the marker positions within the object borders. The newly created objects will be named after the previous markers.

Split Objects at CD Track Indices

The object is split into separate objects at the marker positions within the object borders. The newly created objects will be named after the previous track markers.

Split Objects at Audio Markers

The object is split into separate objects at the audio marker positions within the object borders. The newly created objects will be named after the previous audio markers.

Heal/Unsplit Objects

This command recombines two objects that have been split without any audio calculation. To do this, the objects in the VIP must be positioned exactly beside each another, and the audio material must be connected. The objects must therefore be different but sequential pieces of audio material from the same wave project. If this not the case, the command will not function properly.

Before executing the command, select both of the objects that were split previously. All of the second object's properties, like fades and effects, will be discarded. The resulting combined object will feature the properties of the first object.

Insert

Paste/Insert Clip

The data contained in the clip or VirtClip is inserted at the current position of the playback marker (position line) or at the beginning of the current range.

In the "Link objects" modes, objects behind the insert point are moved to the end. In all other modes, the inserted material replaces the existing objects.

A new area will automatically be defined for the inserted files.

Shortcut: V, Ctrl + V, Shift + Insert

Insert from Clip Manager (ClipStore)

Selected clips are inserted at the playback marker position or in place of the selected range.

Keyboard shortcut: Ctrl + Shift + V

Paste with Time/Ripple

This menu option inserts the clip at the position of the cursor or at the start of the range. The audio material after the cursor position will be moved down by the length of the range (ripple, shuffle).

Shortcut: Ctrl + Alt + V

Overwrite with Clip

The current range will be overwritten with the contents of the clip. The length of the clip will not be reduced. The data previously located at this position will be overwritten.

In virtual projects, the selected range determines the start position and track for the clip contents.

Shortcut: Alt + V or Insert

Mix with Clip

The content of the range is mixed with the content of the clip. This command is available only in the destructive wave edit mode. Please note that with this function both components are entered into the mix at 100%, being added together. This can

cause overmodulation. You may have to adjust the amplitude of the project by normalizing prior to this.

Crossfade with Clip

The content of the range is crossfaded with the content of the clip. This command is only available in the destructive wave edit mode. In this case, the position of the playback marker specifies the end of the clip to be faded with the wave project. The length of the clip sets the length of the crossfade.

Delete

Virtual projects (VIP)

Use this command to delete the selected objects from the virtual project. If a range is selected, then the track automation curves will also be deleted in the range. If objects are only partially in the range, then the corresponding parts of the objects will be deleted.

In connection with the object modes (view page 89), "connect objects..." moves the objects located behind the deleted range forward, and the track will be shortened accordingly. In the other object modes, the range is replaced by silence.

Keyboard shortcut: Del

Deleting Audio Files

The data in the selected range is deleted and the subsequent audio segment is moved back to fill the gap.

Caution: If you are using offline Audio Edit Mode and have deactivated the "Undo" function ("Options -> Program preferences -> Undo Definitions (view page 702)"), the files cannot be restored.

Ripple delete

This command deletes the marked range. Material later than the end of the range will be prolonged by the length of the range.

Shortcut: Ctrl + Del

Extract

Press "Extract" to trim your project by deleting the areas outside of the selected segment.

Virtual projects (VIP)

The extraction function results in all objects outside of the selected segment in a virtual project being deleted. All tracks stay unchanged, even if no object is found in the selected segment. This function is not track-specific.

Individual objects can be cropped using "Object -> Trim Objects (view page 785)".

Extracting Audio Files

In Wave Editing Mode only the part of the audio file below the selected area remains. The data before and after the range is deleted.

Silence

Insert Silence

This command inserts silence to all tracks at the position of the current play or at the start position of the selected range. The objects will be split at this position and the files after it will be added after the silence.

If a range is selected, its length will be adapted as default. The selected range is not adjusted after the operation. It's possible to change the value in the input field. The project is extended by the length of the inserted object.

Strip silence

Use this function to split objects at positions where silence occurs. After applying the command the objects are split in such a way that the silent parts of the objects are separated from them. You can specify a threshold value for both silence and wanted signal in the dialog beforehand. The cut objects are selected to make it easier to delete them using the "Del" button.

Threshold for silence detection (dB): If the signal falls below this value, the object will be cut at the corresponding position.

Threshold value for used signal (dB): If the signal rises above this value, the object will be cut at the corresponding position.

The "**Minimum length of silence**" parameter sets the length which silent passages should be applied so that they can be selected and deleted. This way, you can remove very short passages of silences from the cut.

With the "**Crossfades**" option, you can create automatic crossfades between at the cut positions of the objects.

Use sample space accuracy: With a new, improved algorithm, the processing speed of this function can be significantly increased. If absolute accuracy is not important when cutting, disable the option for a speed advantage.

Replace with silence

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 track stays unchanged.

Keyboard shortcut: Alt+Del

Tempo

Tempo map...

Detailed information about the tempo map dialog can be found in the chapter "Tempo editing > Tempo map dialog" (view page 444).

Tempo map BPM mode/grid adjustment mode

Detailed information about the tempo map grid fit mode can be found in the chapter "Tempo editing > Grid tapping" (view page 448).

Insert tempo change

This opens the Tempo and time signature dialog (view page 454) to insert a tempo change.

Insert time signature change

This opens the Tempo and time signature dialog (view page 454) to insert a time signature change.

Insert grid position marker

Here you can insert grid position markers (view page 444) at the current playback marker position.

Change Global Tempo

In this dialog you can change the tempo of the project by entering a new value. In this case the value represents the change factor whereby "2" means that the tempo is doubled (in BPM) and "0.5" means that the tempo is halved.

Ignore all tempo markers, use project tempo only

With this option you can ignore any previously created tempo markers in your project, so that only the project tempo observed.

Metronome Active

This command activates/deactivates the metronome. This function can also be activated via the "Click" button in the transport controls.

Metronome options

Detailed information about this is available in "File -> Program Preferences -> Metronome Options (view page 95)".

Create click track

The new command **Menu Edit > Tempo > Create click track** creates an audio track, which contains all metronome clicks as objects.

Range

Range All

This expands the range to include the entire project. Double clicking the timeline or pressing "A" selects the range in the selected track; double clicking again or pressing "A" again selects the range over all tracks, and double clicking or "A" once again restores the simple timeline selection.

Shortcut: A

Manipulate Range

Move Range Start Left

This function moves the beginning of the range in the current arranger window one grid unit to the left. The length of the movement depends on the set grid value. This may be adjusted via "Project options -> General" (shortcut: "I", "Ctrl + Shift + #").

Shortcut: Alt + ÷ (number block), right arrow

Move range start to right

This function moves the beginning of the range in the current arranger window by one grid unit to the right. The step size is defined by the set grid value.

Keyboard shortcut: Alt + * (number pad), right arrow

Move range end to left

This function moves the beginning of the range in the current arranger window one grid unit to the left. The step size is defined by the set grid value.

Keyboard shortcut: Alt + - (number pad), Shift + left arrow"

Move range end to right

This function moves the end of the range in the current arranger window one grid unit to the left. The step size is defined by the set grid value.

Keyboard shortcut: Alt + + (number pad), Shift + right arrow

Range to Beginning

Choose this option to extend the currently marked range to the beginning of the project (Wave Project or VIP).

Range to End

Choose this option to extend the currently marked range to the end of the project (Wave Project or VIP).

Flip Range Left

Choose this option to flip the currently marked range to the left (this swaps the start of the marked range to the end of the marked range.)

Shortcut: Ctrl + Shift + left arrow

Flip Range Right

Choose this option to flip the currently marked Range to the right. (What used to be the end of the marked Range is now the beginning of the marked Range.)

Shortcut: Ctrl + Shift + right arrow

Find zero crossings

Beginning of Range -> 0

Choose this option to move the beginning of the currently selected range to the next zero amplitude crossing. "Zero" is the next sample value featuring a ZERO value or that indicates the boundary between a positive and a negative sample value (or vice

versa). This function is especially useful when searching for loop points. Remember to zoom into the waveform display far enough so that you'll be able to see the actual changes in the arranger window.

Shortcut: Ctrl + Page up

Beginning of Range <- 0

Choose this option to move the beginning of the currently marked range left to the previous zero amplitude crossing.

Shortcut: Shift + Page up

End of Range -> 0

Choose this option to move the end of the currently selected range to the next zero amplitude crossing.

Shortcut: Ctrl + Page down

End of Range <- 0

Choose this option to move the end of the range left to the previous zero amplitude crossing.

Shortcut: Shift + Page down

0 -> Range <- 0

Choose this option to move the beginning of the range right to the next zero amplitude crossing, and to move the end of the range left to the previous left zero amplitude crossing.

Range start to left marker

Choose this option to extend the Range to the next marker on the left.

Shortcut: Shift + F2

Note: This command also takes set object audio markers into account as soon as they have been selected via menu "File > System preferences > System/audio -> Design -> View options".

Range Start to Left Object Border

The start of the range is set to the next object edge on the left.

Range end to right marker

The end of the range is set to the next marker to the right.

Shortcut: Shift + F3

Note: This command also takes set object audio markers into account as soon as they have been selected via menu "File > System preferences > System/audio -> Design -> View options".

Range End to Right Object Border

The end of the range is set to the next object edge on the right.

Area Above all Selected Items

Choose this option to extend the range over all selected objects.

Remove Range

This command removes the current range from the grid and marker bar. This function is also available by right clicking in the grid and marker bar.

Range Not Over Tracks

This command selects the selected range from the grid and marker bar only, and not in the tracks. Switch between the "Range not over tracks", "Range over current track", and "Range over all tracks" functions by double clicking in the selected range in the grid and marker bar.

Range Over Current Track

This command selects the selected range from the grid and marker bar as well as in the selected track.

Range Over All Tracks

This command selects the selected range from the grid and marker bar as well as in all selected tracks.

Range Length

Set the range starting from the playback marker to lengths of 1, 2, 4, 8 or 16 beats here.

Split Range

Use this function to split the arranger window into three sections. In this case, the upper section displays the entire project in an overview, while the left lower section displays the start of the range and the right lower section displays the end of the range (enlarged sections).

The range borders may be set precisely in the lower sections, provided the grid setting "Grid/frames" is set in the "Project options" beforehand (shortcut: "I").

Shortcut: B

Back to range: "Shift + B"

Split Range for Video

This function is especially useful when working with AVI videos. The upper section displays the entire project, the lower left section displays the range start, and the lower right section indicates the range end. For the lower sections, a zoom depth of 1 frame is preset to make image-precise cutting and editing possible.

To undo the split, restore the original range by pressing "Shift + B".

Store Range

Sequoia enables ranges to be stored by selecting the corresponding range number.

All current ranges in a project may be viewed via "Tools -> Manager -> Range manager".

Shortcut: Alt + F2 ... F10

"Alt + F4" however, should not be used, since this is a Windows command that closes the current screen. Similarly, "Alt + F9" should not be used either, since this is used for the 4-point cut command in Sequoia.

The "Other..." dialog may be used to define additional ranges featuring freely assigned names.

Shortcut: Alt + F11

Get Range

Open previously saved ranges via this command. Getting a range is also possible during playback. This enables you toggle between various ranges in order to compare them acoustically.

Shortcut: Ctrl + F2 ... F10

Ranges may be named and accessed via "Tools -> Manager -> Range manager".

Get Range Length

Use this function to set the current range to the length of the corresponding saved range.

Shortcut: Ctrl + Shift + F2 ... F10

Range Manager...

The "Range manager" displays all saved ranges of the current project.

Detailed information about the "Range manager" is available in the chapter "Manager -> Range manager (view page 200)".

Shortcut: Ctrl + Alt + Shift + B

Recall Last Range

Use this function to recall the last selected range. Repeated application accesses ranges up to five levels back.

Shortcut: Shift + Backspace

Crossfade

Crossfade Editor

Sequoia contains a highly professional crossfade editor.

The crossfade editor is opened by clicking the toolbar button or via menu "Edit -> Crossfade editor". The preset shortcut is "Ctrl + F".

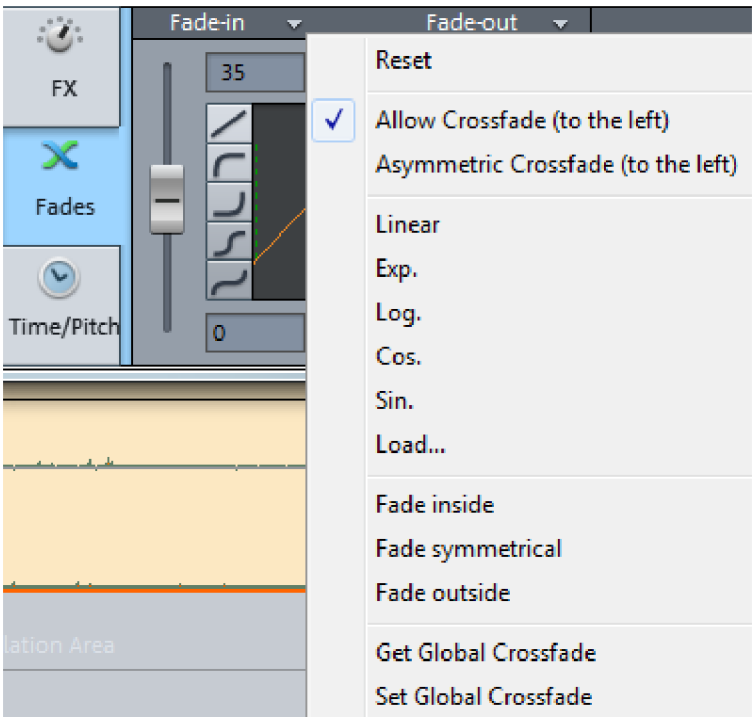
Use the crossfade editor to conveniently and efficiently edit all aspects of a crossfade. All changes are non-destructive.

Detailed information about the crossfade editor is provided in "Crossfade editor (view page 425)".

Auto crossfade mode active

With this option, all newly recorded or cut material in a VIP has an automatic fade-in applied to the start and end.

Every object may be assigned a standard fade-in, and this may be edited via the object editor fade menu via "Get/set global crossfade".



If two objects overlap in this mode, a real-time crossfade will occur at the intersection. The fade settings can be modified using a double click on each object in the object editor.

If required, you can manually edit each intersection with the crossfade editor or object handles.

Crossfade Editing

Move Out Point Left

This function moves the out-point of a crossfades to the left.

Move Out Point Right

This function moves the out-point of a crossfades to the right.

Move OutPoint to Cursor

This function moves the out-point of a crossfades to the current playback marker position.

Move In Point Left

This function moves the in-point of a crossfades to the left.

Move In Point Right

This function moves the in-point of a crossfades to the right.

Move InPoint to Cursor

This function moves the in-point to the current playback marker position.

Crossfade Editor OK

This function closes the crossfade editor and applies the current values.

Crossfade Editor Cancel

This function closes the crossfade editor and does not apply the current values; the original condition is restored.

Revert Crossfade

Use this function to restore the original condition of the crossfade editor.

Previous Crossfade

This function jumps the crossfade editor to the previous crossfade on the same track. This enables crossfades to be edited quickly in sequence, without having to leave the crossfade editor.

Next Crossfade

Use this function to jump the crossfade editor to the following crossfade on the same track. This enables crossfades that follow each other to be edited quickly, without leaving the crossfade editor.

Set Focus to Crossfade Window

This function places the input focus on the crossfade editor. Keyboard entries that follow will then also affect the crossfade editor and not the audio window.

Set Focus to Project Window

Use this function to set the input focus on the audio window. Keyboard entries that follow will then also affect the audio window and no the crossfade editor.

Crossfade Editor Preferences...

Please see "Crossfade editor (view page 425)" for more information about these options.

Allow Crossfade (to the left)

This function displays the progress of the crossfade to the left.

Asymmetrical Crossfade (to the left)

This function displays the asymmetrical progress of the crossfade to the left.

Linear Crossfade

This function displays the linear progress of the crossfade.

Source Destination Cut

The Source Destination Cut (view page 400) is a convenient way to conduct complex cut operations with markers.

Set/Delete In/Out Point(s)

Set In Point

This menu item sets the in-point destination (target) for a Source Destination Cut.

Set In Point (Dest.)

This menu item sets the in-point of the destination track (target) for a Source Destination Cut.

Set In Point (Source)

This menu item sets the in-point of the source for a Source Destination Cut.

Set Out Point

This menu item sets the out-point for a Source Destination Cut.

Set Out Point (Dest.)

This menu item sets the out-point of the destination track (target) for a Source Destination Cut.

Set Out Point (Source)

This menu item sets the out-point of the source track for a Source Destination Cut.

Delete In Point

This menu item deletes the in-point destination for a Source Destination Cut.

Delete In Point (Dest.)

This menu item deletes the in-point destination (target) for a Source Destination Cut.

Delete In Point (Source)

This menu item deletes the "in point" of the source track (source) for a Source Destination Cut.

Delete Out Point

This menu item deletes the out-point for a Source Destination Cut.

Delete Out Point (Dest.)

This menu item deletes the out-point of the destination track (target) for a Source Destination Cut.

Delete Out Point (Source)

This menu item deletes the out-point of the source track for a Source Destination Cut.

Delete Destination Points

This menu item deletes the in and out points of the destination track.

Delete Source Points

This menu item deletes the in and out points of the source track.

Move to In Point

This menu item places the playback marker and visible screen section at the in-point on the destination track.

Move to Out Point

This menu item places the playback marker and visible screen section at the destination track's out-point.

Zoom to In Point

This menu item places the visible screen section at the in point on the destination track.

In Source Destination Cut mode (see the chapter "Source Destination Cut (view page 400)"), the screen is zoomed to the in point of the active track. This may be a source track as well as the destination track. The system settings "**Y -> Program -> Source Destination Cut**" enable you to define the time span in milliseconds for the in point zoom display. The keyboard shortcut "**Ctrl + Alt + Shift + Page up/Page down**" moves the cursor to the in/out point in the active track by the zoom factor. The preset value is a time of 2000 ms.

Zoom to Out Point

This menu item places the visible screen section at the destination track's out point.

In Source Destination Cut mode (see the chapter "Source Destination Cut (view page 400)") the screen is zoomed to the out point of the active track. This may be a source track as well as the destination track. The system settings "**Y -> Program -> 4 point cut**" enable you to define a time span in milliseconds for the out point zoom display. The keyboard shortcut "**Ctrl + Alt + Shift + Page up/Page down**" moves the cursor in the currently selected track by the zoom factor to the in/out point. The preset value is a time of 2000 ms.

Insert

This menu item makes an insert cut including the source/destination in and out points. Existing audio material in the destination range is overwritten with the source material in this case.

Insert with Timestretching

In this case, the source material is stretched between the destination in-point and destination out-point. Existing material in the destination track is replaced with the material from the source track.

This enables the timing of drum loops to be adjusted to the existing arrangement, without the need to apply complicated operations. Timestretching is applied non-destructively.

Keyboard shortcut: Alt+F9

Insert at source position

You can use this cut command in synchronous cut, if the absolute time position is already set in the source material. You don't need to set any Destination markers, markers that were previously set will be ignored. The cut retains the time position of the source material.

Shortcut: Ctrl + Alt + F9

Insert with Ripple

This menu item applies an insert cut including the source/destination in and out-points. The audio material in the destination track behind the cut is moved forward by the cut length (ripple, shuffle).

Delete and Ripple

This command deletes the marked range between the in and out-point. The material behind the removed range will be moved forward by the length of the range (ripple).

Delete Silence

This menu item deletes the material between the in and out-points in the active source or destination track. The cut length range is replaced with digital silence in this case.

Blackfade insert

This menu point leads to an insert cut that takes into consideration the source and destination in and out points. The material in the destination range behind the cut will not be moved.

The preset black fade ("Default_FadeFromBlack", "Default_FadeToBlack") is used. This preset can be customized in the Crossfade Editor (view page 425).

Blackfade insert with ripple

This function moves the audio material behind the cut range in the destination track back by the cut length (ripple) after inserting.

The preset black fade ("Default_FadeFromBlack", "Default_FadeToBlack") is used. This preset can be customized in the Crossfade Editor (view page 425).

Blackfade Delete and Ripple

This command deletes the marked range between the in and out-point. Material later than the end of the range will be prolonged by the length of the range.

The preset black fade ("Default_FadeFromBlack", "Default_FadeToBlack") is used. This preset can be customized in the Crossfade Editor (view page 425).

Blackfade delete silence

This menu item deletes the marked range between the destination in and out-point. The material behind the cut will not be moved.

The preset black fade ("Default_FadeFromBlack", "Default_FadeToBlack") is used. This preset can be customized in the Crossfade Editor (view page 425).

Activate Source

This menu item activates the range with the source material in Source Destination Cut Mode (see the chapter "Source Destination Cut -> Source Destination Cut Mode (S/D special mode) (view page 409)".

Activate Overview (MuSyC)

This menu item activates the overview project in case you are working with a Multi-Synchronous Cut (see chapter "MuSyC - Multi-Synchronous Cut (view page 411, view page 422)").

Activate Destination

This menu item activates the range with the destination material in Source Destination Cut Mode (see chapter "Source Destination Cut (view page 400)".

Source Destination Cut Mode

This menu item activates the source/destination cut mode (see the chapter "Source Destination Cut -> Source Destination Cut Mode (S/D special mode) (view page 409)".

Source Destination Cut Settings

This menu item opens the settings dialog for 4 point cut. Detailed information about this dialog is available in the chapter "Source Destination Cut -> Source Destination Cut Settings (view page 410)".

Multi-Synchronous Cut

Detailed information about multi-synchronous cuts is available in "MuSyC - Multi-Synchronous Cut (view page 411, view page 422)".

MuSyC Preparation...

This command opens a dialog for preparing a Multi-Synchronous Cut.

Detailed information about multi-synchronous cuts is available in "MuSyC - Multi-Synchronous Cut (view page 411, view page 422)".

Take Assistant...

This command opens the MuSyC take wizard.

Detailed information about multi-synchronous cuts is available in "MuSyC - Multi-Synchronous Cut (view page 411, view page 422)".

Object Alignment

Detailed information about multi-synchronous object assignment is available in "MuSyC - Multi-Synchronous Cut (view page 411, view page 422)".

Time Alignment

Detailed information about multi-synchronous time adjustment is available in "MuSyC - Multi-Synchronous Cut (view page 411, view page 422)".

Fast Insert

Detailed information about this MuSyC function is available in "MuSyC - Multi-Synchronous Cut (view page 411, view page 422)".

Play Source

Detailed information about this MuSyC function is available in "MuSyC - Multi-Synchronous Cut (view page 411, view page 422)".

Play Destination

Detailed information about this MuSyC function is available in "MuSyC - Multi-Synchronous Cut (view page 411, view page 422)".

Activate Overview (MuSyC)

Detailed information about this MuSyC function is available in "MuSyC - Multi-Synchronous Cut (view page 411, view page 422)".

Source Destination Cut Settings (MuSyC)

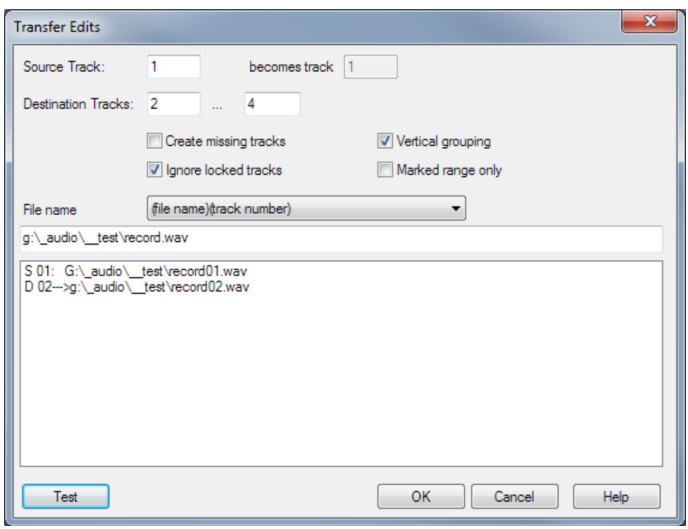
Detailed information about this MuSyC function is available in "MuSyC - Multi-synchronous cut (view page 411, view page 422)".

Transferring cuts

Using this function, cuts from one track can be transferred to other tracks.

For multitrack recordings, you can create a rough cut only on the source track before transferring the cuts to recording destination tracks.

This reduces the CPU load and enables faster editing.



Transferring Edits – Requirements

It is essential that the tracks are named systematically prior to recording projects like this. In this case, allocating an incremental track number or a well-defined track name is necessary. The project name is used as the default name; the program automatically adds the track number to this.

Project name = "Recording"

Track	Recording file
1	Recording01.wav
2	Recording02.wav
3	Recording03.wav

Note: During recording, additional suffixes are added automatically.

Bitdepth: _24/_32

Mono: M

Takes: _T001

Complex names can also be given to the takes. Once names have been assigned to individual tracks, you can specify various name standards for the recorded files in the **track settings** (Alt + I) under "**Recording**" by pressing the "**All tracks**" button. The following combinations are available:

(filename)(track number)

(project name)(track number)

Track name

(track number)(track name)

(project name)(track name)

You can also retrieve track names from the first object names or from the first object files. By selecting "project name_track name", the following name permissions open for the recording files:

Project name1 = "The_Magic_Flute_Recording20090101"

Track	Track name	Recording file
1	01 Mix	The_Magic_Flute_Recording20090101_01Mix .wav
2	02 Tamino	The_Magic_Flute_Recording20090101_02Tamino.wav
3	03 Papageno	The_Magic_Flute_Recording20090101_03Papageno.wav

Note: As soon as you have to change a track name, press the "All tracks" button again to update the file names.

Now you can start the recordings. This can be contained within one single project. However, it's also possible to record various performances in various projects, for example:

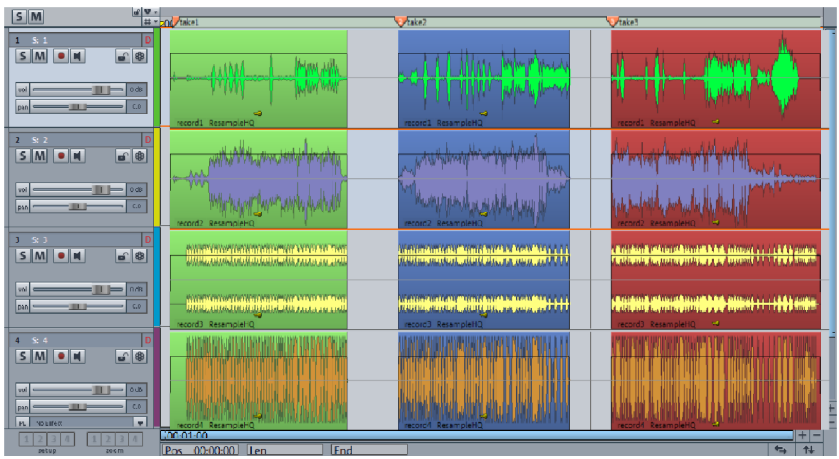
Project name2 = The_Magic_Flute_Recording20090102

Project name3 = "The_Magic_Flute_Recording20090103...

Important: We recommend making recordings using identical settings (bit rate, sample frequency). You should also record all tracks uniformly in mono or stereo in order to avoid problems when transferring cuts later.

Transferring Edits – Example

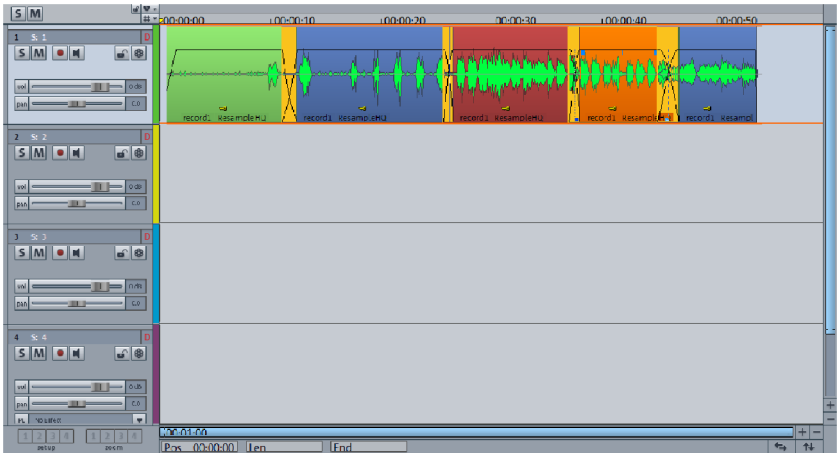
In this case, we are recording three takes into one project, represented by three different recordings.



You may now delete objects located outside of the reference track.

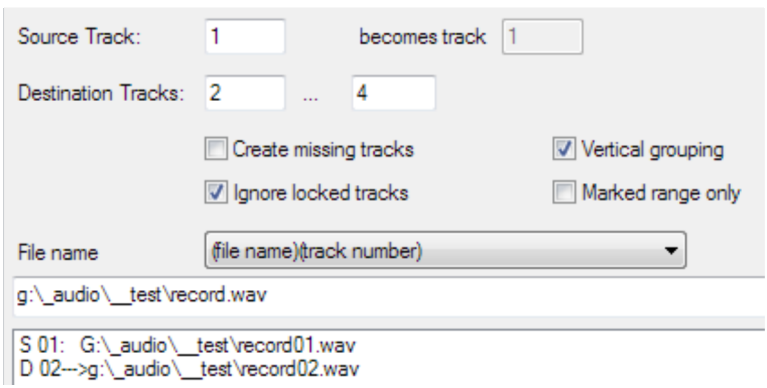
Note: If you have changed the naming conventions via the function "All tracks" in the track settings, you shouldn't delete the empty tracks from the project, since this assignment is still required for the cut transfer.

Now the cut may be made in the reference track. The result is a montage of objects from various takes for the reference track.



Select the reference track and open the "Transfer cuts..." dialog from the "Edit" menu.

Transferring Edits – Dialog



Source track is the cut reference track featuring cuts to be transferred.

Destination tracks are the tracks where the audio material is available and should be cut according to the reference track, for example, track 2...4.

Create missing tracks would create new tracks for the desired track number, if not already present.

Ignore locked tracks leaves the corresponding tracks untouched.

Vertical grouping groups all objects one under the other to prepare them for multitrack cutting.

Marked range only limits the cut transfer to one part of the project.

The file name is generated from specific parameters. The following options are available:

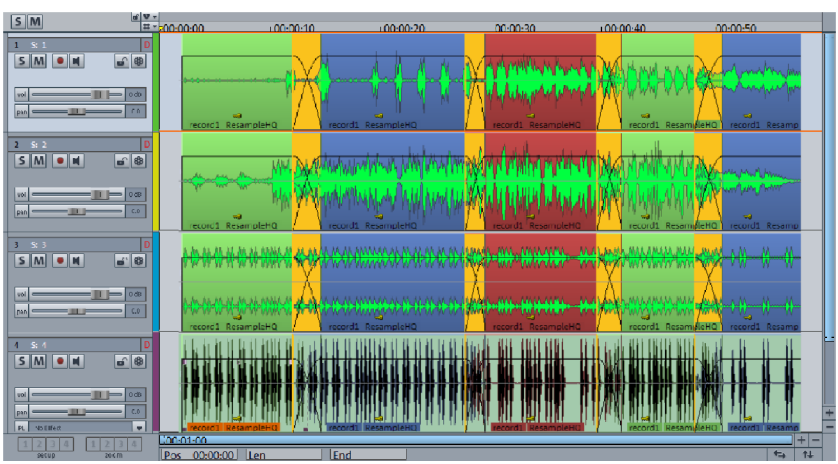
(file name)(track number)	Specify a file name in the field beneath it; the number of the corresponding track is then automatically added to this.
...(track number)...	The only difference in the file name is in the track number.
...(track name)...	The name for the track is as the same as in the corresponding track.
Track record file name	The name for the record file is the same as the corresponding track in the track settings.

The dots "..." mean that other static components may be present in the file name, e.g. the project name (The_Magic_Flute20090101_01Mix.wav).

The display window features the physical files in track 1 that start with **S 01**. The following track features the file names detected that would be **D 02**. This checks already whether this file is present. If this is not the case, the file will be labeled (NOT FOUND).

The display is limited to two tracks. To check all tracks for plausibility, press the "Test" button. If an error occurs, the first corrupt file will be shown. If this happens, check your input and the existing file names in the project index.

Pressing "OK" to close the dialog transfers all cuts and the properties of the reference objects like object volume, resampling, and effects to related tracks. The result is a complete multitrack cut that may be edited as usual.



Auto ducking function

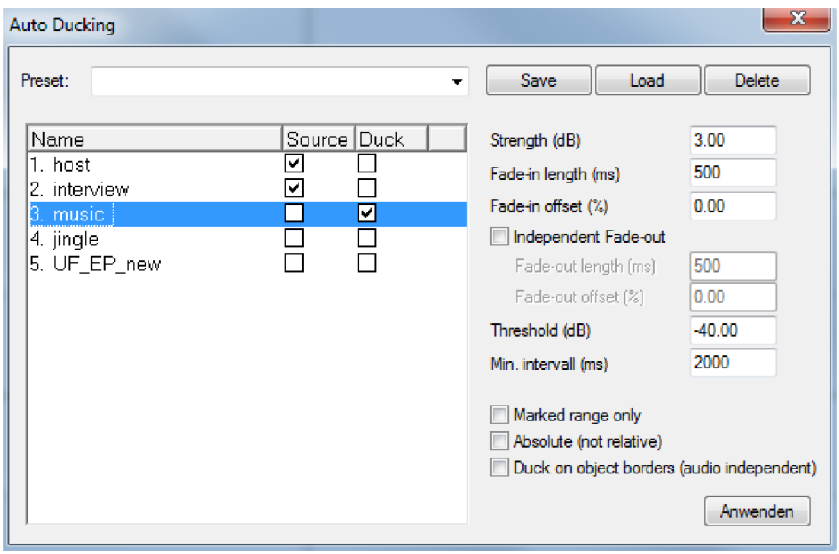
Auto Ducking refers to a dialog based tool used to manipulate the volume automation curve of one or more **target tracks**. Edits are made using the volume of the audio material of one or more **source tracks**.

Auto ducking can be used to create volume curves for automatically fading background music during spoken sections. To do this, proceed as follows:

1. Create a montage from speech parts and music.

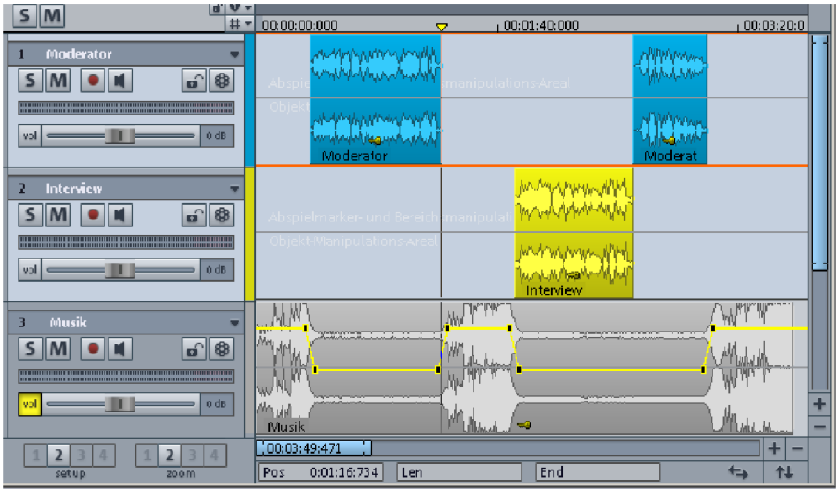
2. Insert the sorted audio material on separate tracks.

3. Open the Auto Ducking dialog (Menu Edit > Auto Ducking).



4. Decide which tracks ("**Duck**") should be influenced by the volume of other tracks ("**Source**"). In this example the background music will be influenced by two speech tracks.

5. Click on "Apply". On the "ducked" track a volume curve will be created, which automatically fades the background music in and out at the points where speech recordings with corresponding levels are available.



Additionally **the following options are available**:

Strength: To what extent the track should be reduced in dB. Negative values allow increases.

Fade-in length (ms): Length of the transition at the start of a volume adjustment.

Fade-in offset: The offset to be used for fade-ins. At 0% the volume change will begin at the start of the source material. At 100% the fade will happen entirely before the source material.

Independent Fade-out: If active the length and offset can be set for fade ins/fade outs dependently.

Fade-out length (ms): Length of the transition at the end of a volume adjustment.

Fade-out-offset: The offset to be used for fade outs. At 0% the volume change will begin at the start of the source material. At 100% the fade will happen entirely after the source material.

Threshold: Value in dB that the source material must exceed in order to trigger a ducking.

Min.Interval (ms): If the source material falls below the threshold, and source material above the threshold is available inside the minimal interval, the volume in this interval will not be affected. This is useful for example to straddle short pauses in speech.

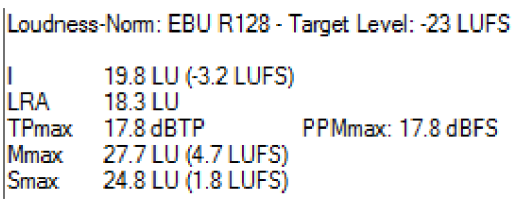
Marked range only: Auto ducking will only be used in the selected range.

Absolute (not relative): When you press the apply button the volume curve is recalculated and old curve points are deleted. If this option is switched off, a relative change will ensue, i. e. curve points that are already available will be adjusted to the value of the change. Take note that reductions will accumulate in relative mode when applied multiple times.

Duck on object borders (audio independent): Ducking is independent of audio material and exclusively affects the object edges of the source track.

Calculate Loudness

This command calculates the loudness (view page 1128) for the entire project or a selected range and displays the loudness value. You can also access this function by right-clicking on an audio file in the file manager.



In WAV files the loudness values are included in the metadata of the file (according to EBU 3344). For all other files they are only show temporarily.

In this dialog you also have the option of clicking on the corresponding button to enable the adjustment of the program loudness with the volume fader.

Batch Processing

Batch processing lets you automate work processes for several files. This makes it possible to define an editing sequence that will be applied to all selected files.

Each "job" is added to the job list. You can define multiple batch jobs which will then be executed one after the other.

Possible editing options:

- Loudness or peak level normalization
- Linear fading (in and out)
- All of the realtime effects included in Sequoia (including restoration and VST effects)
- Time Stretching / Pitch Shifting
- Remove DC offset

- Target format settings: Word width (8/16/24/32 float), sample rate, stereo/mono settings, file format conversion, compression process
- Save settings and name the created file while retaining the original
- Behavior when dealing with critical material - Error handling (view page 754)

For example, you can normalize a whole folder full of 24-bit wave files, apply 5 ms fades at the beginning and the end of each file, compress them with the multi-band compressor, change them all to 16-bit mono and then save them all as MP3 files.

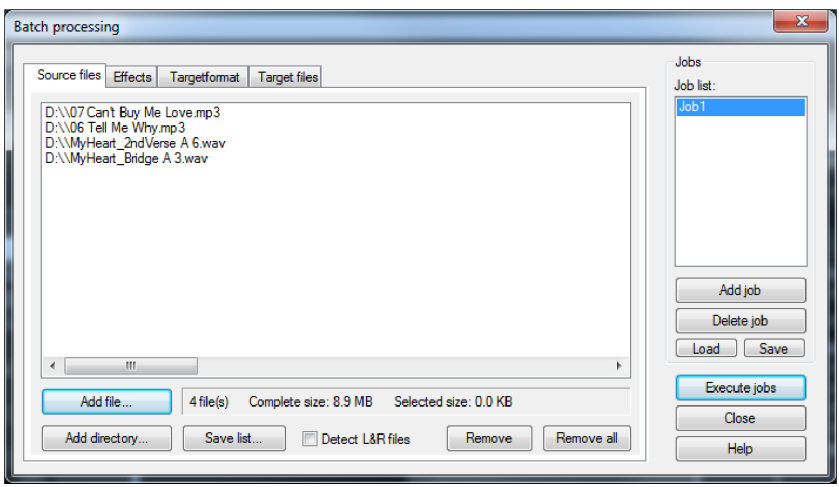
Jobs

On the right border of the batch processing window is the job list. New jobs are created by clicking the "**Add job**" button. If you select a job (with a mouse click), the four settings tabs are displayed (source files, effects, target format, target files). "**Delete job**" removes the selected job. "**Perform jobs**" starts the batch processing.

With **„Load"** and **„Save"** you can load and save jobs in XML format.

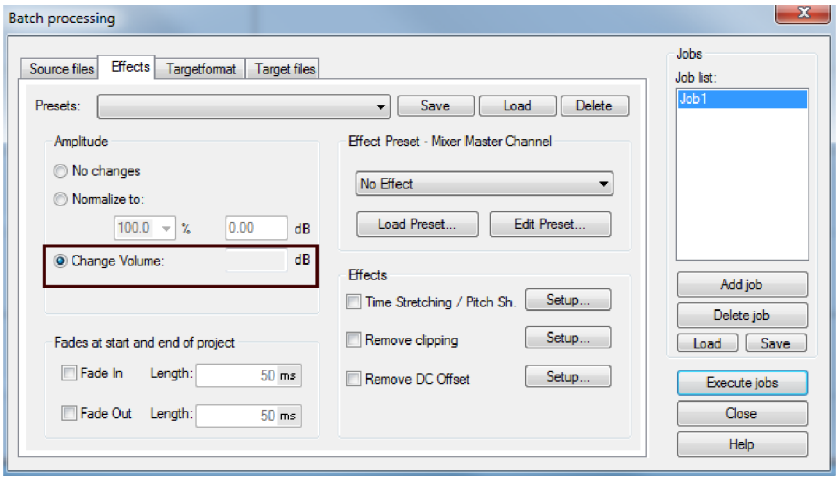
Even if you close the batch processing window temporarily, all created jobs will remain saved until Sequoia is closed.

Source Files



Create the list of the files for editing with the "**Add file**" button. Multiple selection is possible. Select **"Load folder"** to add all of the audio files in a folder (including all subfolders). You can load all file formats that Sequoia imports. "**Save list**" creates a playlist in *.m3u format for selection later. The option "**Detect L&R files**" makes it possible to edit left and right files as stereo. "**Remove**" deletes all selected list entries. "**Remove all**" deletes the complete list.

Effects



1. **Normalize amplitude**: You can enter a target maximum amplitude value in % or dB. For example, a value of 75% corresponds to -2.5 dB. You can also enter volume changes in dB.

Detailed information about normalization is available in "Effects > Amplitude > Normalize (view page 860)".

Loudness adjustment (only Sequoia): Here you can adjust the program settings for loudness in LUFS (e. g. EBU R128 Loudness Norm) for calculating loudness values. This way you can set the loudness value for selected files.

2. **Load effect presets from the master channel:** It is possible to integrate existing master channel effect settings into the batch processing using mixer-preset files.

- The Sequoia mixer enables all of the mixer settings to be exported as a mixer preset. To do this, click the "Save mixer settings" button in the mixer. This is located under the mixer snapshot storage slots.
- When you load the *.mix file into the batch processing function with "**Load preset**", all effect settings for the master channel will be calculated into the file batch. The "**Edit preset**" option opens a special FX routing dialog that provides access to all of the effect parameters.

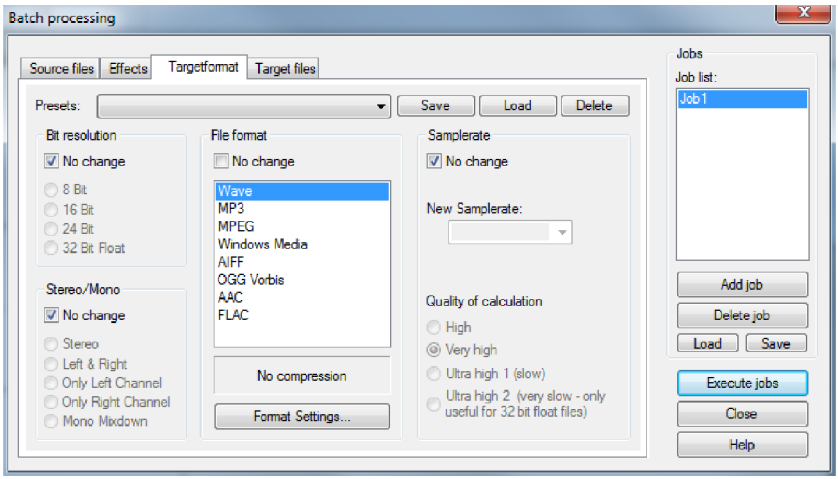
Detailed information about the FX routing dialog is provided in the chapter "Mixer - buses and routing -> Effects routing/plug-ins dialog (view page 388)".

Additional effects: In addition to the master effects of the mixer presets, non-real-time effects such as "Pitch Shifting/Time Stretching", "Remove DC Offset",

and "Remove clipping" can also be applied. The "Settings" button opens the corresponding effects dialog.

3. Fades at start and end of project: Linear fades can be added to the start and end of files. The fade lengths can be set freely.

Target format

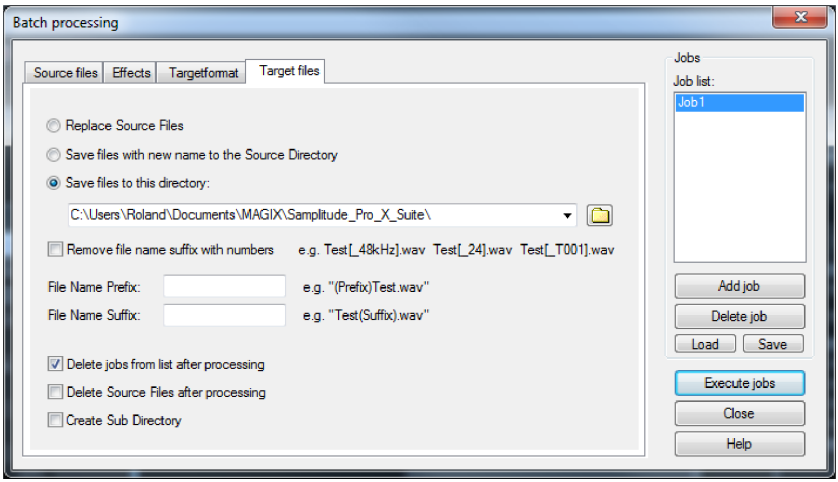


All settings in this dialog can be saved as presets. You can freely choose the bit resolution (8/16/24/32 bit float), the sample rate, stereo/mono/left/right as well as the save format with the corresponding format settings.

You can find out more about sample rates in the "File > Effects > Sample manipulation > Adjust sample Rate (view page 975)".

You can find out more about file formats in the "Menu > File > Export (view page 622)".

Destination Files



There are various ways to save edited files:

Replace source files: The original file is replaced with the edited one. If the file is used in a VIP, the VIP is closed first.

Save files in the source folder with a different name: The changed file will be copied back into the source folder and the file name will be expanded with the added "**file name prefix**" and "**file name suffix**".

Save files in the following source folder: The changed file will be copied back into folder that can be freely selected and the file name will be expanded with the added "**file name prefix**" and "**file name suffix**".

Remove file name suffix with numbers: This option removes numbers added by the Sequoia batch processing function (e. g. "_48kHz" or "_T001").

Delete jobs from list after processing: If you select this option, a job that has been processed will be removed from the list.

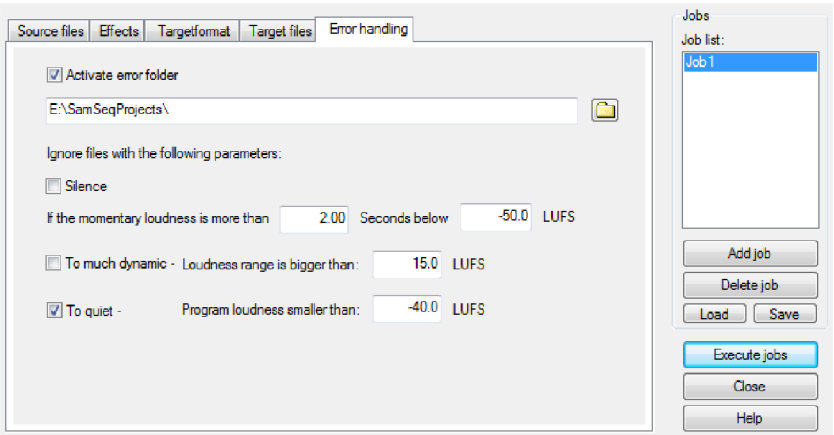
"**Delete source files after processing**" can also be applied.

Keep source folder structure: This option saves all files in the same subfolder structure including the source path. This process creates subfolders as required.

Error Handling

Here you can select a folder where files that do not conform to specific criteria of the loudness norm will be stored so they can be given special treatment.

All other files are processed normally and saved in the defined target folder.



First activate the option „**Activate error folder**".

Silence: In a selected folder you can sort out all of the files that fall below a certain loudness (LUFS) for a defined period of time.

Too much dynamic: In a selected folder you can sort out all of the files that exceed a certain loudness (LUFS).

Too quiet: In a selected folder you can sort out all of the files that fall below a certain loudness (LUFS).

Opening batch processing using the command line

Sequoia.exe -batch [Drive:][Path]BatchProcessingPresetName.xml

If you start batch processing using batch scripts, a switch to the PowerShell will prevent further processing of the script before the batch has been processed. To do this, please proceed as follows:

Option 1: Working in the PowerShell

- Switch to the PowerShell with the command: powershell
- Now enter the the correspondingly adjusted paths:

C:\Sequoia\Sequoia.exe -batch "C:\Batchfolder\Preset.xml" | Out-Null

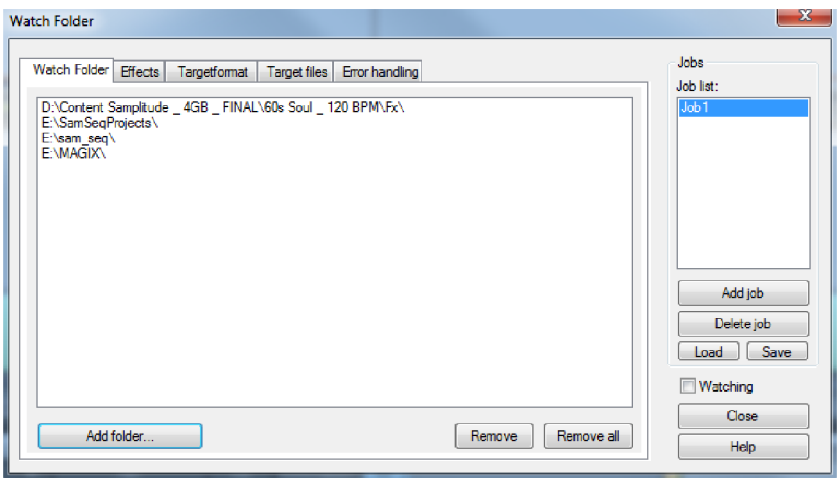
Option 2: Using the PowerShell only to call up the command

- Now enter the the correspondingly adjusted paths:

powershell "C:\Sequoia\Sequoia.exe -batch 'C:\Batchfolder\Preset.xml' | Out-Null"

Watch Folder

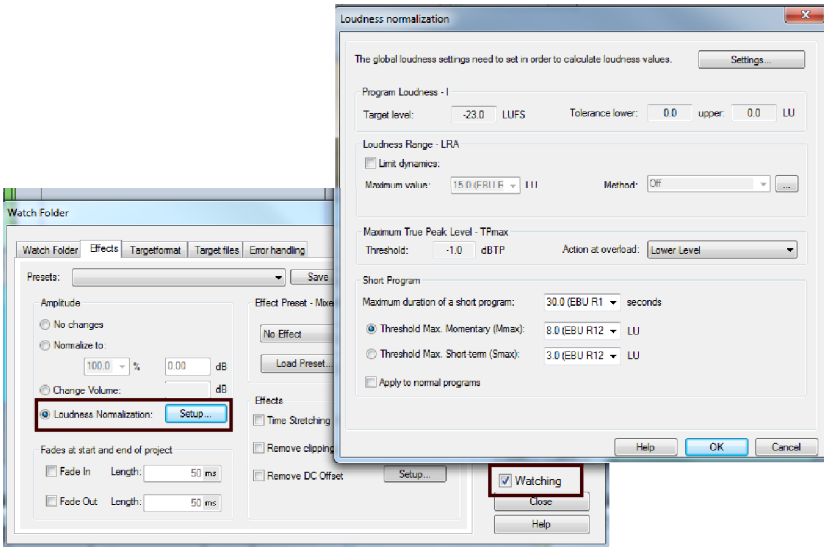
The Watch Folder enables you to edit and monitor files from selected folders in the same way as batch processing.



Use the option **„Add folder"** to put folders that you want to monitor into the Watch Folder and then determine the Effects (view page 752), Target format (view page 753), Target file (view page 754) and Error handling (view page 754).

With **„Load"** and **„Save"** you can load and save jobs in XML format. The file extension for the Watch Folder jobs is: watch.xml

If you activate the option **„Watching"**, the files in the folder will be edited according to the entered settings. Newly added files are also edited accordingly (e. g. with Loudness normalization (view page 862)).



Opening the Watch Folder using the command line

Sequoia.exe -watch [Drive:][Path]WatchFolderPresetName.xml

Command line parameters:

-batch [Drive:][Path]BatchProcessingPresetName.xml

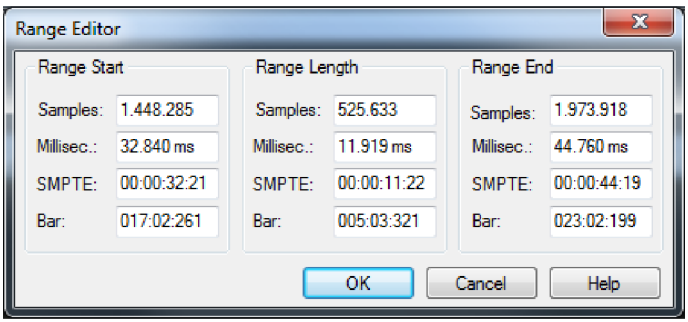
-watch [Drive:][Path]BatchProcessingPresetName.xml

The process can be canceled by pressing Ctrl+C.

More

Range Editor...

This window specifies the currently marked range's start and end positions and length in various measurement units.



Adjustments to the parameters have the following effect:

Change start of range -> Range end remains constant

Change range length -> Range start remains constant

Change range end -> Range start remains constant

Edit time display

Here you can set and the number fields for the range position (Alt + Num pad 1), range length (Alt + Num pad 2), and range end. (Alt + Num pad 3).

Shortcut: Alt + Num pad 1 ... 5

Track Menu

Insert New Tracks

Add One Track

Use this function to add a new track to the last track in the project.

Add Several Tracks

To add multiple tracks, select the command "Add several tracks". A dialog window appears to select the desired number of new tracks.

The maximum track count in Sequoia is limited to 999 stereo tracks. Each of these tracks can also act as a AUX bus and/or as a submix bus.

Insert Track(s)

This command adds one empty track at the end of a selected track.

Insert several empty track(s)...

To add multiple tracks, select the command "Insert several empty tracks". A dialog window appears to select the desired number of new tracks.

New MIDI Track

This command adds a MIDI track to the end of a selected track.

New Folder Track

This command adds a folder track in front of the selected track. If a range across several tracks is selected before creating the folder track, each of these tracks will be added to the new folder track.

You can find detailed information about folder tracks in the section "Working in the Project Window" > "Folder Tracks" (view page 132).

New Submix Bus

Add a submix bus as a track here. The new bus will be added below the selected track.

New AUX Bus

Add an AUX bus (view page Fehler! Textmarke nicht definiert.) as a track here. The new bus will be added below the selected track.

New Surround Bus

A surround bus corresponds to a normal submix bus (view page 384) with surround functionality. All tracks routed to a surround bus receive the Surround Editor instead of the normal Panorama knob, which can be used to adjust the surround position of the output signal of this track.

The Surround master is created simultaneously when a Surround bus is created; the individual channels are routed to the different output devices.If no Surround Master is present in the project when a Surround Bus is created, a Surround Master is created at the same time, whose individual channels are routed to the various hardware outputs. The Surround Settings dialog (view page 477) is called automatically for this purpose.

New Surround Master

You can also mix your project afterwards as a surround version. For this purpose, use this command to create a Surround master. You can then route the tracks individually to the surround master exclusively or additionally. All tracks routed to a surround master receive the Surround Editor instead of the normal Panorama knob, which can be used to adjust the surround position of the output signal of this track.

As soon as you have created a Surround Master, you can also route just the output signal of a single object to this Surround Master and set the Surround panorama independently from the track panorama settings. Read more on this in Working with multiple Masters (view page 494).

New Surround AUX Bus

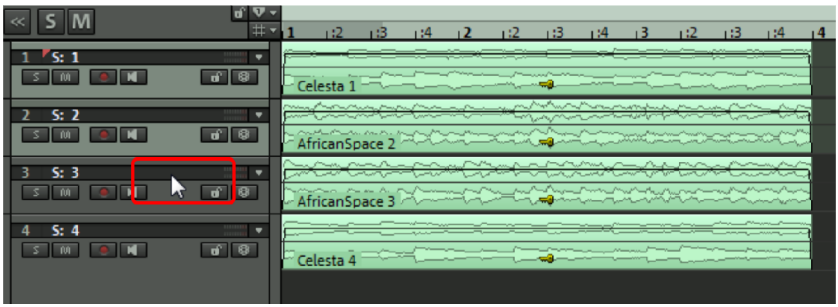
Use this command to create a new Surround AUX Bus.

A Surround AUX bus allows integration of Surround effects. This may be fed by a normal track or by a lower lying Surround bus. The AUX bus itself can again feed a Surround bus with a higher track number.

Copy Track(s)

Use this function to copy selected tracks to the clipboard to paste them into the same or another project (see below).

Individual tracks are selected by clicking on the track header (view page 130). To select more than one track, hold down the **Ctrl key** and click on the **track names**. To select multiple consecutive tracks, select the first track and hold down the Shift key to click the track name of the last track.



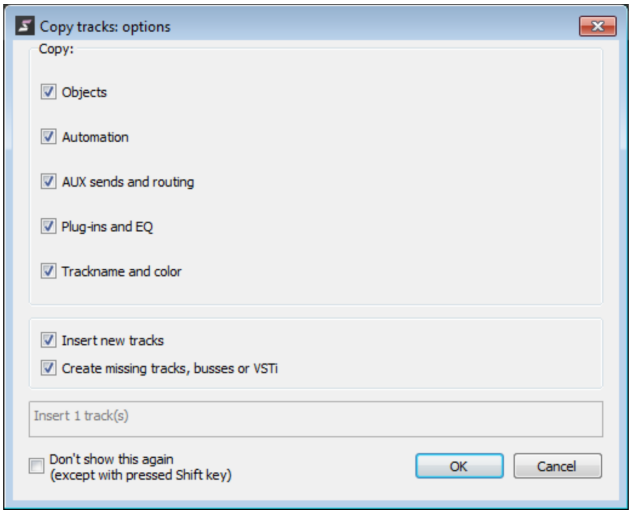
All track properties and contents are copied. When pasting (see the following section), you can specify which of these are to be inserted.

Paste Track(s)

This function inserts all tracks copied into the clipboard under the selected track.

At first a dialog with several options is opened. By this options you can use this function also to transfer properties of one track like plug-ins, automation or routings to another track, keeping the objects on this track untouched.

The dialog remembers the settings, therefore, for further insert operations with the selected settings you can skip the opening of the Options dialog box by setting the check mark **Don't show this again** in the lower right corner of the dialog box. To reach the options dialog again, hold the **Shift key** while chosing the Paste Track(s) menu item.



Copy: Here you can specify which track properties and contents are to be inserted.

- Objects: Any objects including their effects and automations
- Automation: all automation curves of the track. Note that all automation curves are inserted. If you have automated plug-ins in the copied track but do not copy the plug-ins, you get automation curves without a target. These are displayed in the automation menu with ???.
- Aux sends and routing: The values of the aux send faders as well as the audio and MIDI input and output routing. Note also the "Insert missing tracks, busses and VSTi" options below.
- Plugin-ins and EQ: All track effects as well as the plug-in instruments and effects loaded into the track.
- Track names and colors: The track names are important in that the assignment of the copied routing is based on the track names.

Insert new tracks: If you uncheck this box, no new tracks will be inserted, but the target project will be overwritten with the clipboard content. If exactly one track is selected in the target project, the target project will be overwritten with all the tracks from the clipboard. However, if multiple tracks are selected in the target project, only these selected tracks will be overwritten.

Add missing tracks, buses and VSTi: If you paste the copied tracks into another project and the "Aux sends and routing" option is active, this option ensures that the routing targets, aux and sub-mix busses are created if they are not copied with. All track settings of the additional tracks (e. g. plug-ins in the AUX busses, VSTi plug-ins as MIDI Out targets of tracks with MIDI objects, audio tracks for single outputs of VSTis) are taken over.

Attention: AUX and SubMix busses are only created if there aren't already busses with the same name in the target project. Otherwise these will be used, but the effect settings of the busses from the original project will not be taken over!

Note: The audio tracks of VSTi single outputs are also created including their insert and aux send effects, with additional AUX busses with effects if necessary. However, if these tracks are again routed to busses, these busses are no longer copied.

Delete Track(s)

This function deletes all selected tracks.

Track Type

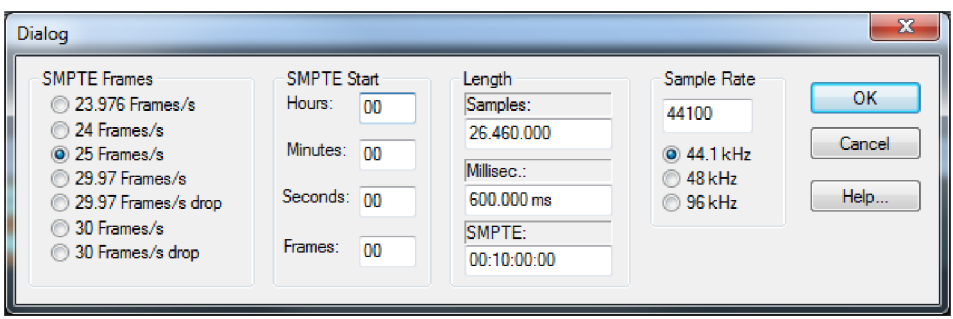
Here you can specify whether the track should be an **audio** track or a **MIDI track**. The track can also be defined as a **submix** and/or **AUX bus**.

Economy track: Switches the active track to "Economy" mode (view page 602).

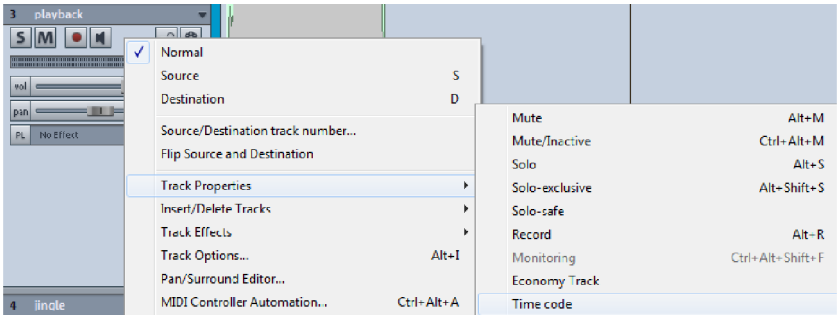
Timecode: Switches the active track to "Timecode" mode. This is how you can create a timecode track:

1. Create a new audio track and name it "SMPTE timecode".

2. Create a timecode for this track using the "Menu effects > SMPTE generator... (view page 1088)" dialog.

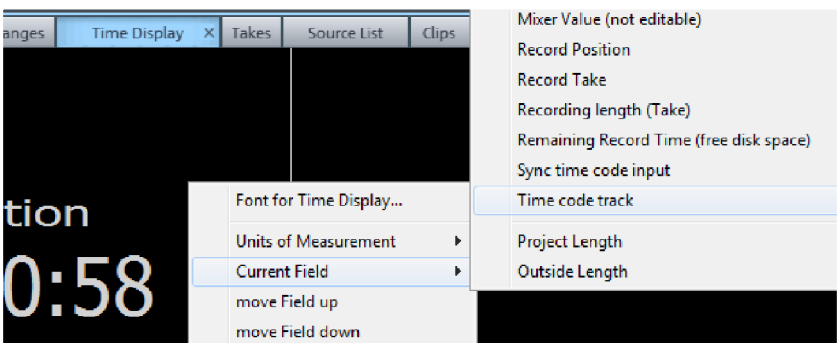


3. The created timecode appears as an object in the selected track. Now define it as a timecode track by right-clicking on the track head and setting a check next to "Track properties > Timecode".



The timecode track background will now appear colored.

4. In the time display, select the current "Timecode track" field with a right click.



This way, you've integrated the SMPTE timecode into the time display. If you are using files with an embedded timecode, specify the track that these files are on as the "Timecode Track".

Input

Stereo in: The channel input is switched to stereo and the signal will be recorded on two channels.

Mono in: The channel input is switched to mono and the signal will be recorded on only one channel.

Mono In / Stereo FX: The track input is switched to mono and a single-channel input signal is recorded, but the track itself remains a stereo track in which the effect calculation is performed in stereo.

MIDI track record: If you click this menu entry the MIDI section of the track editor will open, allowing you to record MIDI files.

VST MIDI out recording: By activating the option "VST MIDI Out recording" on a track you can record all received VSTi MIDI Out data in the corresponding track.

Group track controls

With this function you can group control functions like faders, record or mute buttons from different tracks. If you want to group track controls, select the desired elements while holding down the "Ctrl" key and then activate "Group track controls". Control groups will be automatically called up when one of the elements of this group is activated.

Example: When muting a submix bus, no AUX channels from the original tracks will be sent. An additional control group can be created in order to simultaneously control the faders of several tracks. By clicking and simultaneously using the Shift + Ctrl keyboard shortcut individual faders can be moved in the opposite direction. By clicking together with the shortcut "Shift + Ctrl", individual faders may even be moved in opposite directions.

Note: Please note that only one control element may be present per control group. Overlapping control groups can't exist.

Ungroup track control group

You can use this function to ungroup. To do so, click on an element in the control element and select "Ungroup track control".

Hide Track

This command prevents the selected track from being displayed in the track view. However, it will still remain visible in the mixer and will continue to play there.

The menu item "Unhide All Tracks" shows this track and all other hidden tracks.

An overview of all tracks including their status of visibility is included in the Track Manager. ("View" > "Manager" > "Track Manager") Hidden tracks can be recognized in the track manager by the lack of a tick beside "Arrangement". In the column beside it to the right, you can hide the track in the mixer window as well.

Detailed information about the track manager is provided in the "Managers -> Track manager" section.

Unhide All Tracks

Use this command to show all hidden tracks in the arranger once again.

Track size

Maximize Track

The activated track is maximized, i.e. it is displayed in a large view.

Minimize None

All minimized (and maximized tracks) are displayed in the same size.

Display Subtracks

This command opens the subtracks for a folder track.

Track Freeze

This option renders the selected track as a wave file which replaces all the objects on the active track. The object and track effects are also included which reduces the CPU load resulting from CPU intensive effects and plug-in calculations.

The advantage of Track Freeze is that the frozen track is saved in a separate VIP and can be edited or recalled at any time. This allows you to work flexibly and reduce the load on your processor at the same time.

Detailed information about freezing plug-ins is provided in "Software instruments/VST plug-ins -> Freezing plug-ins (Freeze)".

Track Freeze for AUX Buses/Submix Buses

Submix and Aux bus tracks may also be frozen. This way, you can get a complete group of tracks to release the required CPU load including all effects, fades, crossfades, and automation settings in one step.

Submix Bus

- Tracks routed onto the submix bus are not changed. They remain the same.
- The submix bus input is muted
- The file created while freezing is inserted into the submix track as an audio object.
- Changes made to the tracks after freezing which are routed to the submix bus have no effect on the playback of the frozen submix bus.

AUX Bus

- AUX send settings in tracks are taken into account when freezing
- Tracks routed onto the frozen AUX bus are not changed but remain unchanged instead
- The AUX bus input is muted
- The AUX send fader of the affected tracks is colored blue.
- An asterisk is added in front of the name of the AUX send fader in the mixer.
- The file created while freezing is inserted into the AUX track as an audio object
- After freezing, any changes made to the tracks that transmit to the AUX bus are not updated in the file that was created during the freezing process.

Edit Track Freeze

This command opens the temporary trackbouncing VIP in which the frozen track is saved.

You can make any changes you like here (just like in any other project), from object cuts via effect calculations in the track and object to adding other wave projects.

Frozen Submix/AUX Bus

If you have changed something on tracks that route to a frozen submix bus, you should "unfreeze" the bus using the "Track unfreeze" command and then refreeze it immediately afterwards with "Track freeze" in order to keep it up-to-date.

Unfreeze Track

This menu item removes the wave file created during "Track freeze" and brings back the track that was saved in the temporary trackbouncing VIP.

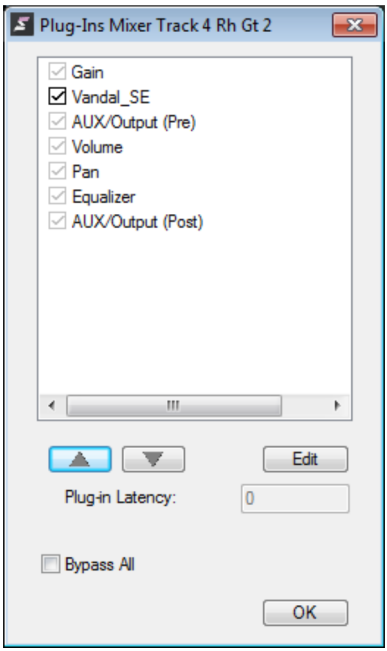
Changes made in the temporary freeze project are also applied to the track.

Note: If you add additional tracks which were created when freezing a track to the project, "Track unfreeze" will no longer be usable.

Track Effect Settings

Effect Routing Dialog

In the Effect Routing dialog you can make all important settings for realtime effects.



Effect Routing Dialog - Routing

All available effects are listed in the effects list in the respective context (track, object, or master). By right-clicking you can directly access the settings dialog of the selected effect. The check boxes for volume, pan, and gain knobs as well as the AUX gain knobs are grayed out because they are only represented in the list for the purpose of ordering the effects.

The order of the effects in the mixer tracks and the mixer master can be configured freely; in the object, it is restricted. You can change the position of an effect or plug-in with the up/down arrows.

On the tracks there are two AUX Send options (Pre and Post). A right-click opens the AUX routing dialog. Depending on whether the AUX send controller is switched to pre or post, the corresponding entry in the signal chain will be used to feed the AUX signal to the AUX bus. VST plug-ins can also be routed pre and post in a similarly flexible manner.

Effect Routing Dialog - Buttons

Edit: Opens the dialog of the active effect. This can also be done by right-clicking on the desired list entry.

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

Copy Track Effect Settings

This command allows you to copy the complete effects settings of the active track into the clipboard. This includes the order and parameter adjustments of all internal effects including those of VST plug-ins.

Insert Track Effect Settings

This command allows you to insert the complete effects settings from the clipboard in a track. This includes the order and parameter adjustments of all internal effects including those of VST plug-ins.

Reset Track Effect Settings

This command resets all track effects settings to default values, i.e. no effects are applied.

Save Track Effect Settings...

Here you can save custom track effect setting as .TRK files.

Load Track Effect Settings

Here you can load previously saved/preset track effect settings in the selected track.

Source/Destination

Detailed information about this is available in "Source Destination Cut (view page 400)".

Normal

Switches the selected track to "Normal" mode.

Source

Switches the selected track to "Source" mode.

Shortcut: S

Destination

Switches the selected track to "Destination" mode.

Shortcut: D

Source Destination Track Number...

Use this dialog to specify how many source and destination tracks should be used during the Source Destination Cut or the multi-source session.

Flip Source and Destination

Source and destination tracks swap modes.

Move Source Up

The current source track changes to "Normal" mode, and the track above changes to "Source" mode.

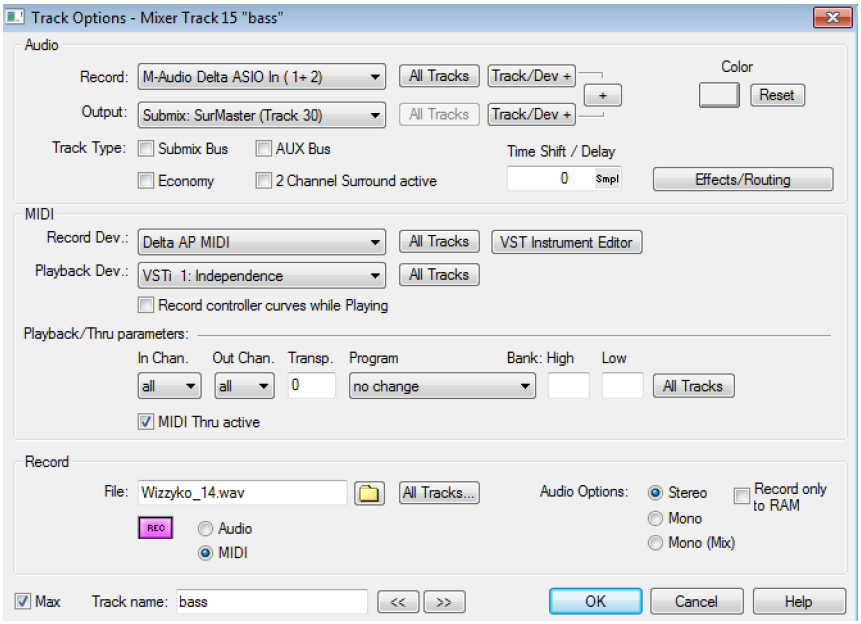
Shortcut: Alt + Page up

Move Source Down

The current source track changes to "Normal" mode, and the track beneath it changes to "Source" mode.

Shortcut: Alt + Page down

Track Options...



Audio

Here you can change settings for recording and playback of audio tracks. Changes to these settings also apply to the settings for audio created by VST instruments.

Record: Select your sound card inputs to record this track.

All Tracks: Sets the recording device selected for this track to all other tracks.

Track/dev+: Sets the recording device for this track to the next - e.g. if inputs 1 + 2 of the sound card are set, it will be switched to 3 + 4. The button with the "+" sign to the right sets the recording and playback devices to the next ones simultaneously.

Playback: Select your sound card outputs to record this track. You can also select a bus track.

All Tracks: Sets the playback device selected for this track to all other tracks.

Track/Dev+: Sets the playback device to the next, e.g. if set earlier to outputs 1 + 2 of your sound card, this will be switched to 3 + 4. The button with the "+" sign to the right sets the recording and playback devices to the next ones simultaneously.

Track type: This indicates if the track is a aux or submix bus. A single track can also be both. "Timecode" (view page 763), "Economy" (view page 602) as well as "2-channel Surround" modes may also be activated for this track.

Timeshift/Delay: Use this field to set an offset for an entire track. Positive values cause the playback of this track as well as the display of the play cursor of be delayed by the entered amount. Negative values result in earlier playback of the track. All other tracks and the play cursor will be delayed in relation to the entered value. To the right of the input field you can select the unit of the time delay.

Color: Select a color to paint the corresponding track and its objects.

Effects/Routing: This button opens the effects routing dialog for the corresponding track.

Track Name: Use this field to change the name of the current track.

MIDI

The MIDI playback section provides access to the various MIDI options.

Recording Device (MIDI): Use this menu to choose a MIDI input device.

All Tracks: Sets the recording device selected for this track to all other tracks.

VST Instrument Editor: This button opens the graphical interface for the connected VST instruments.

Playback Device (MIDI): In this menu you can choose your MIDI playback device. Here you can also select VST instruments as playback devices.

All Tracks: Sets the playback device selected for this track to all other tracks.

Record controller curves during playback: This option enables controller curves to be recorded during playback.

The following are available as additional playback options: In-channel, out-channel, transpose, program change, bank high, bank low. You can also activate **MIDI-Thru** for the affected track. This activates the speaker icon for the track.

Track Name: Use this field to change the name of the current track.

Recording

The section "recording" contains settings for recording from audio and MIDI.

File: Indicate the name of the file to be recorded here.

Assign file name automatically

Click "All tracks" on the right of the file name box and then select the desired command from the popup that opens:

- File name_track number: Transfers the file name of the track to all other tracks and adds the corresponding track numbers to them, for example, funky_01.wav, funky_02.wav, etc.
- Project name_track number: Transfers the project name to all tracks and adds the corresponding track number to them, for example, demo_01.wav, demo_02.wav.
- Track name: Sets the track name of each track as file names for the audio recording, for example, Drums.wav, Bass.wav, etc.
- Track number_track name: Sets the track number and the track name of each track as file names for the audio recording, for example, 01_Drums.wav, 02_Bass.wav, etc.
- Project name_track name: Transfers the project name to all tracks and adds the corresponding track name to them, for example, demo_Drums.wav, demo_Bass.wav, etc.
- Get track name from first object name: This option gives the track name the same name as the first object in that track.
- Get track name from first object file: This option gives the track name the same name as the first file in that track.

REC: Specify whether or not you want to record MIDI or audio here.

Audio options

You may also choose between recording in **mono** or **stereo** on the selected track. A **mono mix** of both device channels is also possible.

The option "**Recording only in RAM**" allows recording to be done directly in the system RAM.

Revolver Tracks

New empty revolver track: This command creates a new revolver track.

Before track objects are rearranged, select the option "**New Revolver Track (copy)**". The track objects will be copied as a new Revolver Track and a star appears before the track name. The original objects may now be edited or repositioned to produce a different version of the track.

Delete Revolver Track: This command deletes a Revolver Track.

Previous Revolver Track: This command displays the previous Revolver Track

Next Revolver Track: This command displays the next Revolver Track

More information on Revolvertracks (view page 135)...

More

Cut tracks

This function cuts out all tracks that are selected in the track header (view page 130). The corresponding tracks are stored in the clipboard in the meantime.

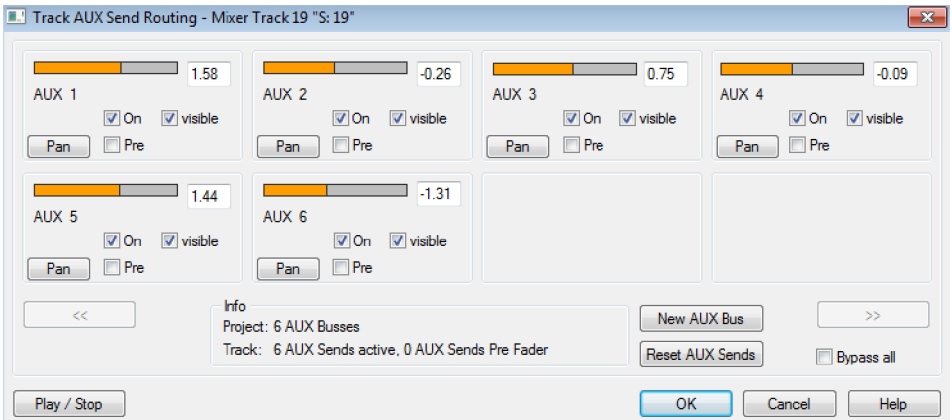
Track effects

These options allow (as with the Mixer channels) the application of real-time effects to the selected track in Sequoia.

More information about effects is available in "Effects - division and working methods (view page 219)".

AUX Sends

Use this dialog to create new AUX sends or determine the panorama out of existing AUX buses. You can also switch the AUX track to pre-fader or post-fader or deactivate it.



Description of the "AUX Sends" dialog

You can enter the send level of each AUX bus numerically or pull the orange bar in the gray field from left to right.

By default, all AUX buses are routed "post" in Sequoia. To switch them to "pre", activate the corresponding box.

New AUX Bus: Clicking this button creates a new AUX bus.

Reset AUX Sends: All AUX sends settings are reset.

- AUX sends: Opens the AUX send routing dialog window
- Dynamics: Dynamics, Advanced Dynamics, Multiband Dynamics, sMax11, eFX_Compressor, eFX_Gate, eFX_Limiter, AM-Munition, AM-Track, AM-Phibia, AM-Pulse
- Frequency/Filters: Parametric EQ, EQ116, FFT Filter, Brilliance Enhancer, Filtox
- Delay/Reverb: Delay, room simulator, efx_Reverb, efx_StereoDelay, Ecox, Variverb II
- Distortion: Distortion, Vandal, Bit Machine, eFX_TubeStage
- Restoration: DeClipper, DeClicker/DeCrackler, DeHisser, DeNoiser
- Stereo/Phase: Pan/Surround Editor, Multiband Stereo Enhancer, eFX_TremoloPan, track phase inversion
- Modulation/Special: Vocoder, efx_ChorusFlanger, efx_Phaser, Corvex
- MIDI: Velocity Dynamics
- Plug-ins: Opens the VST plug-in dialog.

- Essential FX: eFX_ChorusFlanger, eFX_Phaser, eFX_Reverb, eFX_StereoDelay, eFX_Compressor, eFX_Gate, eFX_DeEsser, eFX_VocalStrip, eFX_TubeStage, eFX_Limiter eFX_TremoloPan.
- MAGIX Plug-ins: AM-Munition, AM-Track, AM-Phibia, AM-Pulse, Corvex, Ecox, Filtox, VariVerb II, Vandal.

Track properties

This submenu provides access to all track properties.

Mute: Mutes the active track

Keyboard shortcut: Alt + M

Mute inactive: Mutes and also deactivates the track. This increases performance by removing the need for caching and FX processing for the track.

Keyboard shortcut: Ctrl + Alt + M

Solo: Switches the active track to Solo mode

Keyboard shortcut: Alt + S

Solo exclusive: This setting switches the active track to exclusive solo mode, i. e. only this active track is audible. All other tracks in solo mode are muted.

Keyboard shortcut: Shift + Alt + S

Solo safe: This setting ensures that all channels that are switched to solo are automatically monitored with the AUX Return channels that they feed.

Record: Activates selected tracks for recording.

Keyboard shortcut: Alt + R

Exclusive record: Only the last selected track is activated for recording.

Monitoring: Switches manual monitoring for the corresponding track on/off

Keyboard shortcut: Ctrl + Alt + Shift + F

Lock: Locks the active track.

Keyboard Shortcut: Alt + L

Volume curve active: Shows if the volume automation is active.

Pan curve active: Shows if the panorama automation is active.

Edit volume: Here you can enter a numerical value in dB for the volume.

Shortcut: Ctrl + Shift + K

Edit panorama: Here you can enter a numerical value for the panorama curve.

Shortcut: Ctrl + Shift + P

Track phase inversion: Inverts the phase of the selected track.

Keyboard shortcut: H

VCA Master: As soon as you activate this option, the track will appear as a VCA group.

Mute Bus Inputs: Sets the input of the selected Submix Bus or AUX bus to inactive, i. e. the signals routed to the bus are not routed through to the signal path of the bus.

Global Mute: The mute settings for all tracks are switched on or off.

Global solo: Enable/Disable all solo states of the tracks

Global: Solo-safe: This switches all tracks to solo-safe mode. In this mode, the corresponding track is monitored together with its AUX returns if switched to solo.

Global: Solo PFL: This switches all tracks to Solo PFL mode. In this mode, the corresponding track is accessed at a position before the fader and heard as a pre-fader signal if switched to solo.

Global: Solo exclusive: This switches all tracks to solo-exclusive mode. In this mode, the corresponding track is played exclusively if switched to solo. Tracks already switched to "Solo" will be removed from solo mode.

Stereo/Surround Editor (stereo panorama dialog)

This command opens the Surround/panorama dialog, which features plenty of useful panorama presets. If you are working in the normal "Stereo master" mode, you have the option of activating 2-channel Surround.

The "Mono" button in the track editor sets the track to mono editing from input to the panorama controller. Particularly all track effects preceding the Pan controller operate in mono, which considerably saves CPU resources. However, the routing position of the Pan controller can be freely adjusted in the FX routing dialog. In this case, the

submix and AUX return buses always remain stereo. If stereo objects are located in mono tracks, the mono share (L+R) is played. This can be changed by using the "Mono side channel processing" preset. This preset offers individual options for panorama settings for stereo objects in mono tracks. If you use AUX sends, you can also use the panorama fader of the AUX send routing dialog for panning the mono signal.

You can find more information in the "Mixer" > "Stereo panorama dialog (view page 392)" chapter.

In Sequoia there is also a Surround Master Mode which can be set up through the "Options" menu -> "Project Properties" -> "Mixer Setup". If this mode is active, every track is set to "Surround" mode.

VST Instrument Editor

Software instruments and MAGIX synths (e. g. Revolta) can be integrated seamlessly into a virtual project and controlled with internal MIDI functions and editors. The instruments are integrated into all effects and routing options. If you have a track open in a virtual instrument, you can open the user interface of the VSTi here.

Note: If activating this menu item does not produce a result, it means that a VST instrument has not been loaded into the corresponding track. Detailed information about plug-ins can be found in the "Software/VST Instruments (view page 275)" chapter.

Activate Previous/Next Track

Use this command to activate the next (lower) or previous (upper) track in the VIP from the view of the currently activated track. With the cursor keys you can conveniently scroll up or down.

Shortcut:

Activate next track	Alt + cursor down
Activate previous track	Alt + cursor up

Track visualization

The visualization from the menu window can be set as a master or track display.

Detailed information on the individual visualization displays can be found in the chapter "View > Visualization" (view page 1121).

Object Menu

Unlike with the "Edit" menu all commands from this menu refer to selected objects in the virtual project.

Object Editor...

Use the object editor (view page 165) for object-oriented editing. This enables you to quickly and easily edit each selected object independent of any additional track settings.

Shortcut: Ctrl + O

MIDI Editor...

This menu item opens the MIDI editor. The content of the MIDI object selected in the VIP may be displayed and edited here. You may choose from the matrix editor, drum editor, score editor, event list, and velocity/controller editor.

If a MIDI object is not selected, you will be asked if you wish to create one. If you confirm this with "OK", a MIDI object in the current track is created at the cursor position/start of the range.

Detailed information about this is provided in "MIDI in Sequoia -> MIDI editors (view page 298)".

Wave Editing

This opens the audio file associated with an object in the wave window. The range corresponds precisely to the part of the audio data that the object accesses.

More information about this can be found in the chapter "Sequoia Basics" > "Audio editing in Sequoia" (view page 62).

Destructive Editing

Destructive audio edit mode is the standard mode when the command "Edit a copy of wave content" in the Object menu is used to open audio files that belong to a VIP object.

Audio files (e. g. Wave files) opened directly through "File -> Open -> HD Wave" are opened in the virtual wave editor by default. This can be changed in the program settings (Y key, "Program" > "General > Open wave projects in destructive editing mode").

Regardless of the mode you load your wave project in, **you can switch between destructive and virtual wave editing through "File -> Project Properties -> Destructive Wave Edit Mode"**. If the box is checked, the extension "destructive" will appear in the title bar of your project.

In destructive mode audio files are edited directly on the hard drive. Changes made in the editing window are immediately applied to the audio file. To be able to undo actions in the opened file in this mode using "Ctrl + Z", you have to activate the undo function for audio files in the program settings ("Y -> Program -> Undo").

Note: At any time, a selected range from an audio file in destructive mode can be transferred to a VIP using "Copy" and "Paste".

Real-time Audio Editing

You can also load audio files without a VIP and edit them in a virtual real-time mode. To do this, deactivate "Open wave projects in destructive editing mode". in the program settings (Y key, "Program" > "General). Now the files that are loaded through "File > Import... > Load Audio file" can be opened in an audio file that works virtually similar to a VIP.

The advantage of this editing mode is that it combines real-time editing of the audio material with the features of non-destructive virtual editing that are the standard in VIPs.

In contrast to destructive editing you will notice the following differences:

The features "Cut", "Copy", "Delete", and "Insert" are non-destructive. The positions where these operations are performed are marked with dotted lines. These operations can be performed in real-time because it is no longer necessary to copy data to the hard drive for undo functions. The edits performed are only applied to the audio file when it is saved.

"Wave editing" mode – volume and panorama curve

Volume and panorama editing is also virtual in "Wave editing" mode. Activate "Volume Draw/Automation Draw" Mode. You can now draw level and panorama curves with the mouse, just like in the VIP.

Real-time Audio Editing - Master Section

If you open the mixer window by pressing the "M" key, you will see a minimized mixer on the master channel. All of the real-time effects that you are already familiar with from the VIP master section are now available for real-time audio editing.

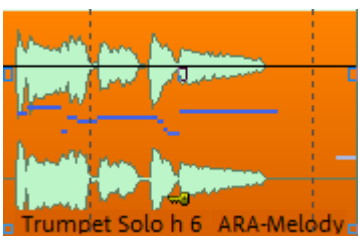
This allows you to use several effects simultaneously during real-time editing. However, during destructive editing only one effect at a time can be written to the audio file.

"Wave editing" mode – auto-crossfade

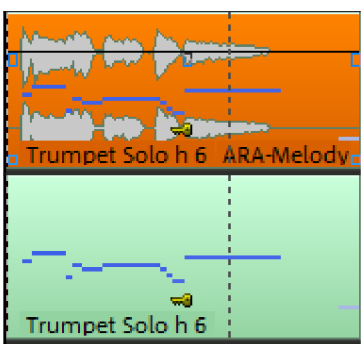
The auto-crossfade option may be activated during any cut or insert operation in non-destructive wave projects

Audio to MIDI

If an audio object has been analyzed or edited using Melodyne, you will also see the pitch on the audio object as a piano roll, similar to how MIDI objects in VIP are represented.



Selecting "Audio to MIDI..." from the object context menu or the "Object" menu, you can create a new MIDI object.



The MIDI object will be added to the track directly below the audio object. If there are already objects at this time position, on this track the MIDI object is inserted to a newly created track instead.

New MIDI Object

This creates a new MIDI object in the selected track. A pop-up menu opens when the function is selected to choose from multiple templates.

Presets include **MIDI drums** and **MIDI phrases** (filter sweep, 6 octave sequence, etc.) or an **empty template**.

These are also available in the "Templates" folder in the program directory. The pop-up menu may also be expanded with other templates simply by copying the desired MIDI files into the "Templates" folder or exporting them directly from Sequoia as a template.

New MIDI object in the range

This function creates a new MIDI object in the current track within the selected range.

New Synth Object

This command creates a 4-bar loop object at the current playback marker position. The instrument interface for the synth object that is created can be accessed by double-clicking on the object. Object synths include BeatBox 2, Loop Designer and Robota.

Edit

Cut Objects

Use this command to delete selected objects from the current project and insert them into the VirtClip. The content previously in the VirtClip will be overwritten.

Copy Objects

This command copies the selected objects into the VirtClip. The previous content in the VirtClip will be lost.

Insert Objects

Use this command to add objects from the VirtClip into the project, starting from the current position of the play cursorplayback marker this may cause objects to overlap.

Delete Objects

This deletes the selected objects from the current project. The content of the VirtClip will remain.

Shortcut: Ctrl + Del

Duplicate Objects

All of the selected objects are copied and positioned behind the original object dependent on the set grid.

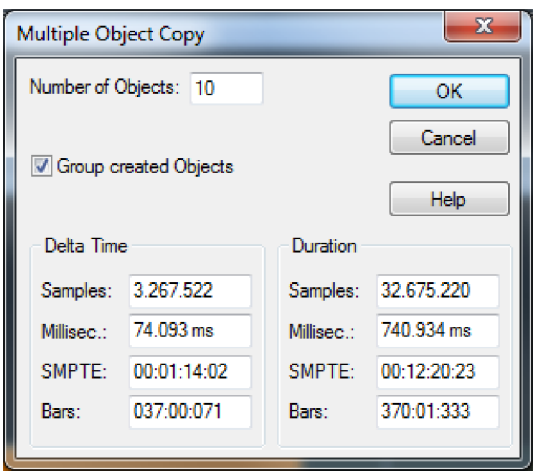
Duplicate Objects Multiple

Use this function to duplicate selected objects sequentially. In the dialog window that opens, enter the number of copies, the corresponding distance, and the entire length.

The following parameters are available:

Number of Objects: Specify the number of duplicates here.

Group Created Objects: Decide here whether the duplicated objects should be assigned to a group.



Time Difference: Set the distance between each object's starting point here. The default length is set to that of the object you wish to duplicate. This setting connects the objects seamlessly with one another so that a loop is created.

Duration: As an alternative to the time difference, the entire length of all copies may be set here. This is calculated from the product of the time difference and the number of copies.

Creating looped objects

Use this command to define a loop inside an object. Use the previously selected range above the object to set the loop start and end points. By dragging the length handle at the bottom right of the object you can make the loop exceed the previous object end point.

If you only selected an object, but not the range above it, the entire object will be looped by dragging to the right.

Loop objects are suited, for example, to quick creation of full drum tracks from a drum loop.

Loop points may also be set easily via the object editor.

Shortcut: Ctrl + L

Split Objects

Use this option to split all selected objects at the cursor position and separate them into individual objects.

If the range is selected, the cut will be made at each border. Ensure that the object that is to be edited is contained within the active range. If necessary, all objects may be activated by double clicking on the range selected in the range bar. In this case, it is not necessary to select the objects beforehand.

Shortcut: T

Split Objects with Alternate (Linear) Fade

Use this option to split all selected objects at the cursor position, thereby separating them into independent objects. The curve defined as Default_Crossfade_ALT in the crossfade editor will be applied as the crossfade.

Shortcut: Shift + T

Detailed information about Sequoia's crossfade editor is provided in "Crossfade editor -> Crossfade presets (view page 436)".

Split Objects at Project Markers

The object is split into separate objects at the marker positions within the object borders. The newly created objects will be named after the previous markers.

Split Objects at CD Track Indices

The object is split into separate objects at the marker positions within the object borders. The newly created objects will be named after the previous track markers.

Split Objects at Audio Markers

The object is split into separate objects at the audio marker positions within the object borders. The newly created objects will be named after the previous audio markers.

Heal/Unsplit Objects

This command recombines two objects that have been split without any audio calculation. To do this, the objects in the VIP must be positioned exactly beside each another, and the audio material must be connected. The objects must therefore be different but sequential pieces of audio material from the same wave project. If this not the case, the command will not function properly.

Before executing the command, select both of the objects that were split previously. All of the second object's properties, like fades and effects, will be discarded. The resulting combined object will feature the properties of the first object.

Trim Objects

This function moves the object borders of a selected object to the edges of the current range. To do this, the range must be contained completely within the object's borders.

Shortcut: Ctrl + T

Trim MIDI Objects

This function removes the MIDI data of an object that is not located within the current object's borders ("virtual MIDI data"). For example, this type of object virtual MIDI data results from dragging the left object border of a MIDI object after a MIDI recording inward (to the right). This function can be very useful if you wish to edit multiple MIDI objects at the same time in the MIDI editor, since overlapping virtual MIDI ranges can be very difficult to keep track of.

Glue Objects

A new object is created using those selected. Trackbouncing is used in this case. If you have selected just one object, this will be glued together with the object to the right. This function corresponds with the "Glue" symbol in toolbar.

Shortcut: Ctrl + Alt + G

After gluing, original cut may be subsequently edited the using the "Freeze object -> Edit object freeze" function. When selecting this function, a single-track VIP that contains the original objects will open as well. These may then be edited individually.

Use the "Freeze object -> Unfreeze object" function at any time to restore the original state of the object.

Mute Objects

This mutes the selected object.

Shortcut: Ctrl + M

Lock Objects

This protects objects from being unintentionally moved. When this function is active, the key symbol in the lower portion of the object turns red. If the key symbol is grayed out because of the display settings, the objects will be displayed crossed out instead.

Note: Please take note of the "Locking options". These specify which functions may be locked for set objects.

An object may also be locked by clicking the key symbol in the lower portion of the object.

Unlock Objects

This unlocks an object. All selected objects will be unlocked.

An object may also be unlocked by clicking the key symbol in the lower portion of the object.

Lock Definitions...

Detailed information about this is available in "File -> Program Preferences -> Lock Definitions (view page 708)...".

Edit audio file copy

With this command you can create a copy of the audio file at the base of object, which you can then edit. You will then see the selected copy in the audio editing window to the right next to the original file.

Edit Root VIP

Audio objects that were compiled using an internal mixdown of a virtual project to a wave project and then inserted into a new project contain information about the VIP that was used to generate them (bounced). The "Edit root VIP" function allows you to reopen this VIP for editing.

Sample: A CD is compiled from any number of songs that were created in Sequoia and were converted to individual wave files using the trackbouncing function (view page 643) and are now visible in the arranger window of the new project. It's possible that you may want to change something else in the song retroactively. "Edit root VIP" opens the original project to make the desired changes. Once the modified project is saved, it will be re-bounced and updated in the project that is used to compile the CD.

Calculate MIDI Objects and Track Effects

This command permanently calculates all MIDI channel settings, transpositions and track effects into the MIDI data.

Demix MIDI Objects by MIDI Channels

This function will create a new track for each channel of the selected MIDI object. The objects will then contain only the MIDI files of the specific MIDI channel.

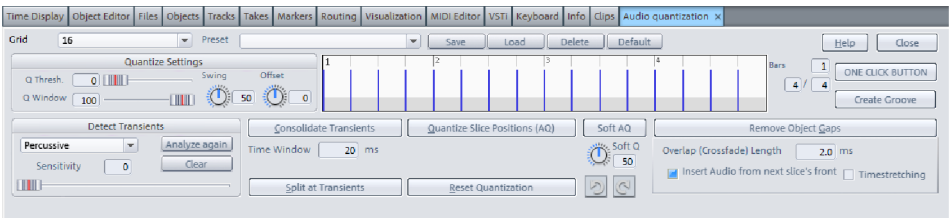
Extract Controller Curves from MIDI Objects

This option extracts the continuous controller data (MIDI CC) from MIDI objects and transfers it to the VIP as track automation.

Quantization

Audio quantization

Audio quantization is a special function for adjusting and snapping a recording to a grid.



These commands reflect the typical working procedure when "adapting" a multitrack drum recording to the VIP grid and are applied to the object selection. The "One click button" automatically executes the "Determine transients", "Consolidate transients", "Cut at transients", "Quantize object positions (AQ)", and "Remove Object Gaps" commands sequentially.

During determination of transients, you can set the sensitivity of the detection with the "Sensitivity" controller. The higher the value, the more beat markers are created.

By using "Recalculate" and "Delete", you can recalculate or delete an already created audio material analysis.

If you would like to connect the slices you have created to crossfades, then enter the length of the fade into the respective field "Overlap (crossfade) length".

Several gaps between slices can be closed either by object time stretching or applying audio material from the slice to the right of the gap.

For a detailed description of the audio quantization wizard, see the menu reference > "Object" menu in the manual PDF.

Define transients

Audio quantizing objects is particularly well suited for modifying multi track drum recordings.

"F3" and "F2" allow you to jump to the next or previous transients in the audio signal. By quantizing the VIP object position and dividing objects at beat markers/transient positions you have the option to perform highly flexible dynamic drum quantizations.

All following commands refer to selected objects and can also be accessed using the Audio quantization wizard.

"Specify transients" marks the peaks of the selected objects as "AQ" with special beat markers. A beat marker is an audio object marker, which is written into the audio file. To make these markers visible, turn on "Objects -> Transients (AQ)" ("Shift + Tab"). All other audio markers may be displayed via "Objects -> Audio markers".

Detailed information about audio markers can be found under "Menu > Play/Rec > Markers > Audio markers (view page 844)".

Consolidate Transients

Transients within a certain range (e.g. 20 ms) are moved to the first transient position with this command. The effect is that "Split at transients" doesn't create too many splices, especially if you have several tracks with slightly different transient positions for the same drum hit, i.e. due to microphone distance related signal latency. The time window may be specified with the audio quantization wizard.

Create Groove

"Create groove" button in the audio quantization wizard creates a groove template based on the AQ markers (transients) according to the selected audio objects in the selected range.

Create MIDI trigger from transients

This function creates a new MIDI track below the track which contains the selected and analyzed objects. The transients found there will be displayed as MIDI events with corresponding velocity.



If you integrate a VST instrument in a newly created MIDI trigger track now you can double up or replace your existing audio track with additional sounds.

Create Groove Template from Transients

This function stores the transients created for the objects as a groove template in the program folder **"FX preset -> Grooves**".

Select groove templates in the audio quantization wizard under "**Quantization settings -> Grid**". The length and start of groove template is always set to full beats.

Split at Transients

All selected objects are split over multiple tracks at the beat markers (AQ). This quantizes each drum hit individually.

Example: To split all drum tracks according to the bass drum and snare tracks, carry out transient recognition for the objects in those tracks. Next, select all drum tracks and apply the "Cut at transients" command. Now all objects will be split at any bass drum and snare hit.

Quantize object position (hard)

The menu command "**Hard quantize audio object position**" equals a soft Q value of "100" or "Quantize object positions (AQ) (view page 791)" in the Quantization wizard.

Soft Quantization of Object Position - Soft Q

This value sets the strength or "Soft Q" value of the quantization.

- "100" moves the event precisely to the quantize grid point,
- "50" shifts the event to the middle between the current position and the quantization grid point,
- "0" means no movement, i.e. quantization off

The menu command "**Hard quantize audio object position**" equals a soft Q value of "100" or "Quantize object positions (AQ)" in the Quantization Wizard.

Offset

The value range in this parameter stretches from -100 to +100. By changing the offset values, you move the whole quantization grid. Select a negative value for the offset, and the quantization grid will be moved by the corresponding value to the left, or forward in time. On the other hand, if you select a positive offset value, you will set the quantization grid by the corresponding value to the right, or backwards in time.

A value of -100 corresponds to a shift of half a grid length to the left; +100 corresponds to a shift of a half a grid length to the right.

Quantize object position (AQ)

Now all the selected objects will be adjusted to the corresponding quantization settings that you previously set in the Audio quantization wizard. During this process the **"Soft AQ parameter"** will not be observed.

When quantizing the object is hard quantized, i. e. it will be shifted to the next snap point.

If you are not satisfied with the results, you can "Reset Audio Object Quantization" settings, which will set the objects back to their original position

Soft AQ

"Soft AQ" executes soft quantization according to the "Soft Q" parameter settings.

Remove Object Gaps

After object quantization has been applied, gaps between the objects can occur, which may lead to audible dropouts in overhead signals. The command "Remove gaps between objects" provides options for closing these gaps. The function "**Use next object slice audio material**" may be used to move the object start position of the object to the right of the gap to the left until the gap is closed, or the gaps between the objects may be filled by **Time Stretching** the objects next to the gaps. Each method can be selected in the audio quantization wizard. The **overlap (crossfade) length** setting of the first option, i.e. "**Apply next audio object slice's audio material**", is applied.

Undo/Redo/Reset Quantization

Use the **round arrow buttons** to undo/redo the last quantization applied. In this case, the counter-clockwise round arrow serves as the "Undo quantization" function, while the clockwise arrow serves as the "Redo quantization" function.

Reset quantization: This function resets all quantization processes of the object positions.

Reset Quantization

The "Reset quantization" resets the selected objects to the original position and also undoes the quantization.

Quantize Settings - Preset

Different presets are available for selection:

- **5-tuplet**: Quantization occurs in fifths
- **Magnetic quantize**: The "window" value is set to "50", i.e. only 50% of quantization will be included. Only those events will be quantized that are located within a range of 25% of the snap range to the left and right of the grid point.
- **Soft quantize**: The "Level" parameter is set to "50", i.e. quantization occurs at a half interval between the current position and the next grid point.
- **Swing**: The swing parameter is set to "75", i.e. in contrast to the binary rhythm, which features a "swing" value of "50", uneven/un-highlighted counting times will be set to delay. This highlights the "swing" feeling.
- **Triplets**: Quantization occurs in triplets
- **16th offbeat**: The quantization grid's timing is moved back a 16th note
- **8th offbeat**: The quantization grid's timing is moved back a 8th note
- "**New groove**" and "**More life for hi-hat**" provide groove templates

Of course, you can also set your own values and save them as a preset.

Quantize Settings - Default

The "**Standard settings**" button provides the option of resetting the preset values:

Q threshold: 0

Q window: 100

Swing: 50

Soft Q: 25

Offset: 0

Quantization Setting - Grid

Use the "Snap/Grid" parameter to set the step length and thereby the quantization values for audio quantization. A value of "1" provides a snap value of a whole note, "2" is a half note, "4" is a quarter note, etc. Here there is also a selection of dotted notes, triplets and special groove templates.

Detailed information about groove templates is available in "MIDI editors -> Quantize -> Quantize settings - Groove template" (view page 314)

Q Threshold

The parameter "Q threshold" may be used to slightly vary quantization by excluding notes from quantization that are very close to the next quantization value.

Q Window

"Q window" refers to the interval to the left and right of a grid point; events will be quantized within this range. No quantization will take place beyond this, and for this reason, the events outside of this window will remain at their position. The quantization area is dependent on the values of the parameters "Q snap" and "Q threshold".

Example: Grid: 4 max. Window: 4

- 100: The Q range covers the entire area between the grid points on the quantization grid. All events will be quantized
- 50: The Q range covers half of the quantization interval. Events with gaps of ¼ of the grid width left and right of the grid point (1/16 note values in this example) are quantized
- 0: no Q range -> quantization off

Swing

Set swinging, ternary playing with this value. This enables you to enter the division for an uneven/unaccented grid points.

- 50: "50-50/1:1" split. The unaccented eighths are exactly half way between the even eighth notes ("even", binary playing method)
- 67: "67-33/2:1" tuplets. The beat is split up into three counts, whereby the note is assigned 2 beats (67%) and the off beat note one count (33%)
- 75: "75-25/3:1" split. For example, a pointed eighth and a sixteenth is created from two eighth notes

Show Grid/Time Signature

Displays the desired beat signature and number of beats. The display window will change accordingly.

Example of use:

1. Enable display for audio markers/transients in the view options (System options ("Y" key) > "Design" -> "View Options" -> "Objects" -> "Audio markers/Transients").
2. Load a VIP with a multitrack drum recording where the drummer has played to the click, so overall the VIP grid already matches the drummer's tempo.
3. Select a range for audio quantization. We recommend about 8 to 16 bars to guarantee a quick calculation. The analysis of transients for an entire song can take a long time. Trim the objects to the selected range size.

4. Select all drum objects, open the audio quantization dialog, and move the "Sensitivity" slider. Beat markers will now appear in the objects at the positions of the transients. Set the slider so that the main hits correspond to the beat markers. You can use the F2 and F3 keys to move the playback marker to the previous or next marker.
5. To cut objects for all tracks at the beginning of transients, specify the beat markers with the "Consolidate transients" command. Sequoia adjusts the beat markers of all tracks within a period defined in the "Time window" parameter to the beat marker position farthest to the left.
6. To separate all quantization objects, use the "Split at transients" command.
7. Use "Quantize object positions" to adjust the individual positions of the slice objects. You also have the option of changing the quantization settings here.
8. In some cases the quantization will result in gaps between the slice objects. These can be filled with the option "Remove Object Gaps". The relevant objects will be stretched.

MIDI Quantization (Standard)

This command conducts standard quantization of the lengths of the MIDI notes of all selected MIDI objects according to the MIDI quantize settings.

Detailed information about quantization can be found in the chapter "MIDI in Sequoia > Quantization".

MIDI Humanize Q

This command applies the current **Humanize** value in the quantization options.

Detailed information about quantization can be found in the chapter "MIDI in Sequoia > Quantization".

MIDI Quantization Settings

This command opens the dialog for MIDI event quantization.

Detailed information about quantization can be found in the chapter "MIDI in Sequoia > Quantization".

Extended MIDI Quantize

Start Q / MIDI Start Quantization

This command quantizes the start positions of the MIDI notes for all selected MIDI objects according to the MIDI quantization settings. The grid view corresponds to the start quantization value.

Detailed information about quantization can be found in the chapter "MIDI in Sequoia > Quantization".

Start and Length Q / MIDI Quantization (Start and Length)

This command quantizes the start and length of the MIDI notes for all selected MIDI objects according to the MIDI quantization settings.

Detailed information about quantization can be found in the chapter "MIDI in Sequoia > Quantization".

Length Q / MIDI Length Quantization

This command quantizes the length of the MIDI notes for all selected MIDI objects according to the MIDI quantization settings.

Detailed information about quantization can be found in the chapter "MIDI in Sequoia > Quantization".

Soft Q (approximation quantization)

This command lets you can take the current Soft Q (power) value into consideration while quantizing.

Detailed information about quantization can be found in the chapter "MIDI in Sequoia > Quantization".

Quantize Note Ends

This command quantizes the end of the MIDI notes for all selected MIDI objects according to the MIDI quantization settings.

Detailed information about quantization can be found in the chapter "MIDI in Sequoia > Quantization".

Reset MIDI Quantization

Use this command to restore the original offset of start and length values of MIDI notes for all selected MIDI objects. This enables you to undo quantization at any time (even after the VIP has been saved).

Detailed information about quantization can be found in the chapter "MIDI in Sequoia > Quantization".

MIDI Input Q (global)

Use this option to quantize the MIDI recording destructively. The original position may be restored via the menu item "Object -> Quantize -> Reset Quantization".

Object Effects

The following effects can also be accessed at the object level:

Dynamics: Dynamics, Advanced Dynamics, Multiband Dynamics, sMax11, eFX_Compressor, eFX_Gate, eFX_Limiter, AM-Munition, AM-Track, AM-Phibia, AM-Pulse

Frequency/Filters: Parametric EQ, EQ116, FFT Filter, Brilliance Enhancer, Filtox

Delay/Reverb: Delay, room simulator, efx_Reverb, efx_StereoDelay, Ecox, Variverb II

Time/Pitch: Resampling/Timestretching, Elastic Audio.

Distortion: Distortion, Vandal, Bit Machine, eFX_TubeStage.

Restoration: DeClipper, DeClicker/DeCrackler, DeHisser, DeNoiser.

Stereo/Phase: Switch object channels, Multiband Stereo Enhancer, invert phase (both channels, left channel, right channel)

Modulation/Special: Vocoder, efx_ChorusFlanger, efx_Phaser, Corvex, object reverse.

Plug-ins: Opens the VST plug-in dialog.

essential FX: eFX_ChorusFlanger, eFX_Phaser, eFX_Reverb, eFX_StereoDelay, eFX_Compressor, eFX_Gate, eFX_Limiter, eFX_DeEsser, eFX_VocalStrip, eFX_TubeStage, eFX_TremoloPan.

MAGIX Plug-ins: AM-Munition, AM-Track, AM-Phibia, AM-Pulse, Corvex, Ecox, Filtox, VariVerb II, Vandal.

Select objects

Select objects under playback marker/range

Selects all objects found under the playback marker or selected track ranges.

Select all objects

Selects all objects in the arranger window.

Shortcut: Ctrl + A

Select objects in active track

All objects in the active track will be selected. This function may also be applied by double clicking a free area in the active track.

Object lasso

To select several objects, click on an empty area in the lower portion of the track and then drag the mouse over the objects you wish to select. Sometimes there is not enough space between the objects. In this case, adjust the mouse behavior with the object lasso.

Once activated, click on an object and pull up a selection frame without moving the object you clicked on (which would be the case in "Universal" mode). After selecting, the mouse reverts to regular behavior. To use the lasso selection method several times in sequence, the object lasso must be reactivated each time.

Shortcut: Ctrl + Alt + L

Select previous object

Use this command to select the previous object on the same track.

Shortcut: Ctrl + Alt + Q, <

Select next object

Use this command to select the next object on the same track.

Shortcut: Ctrl + Alt + W, >

Select to previous object

Use this command to add the previous object on the same track to the current selection.

Shortcut: Ctrl + Alt + Shift + Q

Select to next object

Use this command to add the next object on the same track to the current selection.

Shortcut: Ctrl + Alt + Shift + W

Switch selection

All deselected objects are selected and all selected objects are deselected.

Deselect objects

All selected objects are deselected.

Shortcut: Ctrl + Shift + A

Groups

Group Objects

This function assigns all selected objects to a group. As soon as one object is selected, all objects in the group are automatically selected so that you can work on them collectively.

Shortcut: Ctrl + G

Ungroup Objects

Use this function to ungroup a selection of objects. This turns all selected objects into independent objects again.

Shortcut: Ctrl + U

Temporarily Exclude Object from Group

This command removes most recently clicked object from an existing group. By clicking an object again and repeating the function, the removed object is moved back into the group.

Shortcut: Ctrl + Shift + U

or Shift + Ungroup button

Temporarily Exclude All Objects from Groups

This command temporarily removes all objects from their groups. In this case, the "Ungroup" button will blink. If the function is accessed again or if the blinking buttons are pressed again, the groups will be restored and the button will stop blinking and return to inactive status.

Shortcut: Shift + Alt + Ungroup button

The shortcut "Ctrl + Alt + Ungroup" resets the grouping history.

Additional information about grouping objects is available in the chapter "Working with objects -> Group/ungroup objects (view page 146)"

Moving objects

Move objects

This opens the dialog window to enter the starting position of an object in number of samples, milliseconds, SMPTE, or beat units.

Arrange Objects

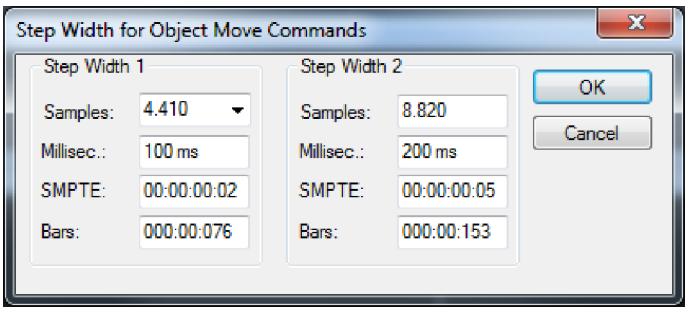
Use this function to edit the numerical interval between the selected object and the object in front of it on the same track. With multi object selection the entire distance between all objects on a track can be set.

Object/Fade Step Size

Select an object for simple object editing.

If you have selected two objects for editing, it is possible to edit it them separately as well as together.

Each of the commands below is available in two step sizes. The step sizes can be set in the dialog that follows.



Use the **"Ctrl" key to select the left object** for editing and the **"Alt" key to select the right object**. During this process the commands below are automatically carried out with **Step size 1**. To edit with a **step size of two**, **hold down the Shift key** as well. Now press a key on the number pad (0-9) to activate the respective function:

Left object to the left	Ctrl + 1
Right object to the left	Alt + 1
Left object to the right	Ctrl + 2
Right object to the right	Alt + 2
Object(s) to the left	Ctrl + Alt + +1
Object(s) to the right	Ctrl + Alt + +2
Object(s) track downwards	Ctrl + Alt + Shift + Page down
Object(s) track upwards	Ctrl + Alt + Shift + Page up
Object start to the left	Ctrl + 3
Object start to the right	Ctrl + 4
Object end to the left	Alt + 3
Object end to the right	Alt + 4
Crossfade to the left	Ctrl + Alt + 3
Crossfade to the right	Ctrl + Alt + 4
Fade-in handles to the left	Ctrl + 5
Fade-in handles to the right	Ctrl + 6
Fade-out handles to the left	Alt + 5
Fade-out handles to the right	Alt + 6

Increase left volume	Ctrl + 8
Decrease left volume	Ctrl + 7
Increase right volume	Alt + 8
Decrease right volume	Alt + 7
Increase volume	Ctrl + Alt + 8
Decrease volume	Ctrl + Alt + 7
Left object content to the left	Ctrl + 9
Left object content to the right	Ctrl + 0
Right object content to the left	Alt + 9
Right object content to the right	Alt + 0
Object(s) content to the left	Ctrl + Alt + 9
Object(s) content to the right	Ctrl + Alt + 0
Additional key for step size 2	Shift key

Please note the useful "**Object content to the left/right**" functions. Here the object length and object position are maintained while the assigned audio material from the corresponding audio file is inserted into it.

Object Hotspot to Play Cursor Position

Use this command to move the hotspot of the selected object to the playback marker position. Once you have activated a range, you can use this function to set the hotspot of the selected object to the start of the range. If you selected multiple objects, all other objects are sorted relative to the new position of the first object.

Shortcut: Ctrl + Alt + P

Object Begin to Play Cursor Position

Use this command to move the starting point of the selected object to the playback marker position. Once you have activated a range, you can use this function to set the starting point of the selected object to the start of the range. If you selected multiple objects, all other objects are sorted relative to the new position of the first object.

Object End to Play Cursor Position

Use this command to move the end of the selected object to the playback marker position. Once you have activated a range, you can use this function to set the end of

the selected object to the start of the range. If you selected multiple objects, all other objects are sorted relative to the new position of the first object.

Object to Original Time Position

With this command you can move the selected objects to their original track positions.

Shortcut: Ctrl + Alt + O

Set New Original Time Position

This command sets the current object position as the new original position.

Edit Original Position

This command opens a dialog where you can edit the original position. To do this simply enter the new original position for the object in the input field.

Object step size 1

See "Object/Fade step size".

Object step size 2

See "Object/Fade step size".

Hotspot

Setting a Hotspot

With this function makes it possible to give an object another special snap point hotspot instead of its object front edge.

The hotspot is not placed at the current playback marker position inside a selected object.

Shortcut: Shift + H

A vertical line represents this in the object. From now on this object snaps to its hotspot at the corresponding snap markings.

You can make the selected object jump from its hotspot position to any changed position of the playback marker by using the function "Object hotspot to playback marker position (Shortcut Ctrl + Alt+ P)" (view page 801).

Delete Hotspot

This command deletes the hotspots for the selected object.

Takes

Take Manager...

Detailed information about the take manager is available in the chapter "Managers > Take manager (view page 201)".

Shortcut: Ctrl + Alt + Shift + T

Take Composer

This menu item opens the take composer.

Detailed information about the take composer can be found in the chapter "Manager -> Take Composer (view page 203)".

Object Color/Name

Object Name...

This command opens a dialog to assign names to all of the selected objects.

Shortcut: Ctrl + N

Object Background Color...

The selected background color from the color dialog is applied to all selected objects.

Object Foreground Color...

The selected foreground color from the color dialog is applied to all selected objects.

Note: If you have selected "Red/Blue alternation" or "WaveColor colors" for the waveform color in the display options ("View -> VIP display mode -> Definition (view page 694)"), there will be no change to the display of the object foreground color. The change to the waveform color will only be active when you restore the "Default color settings".

Spectral display

- The Spectral view is split for the left and right stereo channels (stereo representation)
- Activate Spectral view by going to "File > Program Preferences > System/Audio > Design > View options".
- Spectral display and WaveColor view are now available for individual VIP objects: "Object > Object color/name"
- Various color palettes are available
- Graphical data is saved as *hs files.

WaveColor Display

See „File“ > „Program Settings“ > „Display Options“ > „Waveform Color“.

Object Freeze

This function renders each selected object as a new wave file. The audio return signal of a software instrument replaces the original MIDI object with an audio object. In this case the audio return signal must be routed to the MIDI track.

The original object always remains preserved and can be re-edited using the "Edit Object Freeze" function or put back into the arranger with the command "Unfreeze Objects".

If more than one object is selected, the "Freeze Objects" function will be applied to each individual object. The frozen objects are grouped. Fade-in, fade-out, and object volume are not calculated, since these properties are applied to the object properties of every new resulting object.

Edit Object Freeze

This command opens the Freeze VIP where the frozen object is saved. This VIP contains tracks with the original object.

Changes made in the temporary freeze project are applied to the object as well if requested.

Note: The length of the freeze VIP cannot be changed. This is because the length is set by the object to which the "object freeze" or "glue objects" function was applied.

Unfreeze Object

This menu item removes the wave file created during "Object freeze" and brings back the original object including the settings that were saved in the temporary trackbouncing VIP.

Note: Do not create additional new tracks in the freeze VIP, since using the "Unfreeze object" function will no longer be possible.

Remix Agent – Tempo and Beat Recognition

The Remix Agent is a powerful tool for analyzing the tempo and beat of your music. First, automatic tempo and beat recognition is carried out for manual editing later. You can then split the object into remix objects, adjust the arrangement tempo and object tempo to one another, and write the tempo and beat information into the audio file.

Remix Agent – Applications

- Provides beat-precise splitting of songs so that you can rearrange the remix objects any way you like in the multitrack project.
- Adjusting the tempo of the arrangement to the tempo of the newly integrated song/CD track.
- Integration and adjustment of newly integrated song parts to the tempo of the arrangement

Remix Agent – Requirements

Tempo and beat recognition is applied on audio material with a length between 15 seconds and ten minutes. This should be rhythmic music.

Launch Remix Agent

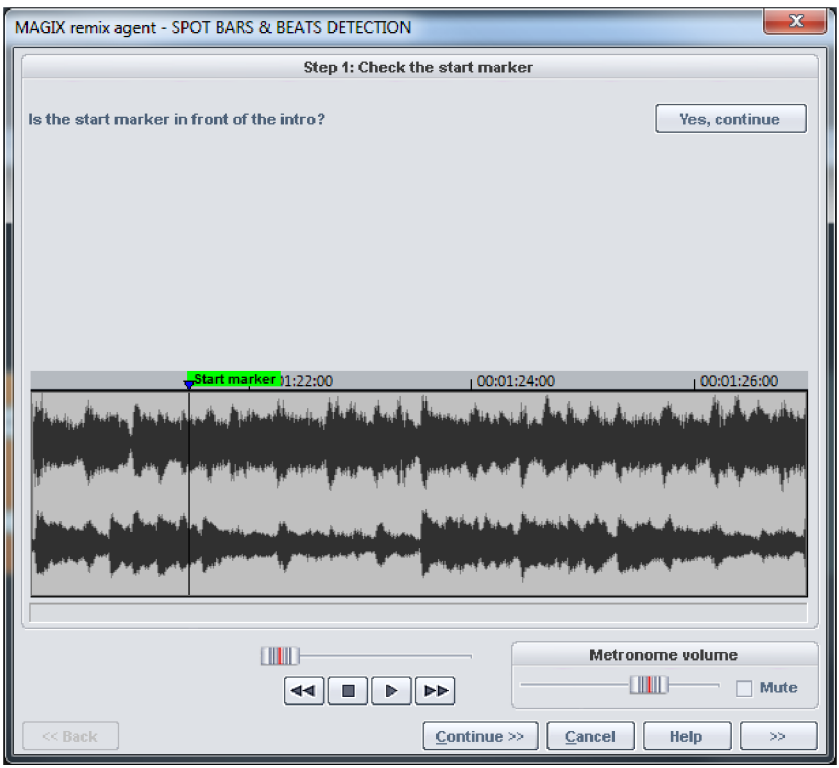
Start the Remix Agent from the "Tools" menu or from the context menu by right clicking the object.

Note: To use the Remix Agent in the wave editor, make sure that there is no tick beside "Destructive wave edit mode" ("File -> Project properties").

Remix Agent – Working Method

The Remix Agent works in fours steps:

1. Checking the start marker
2. Tempo recognition
3. Specifying beat starts
4. Application of BPM and beat recognition



1. Checking the start marker

Place the playback marker at the position where detection should start in the remix object by clicking the timeline. If the object contains a longer intro without any beats or quiet synthesizer sounds in their place, the start marker should be placed after the intro.

The start marker should always be placed just before the beat or even better, just in front of the starting beat of a bar.

Alternatively, you can set the playback marker start position before opening the Remix Agent in the arranger of the virtual project or in the wave editor for the position where detection should start.

2. Check the automatic tempo and bar recognition

After opening the Remix Agent by pressing the "Continue" button, it will begin to analyze the audio material and try to determine the tempo. The object is then played with the result sounding like a metronome click and numbered green beat lines appear in the waveform display.

Note: If the tempo or bar information of the object to be analyzed is already available, these are displayed as dots at the respective positions above the display of the waveform.

Below the waveform display, a display is provided on the left-hand side for the detected beat. In the middle, you'll find a small transport controller to simplify navigation. The fader acts as a position controller. To set the metronome volume, additional fader and mute buttons are provided on the right-hand side.

Correction of beat positions and tempo

Automatic tempo recognition doesn't always work from the start. If you don't hear the metronome clicking in time with the music, click the "No" button in the upper section of the dialog in order to access the manual tempo input dialog.



To correct the metronome speed and any time shifting that may occur between the metronome clicks, use the tempo correction as well as the "Tap tempo" button:

Tempo correction: The Remix Agent provides various speed settings; the speed the detected by Remix Agent as the most probable is preset. If the determined speed isn't correct, select a different, more suitable one from the list. The next time the object is played, it should be synchronous with the metronome clicking.

On/off beat correction: Sometimes the tempo is right, but the beats have been displaced. "On/off beat correction" provides a number of alternatives for moving the beats according to the complexity of the rhythm. Try out the different settings until you hear the metronome clicks in time with the beats.

Tap tempo: Alternatively, click rhythmically on the "Tap tempo" button or press the "T" key. Additional blue lines are displayed in the wave display. After at least four taps, the Remix Agent attempts to select the correct tempo from the list in "Tempo correction". The display next to the "Tap tempo" button displays the current status. Keep tapping until the red display showing "Unlocked" changes to the green "Locked" setting.

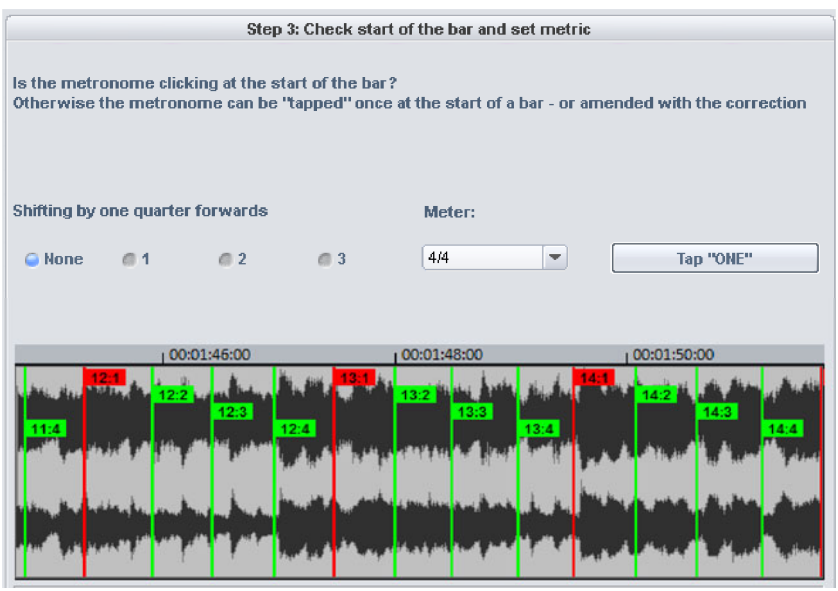
Use the **"0" key** to manually set the quarter beats while the music plays. Stray markers are automatically removed in so that the set tempo remains fundamentally intact.

You can move the markers with the mouse. If you hold the "Ctrl" button simultaneously, subsequent markers are also moved.

If the metronome clicks now correspond with the music, continue to the next step.

3. Determining the start of a bar

Now set the beat type. 4/4 beat is set as default. You can correct it here if need be. The beat at the beginning of the beat should always be synchronized with the high, stressed metronome click or the red line in the waveform display.



This can be corrected in one step: If the start of the bar is audible, click "**Tap one**" once with the mouse or press the**"T" key** on the keyboard.

Alternatively, select how many quarter notes the "One" is to be moved back.

Use the **"0" key** to manually tap the position of the beginnings of the bars during playback. This is an efficient option for correcting the bars of longer sections.

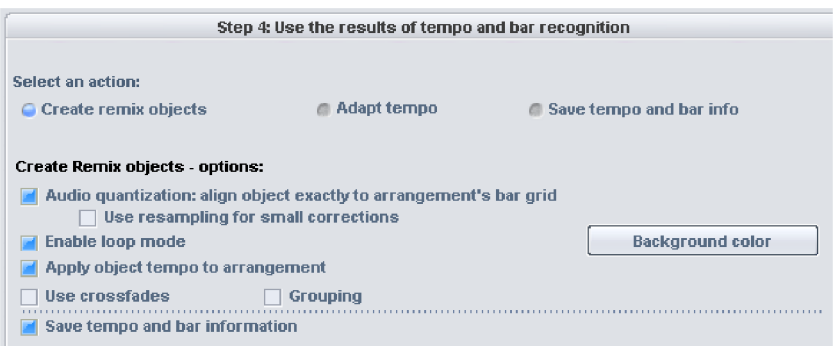
Continue to the last step once the starts of the bars are now correct.

4. Application of BPM and beat recognition

This allows you to specify what should be done with the analyzed audio material. The following option is available:

- Create remix objects from the analyzed audio material
- Adapt the project tempo to the tempo of the audio material or vice versa
- Save only the tempo and beat information in the audio file for possible editing later on.

Create remix objects



Use this option to split the song into individual objects by beat, which can then be used later on in the virtual project.

Note: The "Create remix objects" option can only be processed if the Remix Agent has been opened from within the virtual project.

Create remix objects – audio quantization

If you have selected this option, the new objects will be fit directly into the beat grid of the arrangement.

Especially with live songs, there are slight tempo variations, making it possible that various beat lengths appear. Object Time Stretching is automatically activated and applied to correct the difference in length so that the objects still fit into the rigid beat grid of the arrangement.

The "**Use resampling for small corrections**" option makes sure that the higher quality "resampling" algorithm is used on smaller corrections rather than the "Time Stretching" algorithm.

Note: If you change the tempo of your multitrack project later and adjust the audio objects in the VIP to the new tempo, there will be clearly audible pitch changes in the remix objects.

Remix objects in "Loop" mode

If you select this option, the new objects will be set to "Loop" mode. This way, newly created remix objects can be stretched out as much as you like using the right object mouse handle.

Create remix objects – set arrangement tempo to the object tempo

Here, the arrangement of the virtual project takes on the found BPM value. To use the split song as the basis for a new composition in the remixes, use this option.

Use crossfades: Remix objects may be faded into one another with this function. The parameters of the transitions can be set in the crossfade editor.

Group: This function groups the remix objects.

Save tempo and beat information in audio file: If you select this option, the tempo and beat information are written to the audio file.

Background color: Click this button to set the background color of the remix objects you want to create.

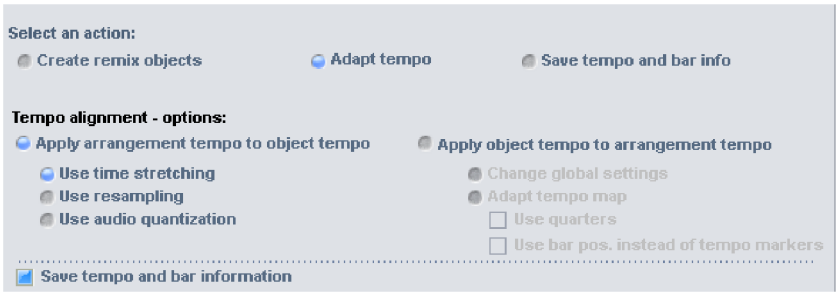
Adapt tempo

This option provides you with the opportunity to adapt either the tempo of the analyzed audio material to your project or the project speed to the speed of the analyzed audio material.

Note: The "Tempo alignment" option can only be processed if the Remix Agent has been opened from within the virtual project.

Apply arrangement tempo to object tempo

This adapts the object length to that of the existing arrangement.



You can choose from three different methods:

Use Time Stretching: The pitch of the song remains constant in timestretching, but the sound quality may suffer.

Use Resampling: Resampling changes the pitch (similar to changing the speed of a record player), but retains most of the sound quality of the song.

Note: If you change the tempo of your multitrack project later and adjust the audio objects in the VIP to the new tempo, there will be clearly audible pitch changes in the remix objects.

Use audio quantization: During audio quantization, the tempo adjustments are calculated into the audio file so that it seems as if remix objects were created and then immediately compiled into a new audio file. If recognition is unreliable, the result may feature extreme tempo variations. In this case, it is particularly important to set the start marker at such a position (before opening the Remix Agent) so that the tempo may be detected reliably. The advantage of audio quantization is that smaller tempo variations may be balanced in the music. The starts of the bars in the music correspond with the bar starts in the arrangement: they do not slowly drift apart.

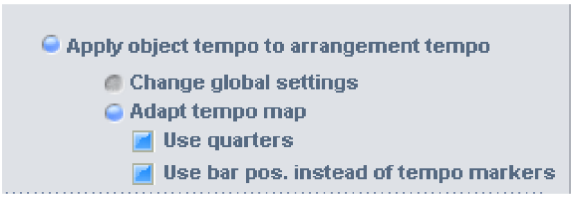
Save tempo & bar info into the audio file: If this option is active, the tempo and bar information are saved in the audio file. The objects in the virtual project won't be changed.

Set arrangement tempo to object tempo

Just like "Create remix objects", the arrangement of the virtual project assumes the BPM value determined in the Remix Agent. If you would like to use the split song as the basis for a new composition in the remixes, then select this option.

Change global settings: The BPM value of the arrangement in the VIP is set to the detected value.

Adapt tempo map: Sets a tempo marker in the virtual project's arrangement for each bar from the position of the playback marker to the end position of the remix objects.



- **Use quarter notes**: If this option is active, four markers are set (quarter notes) instead of one (whole notes).
- **Use beat position instead of tempo marker**: Beat position markers are set instead of tempo markers.

First example: Synchronizing a MIDI arrangement

1. Place the song object you want to synchronize the MIDI arrangement and the MIDI object to on top of one another on the same track in the virtual project. Open the remix agent for the song object. Specify the beat and quarter positions using the remix agent in step 1-3.
2. In step 4, select "**Adjust tempo**", then "**Apply arrangement to object tempo**" and "**Adjust tempo map**". This specifies that you want to create a tempo map.
3. Now select "**Use quarter beats**" and "**Bar position instead of tempo marker**". The tempo map receives a synchronization point at each quarter and not only at the beginning of each beat. If you click the "**Apply**" button and open the MIDI editor for the MIDI object, the beats will match one another and all notes will be shown and played in sync to the song. Even the metronome click will be in time with the song.

Second example: Mixing two song objects

1. First, adapt the project tempo of the virtual project to that of the first song object that you want to fade from. Select the options "**Apply object tempo to arrangement tempo**" and "**Change global settings**".
2. Next, adjust the speed of the second song object to that of the project by applying the "**Set object tempo to arrangement tempo**" option.
3. Since the project speed now matches the song speeds of both songs, you can easily fade between the two songs.

Save tempo and bar info

If this option is active, the tempo, and bar information are saved in the audio file. The objects in the virtual project won't be changed.

Generate beat markers in the current range: Use this option to place markers at the beginning of the beats of a song. This corresponds to the positions of the strokes displayed in red in the waveform display.

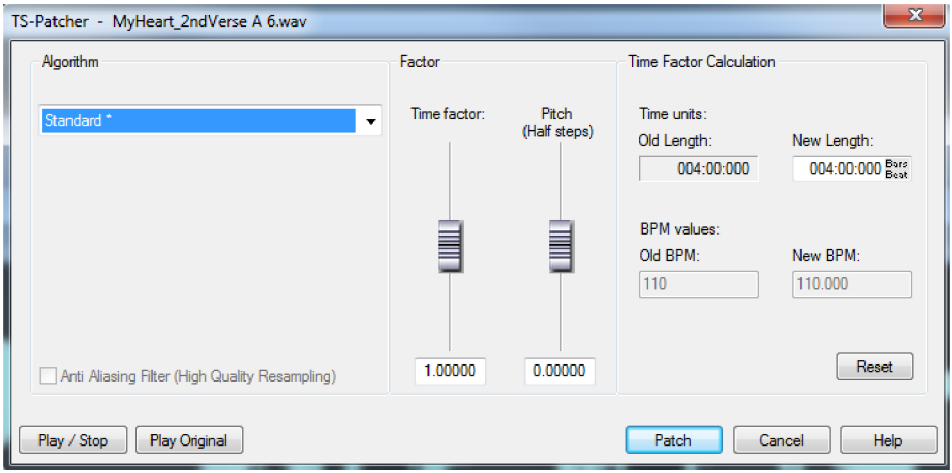
Generate quarter markers in the current range: Select this option to place markers at the beginning of the positions of the quarter beats starts. This corresponds to the positions of all the strokes in the waveform display.

WaveColor audio search

This function uses the WaveColor algorithm. This allows you to find the same or similar sounding ranges in an audio file very quickly. To do this, double click on the object you wish to use while holding down the "Shift" key, and this will then open it as a wave project. Now, select the range that you want to examine and copy it to the clipboard (keyboard shortcut: "C"). Now start the audio comparison function.

The dialog that opens contains a "Sensitivity" slider which lets you adjust the threshold value for the algorithm. Depending on the sensitivity level you choose, a different number of markers with the name "Match" will appear in your project at those points where the algorithm identifies similar-sounding passages. Close the search dialog. You can now jump to the ranges that were detected by clicking the corresponding "match" marker.

Time Stretch/Pitch Shift Patcher...



Time Stretch/Pitch Shift Patcher - Overview

Use this tool to patch wave files for use in Sequoia. In this case, additional information and settings may be written into the wave project. These enable the processing of Time Stretching and Pitch Shifting Algorithms at optimum quality. Without the patcher, you would have to search and set the settings and additional information for each wave file you wish to edit.

The following may be patched individually:

- Algorithms for Time Stretching/Pitch Shifting
- Time factor/pitch
- Time factor processing (new length/new BPM)

Timestretch/Pitchshift Patcher - Patching Process

Patching only works when you have opened the audio file in audio editing (view page 62). It can also not be open in a virtual project at the same time.

Select the algorithm you want to use from the dialog. You can also specify the tempo of the wave file using the "Time Factor" controller.

You can now test the selected algorithm in conjunction with the "Play/stop" and "Play original" buttons. If you click the "Patch" button, Sequoia will write the information permanently to the wave file.

Algorithm for Time Stretching/Pitch Shifting

In Sequoia you can use the algorithms élastique Pro, élastique Efficient, Resample and Monophonic Voice for realtime time stretching and pitch shifting.

Detailed information about this is available in "Effects" > "Time / Pitch" > Resampling/Time Stretching (view page 912)". When you find a time stretching algorithm that meets your requirements you can use the Patcher to save it in the audio file. This ensures that this selected algorithm will be used automatically when you apply time stretching/pitch shifting to your audio files.

Beat Markers

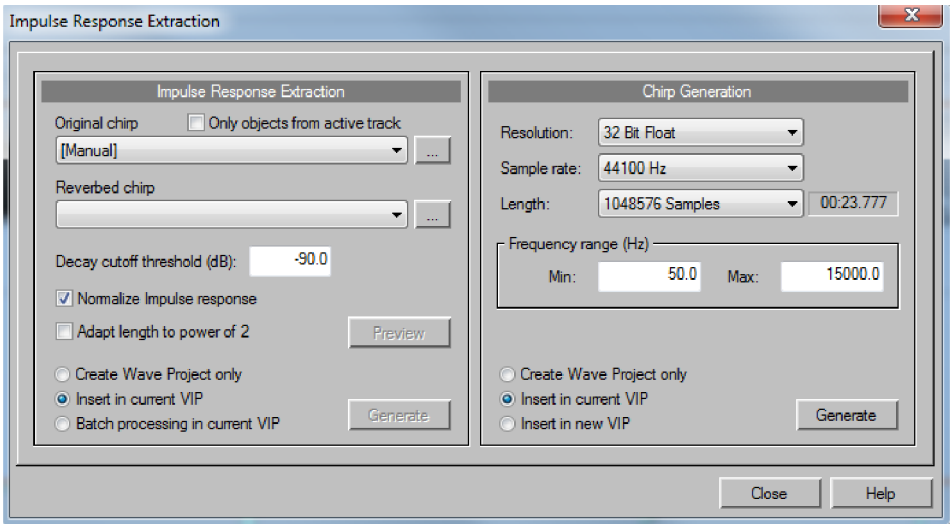
Beat markers synchronize the audio material so that the groove remains perfectly intact. When using the beat marker-based algorithms, the beat markers are also saved in the wave project.

Note: Unlike with the Time Stretching dialog in the "Effects" menu, the patcher is non-modal, meaning that you can move the beat markers in the wave project and simultaneously control the result of the Time Stretching.

BPM Value

The desired BPM value (beats per minute) can be patched here. This is useful for when the Time Stretching factor must be determined later to adjust the wave project to the tempo of an existing arrangement. If the "New BPM" field in the dialog is grayed out, you can change the value using the "Time factor" controller.

Impulse Response Extraction...



Use this dialog to create your own impulse responses from effect processors or natural spaces.

The chirp process used in Sequoia is superior in terms of determination of impulse response. Among other things, it distinguishes itself in terms of its robustness with regard to evenly measured noise or nonlinear distortions.

The impulse responses extracted here may be used in the Room Simulator (view page 901).

Impulse response extraction - General Operation

1. Generation of a chirp
2. Collection of the spatial or reverb impulse by processing the chirp in reverb devices or by playback in real rooms; micro-recording of the sound of the room
3. Calculation of the impulse response from the original and reverberated chirp

Generation of a Chirp

A chirp signal is a sine shape signal that features a constantly changing frequency.

The right side of the impulse response detection dialog features the **parameters for chirp generation**:

Resolution: Select between 16-bit and 32-bit float resolutions here. We recommend a 32-bit float resolution.

Sample rate: Select the sample rate. Note that the sample rates of the original chirp, the reverberated chirp, and the virtual must all match.

Note about sample rates: To use 96 kHz or 192 kHz, note that the specification of natural impulse responses of the speakers must be capable of playing back the high frequencies so that extremely low ultrasonic components of the impulse responses are also able to be detected. Normally, these operate at 44.1 or 48 kHz. The Room Simulator resamples the impulse responses at high sample rates internally to 96 kHz ot 192 kHz.

Length: Select the length of the chirp in samples, staged in powers of two.

The longer the chirp length, the better the results, since the influence of consistent distortions like hissing, traffic noise, or non-linear distortions in the result is reduced. On the other hand, the CPU load increases while processing the impulse response.

Tip: The chirp should be at least as long as the length of the reverberation.

Frequency range (Hz): Specify the start and end frequency of the chirp here.

Note: The frequency range should not be arranged according to the frequency range of the loudspeaker. Frequencies in the sub bass range which cannot be emitted by the loudspeaker should also be avoided.

Options

Create wave project only: The chirp is written to a newly created wave project.

Insert into current VIP: The newly created wave project is added to the current virtual project.

Insert into new VIP: The new wave project is added to a new virtual project. This enables the chirp to be played immediately while the reverb in the room is recorded simultaneously in the same VIP.

Generate: The chirp is generated and written to a wave file. According to the option selected, this will now be visible in the virtual project as an object or the wave project. After generating the chirp, the dialog may be closed again.

Reverberation of the Chirp Generated

Natural rooms

To detect the room or reverb impulse, the chirp signal is played back in mono with a speaker or a combination of speakers in the room. The speaker or the combination of speakers should be set up in the room where the acoustic sound sources such as instruments or singers are located; for example, in an opera house, this would be on the stage or in the orchestra pit.

Depending on their design, individual speakers feature different orientation than typically natural sound sources, and they radiate less sound energy from the rear. For this reason, the complete loudspeaker combination should be used to transmit the chirp.

One proven speaker combination is to direct one loudspeaker towards the audience and direct another towards the rear (in the opposite direction).

Digital reverb devices

This processes the chirp signal with the reverb device.

The un-reverberated, dry component ("Dry" parameter) should always be set to zero in this case so that the original chirp is not mixed in. Otherwise, the resulting impulse response will be overlaid by a high-level Dirac impulse.

Note: Record the reverb until it dissipates completely so that the recalculation is successful. Impulse noise should not occur during dissipation.

Simultaneous playback and recording of the chirp in Sequoia

- In the dialog "Recording Options" (shortcut: Shift + R), set a check next to "Playback During Recording" and select the entry "Peak Meter Monitoring" for the monitoring mode, since other monitoring settings may cause acoustic feedback.
- Activate the "Monitor" button in the chirp track and switch the "Visualization" on via the toolbar in the arranger.
- Start a test recording that features the complete chirp and adjust the peak control. The recording of the reverberated chirp should be controlled well enough that overmodulation does not occur. After the optimal peak control levels have been set, record the reverberated chirp.
- If another visualization window featuring a spectrogram display is opened and the value range is set to -90 dB, you can check the reverberated chirp for nonlinear distortion, noisy impulses, or aliasing.

Recalculation of the impulse response from the original reverberated chirp

Open the dialog for impulse response extraction again via the "Object" menu. The dialog elements for calculating the impulse response are featured on the left hand side of the dialog.

Original chirp: Select the wave project or the object here that features the original un-reverberated chirp.

If the original chirp is not available, the parameters of the original chirp will have to be set manually on the right hand side of the dialog. Note that each deviation of the set parameters from the original chirp may cause the measurement system to fail.

Reverberated signal: The wave project or object that contains the reverberated chirp has to be selected. When working with a virtual project (see below), this list is filled automatically. The letter "T" (for track) and the track numbers are placed in front of the object names in the list.

Decay cutoff threshold: A threshold in may be set here dB which specifies the point in the signal value at which the impulse response should be cut.

Adjust length by power of two: Use this option to hide the impulse responses so that the length in samples slightly smaller than a power of two. This provides performance optimized when applying the impulse response in the room simulator.

Create Wave Project only: The resulting sample with the impulse response is only created as a wave project.

Insert in current VIP: An object with the impulse response is loaded into the current virtual project.

Batch processing in current project: To apply batch processing, assign the reverberated chirps as objects in a virtual project below the original chirp track-for-track. If this option is activated, impulse responses will be calculated for all objects found in the list of reverberated chirps. The requirement for this of course is that the same original chirp has been reverberated. New objects with impulse responses appear correspondingly shorter behind objects featuring reverberated chirps. Use the "Mute" button to remove objects with reverberated impulse responses from the batch processing list.

Preview: Calculates and plays the impulse response.

Generate: Calculates and saves the impulse response.

Impulse Response Extraction - Tips and Tricks

Digital reverb devices

- Use the digital in and outputs, provided these are present.
- If only analog in and outputs are available, then these should be used with a 24-bit converter.
- Underdriving worsens the signal noise interval.
- Digital distortion should definitely be avoided.
- The parameter for the mixing the reverberated signal ("dry" component) should be set to zero.
- Draw the chirp completely up until the reverb decays.
- Don't allow any switching on and off clicking sounds to enter the recording of the reverberated chirp.
- The chirp length normally does not need to be longer than a minute (this does not apply if chirp residue occurs in the results. Details about this are provided in the corresponding section below.)
- In practice, a length of 47 seconds has proven to be normal in rooms.
- A bandwidth of 50 - 15000 Hz is usually enough.
- For chirp playback and recording of the reverberated chirp, you should use converters with identical properties (not different devices).

Multi-channel impulse responses

- When recording multi-channel impulse responses for the Surround room simulation, a real multi-channel recording is not necessarily required.
- The impulse responses for the various directions can also be recorded after one another, since these may be synchronized later. The recordings can also be synchronized retroactively.
- For this synchronization to work properly, you should record in stereo format and feed the second stereo channel with a reference microphone with a constant position.
- The stereo recordings of the impulse responses can now be synchronized perfectly in the virtual project with the reference signal on the second channel. After this, you can extract Surround impulses by bouncing stereo recordings and muting the reference microphone channels.

Natural rooms

- Light analog overdrive in playback of the speaker or amplifier used for the chirp is not problematic, at least if longer chirps are used.
- Avoid digital distortion in the AD/DA converter.
- The louder the chirps are played, the better the signal noise interval will be in the measurement.
- The longer the chirps are, the better sustained noise such as traffic or hissing will be removed from the resulting impulse response. The signal noise interval

improves enormously compared to the signal noise interval of the reverberated chirp, i.e. longer than a minute.
- You can also use longer chirps to extract impulse responses with very good signal noise interval at lower playback volume.
- Avoid impulse noises during measurement that sound like dropping objects, steps, chairs, etc. in order to exclude temporally reversed chirps from the impulse response.
- The linear distortions from microphones, amplifiers, speakers transfer to the impulse response. For this reason, use components that are as high-quality as possible. The speaker is normally the weakest link in the chain.

Impulse Response Extraction - Problems and Solutions

Chirp residuals in the impulse response

In some cases, the calculated impulse responses contain chirp residuals. The following describes two typical causes and solutions.

Cause 1: Aliasing

When using older digital reverb devices that work with lower internal sampling rates, aliasing may occur. The spectrogram can help to display the reverberated chirp, which may be easily recognized in the frequency range, provided the value range of the visualization is set to -90 dB.

Solution

Experiment with different sample rates to test whether the aliasing component is able to be reduced by using digital in and outputs instead of analog ones.

Otherwise, only use of a very long chirp will help. The reminaining chirp is then subjected to a low frequency range (e.g. 12000 Hz to 12100 Hz) and may be removed with a steep band-pass filter. The FFT filter provided by Sequoia is suited to this process ideally.

Cause 2: Incorrect impulses during recording, reverberation, or signal transmission

Impulses or impulse-like acoustic sounds also produce chirp residue in the extracted impulse response. The process reacts very sensitively to these noises. Impulses featuring a level less than approx. -50 dB will sound noisy.

Some of the causes of incorrect impulses:

- Crackling that is present during playback of the chirp or recording of the reverberated chirp due to driver problems, etc.
- Crackling when the chirp is not played from the start.

- Inconsistency due to premature ending of recording, e.g. due to incompletely recorded reverberation after the chirp has ended.
- Impulse-like noises when recording natural spaces, e.g. dropping objects, steps, moving chairs, etc.
- Incorrect cable, outlet, etc.

Solution

In order to isolate noisy impulses like crackling, listen closely to the reverberated chirp. If the reverberated chirp is overlaid with impulse response sounds, incorrect impulses occurred on the playback side. If the reverberated chirp is overlaid with impulse sounds, then incorrect impulses occurred on the recording side.

In the spectrogram display of the visualization, impulses may be recognized as vertical lines.

Cause 3: Different properties between the DA converter for the playback of the chirp and the AD converter for the recording of reverberated chirp.

If you're working with different sound cards/devices, a chirp-like noise may occur shortly prior to the start of the impulse response. The cracking noise at the beginning of the impulse response may sound very "chirped".

Solution:

You should always use the same device for recording and playback.

Recalculation of Impulse Responses when Using Longer Chirps

The recalculation of impulse responses from longer reverberated is extremely processor-heavy. The memory requirement is also extraordinarily high. For example, for chirps with a length of 6 minutes at 44.1 kHz, approx. 700 MB RAM is required.

Make sure that enough RAM is present in this case.

The progress display does not behave linear in this case; it moves forward increasingly slowly during the process, since the number of operations required increases constantly towards the end of the calculation.

A a large number of calculations are planned, then batch processing is recommended, e.g. overnight.

Impulse Response Process - Limits of Simulation of Reverberation Devices

The impulse response process enables linear and time-variable systems to be simulated. Effects such as chorus or flanger change the transfer function dependent on time, i.e. they behave according to the time variation, meaning they cannot be simulated. The same applies to effects of any kind that feature parameters that are modulated by LFOs.

Distortion or compressors feature nonlinear functionality and are also impossible to simulate. The reverb presets of multi-effects devices often still use other effects than actual reverberation, e.g. chorus effects that cover resonance. In this case, exact reproduction of the reverb preset is not possible.

The benefit of the impulse response process, on the other hand, is its abaility to simulate natural rooms with extremely high quality.

More

Duplicate Objects (in Place)

Use this function to duplicate all selected objects. The duplicated objects lie congruently over the original objects. The duplicated objects may be moved to the desired position by dragging the mouse in "Object/curve" or "Universal" mode.

Duplication is also possible via drag & drop while holding down "Ctrl".

This enables objects to be duplicated quickly without having to access the VirtClip.

Extract

Use this command to delete all unselected objects.

Note the difference between this and the "Extract" function in the "Edit" menu (view page 726), which only applies to the selected area.

New Object

This function creates a new object. The last activated object is inserted at the position of the playback cursor in the selected track.

Play/Rec Menu

Play Once

This plays the wave project or the selected range once. This function corresponds to the "Play" button in the transport control (view page 65) as well as the "Play once" button in the toolbar.

Shortcut: Spacebar

Play Loop

This plays the wave project or the selected range in a loop. This function corresponds to the "Play" button in the transport control (view page 65) as well as the "Play loop" button in the toolbar.

Shortcut: Spacebar

Play in Range/Loop

The current range is played from the start of the project as a loop. This mode is especially useful for testing loops in the instrument sample and corresponds to the "Play in range/loop" button in the toolbar.

Shortcut: Shift + P

Play with Preload

Playback is prepared with all buffers loaded. Use this function prior to starting precise synchronization manually.

Shortcut: Shift + Spacebar

Play Only Selected Objects

Use this command to play the selected objects only. All unselected objects will be muted temporarily. If the play cursorplayback markeready located at the first selected object's time position, it will be moved there.

Shortcut: Ctrl + Spacebar

Play Cut

Play to Cut Start (In Point)

A defined time segment is played until the start of the selected range. The duration of the section can be set through "File -> Program settings -> Set pre-roll time (view page 709)" or for the crossfade editor through "Options -> Program settings -> Program -> Crossfade editor -> Playback -> Pre-roll time (sec)".

In the crossfade editor, the upper object is played with the pre-roll time at the cut position.

Shortcut key: F5

Play from Cut Start (In Point)

A defined time segment is played from the start of the selected range. The duration of the section can be set through "File -> Program settings -> Set pre-roll time (view page 709)" or for the crossfade editor through "File -> Program settings -> System options -> Program -> Crossfade editor -> Playback -> Post-roll time (sec)".

Fade-in objects will be played back in the crossfade editor without a fade.

Shortcut key: F6

Play to cut end (Out Point)

A defined time segment is played until the end of the selected range. The duration of the section can be set through "File -> Program settings -> Set pre-roll time (view page 709)" or for the crossfade editor through "Options -> Program settings -> Program -> Crossfade editor -> Playback -> Pre-roll time (sec)".

Fade-out objects will be played back in the crossfade editor without a fade.

Shortcut key: F7

Play from Cut End (Out Point)

A defined time segment is played from the end of the selected range. The duration of the section can be set through "File -> Program settings -> Set pre-roll time (view page 709)" or for the crossfade editor through "File -> Program settings -> System options -> Program -> Crossfade editor -> Playback -> Post-roll time (sec)".

In the crossfade editor, the upper object is played with the post-roll time at the cut position without a fade.

Shortcut key: F8

Play Over Cut

Here a cut operation is simulated where a defined time range is played back up to the start of the selected range, the selected range is skipped over and a specified range after the end of the selected range is played back. The duration of the pre-roll/post-roll time can be set through "File > Program Settings > Set Pre-roll Time (view page 709)" or for the crossfade editor through "File > Program Settings > System Options > Program -> Crossfade Editor -> Playback -> Pre-roll/ Post-roll Time (sec)".

Keyboard shortcut: F4

Play into Cut

Playback begins allowing for the lead-in time before the selected range starts and ends allowing for the lead-out time at the end of the selected range.

Play Fade Out

The fade-out of the first (upper) object plays back in the crossfade editor with pre-roll time.

Shortcut: Shift + Spacebar

Play Fade In

The fade-in of the first (lower) object plays back in the crossfade editor with post-roll time.

Shortcut: Ctrl + Spacebar

Play Upper Content

Plays the audio material in the upper half of the crossfade editor.

Shortcut: Alt + Shift + Spacebar

Play Lower Content

Use this command to play the audio material in the lower half of the crossfade editors.

Shortcut: Alt + Ctrl + Spacebar

Restart Play

Use this command to move the playback marker to the original position during playback or back to the range start and then start playing from this position once again.

Stop

Choose this option to stop playback. The playback marker will jump to the start position. Whether or not the original position is the previous starting position or current stop position may be set in the playback parameters (shortcut: "P").

Shortcut: Spacebar

Stop and Go to Current Position

This option cancels playback; the playback marker will remain at the current position.

Shortcut: Number block "0" or ","

Playback Mode

Loop Mode

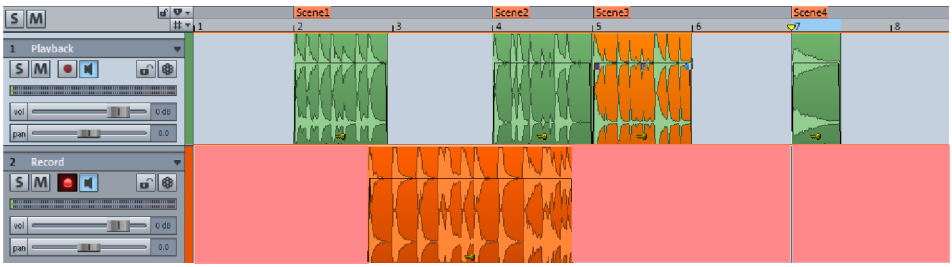
This mode plays the selected ranges in a loop. This corresponds to switching on the "Loop" button in the transport control.

Change Play Direction

This command changes the playback direction, even during playback.

Cue Mode

Cue mode is a special live playback mode often used for special cue sounds in radio broadcasts and theater. The objects (cues) in this audio track can be played back in sequence while another track is recorded simultaneously.



You can control cue mode in two ways:

1. Control using object edges

Starting playback with the space bar highlights the next range and then plays it in Cue mode. Playback stops automatically at the object's end. Upon resuming playback the following object starts. Related objects are treated as an object.

You can play back multitrack projects in cue mode. The object on the first track will be used for the Start/Stop range markings.

2. Control with track markers

With the help of the CD track markers you can use and control cue mode for complex arrangements independent of the object edges of the first track.

To do this, go to „System Options > Playback (view page 59)“ and activate "At Track Indices" in the Cue Mode section.

Now position CD track markers at the desired positions in the timeline.

Once play is activated, playback will begin from the next CD track marker and end at the next one after that, which in turn is the starting put for the next playback range.

Note: For smooth cue mode functioning deactivate the loop mode.

In cue mode, the recording mode in the transport console changes to **Record without playback (band control)** (view page 833). The cue track is highlighted with a special background color.

If recording is done in cue mode, automatic scrolling in the arranger window will remain switched off.

The playback marker can be repositioned at any time within the currently played cues.

In "System Options -> Playback (view page 59)" you can ensure that the **first track in cue mode is switched to "Solo"**. Here you can also set the **pre-roll (ms)** to determine the delay between the execution of the record command and the actual start of the recording.

Scrubbing Active

- Scrubbing Active Keyboard shortcut: "**Alt + Shift + Arrow down**"
- Jog (Absolute)
- Two Speed
- Shuttle (Relative)
- Scrub Left Keyboard shortcut: "**Alt + Shift + Arrow left**"
- Scrub Right Keyboard shortcut: "**Alt + Shift + Arrow right**"

Detailed information about scrubbing is provided in "Playback -> Playback options -> Scrubbing (view page 831)"

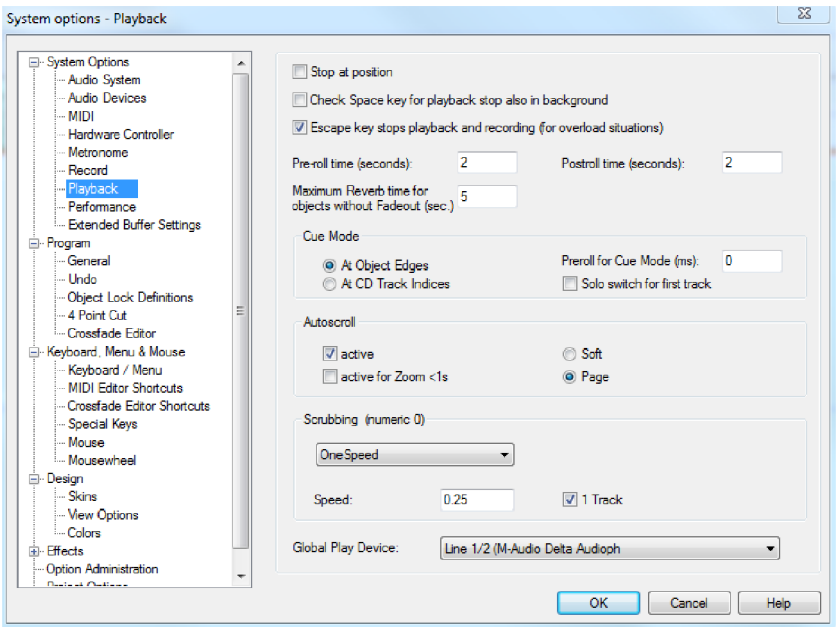
Playback at speed 1 - 4: Select from four different playback speeds. Playback begins immediately after the corresponding object is clicked.

- Play with Speed 1: Playback begins at the cursor position at 1/4 of the original speed.
- Play with Speed 2: Playback begins at the cursor position at half speed.
- Play with Speed 3: Playback begins at the cursor position at original speed.
- Play with Speed 4: Playback begins at the cursor position at double speed.

Recalling one of the four playback speeds with the Shift key pressed opens a dialog window, where you can specify an individual speed value.

Playback Parameters

This option opens the playback parameter window.



Keyboard shortcut: P

Check the "**Stop at current position**" option 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 Sequoia 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)**".

The **pre-roll time** and **post-roll time** are necessary parameters for editing in Sequoia. Pre-roll time is used to define the time section in seconds which is to be played before the start of the selected range. Post-roll time, on the other hand, is a defined time segment starting from the end of the selected range.

The **maximum reverb time** for **objects without fade-out** can be set up to 60 seconds here.

Note: Please note that the length of reverb time can lead to performance problems.

To use **cue mode** you have to specify whether playback control will be carried out at the **object edges** or the **CD track indices** (track markers). You can also ensure that the **the first track in cue mode is set to „Solo“**. Here you can enter the **pre-roll in milliseconds** which determines the delay between the execution of the record command and the actual start of the recording.

Detailed information about cue mode is available in "Play/Rec -> Playback mode -> Cue mode (view page 828)".

With **Autoscroll** the graphical display constantly moves even before the playback marker leaves the visible section. This provides a constant overview. Switch to autoscroll mode by checking "**Active**". "**Active for zoom < 1 s**" causes autoscrolling to begin at very high zoom levels of less a second

You can choose between "**Page**" and **"Soft" autoscroll 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. At smaller buffer sizes (view page 594) (e.g. < 4096 samples), scrolling will be softer.

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

If you press "**0**" on the number pad or the key combination "**Alt + Shift + Page down**" and hold these keys down, Sequoia switches to "Scrub Mode".

This provides special control via the mouse over the selected track's playback speed.

The keyboard shortcut for scrubbing to the left is "**Alt + Shift + Left arrow**"

The keyboard shortcut for scrubbing to the right is "**Alt + Shift + Right arrow**"

There are four different scrubbing modes to choose from:

Shuttle: The relative distance between the playback marker and the mouse position is used to control the speed. The playback marker follows the movement of the mouse. The further you move the mouse from the playback marker, the faster playback will be. This means:

Scrub control faders at the left edge = double speed backwards,

Scrub control fader in the middle = no movement,

Scrub control faders at the right edge = double speed forwards.

Absolute: Use the absolute position of the mouse in the window to control the speed.

Two speed: Two speeds are provided for scrubbing. The object plays back slower or faster depending on the distance between the scrub control fader and the mouse position, in which case slow scrubbing at a speed of 0.25, or 1/4 of the original speed is preset, and fast scrubbing is set to 1.0, i.e. the original speed. Change the value for slow playback in the "Speed" field.

One speed: The preset scrubbing speed is 1.0, i.e. original speed. The "Shift" key reduces this value by half. The "Ctrl" key uses the speed set in the "Scrubbing speed" field.

Scrubbing speed: This specifies the factor of the original speed applies to the scrubbing speed. 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.

Scrubbing only on active track (1 track): If this check box is active, Sequoia will only scrub the active track.

Global play device: Specifies the sound card driver for playback.

Record

If you select this menu point, recording is started immediately for the live, active track. You can record both the audio as well as the MIDI into each active track here.

Detailed information on recording can be found in the chapter "Sequoia Quickstart -> Workshop: recording (view page 92)".

Shortcut: R

Record Mode/Punch In

Standard Mode (Playback while Recording)

This option corresponds to the option in the recording options with the same name. This mode allows tracks to be recorded in addition to existing audio tracks while playback is running.

Switch on the new recording tracks and activate monitoring.



These tracks are now active in "Input" mode, i.e. the input sound desired for recording will be audible for these tracks. Recording begins immediately at the playback marker position when the recording button was clicked.

If you have selected a range behind the playback marker on the grid and marker bar and have activated "Loop" mode, then recording will only begin will only be activated once the start of the range is reached. Until this the track will signal that it is ready to record (the recording button will blink during playback), then the selected range will be recorded in "Loop" mode.

Record without Playback (Read After Write)

If you use "**Record without playback**" mode, recording will start at the playback marker position when the record button is clicked. In this case you will only here the input signal of the track that is in record mode and the playback marker will not follow recording in the timeline.

Now if you start playback from the position of the playback marker using the "Play" button on the transport console, you can hear all of the tracks being played back from this position. The recording continues independently of this until the "Record" button is clicked again. Now you will see the recording you just made as a new object in the arranger.

Punch Marker Mode

This button activates Punch marker Mode. "Punch-in/Punch-out" is a recording process that can be started and ended while playback is running.

There are two principal approaches for punch recordings:

- **Punch "On-The-Fly"**: In this mode, the recording can be started (punch-in) and stopped again (punch-out) at any time during playback. Multiple punch processes can be made during a single pass in order to improve various positions in one recording. To do this, start playback with the spacebar. Now you can click the "Record" button in the transport console to "punch in" and "punch out" again.

Note: If you check "Prepare all tracks for track punch record" in the "System options -> Record (view page 671)" menu, you will also be able to punch your recording by using the recording buttons for the individual audio tracks (see below).

- **Punch with markers**: To execute the punch procedure with markers, activate the buttons "In" (sets the punch-in marker) and "Out" (sets the punch-out marker) at the desired positions. Start the process with the record button on the transport console or of the corresponding track. The actual recording will only take place within the punch range. When the recording is started and the playback marker is not yet in the punch range the "Record" button will blink. During punch recording, it remains red.
- **Multiple punch recordings in one pass**: It is possible to make multiple punch recordings in one pass with the help of punch markers. This is done by positioning the playback marker at the desired point and pressing the "In/Out" buttons in the transport control while pressing the "Alt" key.

Note: When working with punch-in/out, it is useful to have "Auto crossfade" mode active. "Auto crossfade" will create smooth transitions automatically between takes

- **Punch recordings** can also be executed **as loops**. Select a range across the planned punch section, press the "Loop" button and activate the recording. This will be looped until the process is canceled with the spacebar. Punch-in and punch-out markers are set automatically. A punch recording is at the punch position per pass. To find the best take from these looped runs, use the "Take manager (view page 201)".
- If the recording option "**Prepare all tracks for track-punch record**" is active ("System Options -> Record (view page 671)"), audio tracks that were not active when recording was started can be added to a recording, or tracks can be removed from the recording. To do this, click on the record button of the desired track. Punching on single track requires that the track has been assigned its own sound card input which is not already being used for recording. This is indicated by a red border around the record button in the track header.

Note: Tracks that have been displaced in "Punch marker" mode by individual track punching will not be influenced by punch markers.

Record Pause

Recording is set to pause.

Recording Fail Start Function

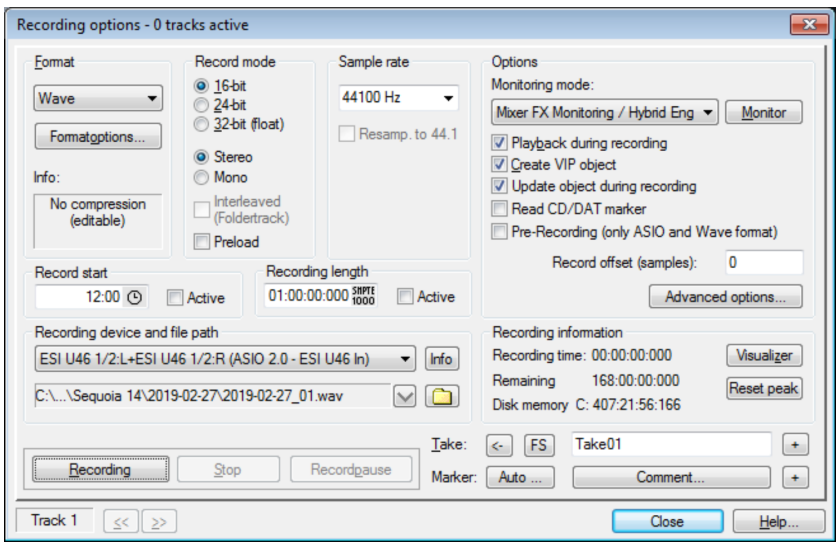
The new take is created with the same name as the last recorded take. A new object featuring the extension "FS" will be made in this case.

New Record Take

A new take is added while the recording is running. A new object is created.

Recording Options

This menu option opens the "Record" window. All recording settings can be changed here.



Keyboard shortcut: Shift + R

Format: Set the corresponding recording file format (Wave, MP3, MPG, WMA, Real Audio, AIFF, Ogg Vorbis, and FLAC). In the info box below you'll find information about each of the formats available. This way you can see if the format is editable in Sequoia and which type of compression it uses.

Record Mode: Select the desired bit resolution for your recording and choose between stereo, mono or interleaved recording. Interleaved is only active if all armed tracks are combined in one folder track. Read more on this in Interleaved recording (view page 106).

Preload: The recording is prepared and all buffers will be loaded. After that the message "Data loaded. Waiting to start..." appears. As soon as you press "OK", the recording starts without any delay.

Sample rate: This option allows you to select the sample rate of the recording. Take note that your audio card must support the selected sample rate.

Resampling to 44.1: If you have set a sample rate of 44,100 Hz and then activate this option, Sequoia will apply resampling at 44.1 kHz. The quality of resampling can be set in "**System options -> Effects -> Resampling/Bouncing.** (view page 681)".

Recording start: Set the recording start time in this field. The recording is then controlled automatically by the internal system clock to a specific time.

Recording length: Specify the recording length here. Using "Recording start" and "Recording length" you have the option of executing timed recordings even when away. "Recording" is armed as soon as Record has been activated and starts at the set time for the set length. If no length has been selected, recording continues until the hard disk is full and then stops automatically.

Record device: Select the sound card driver that should be used for recording in this dialog. If there is no entry here or a wrong one, then the card is not installed in Windows correctly. Use the "Info" button to find out the recording properties of the recording device.

File path: Specify the path to the files where the data should be recorded. The yellow folder button opens a query for selecting the path and file name.

Record standard output: Enable this option to record the standard output of the operating system, such as DVD sound, the browser or game audio playback. An actual digital recording is implemented, which means that the exact digital data delivered to the sound card driver by the playback software (e.g. player in browser) is stored.

To prevent this option from being inadvertently left active, it must be explicitly activated again for each such capture process.

Note: Some audio devices integrated into the mainboard also offer the entire audio output as a recording device, often called "Stereo Mix" or "What You Hear". This also works, but the result of such a recording is not a real digital recording, because the analog output signal is fed back to the analog input in the audio device, so a double conversion - from digital to analog and back - takes place.

Tip: Since this recording really records everything that is played back in the operating system, you should disable system sounds, website notifications and the like.

Further Recording Options

Monitoring mode:

> Please also read the chapter "System options -> Global audio options -> Monitoring settings (view page 596)" for more details.

Monitor: Use this button to activate the LED control displays. Please note that the recording device must be selected prior to beginning this. While recording, the LED control displays move slower, but they nevertheless show each maximum level.

Play while recording: Activate synchronous recording and playing here if this supported by your sound card.

> **Hint**: If you want to record and play over multiple sound cards, small differences may result during playback of longer passages. This is as a result of sample rates of cards which are not 100% synchronous. Ideally you should use the same card for recording as you do for playback if possible. If your card creates a delay between the start of recording and the start of playback, you can balance it out in the "Record Offset" field. To do so play a sample with a noticeable impulse and record it using a loop from the sound card's output to the sound card's input. Then zoom into the Arranger to such a depth that you can precisely pinpoint the offset value.

Create VIP object: Specify via this mode whether or not all recordings should be integrated automatically into a virtual project. The created objects are named with the specified object names.

Update object during recording: This object updates the graphical display of the object while recording.

Read CD/DAT marker: DAT devices and several professional CD players output digital marker information via an SPDIF output e. g. CD track markers or DAT markers). With this recording option all marker information is read from the soundcard's SPDIF input, provided the selected audio device supports it.

Pre-recording (ASIO only): This recording option function inserts audio material that you have added at the beginning of the recording to the beginning of the current recording.

> More information about pre-recording is available via the menu item "Program preferences > System/Audio > Record (view page 671)".

Recording offset (samples): If recordings exhibit a constant, undesired shift in relation to the existing audio material in the arrangement, set an offset here which can be used for positioning all recordings.

Advanced options: Detailed information about advanced options is available viaFile > Program preferences > System/Audio > Record (view page 671)".

Recording information: This area provides information about the last recording that was made, i.e. **Recording time** (length of the recording), **Remaining time** (remaining recording time), and **Drive space** (remaining storage space on the hard disk being used).

Visualizer: Opens or closes the visualization window.

Reset Pk.: Reset the visualization's peak hold display.

Record: Click this button to start the actual recording.

Hint: Please note that during active external synchronization as a "Slave", recording does not start immediately, but rather when the master starts.

Stop: Ends a running recording process.

Recording pause: Use this button to interrupt the recording process. The playback marker continues to run. Pressing "Recording pause" again restarts the recording from wherever you want it to continue.

Take: Each object or take number (Take1, Take2, etc.) is given a marker so that they can be found again quickly. If you press "**FS (False Start)**" while recording, the recording object will be split at the current position – the recording itself, however, will continue. This allows the first part of the take to be named "FS", e. g. Take02FS. The continuous take is called "Take02" in this example.

To **rename the current take**, enter the description in the take name field and then press the "**<-**" button.

The take manager (view page 201) is ideal for managing the individual takes.

Marker: While recording, the option is available to select a marker at the current position of the playback marker, e. g. to correct an erroneous passage later on. If you want to set a further marker, press the "**+**" button next to the "Comment" button. With the "**Comment**" button you can set and name additional markers. These will then appear in the marker bar but will not be recorded in the takes. The "**Auto**" button adds additional markers with numbers and names to your current take.

Close: Use this button to exit the recording options window.

Monitoring

Please also read the chapter "System options -> Global audio options -> Monitoring settings (view page 596)" for more details.

Move Play Cursor

to Beginning

This option positions the playback marker at the beginning of the project.

Shortcut: Home

to End

This option positions the playback marker at the end of the project.

Shortcut: End

to Range Start

Sets the start position of the playback marker at the beginning of the currently selected range.

to Range End

Positions the start position of the playback marker at the end of the currently selected range.

to Start of Section

This option positions the playback marker at the beginning of the visible section.

Left/Right Move in Page/Scroll Mode

Use these commands to move the playback marker to the right or left.

In "Page" mode, the playback marker moves across the screen to the end of the visible section; at that point, the window view shifts to the start of the following section.

In "Scroll" mode, the playback marker jumps to the middle of the section and remains there as the project moves through it.

Shortcuts:

Movement in "Page" mode:	Right arrow/left arrow
Movement in "Scroll" mode:	Alt + left/right arrow

Note: Once a range open is open, use the "left/right arrow" keys to control the beginning of the range.

Object Border Left

The playback marker jumps to the left to the next object edge in the selected track.

Shortcut: Ctrl + Q

Object Border Right

The playback marker jumps forward to the right to the next object edge in the selected track.

Shortcut: Ctrl + W

Marker Left

The playback marker jumps to the left to the next marker in the arranger window.

Shortcut: Alt + Q

Marker Right

The playback marker jumps forward to the right to the next marker in the arranger window.

Shortcut: Alt + W

Recall Last Stop Position

If the option "Stop at current position" has not been selected in "System options -> Playback", then the playback marker may be positioned at the last stop position with this command nevertheless.

Shortcut: Ctrl + Alt + ,

Repeat Last Position(s)

This command jumps the playback marker to a maximum of 5 previous stop positions.

Shortcut: Backspace

To peak value of all selected objects

This command sets the playback marker start position at the peak level of all selected objects.

Markers

You will find a number of functions for managing markers. The function keys and numbers can be used to easily save, name, and reopen marker positions.

Detailed information on working on ranges and markers can be found in the "Functions in the project window (view page 109)" chapter.

Marker with Name...

With the option "**Marker with name...**" you can add markers and name them.

Shortcut: ?

Marker with name and numbering...

In the "**Marker with names and numbering...**" dialog " you can freely number and name additional markers.

The "-1 sec" button drags the marker one second forward in the marker list.

Markers with Automatic Numbering

The command "**Markers with automatic numbering**" automatically creates markers with sequential numbers.

Keyboard Shortcut Shift + #

Place Marker at Recording Position

The function "**Marker on record position**" sets a marker at the current recording position during each recording.

Shortcut: Alt + ?

Defined markers are visible at the top border of the project window and can be moved with the mouse. The mouse pointer changes to a double arrow (<->).

Set markers 1 - 10

You can set **markers** at the current playback marker position by selecting a number between one and ten from the menu and entering the corresponding keyboard shortcut.

Shortcut: Shift + 1 ... 0

Jump to markers 1- 10

You can move the playback marker to **marker positions 1-10** by selecting 1 - 10 from the menu and using the corresponding keyboard shortcut.

Keyboard Shortcut: 1 ... 0

Marker on Range Borders

Use this function to set a start and end marker at the borders of a selected range.

Set Markers on Silence

This function automatically sets markers at positions of selected low-volume or silent audio objects. Open the dialog to specify a threshold and a minimum time for detection.

Min. time (ms): Set the minimum time threshold that must be exceeded in order to set a marker.

Threshold (dB): Specify the threshold in decibels here.

Note: When working in destructive wave editing mode, the height of the range in the wave project is set to the level of the threshold to make it easier to see.

Start number: Enter the marker number that is counted upwards to "Marker after silence" here.

Prefix: Enter additional characters/letters which will precede the "Markers after silence" here. This makes it possible to easily differentiate them from markers that may already exist.

Add time (ms): Enter the time range in order to move the "Markers after silence" forward here. The selected object is moved forward by exactly the amount of the time range and filled with silence.

Delete all markers with prefix: Deletes all markers that have prefixes.

Delete all markers: Deletes all markers in the project

Delete VIP objects: This option cuts the object at the set value and may then be deleted.

Marker list input

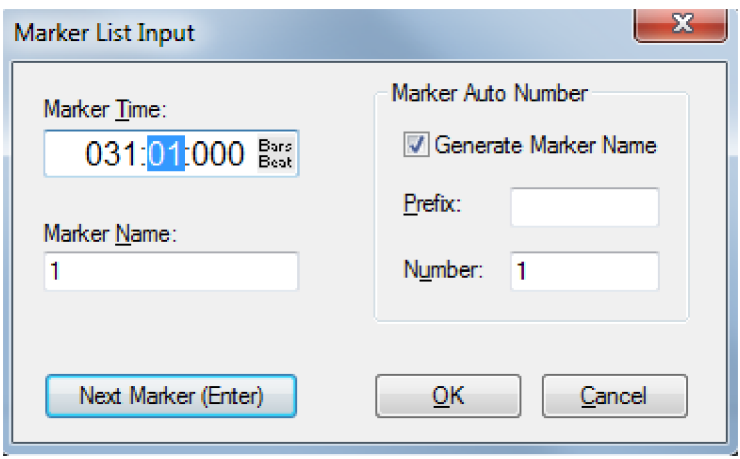
Shortcut: Ctrl + Shift + L

Markers can be quickly and efficiently defined by inputting their time position.

This method is useful if a marker list needs to be entered in Sequoia manually.

Steps:

- Enter the time position of the marker into the "Marker time" editing field. For example: for the time "001:02:010" enter "00102010"; in the editing field the numbers move automatically to the left, and colons do not have to be entered.
- If desired, you can change the specified marker name by pressing "Tab".
- "Next marker (enter)" moves to the next marker, or confirm your entry with "OK" and end the marker list entry.



A new marker name will be suggested for each new marker that you place in the marker list. The automatic marker numbering may be changed by the information in the input fields.

- The "**Prefix**" field displays the set prefix of the marker name, and the "**Number**" field contains the current marker counters, which are incremented by one with each new marker.
- By switching off the automatic marker counter "Add marker names", the set marker name will not be changed again and will be used as the name for future markers. The marker number counts upwards with each new marker that is added.

A specific marker manager is provided to help manage markers. Detailed information about the marker manager is provided in the chapter "Manager -> Marker manager".

Rename Marker

If a marker is selected in the marker list (by clicking it), then this function may be used to rename it.

Delete Marker

This deletes the marker at the current position when selected from the marker list by clicking its front edge.

Delete All markers

This function deletes all markers in the active project. The audio markers, however, remain in the wave objects.

Delete Markers in Range

All the markers in an activated range are deleted.

Set New Audio Marker

This command sets an audio marker at the current playback marker position of a selected audio object. If a marker was previously set at this position, then its name will be applied as an audio marker.

Copy Audio Marker to VIP Marker

This copies all available audio markers of the selected object to the marker bar in the arranger window. Application of the VIP markers only occurs within the object boundaries.

Copy VIP Marker to Audio Marker

This copies existing markers from the marker list inside the object borders to the corresponding position of the audio object in the wave project.

Audio marker manager

Audio markers are linked to the audio material, and are visible on the upper side of an audio object. The purpose of the audio marker is to mark positions within the audio

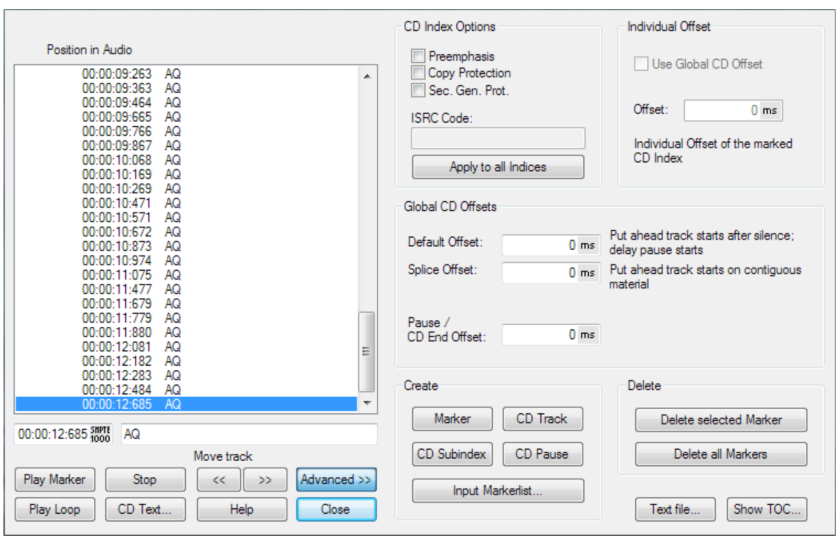
material, this marking will therefore remain regardless of the positioning in the virtual project.

Make Audio markers visible by marking the "Audio markers" checkbox in the view options (Shortcut: "Shift + Tab") in the "Objects" section.

The audio markers displayed in the object of the virtual project are identical to the markers in the corresponding Wave project. If you want to place a new project marker in a Wave project, as is done automatically when recording a new take, the audio markers are visible at the same position in the audio material in all related objects.

The following markers can be created and edited via the audio marker manager:

- By right clicking under the upper edge of an audio object in the virtual project a menu will appear, in which all audio markers are listed. You can go straight to them by clicking. In the same menu you can place new Audio markers at the playback marker's position or open the Audio marker manager.
- The position of the Audio marker selected in the Audio marker manager can be changed in the editing field. If you click there a double ended arrow will appear at the selected point, with this you can change the corresponding value. The value content can be chosen in the field to the right.



- An additional option for setting audio markers at the playback marker position is to click the "Advanced" button and then "Create" and "Marker".

Detailed information about the remaining fields in the "Advanced" dialog are available in "CD/DVD -> CD track/Index manager (view page 1110)".

Note: All time information in the audio manager and in the audio marker menu relate to time positions in the audio material, and not to positions in the virtual project.

Set Start Marker

This function sets the punch start marker (punch-in) to the beginning of a selected range.

Set End Marker

This command sets the punch end marker (punch-out) at the end of a selected range.

Delete Punch Markers

This command deletes the punch-in marker and the punch-out marker from the marker bar.

Additional Punch Start Marker

Adds another punch in marker.

Additional Punch End Marker

Adds another punch-out marker.

Delete Additional Punch Markers

This command deletes additional punch markers from the VIP.

Insert Tempo Change

Detailed information about setting tempo markers can be found in the chapter „Tempo Editing“ > „Tempo Map Dialog“ (view page 454).

Insert Time Signature Change

Detailed information about setting bar markers can be found in the chapter „Tempo Editing“ > „Tempo Map Dialog“ (view page 454).

Insert Grid Position Marker

Detailed information about setting grid position markers can be found in the chapter „Tempo Editing“ > „Grid Position Marker“ (view page 454).

Marker manager

The marker manager lists all of the markers contained in the current project and makes it possible to jump directly from the list to them or to play them.

To view the marker manager, click the "Manager" button in the toolbar and select the "Markers" tab.

Shortcut: Ctrl + Shift + Alt + M

Detailed information about this is can be found in the chapter "Manager -> Marker Manager (view page 196)".

Video Recording

The function "Video recording" enables you to record audio at the same time as the video track. Recording can be done on any available recording device present in the system. For example, a reference video track can be recorded in order to document additional stage sound for editing and mixing later.

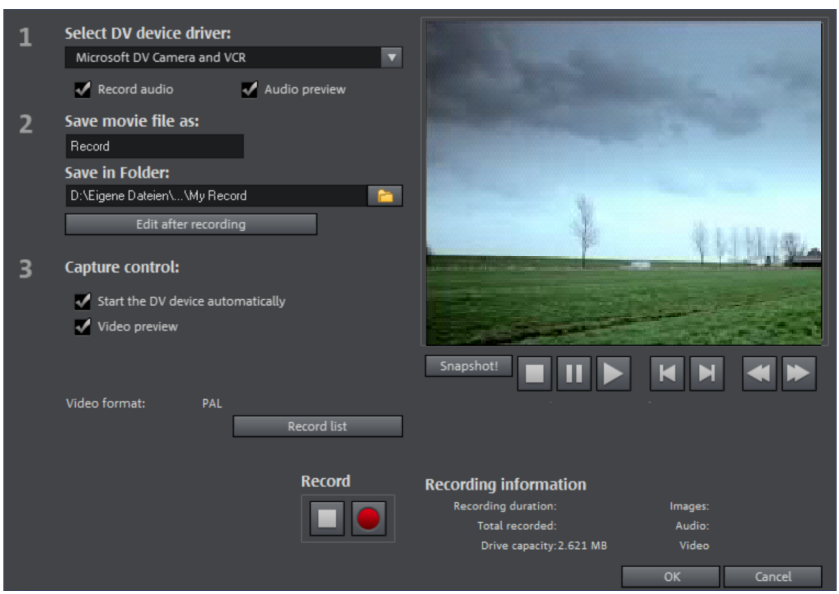
The following settings are available:

Video recording: Select this option if you want to make an analog video recording.



Display for recording analog video

DV recording: Select this option if you want to record a DV video.



Display for recording DV

Click the record button to start video recording. Sequoia shows important recording information during recording.

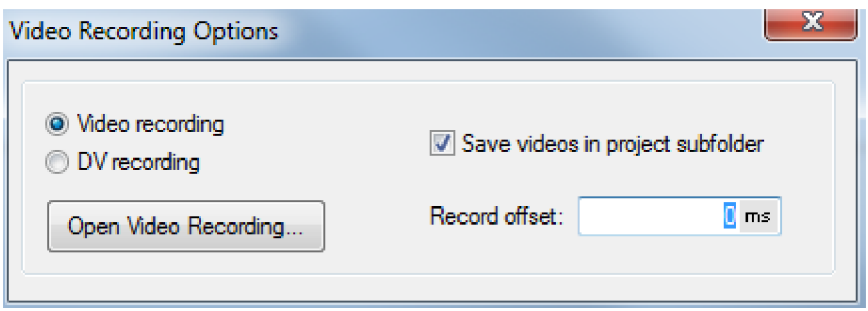
Save in the following folder: Enter the path and file name for the video file to be recorded.

If the "**Recording control > In an external program**" option is selected, recording will always take place in Sequoia. If this option is deactivated, the video recording will be saved only as a file in the indicated directory.

Note: Information about exporting videos can be found at "Export video (view page 627)" and "Export to MAGIX Video Pro X (view page 641)".

Video Recording Options

In the video recording options you can choose between recording devices.



Video recording: Connection to webcams or video capture sticks. You can also use professional video cards from Blackmagic, such as Intensity Pro, DeckLink (SDI, Studio, HD Extreme 3D).
DV recording: Connection to DV cameras or DV converters (such as Canopus ADVC-110) using Firewire connectors.

Here you can also set whether recordings should be saved in the project subfolder and determine record offset.

Retrospective Recording (MIDI Pre-Recording)

Sequoia creates a MIDI object in the recording-ready MIDI track that may be adjusted in terms of **buffer** length via "**System options -> MIDI**".

Retrospective recording intermittently saves MIDI events and audio signals in an adjustable memory buffer in RAM. This takes place while running, even if a recording is not being made via the record button. This enables creative moments to be captured and then integrated into the project as a file/object. In this case, it doesn't matter whether the project is playing or not.

The MIDI object may be added at the cursor position or synchronous to the last playback. A **prerecording** of 2 seconds is also saved during MIDI recording as an extra take. By changing the takes in the **take manager**, e.g. from "MIDI Take3" to "MIDI Take3 **PreRec**", you can drag the object to the left to restore the rhythm played prior to the actual recording. The object borders will be adjusted to exchange the take.

Note: If you would like to add an **audio recording** from the buffer into the project retroactively, drag the newly created object to the left after recording (according to the **set pre-recording time in the recording dialog of the system options**).

MIDI Record Mode

The MIDI recording modes specify how newly recorded MIDI data is inserted into the virtual project if there are already MIDI objects at the recording position.

Normal

In this recording mode, a new MIDI object is created over the existing object for each recording. All objects are maintained. This way, you can record multiple takes of a passage and then compare them to the object with "Alt + right click".

Overdub

In this mode, the data for each new MIDI recording is mixed with already existing takes.

Replace

In this mode, MIDI files of the existing object will be replaced with newly imported ones. If you record over several objects, these are combined into a single new object.

MIDI Panic – All Notes Off

This command sends a note off command to all MIDI devices that are activated in the MIDI options for all 128 notes on all 16 channels. The sustain (Controller 64) is also shut off, and the pitch wheel and modulation are reset to 0. All VSTis used in the project will also receive the all notes off command.

If MIDI tracks or objects exist in the project, the same function is also accessible by clicking the stop button in the transport console or toolbar when the project is stopped.

Automation menu

The content of the "Automation" menu corresponds to the Automation context menu of the selected track. For a description of the menu items, see chapter "Automation", section Automation context menu.

No effect (Track)

If you select this display option the track automations are deactivated.

Edit selected curve

Create

The curve generator (view page 470) creates a beat-based volume curve and pulses according to eighth, quarter, half, or full notes. You can enter a minimum and maximum value or define delay values as an option for the beat-based envelope curve calculation.

As an alternative, you can also let the curve generator write an envelope curve scan.

Invert

This command inverts the activated automation curves.

Thin Out

This function "thins out" the activated automation curve by reducing the number of automation events. During recording, the automation events are placed in very short intervals. The "Thin out" command reduces the number of curve points. The course of the automation is then displayed and reproduced more accurately.

Transfer Track Automation to Object

You can use this function to transfer volume and panorama automation to objects.

Convert object automation into track automation

With this function you can transfer the volume and panorama automation of objects to tracks.

Inactive

Use this command to deactivate the selected automation curve.

Curve Color...

This command displays a color palette to specify the color of the active curve.

Logarithmic

This function switches between the linear and the logarithmic display of the automation curve.

Copy, Paste, Delete

The active automation curve may also be copied, pasted into other tracks, and deleted.

Delete All Curves

This command deletes all automation curves in the corresponding track.

(Track) Automation Mode

Specify the track automation mode here. "**Copy to All Tacks**" applies the currently selected automation mode to all tracks/channels. Select "**Write Only Existing Curves**" to apply the automation data to existing automation curves only.

Detailed information about the automation modes is available in the chapter "Automation -> Automation modes (view page 458)".

Hide Automation

Use this option to deactivate the display of automation curves for all tracks.

Show Track Automation

Select this display option to display only the track automation.

Show Object Automation

If you select this display option, only the object automation will be displayed for all tracks.

Show Only Selected Curve

Views only the selected curves for the corresponding track. This helps to maintain a clear overview if several automation curves have been created.

Show all curves (not selectable)

The deselected automation curves are grayed-out. They cannot be activated with the mouse.

Show all curves (selectable)

The deselected curves are also shaded, but may now be activated with the mouse.

Select previous curve

With this you can select the last edited curve.

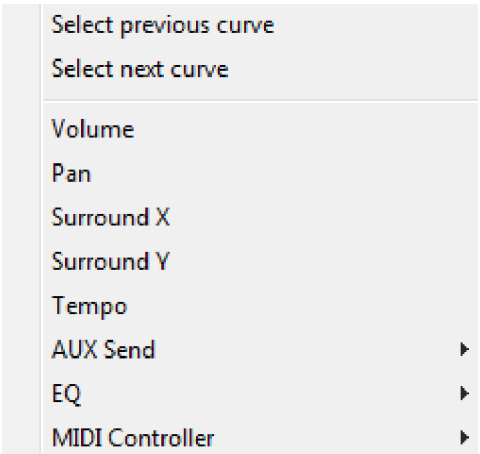
Select Next Curve

With this you can call up the next curve from the selection list.

The last section of the context menu lists the ranges for which automation parameters are available during automation of the corresponding track.

Select Automation Parameters

In the lower section of the "Automation" menu you can select various automation parameters and then control them with the automation controller in the track editor.



Effects Menu

This menu entry provides fast access to the relevant real-time effects at object level. Correspondingly, menu entries for all object effects are provided here.

Note: All effects opened using this menu are calculated destructively, provided the option "**Apply effects offline**" is active. The option to work with a copy is available in order to preserve the original audio material. The "**Create copy**" option is already selected in the corresponding dialog.

Effect Dialog Bar

In several effect dialogs you have direct access to the automation button which also provides access to the automation menu. This is made possible by a newly created header in the respective dialogs.



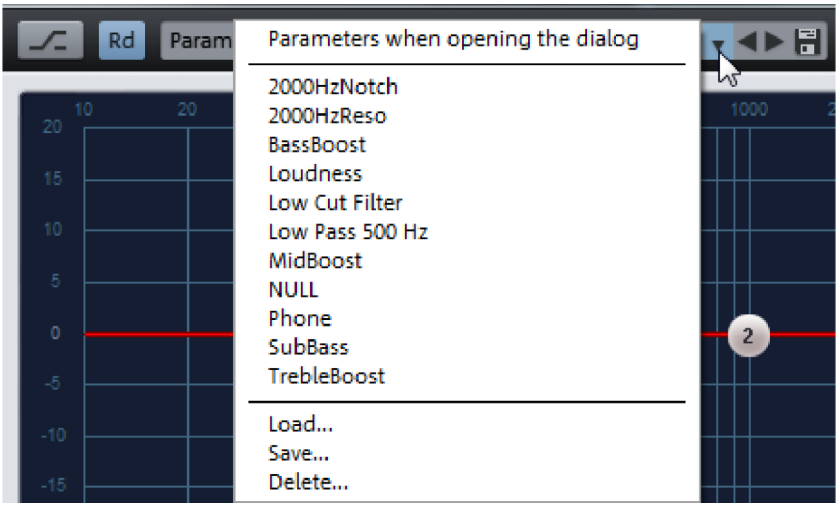
In the presets input field you can load, save or delete settings.

By selecting "Parameters when dialog is opened", you will undo all changes made in the dialog since it was opened.



By closing the dialog, you will apply all current settings.

Clicking on the downward pointing arrow, you will see all presets at a glance. Below the preset list you can open dialogs for loading additional presets and for deleting selected presets. You can navigate through presets using the two arrows pointing in horizontal directions. Current settings are saved to disk.



Here is an overview of the other control elements:



Bypass: The effect is removed from the signal route. This enables the unedited signal to be easily compared to the result of the current effect settings.



Mode: By clicking on this button you can toggle between the selected automation write/read mode. By right-clicking you can get to the Automation context menu.



A/B comparison:This control element is only displayed when it is supported by an effect. You can copy each selected setting using the arrow symbol to a different location. This way you can experiment with successful settings without losing them.



Reset: This control element is only displayed when it is supported by the respective effect. All parameters are reset to default settings.



Play button.



Solo button. This control element is only displayed when it is supported by the respective effect. The Solo mode will be activated for the corresponding effect.



Help: You can get extra information in the help section.

Notes concerning offline effects editing

- When editing objects via the context menu item "**Edit audio file**", offline effects always have an effect on all of the objects that reference the same audio material. If this is not desired, access the command "**Edit a copy of the audio file**" prior to destructive editing via the selected object's context menu. Sequoia will create a copy of the audio material and add this to the project folder. The corresponding object now refers to the copy that has been created. Following this, offline effects may be applied to the selected object without influencing other objects.
- Activate the option "**Create copy**" in the offline effects dialog to use the undo function later (this allows you to undo destructive editing).



- The following functions are available exclusively for audio files: "Sample Manipulation -> Sample Amount *2, /2", "Show/Hide", "To Zero" and "Build Physical Loop".

Note on the "Undo" function: Please tick "Undo active" via the "Options menu -> Program Preferences -> Undo definitions (view page 702)" for virtual projects and for audio files in order to activate the undo function.

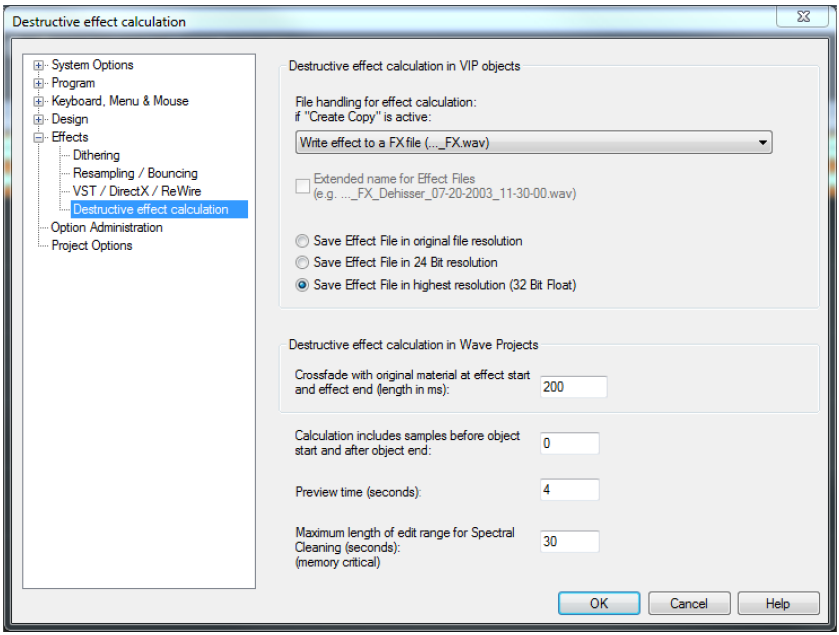
- Note that most functions only work within the selected range in the wave window. To use the functions on the entire audio file, the entire file must be selected (keyboard shortcut: "A"; menu item: "Range -> Range all").
- On the other hand, selected objects in VIP projects are changed independently of the selected range across the entire range (the only exception here is the "Get noise sample" function. (The "Get noise sample" function is an exception). If you do not wish to change the whole object, spit it at the range edges using the "T" key and activate "Auto-crossfade" mode to avoid crackling transitions.

OK: The edit is applied to the selected range of the sample or the selected object.

Cancel: Click this button to close the dialog window without applying any editing.

Advanced options for offline effect calculation

These settings may be opened in the corresponding effects dialog by clicking the "**Advanced options**" button. Alternatively, this dialog may be accessed via "File -> Program Preferences -> Destructive effect calculation...".



In this dialog, choose to write a copy with offline effects from the many options.

Destructive Effect Calculation in Wave Projects

If this is not desired, access the command "**Edit copy of wave project**" prior to destructive editing via the selected object's context menu. Sequoia will create a copy of the audio material and add this to the project folder. The selected object now refers to the copy that has been created. Following this, destructive effects may be applied to the selected objects without influencing other objects.

The dropdown menu of the dialog provides selection between three styles of saving when executing offline effects. Saving is required in order to be able to undo offline effects. A check must be placed beside "**Create copy**" in the effect dialogs for this to be available.

Note: Deselect this function if you're sure that the undo function will not be required. A lot of time and hard disk capacity can be saved if a copy of the wave is not created.

- **Append effect to original file**: The object with the calculated effect is added to the original file. However, this has some disadvantages: When working in an integer wave file, the effect is also saved at this resolution, which is not always desirable when working with 32-bit resolution. The length of the wave file is also changed, which may cause problems when working with looped objects or in various VIPs.
- **Write effect to an FX file (._FX.wav)**: The result of the effect calculation is written to a separate file with the extension "_FX.wav" so that the original wave project remains untouched. This also makes it possible to execute the effect calculation in the 32-bit float format, to save it, and thereby keeping the full quality of the effect. The file is not overwritten with every new effect calculation, but rather the new effect is simply added to the effect file.
- **Generate a new FX file for each effect**: All offline effects are saved in separate files with sequential numbers. Alternatively, these files may be given detailed names featuring effect descriptions and the date.

The effects file may also be saved in the format of the output file, in a 24-bit format, or in a 32-bit float format.

Offline effects calculation in wave projects

When opening a offline effect in wave projects, the result of the effect calculation is always added to the opened wave project. The options of the offline effect calculation on VIP objects described above are not taken into account.

Temporary files for the "Undo" function are only created with offline effect calculation if the "Undo" function for wave projects is also activated (Shortcut: "Y" – Program -> Undo) and "**Create copy**" is ticked in the corresponding effect dialog.

If an effect is applied to a specific range in the wave project, a crossfade may be inserted between the effect and the original at the beginning and end of this range. Enter the **length of the crossfade in milliseconds**.

Moreover, you can calculate a number of **additional samples before object start and after object end**.

Preview Time (seconds): Preview time is the duration calculated for listening to effects in the preview function.

Maximum length of editing range for Spectral Cleaning (seconds): Enter how long (maximum) the audio material should be edited during spectral cleaning.

Amplitude

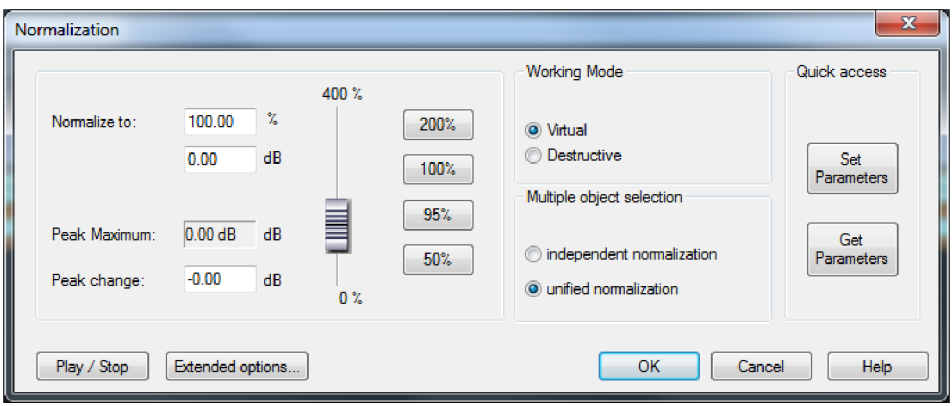
Normalize

Use this function 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. In this case, the maximum is first detected and calculated with the selected percentage value. All values are then weighted by the new factor.

Use this function to fully modulate samples or overmodulate targeted audio segments. Note that the noise level also rises during normalizing.

This function is especially useful when converting from high to lower bit rates. This guarantees that the otherwise low dynamic range is used to its full extent at lower rates.

Shortcut: Shift + N



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: Set the value which the audio material should be normalized to by entering it into the input field, moving the fader, or selecting one of the presets (50, 95, 100, or 200%). The value will be shown in % and dB (100% = 0 dB = max.). Values above 0 dB will cause digital clipping.

Level maximum: This indicates the highest resulting volume for the selected range or object in the corresponding setting.

Level change: Displays the level change in dB according to the selected normalize level and the detected level maximum.

Working Mode

Virtual: This function performs real-time normalizing on the selected objects. In this case, the sample data is not changed, but rather the object volume is adjusted so that the loudest part of the object reaches the selected normalization level. This virtual normalization is non-destructive, unlike destructive normalization, which changes the audio material irrevocably. The object to may be restored to its original volume setting by selecting "Reset", which is located below the volume fader in the object editor.

Destructive: The audio material is physically altered. A tick is set beside "Create copy" in the dialog.

Multiple Object Selection

When multiple objects are selected, there a number of different techniques to normalize them.

Independent normalization: Each object is normalized according to its own maximum level.

Uniform normalization: The maximum level is detected from all objects and each object is normalized according to that value. This is active by default.

Quick Access

Set parameters: Use this button to make the settings of the normalization function accessible via quick access.

Get parameters: Use this button to get the current quick access parameters in the dialog.

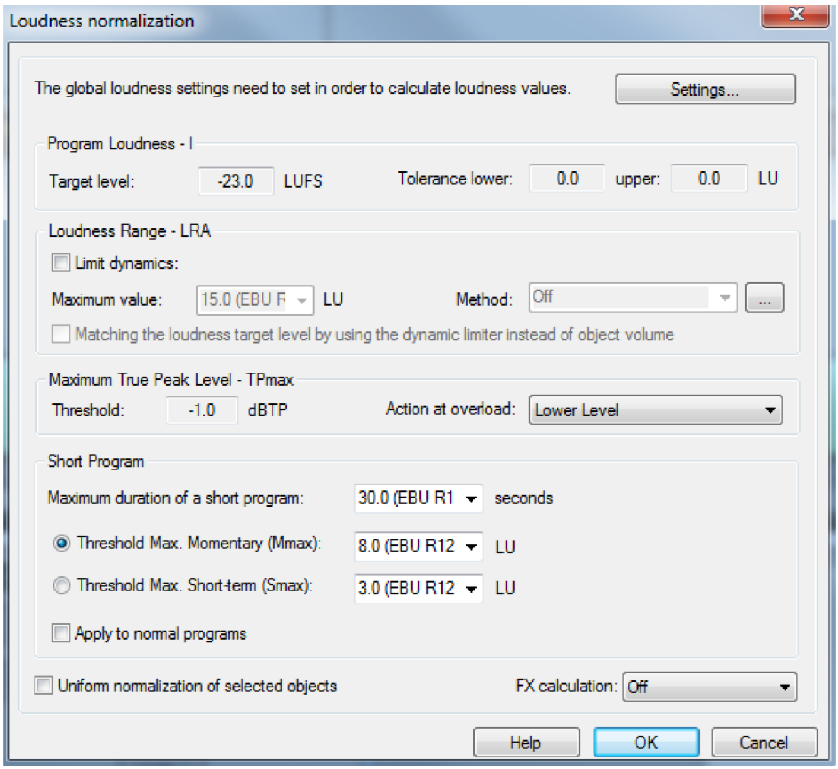
Normalize (Quick Access)

This function enables the specified settings in the "Normalize" dialog under "Quick access -> Set parameters" to be used immediately.

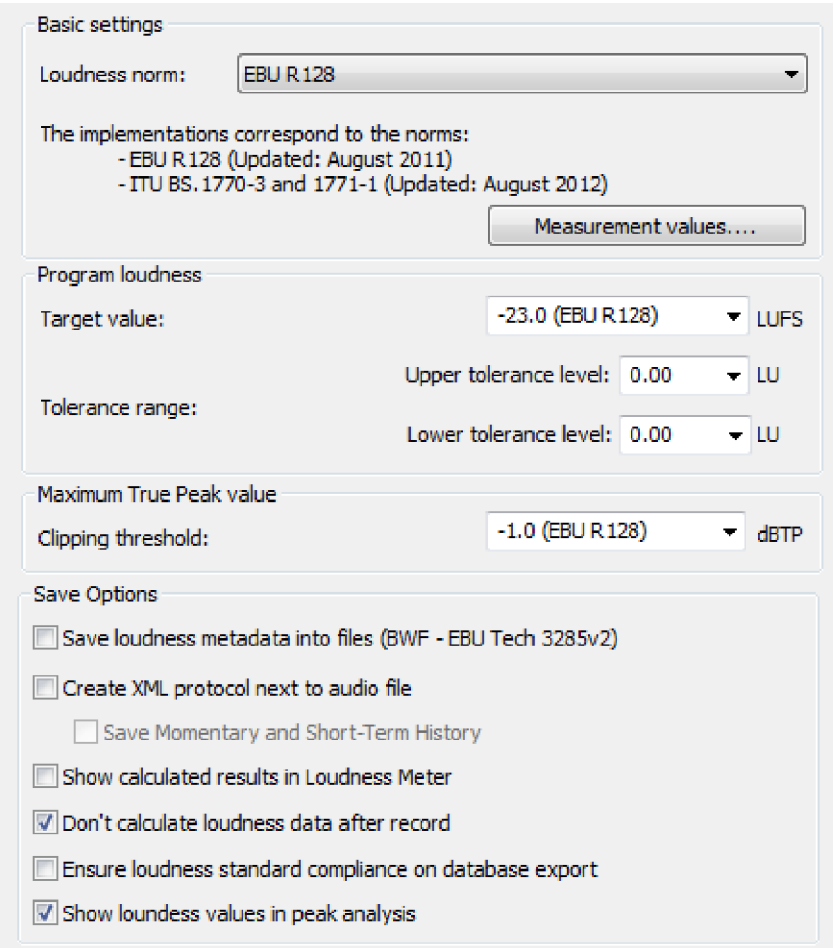
Shortcut: N

Loudness Normalization

Loudness normalization is performed in this dialog. First you can set the target value for loudness as well as the upper and lower tolerance values in LU.



The basis for calculation will be the values entered for **Program Loudness** that can be accessed using the "Settings" button.



For the **Loudness Norm** you can choose either

EBU R128 or

ITU-R BS.1770/1771

in their respective current versions or set a norm that is customized for your specific requirements.

If you don't want to work with one of the preset standards, additional parameters can be set using the Measurement Parameters (view page 1132) button (User Mode).

Here you can also set the **target value** in LUFS for the program loudness and specify the tolerance range.

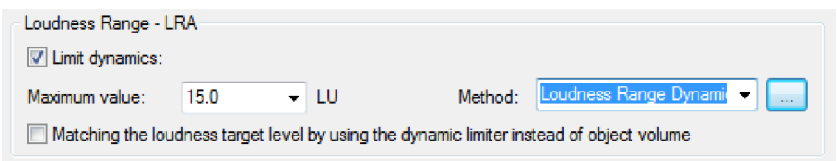
In User Mode the **peak threshold** (Maximum True Peak Level) can be adjusted with a selection or entry.

Save options

You have the following options:

- Write loudness values into files when saving (BWF - EBU Tech 3285v2).
- Create XML protocols and save parallel to file: With this option XML files are created during exporting, bouncing, batch processing or calculating loudness values. You can name these XML files like audio files, e. g. "file.wav.xml" or "file.mp3.xml". These files contain separate data for stereo and Surround streams.
- The loudness values for Max Momentary and Max Short-Term progressions can be saved as XML files with the additional option "Save Momentary and Short-Term Progression".
- Display Calculated Values in Open Loudness Meter
- Don't calculate loudness data after recording.
- Loudness normalization for export to database
- Display loudness values in level analysis.

Dynamic limiting (Loudness range normalization)



Dynamic limiting is a part of the dialog for loudness normalization and enables the

limiting of object dynamics in order to even out loudness variation within the audio material.

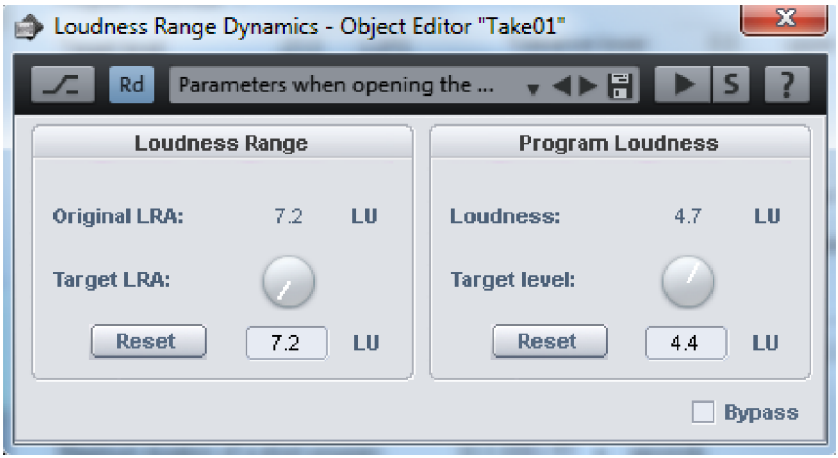
Activating the check box „**Limit dynamics**“ turns the algorithm on.

Set the „**Maximum value**“ for the dynamic range. The target value for the loudness range depends on the listening environment. For example, a loudness range of approximately 6 LU is good for a car whereas a setting of more than 20 LU can be used for a movie theater.

„Process“ provides a selection field where you can choose between the following three settings:

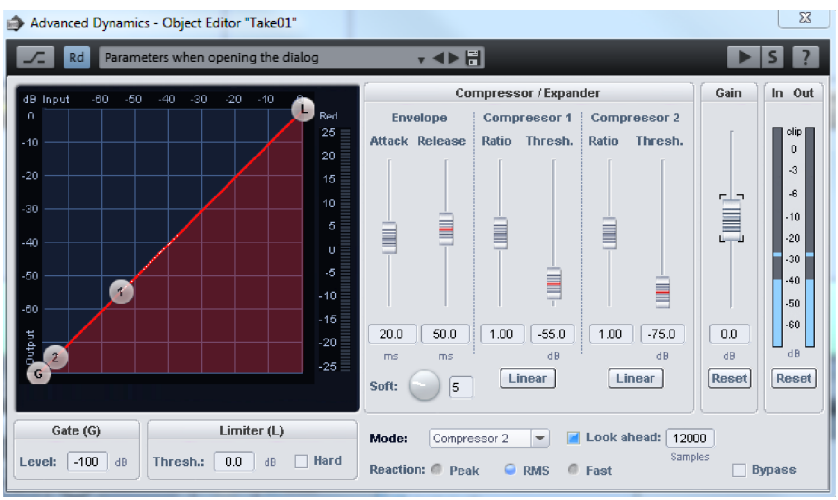
- Off
- Loudness Range Dynamics: This setting calculates an automation curve that dampens loud material and boosts quiet material. The automation points can

also be adjusted manually. The „Loudness Range Dynamics“ plug-in is inserted into the Object Editor (view page 165) and can be used to control the loudness range (LRA).



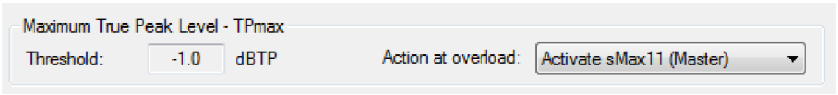
Loudness Range Dynamics in the Object Editor

- Leveler (Advanced Dynamics): The "Advanced Dynamics" plug-in is inserted into the Object Editor. Here the dynamics are automatically set to achieve the target loudness range.



The option "Matching of loudness target value using the dynamic limiter instead of the object volume" can be used to let the loudness target value be changed by the Leveler or Loudness Range Compressor. In this case the object volume is not changed.

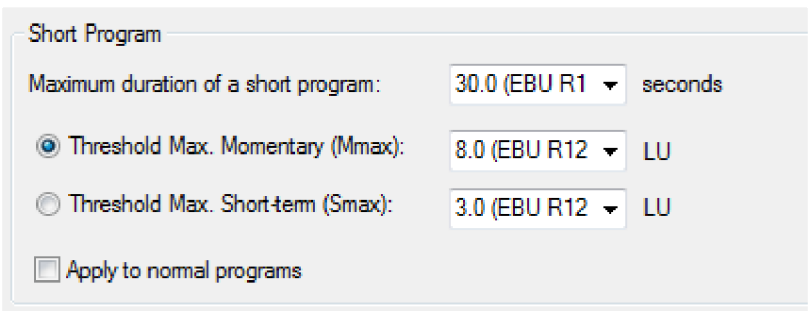
Action at overload



When the clipping threshold for the maximum peak level (Maximum True Peak Level) is exceeded, you have the following the options:

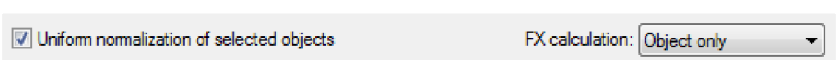
- Show warning
- Decrease level
- Activate sMax11 (Master)

Short programs



The term "short programs" is used in broadcasting to describe short pieces such as trailers or commercials.

- Maximum duration of a short program: Here you can select or enter the length of time for the short program. You can also shut the maximum duration off (0.0 Off).
- Threshold Max. Momentary (Mmax): Maximum value for the Momentary value (view page 1132) of the short program. If this value is exceeded, the loudness normalization will compensate by decreasing the loudness of the piece.
- Threshold Max. Short-term (Smax): Maximum value for the Short-term value (view page 1132) of the short program. If this value is exceeded, the loudness normalization will compensate by decreasing the loudness of the piece.
- Apply to normal programs: This option applies the Mmax and Smax settings globally and not just for short programs.



- Uniform normalization of selected objects: This option corrects all selected objects using the same value. This retains the level ratio between the objects. Here it is important to make sure that the loudest object influences the value for all the others. This can lead to other objects falling under the target program loudness.

- Effect calculation only for object: The object is bounced before the loudness analysis is performed. This means that the object effects are taken into account during the calculation of the loudness value.
- Effect calculation including master: The object, track and master are bounced. This means that all of the settings are included during the loudness calculation.

Fade In/Out...

Use this function to show or hide selected ranges of samples in a wave project. The amplitude will change according to the progression from the start value at the beginning of the range to the end value and the end of the range.

A simple fade-in process may be activated in the dialog via the parameters "Fade start in % = 0", "Fade end in % = 100". A simple fade-out may be executed by setting "Fade start in % = 100", "Fade end in % = 0".

You can have the transition display in three different curve envelopes: exponential, linear, or logarithmic.

Note that in virtual projects, powerful real-time fade and crossfade options are available that usually make the use of destructive fade routines unnecessary.

Set Zero

The level in the selected range is set to zero. This allows unwanted parts within a sample to be removed.

Dynamics Effects

What compressors are available in Sequoia?

There are four different modules for dynamic editing available in Sequoia:

1. **Essential FX Compressor (view page 985):** A simple but effective tool for reducing dynamics with soft curve characteristics and an adaptive regulation process. The "essentialFX Compressor" is a very musical compressor.

2. **Advanced Dynamics (view page 869):** This module is a very comprehensive tool. It is a combination of a classic dynamics module (compressor/expander/gate) and an approximating limiter. The result is an undistorted, optimally modulated signal with defined volume. Setting the parameter is done either by direct input or by graphically changing the characteristic. Level detection can be derived in either "**Peak**", "**RMS**", or "**Fast**" mode.

3. **Multiband Dynamics (view page 875):** This module offers extensive dynamics editing options. The advantage of dynamics manipulation across **multiple frequency bands** versus a standard process is that the danger of pumping and other side-effects sinks drastically. For instance, it can prevent a level peak in the bass range from overpowering the entire signal. Multiband technology also lets you specifically edit individual frequency ranges.

4. Sequoia's **AM-Track**, **AM-Pulse**, **AM-Phibia**, and **AM-Munition** also provide specific dynamics editing functions that simulate analog effects processors. Read more about this in the extensive chapter "MAGIX plug-ins (view page 977)".

5. **sMax11 (view page 883):** With this module you can increase the volume of the audio signal by setting an input amplification (Gain-In). Essentially, this involves a hard or brickwall limiter with input amplification.

6. Essential FX Limiter (view page 989): This plug-in 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.

Which compressor should be used and when?

When to use a certain dynamics module depends on how specific the change needs to be. The "essential FX Compressor" module works well on mixer channels to quickly and efficiently influence the dynamics.

However, the sound of an object should normally be changed very specifically. In this case, the dynamics of an instrument can be varied between the verse and the chorus depending on the playback method. This is a desired relation that can be preserved or even forced using precise compressor settings. We recommend using "Advanced Dynamics" to change the dynamic behavior of the source material as optimally as possible.

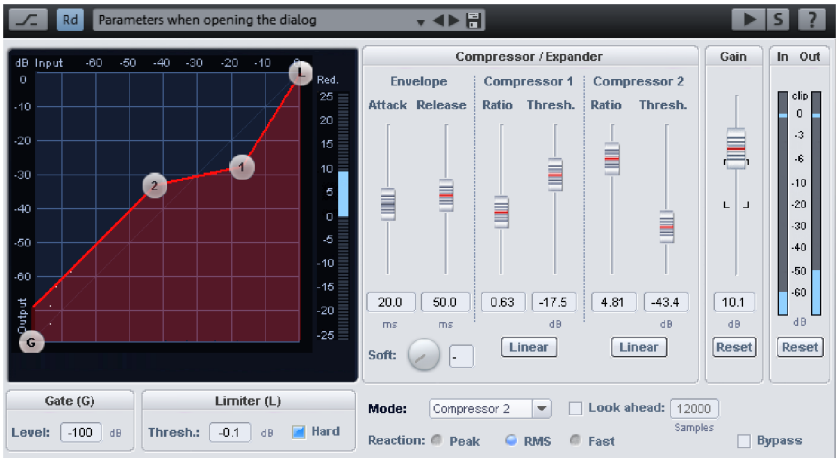
Experiment with "AM-Track", "AM-Pulse", "AM-Phibia", and "AM-Munition" to add characteristic pre-amps or tube sounds to your arrangements.

The master module in the mixer features "Multiband Dynamics" to give the entire mix its final balance. Because an increase in volume using a multiband compressor can lead to clipping, it is necessary to use a limiter to suppress the clipping samples.

We recommend using "Multiband Dynamics" for each object/CD track when mastering CDs. This enables you to react to the differences of CD tracks with different multiband dynamics settings.

The "sMax and the "Essential FX Limiter" are maximizer/limiters that increase the loudness of the audio signal.

Advanced Dynamics

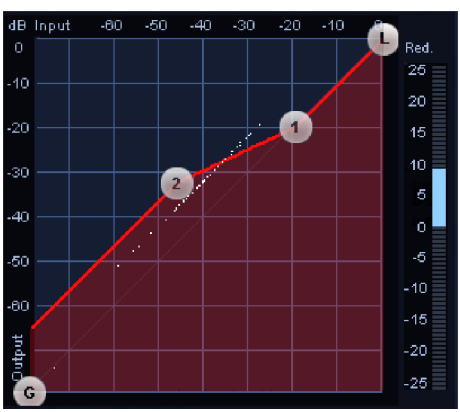


Advanced Dynamics - general functional elements and presets

Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.dy2**.

You can also load all dynamics presets (*.dyn). The parameters are converted to those available in the "Advanced dynamics" module. If saved from within the Advanced Dynamics processor, the preset will be saved with the *.dy2 extension.

Dynamic scope: When you play the selected object/sample, you'll see the in and out level of the signal as a white, scattered line.



Reduction display: To the right of the graphic display you'll see the level reduction of the played audio material.

In/Out displays: These meters show the input and output displays in dB.

Reset: Reset the reduction, in and out displays.

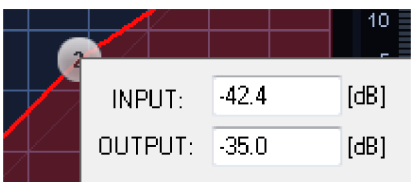
Mode: Modes are typical applications of "Advanced Dynamics". They constitute default settings that makes the graphical editing of the reference line easier. For this, the number of editable parameters are limited by the choice of mode, so that e.g. with pure limiter application, the other curve bases no longer change. The following modes are available to you:

- Simple compressor
- Two-point compressor
- Expander
- Gate & limiter
- Limiter

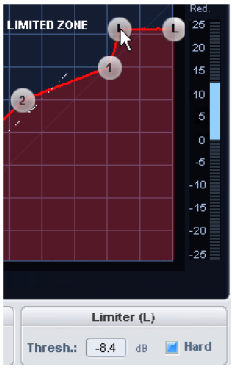
Advanced Dynamics - editing of reference curves (graphically)

To graphically edit the static transfer characteristic, press the left mouse key in the reference line field on one of the curve bases and move it with the key held down. Moving the base points at one end influences the other points of the corresponding parameters.

In order to directly position the point in the reference line field, press the right mouse key. A small dialog appears, where you can read and write your input and output values.



If you turn on the "Hard" parameter in the limiter and drag the curve point for the limiter threshold down, a "Limited zone" label will appear for the area below the limiter threshold. The limiter pushes the levels below the "Limited zone".



To hide a handle, adjust the corresponding threshold to 0dB using the slider; the reference line will not change in this case. In order to make it visible again, the threshold must be below the limiter threshold.

Advanced Dynamics - editing the reference line (parametric)



Gate (G)

Level [in dB] determines the minimum level at the input. All signals below this level will have an output signal level of 0 dB.

Limiter (L)

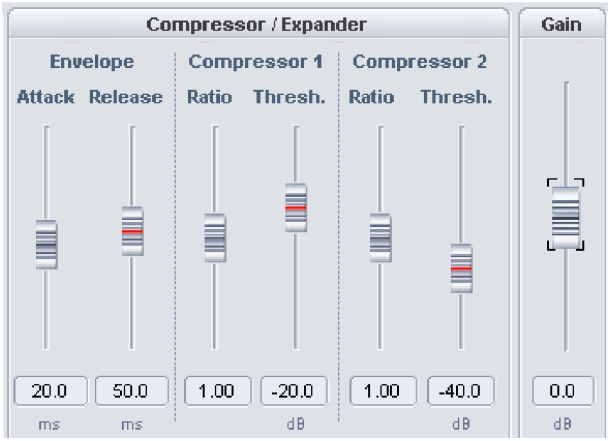
Level [in dB] determines the maximum level at the output. Note the "Hard" setting.

Hard: When you switch on this parameter, the output signal limits exactly to the set threshold value. This means that no sample lies above the set limiter level. Of course, the signal is not simply cut off at this limit. The used algorithm brings the signal as close as possible to the limit without changing the original sound. If the "Hard" setting is not active, the algorithm only controls according to the set transfer reference line and behaves like an analog limiter.

Soft: This parameter specifies the curve of the reference line in values from 0 to 20, whereby 0 means no curve. Curving is useful if the continuous change between uncompressed and compressed signals becomes apparent, i. e. the signal level alternates around the inflection point. With "Soft", a smooth transition is achieved.



Advanced Dynamics - envelope



The time constants affect the sound characteristic significantly. In this way, specific time constants may lead to distortion or "pumping" effects.

Attack: Here you can set the time frame between the crossing of the threshold and the maximum extent of the effect in milliseconds.

Release: Here you can set the time frame between the falling off of the threshold and the complete dissipation of the effect in milliseconds.

Compressor 1/2 threshold: Here you can set the input threshold in dB, above which the respective component is activated.

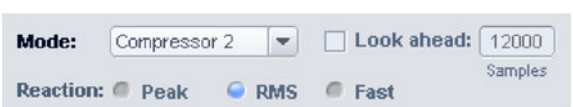
Compressor 1/2 ratio: With this parameter you can control the compression ratio above the threshold. The "Linear" button resets the ratio to 1:1.

Gain: Use this fader to set the whole static reference line in dB.

All changes to these parameters directly influence the static transfer characteristic line. However, you should note that all these parameters influence each other.

Advanced Dynamics - Dynamic parameter

The "**Reaction**" function is applied to the source material similar to the compressor.



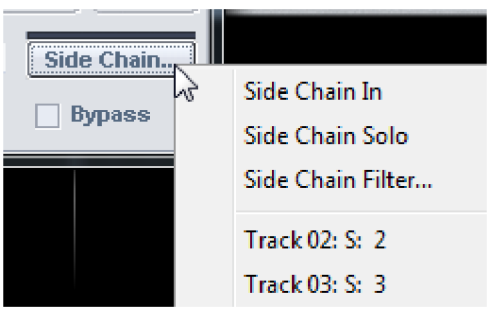
Peak: This mode uses the signal's peaks as a basis for calculation. The dynamics module reacts to level peaks quickly and efficiently.

RMS (Root Mean Square): This mode is involves the signal's average volume, which corresponds with the dynamic behavior of analog dynamics modules. It sounds relatively round and balanced, less focused. The time constant for signal detection of the control signal only refers to the setting of the "Attack" fader. Both time constants are analyzed for signal equalization when manipulating signal dynamics.

Fast: Use this option if the object dynamics should be modified only a little. The maximum level at the output is never higher than the limiter.

Preview: If this switch is activated, the dynamics section works in preview mode. This can help prevent artifacts (pumping) and overmodulations. On the other hand, without a preview, this option will not automatically "level" steep attack phases in the waveform, which may result in sharper-sounding audio material. To emulate the sound of analog dynamic modules, it is recommendable to work without preview. The length of the preview may be entered in the "Samples" field.

Side chain...: If "**Advanced dynamics**" is used as a track or master effect and the track is not a Surround master, the "side chain" button will appear. Enabling this option is used to request an external signal to calculate the internal control signal.



Clicking the "side chain" button opens the side chain menu.

Side chain in: Activates and deactivates the side chain function.

Sidechain sole: With this option you can listen to the sidechain signal. The compressor will be switched to "Bypass", and only the input of the side chain will be played. When closing the plug-in dialog, the "Side chain solo" function is automatically reset. "Sidechain solo" is also ideal for acoustic control when applying the sidechain filter.

In the lower half of the menu all tracks are listed, of which you can set one or more as sidechain inputs. **In the AUX section of tracks that used as sidechain signals or mixer**

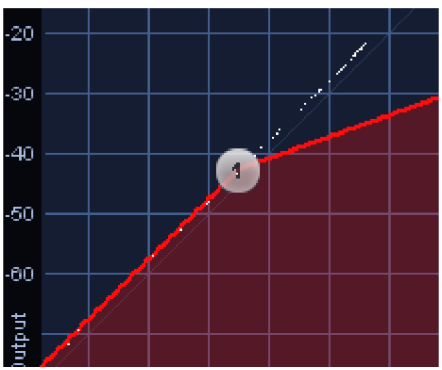
channels a new AUX send fader will be assigned for the target track. Internally, a side chain bus (AUX bus) is created. To differentiate from the "normal" AUX buses, a ">" is added. "Pre-fader" is set to catch the side chain signal. This means that the sidechain remains independent of the fader position of the sending channel.

Right clicking on the AUX send button selects this as "Side chain send". Thereby you can use further track as the source for the sidechain signal at a later date.

Sidechain filter: The sidechain filter can be filtered by a parametric equalizer.

Advanced dynamics - Dynamic scope

Dynamic scope appears when you play the selected object/sample in an open dialog. You can then see a scattered white line, a "snow cloud" whose individual points represent the input and output values in dB. This way, the change to the samples that is achieved via advanced dynamics is constantly visualized.



You can clearly see the way dynamic scopes work in the following example:

Select "ZERO" and press "Play;, now the effect won't change the signal. The scope shows all samples on red lines and this means this means that all input values equal to the output values. Slide the gain fader upwards to see that the "snow cloud" follows the reference line. If you select very high time constants, this takes place relatively slowly.

Selecting a relatively short attack time (10 ms), a long release time (400 ms) and a severely arched reference line can also move the effect towards "pumping". The "restlessness" of this dynamic scope mirrors the "pumping" graphically.

Advanced Dynamics – tips and tricks

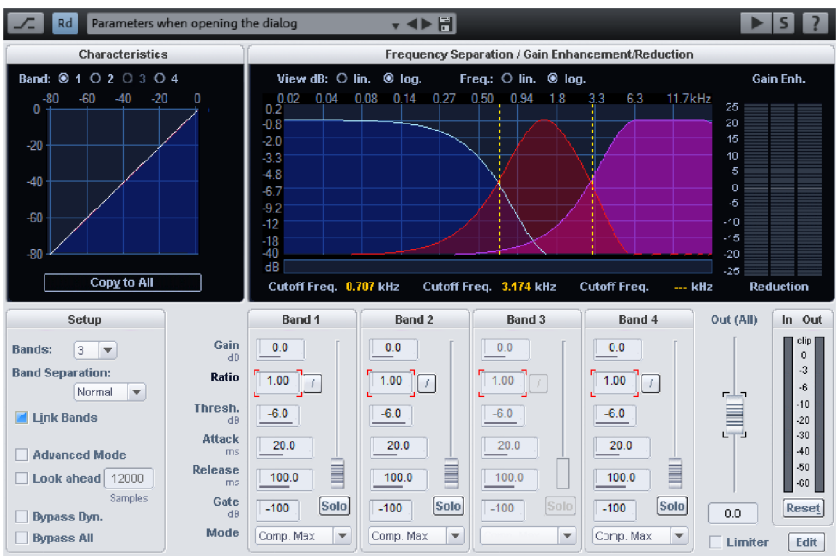
- Select one of the provided presets as a general setting. You can now adapt the reference line to the source material with the help of the dynamic scope and very precisely separate the signal components.

- If the scattered line of the dynamic scope moves very jerkily, this is probably because of the settings of the time constants. Set these according to the sound preferences and then save the personalized presets.
- In order to couple an "analog" limiter with the hard limiter algorithm, define the Limiter level at the maximum desired sample level. Once set, an "analog" limiter may be set by using an extreme ratio setting (compressor 10.00 or expander 0.10).

Multiband Dynamics

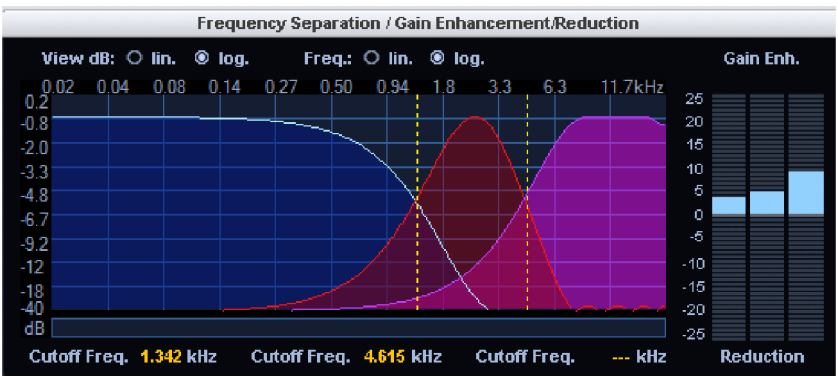
The "Multiband Dynamics" section lets you edit the **dynamics with up to four independent frequency bands**. The entire signal is split into individual frequency bands in this case. The dynamics are then edited separately for each band.

The combination of the individual bands is 100% phase-neutral, and thanks to the FIR Complement Filter, technique no discoloration of the frequency response occurs. In other words: If there is no dynamic editing in the individual bands, i.e. the signal is only split into the bands and then reassembled, the audio material will be perfectly compiled to the state it was before. The advantage of dynamics manipulation across multiple frequency bands versus a standard process is that the danger of pumping and other side-effects sinks drastically. For instance, the function is able to prevent bass peaks from reducing the entire signal. Multiband technology also lets you specifically edit individual frequency ranges.

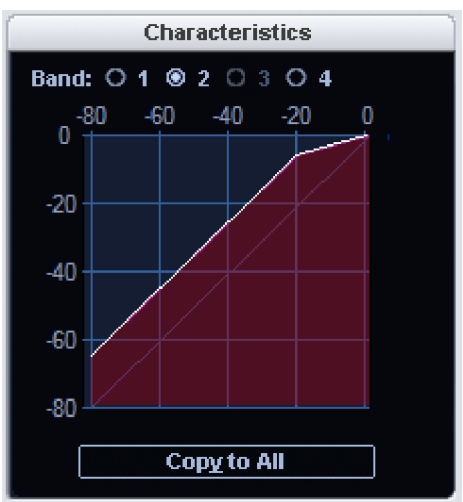


Multiband Dynamics - overview

In the "**Frequency separation / Gain Enhancement/Reduction**" dialog, you'll find the parameters for the filter bank. The graphic display shows the frequency of separate bands. The left axis label shows damping in -dB, the top label shows frequency in kHz. The individual curves are color labeled.



The graphic element to the side shows the **dynamics characteristic curve**. It illustrates the ratio from the input level (upper axis in dB) to the output level (left axis in dB).



The reference curve for the band currently selected is always displayed.

The dynamics section of the individual bands is equipped with the following parameters: "Gain", "Ratio", "Threshold", "Attack", "Release, "Gate Level" as well as the modes "Compr. Max", "Compressor", "Limiter", "Limiter 100", "Expander", and "Gate".

Multiband Dynamics - general controls

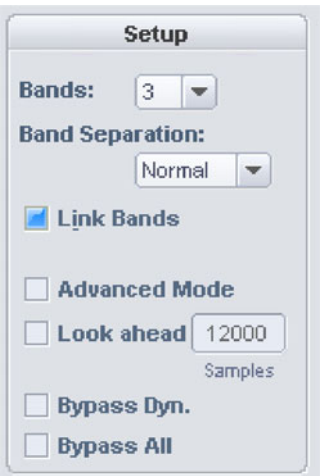
Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.mdy**

Reference line

Band selection: Here you can select a band for dynamic editing. The reference curve of the currently selected band is displayed in white, while the other bands appear as lines in their corresponding color.

Copy to all bands: After pressing this button the parameters of the chosen band, i.e. the parameters whose values appear on the screen, are applied to all bands. If you also have "Link bands" selected, the values for all bands remain coupled to one another.

Setup



Bands: Here you can select the number of bands, between 1 and 4. Set to "1", no separation into different frequencies takes place, and the algorithm works like the standard dynamic section. The algorithm CPU load rises with increasing number of bands.

Band separation: See below

Link bands: When this option is active, the changes to the dynamic parameters affect all bands. Often, it isn't necessary to set dynamic parameters for each band separately. Especially with rough settings, it often makes sense to first adjust all parameters together.

Advanced Mode: When this option is selected, the same internal processing routines are set as in the "Advanced Dynamics" module.

Important: If you have activated "Link bands", it does not mean that all parameters in all bands will have the same values. Only those parameters are the same which were changed after "Link bands" was selected. If all bands should feature the same settings, click "Copy to all bands".

Preview: When this option is selected, the dynamics section works in preview mode. Without preview, artifacts (pumping) and overmodulations can appear. On the other hand, without a preview, this option will not automatically "level" steep attack phases in the waveform, which may result in sharper-sounding audio material. This setting affects all bands.

Bypass dyn. (bypass dynamics):

Dynamic editing of separate bands is removed from the signal route. This function serves to compare the editing results with the original.

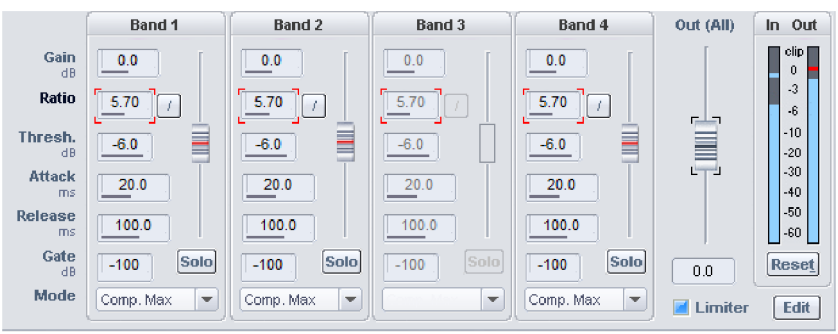
The difference to the "Bypass all" switch is that with "Bypass Dyn." in solo mode, individual bands can be compared separately, with or without dynamic editing as the filter bank is still activated.

Bypass all:

Here, the complete multiband dynamic section is removed form the signal path. Since the entire filter bank is "bridged", Solo mode can no longer be used. The gain knob for the sum is also no longer active.

Multiband Dynamics - set up and parameters

Parameters for the dynamic sections



Note: Depending on the parameter you select by clicking on the corresponding field (gain, ratio, threshold, attack, release or gate), the fader to the left of the input fields controls the corresponding parameters.

Gain (dB): Here you can set the level at the output end of the dynamics phase for each band separately. Please note that the graphic display can cannot be fully display the influence of this control. If you set the controller higher, clipping may occur - In this case the reference line has to be continued upwards out of the display border.

Ratio (factor): This parameter controls the strength of the corresponding effect, 1.0 means that there is no effect. Clicking the button beside it resets the ratio to 1.00.

Threshold (-dB): This is the input threshold value above and below which the corresponding effect is applied.

Attack (ms): Here you can set the time frame between the crossing of the threshold and the maximum extent of the effect.

Release (ms): Here you can set the time frame between the falling off of the threshold and the complete dissipation of the effect.

Gate Level (-dB): This parameter specifies below which volume the level should be set to 0.

Solo mode: In "Solo" mode the frequency bands can be monitored one at a time. This function particularly alleviates adjusting the filter parameters. For instance, you can locate a critical frequency range of a mix before editing the dynamics.

Mode

Comp. Max.: Dynamics of the frequency band is limited in such a way that loud passages above a certain threshold are amplified less, raising the overall volume. These settings are often used to add assertiveness to audio material or to increase the loudness. Among other things, this type of dynamics editing effect is referred to as the "Maximizer" or "Loudness enhancer". Use the "Ratio" input field to set the degree of compression via the "Threshold". You can also influence the "Attack" and "Release" via the corresponding input fields.

Compressor: The effect on dynamics is similar to a classic compressor: The dynamics of a frequency band is limited in that loud passages above a certain threshold are limited in their amplification. Use this settings if you want to achieve sound alteration through compression without raising the overall volume. The level of compression is adjusted by the "Ratio" controller; the activation threshold is determined by "Threshold". "Attack" and "Release" influence the build-up and decay times. A gain control is added to the compressor to compensate for volume reduction.

Expander: The dynamics of a frequency band are increased, loud sequences remain loud, quiet sequences become even quieter. Dynamics expansion is often used for recording speech with a high noise level. The expansion causes the level of the speech to be raised while the noise is suppressed. Please note that there is also a powerful DeNoising algorithm (view page 939) provided.

Gate: Here, very quiet passages below the Threshold Level are dampened or set to zero. This way, you can effectively suppress noise during the pauses between takes. Even at high compression (ratio > 5), the gate function is still useful to avoid strong increases in the quietest of passages and background noise. If different threshold values are entered in different bands, with a bit of skill, drum loops may be "chopped up".

Limiter: Only the loudest passages are limited (above the threshold). Quiet passages remain unchanged. Limiters are used to reduce the occurrence of big level peaks without reducing the master dynamics.

Limiter 100%: Performs the same editing as the Limiter, but the level is immediately raised to 0 dB - this corresponds to a subsequent normalization. This corresponds to a subsequent normalization.

Hint: If you want to use the Limiter as protection against overmodulation, you should take into account that the Limiter can only reliably prevent overmodulation in a single band. If the bands are mixed together, the sum can again result in overstepping the threshold level set for the bands.

Out (All)

Here you can set the overall output of the entire algorithm. This parameter cannot be set separately for each band. The graphic display does not reflect this setting. Use the "Volume" fader to balance the volume difference via the dynamics process.

Limiter On/ Edit:

Here you can add or edit a peak limiter. It prevents overmodulations at the output during volume increases when a multiband dynamics section is used.

Separation frequencies

Click on the graphic display of "**Frequency separation/Level increase/reduction**" in order to change the separation frequencies for individual filter bands by dragging with the mouse. The number of separation frequencies depends on the number of selected bands ("Number of bands" parameter). The border frequencies for the deepest and the highest band (high-cut and low-cut) are shown. The border frequencies are those frequencies where the filter dampening is -3 dB. For mid bands (band pass 1 and band

pass 2) the mid frequencies and bandwidths are displayed. The bandwidth in this case is the distance between the two separation frequencies. These also correspond to the cut points of the neighboring frequency curves.

Band separation (in the setup area)

A higher setting for "Band separation" ("Normal", "High") has a similar influence on the various properties of the filter, whereby the "precision" also increases:

1. The edge steepness of the filter curves increases and the transition range between two bands decreases.
2. Dampening in the stop band increases (low: ca. 25-35 dB, normal: ca. 35-45 dB, high ca. 55-75 dB).
3. The ripple of the frequency range of the bands decreases. However, this is not a problem as the ripple of the individual filter bands compensate each other when put together thanks to the complementary filter technology. In any case, the output signal does not contain ripples.
4. The latency caused by the effect increases.

Multiband Dynamics – filter bank for experts

Those who know a little about the program will be curious why the setting/display of values doesn't appear as it does in the parametric equalizer. The answer: in this case, a different digital filter type is used whose frequency response cannot be fully written using these parameters. While analog filters and "normal" digital filters always contain a logarithmic level drop (the drop can, for example, which is set to -12 dB per octave), the filters used in this case have a linear frequency-response curve, i.e. the level drop in dB increases in size the further the distance from the separation frequency. This method also has the advantage of 100% phase accuracy, among other things. The display of the mid frequencies is derived from the "traditional" filter parameters and this way simplifies "acclimatization".

Tips & tricks: strategies to manage the flood of parameters

A multiband dynamics processor (naturally) contains many parameters.

If you need to alter the dynamics of all bands, rather than one individual band, the following approach may be useful:

1. Global presets for all bands

1. Select the mode that corresponds the most with your needs. If you wish to increase the volume of the audio material, select "Compr. max.". If you wish to improve sound characteristics, increase the transparency, improve the clarity of speech, freshen up older recordings or create more powerful bass without increasing the volume, use the "Compressor". The selection of this mode only applies to the selected band.

2. Click "Copy to all bands". Selecting this mode ensures that all bands feature the applied setting.
3. Activate "Link bands". Any changes you perform in the current band are automatically mirrored on the other bands.
4. Change the parameters of all bands until the audio sounds right.

2. Fine-tuning of individual bands

1. Turn off "Link bands".
2. Activate "Solo" mode for a band. Individual bands may now be isolated, making the task of optimizing settings for each band easier. The optimum settings for the higher frequency bands have lower values because the waveforms are shorter here.
3. If you can't find satisfactory settings for some of the bands, try to change the split frequency setting for the selected band. A narrow band setting may help prevent "pumping" artifacts in the sound.

To edit a specific and critical frequency range only, a different approach may be taken:

1. Turn off "Link bands".
2. Activate "Solo" mode for the band that contains the critical frequency range.
3. Change the separation frequencies for the band so that they can effectively filter out the critical area. Now the dynamic editing can begin. First, select a suitable mode.
4. "Limiter" mode or "Compressor" mode are suitable for the dynamic limiting of critical frequencies, i.e. for sibilant sounds.
5. Use the "Bypass dyn." checkbox to compare the processed and unprocessed bands.
6. Now turn off the "Play solo" option and compare the original and edited signals using "Bypass all".

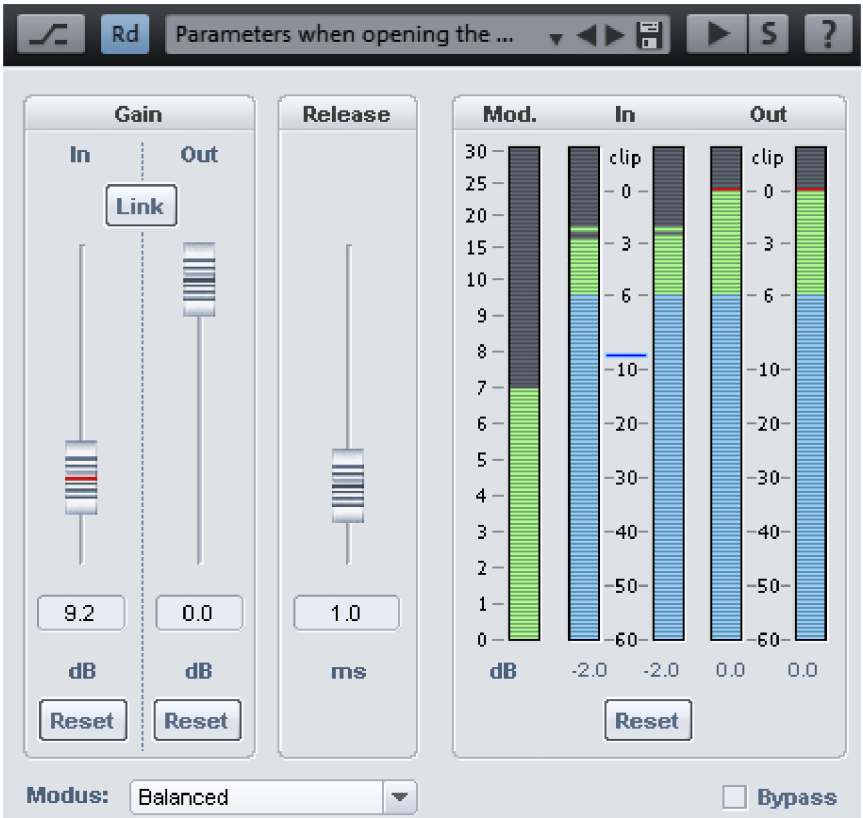
A third option is to first compare the supplied presets and then use the presets as a starting point for searching for optimum settings.

In most cases, even loading a relevant preset will produce good results. But if you really want the very best sound or volume increase result, you'll often have to apply these settings manually.

Comparing at constant volume levels

Use this method to compare the sound differences without allowing the dynamics processing to alter the volume levels. Use the "Out (all)" option to adjust the volume until the audio signal level matches that of the bypassed level, i.e. when the "Bypass all" option is used. If the multiband compressor is used in the "Master" area of the mixer, the peak meter of the mixer may also be used as support.

sMax11



The sMax11 maximizer provides a tool for increasing the loudness of the audio signal.

This is done by entering the input amplitude (**gain-in**). The signal will be amplified by this level. The sMax11 simultaneously ensures that the signal doesn't exceed the output level (**gain-out**) that has been set. This requires regulation of the response time set via the **mode** and **release** time. Essentially, this involves a hard or brickwall limiter with input amplification.

A typical workflow can go something like this:

- First, turn on the **Link** option. This way, volume will stay constant and you can better compare your changes.
- Now, raise the input amplification **Gain in** until the sound changes become unacceptable.
- Adjust **Gain in** to a lower level.
- You can also minimize distortions by raising the **Release** time. However, the compression effect and the volume increase are thereby minimized.
- Switch **Bypass** on once in a while.

- When you've found the optimal setting, switch the **Link** option off.
- Finally, switch the **Output gain** to 0 dB.

sMax11 - General controls and presets

Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.max**.

Play/Stop: This button activates/stops the real-time preview function. This lets you check the acoustics of the respective filter settings.

? (Help): Opens the help file for this effects dialog.

sMax11 - graphical displays

Mod. display: This meter indicates the modification of the signal in dB. The portion of the input amplitude is not included. Only the level with the non-linear changes (required for catching level peaks) will be displayed.

In/out displays: These meters display the input and output displays in dB.

Reset (meter): This button resets the meter displays.

sMax11 - Special controls

Gain-in: This value specifies how much the signal should be increased (maximal).

Reset (Gain-In): Resets the "Gain-in" parameter.

Gain-out: This value sets the maximum output level.

Reset (Gain-Out): Resets the "Gain-out" parameter.

Link: This option connects both gain values.

Note: If the link option is active, then "Gain-in" may be increased, for example, without increasing the output volume. This makes it easier to recognize distortions that appear as a result of a gain-in value set too high.

Release: Set the time frame for the complete dissipation of the effect on the signal in seconds.

Mode: The setting of the mode influences the effect's control behavior: Select the mode most appropriate for your application:

- **Balanced**: Lowest level of distortion with transparent sound, robust to use Appropriate for example for speech recordings.

- **Fast**: Addresses slightly faster than "Balanced". The mode corresponds to the behavior of the hard limiter in the advanced dynamics compressor.
- **Aggressive**: Very short attack time, suitable for especially sharp transients for this reason In signals with dominant percussive components, the sharpness of the beats is preserved. Distortions are masked by the beat.
- **Hard clipper**: Controls are shut off in this mode. Signal peaks are simply cut off, which may lead to significant distortions. This mode is appropriate for example to accentuate the transients of individual beats even stronger.

sMax11 - True Peak Maximizer

Only in Sequoia/Samplitude Suite:

True Peak: The maximum peak level (Maximum True Peak Level) can be selected as a **Clipping Threshold** that is set in the Loudness Normalization (view page 862). In this way the sMax11 can be expanded to include the True Peak function as defined by the ITU-R BS.1770. Inter-sample peaks are also taken into account during the limiting of the signal so that the reconstruction of the audio material at the analog output does not clip or distort.

efx_Compressor

The **(essential) eFX Compressor (view page 985)** is a simple, efficient tool for reducing dynamics with soft characteristics and adaptive control processing. It is a very musical compressor.

eFX_Gate

The eFX_Gate (view page 988) is a classic "analog" gate, which addresses the signal quickly and accurately while avoiding the typical raw artifacts of digital devices.

eFX_Limiter

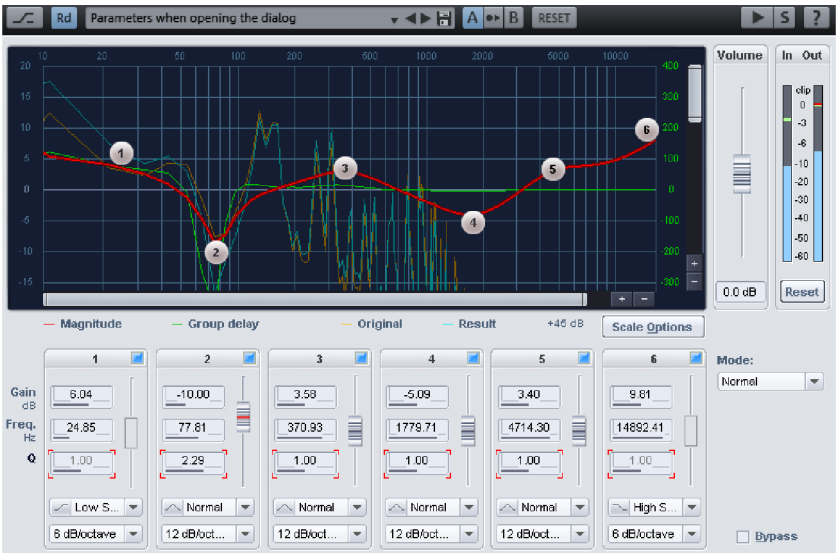
The eFX_Limiter (view page 989) is a simple yet effective tool for increasing the loudness of an audio signal. This creates a compressed but loud signal without allowing the defined output volume to be exceeded.

AM-Munition, AM-Track, AM-Phibia, AM-Pulse

More information is available in "MAGIX plug-ins (view page 977)".

Frequency/Filters

EQ116



EQ116 is a 6-channel fully parametric equalizer for audio signal frequency response.

EQ116 - Graphic displays

Filter graphic: This graphic provides a view of a number frequency-related details:

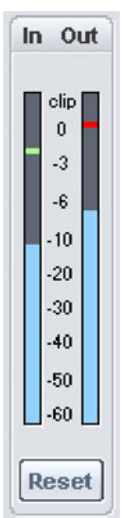
- **Amplitude response:** The resulting amplitude envelope from the individual band settings
- **Group runtime/Phase response:** One of these envelopes may be shown optionally. Toggle between these in the scale options dialog. The group delay describes the frequency-dependent time delay in passing the signal whilst the phase response shows the dependence of the phase of the frequency.
- **Original:** This curve displays the original frequency response.
- **Edited:** This curve shows the frequency response after processing by the effect.

The display of the filter graphic may be adjusted by accessing the "**Scale options**" and by using the vertical and horizontal scroll bars.



Hide real-time curves: This option hides the "Original" and "Edited" real-time curves.

Real-time offset: Adjust the offset value in dB for the real-time curves for optimal display. Click in the display window after every change to the values to make the adjustment visible.

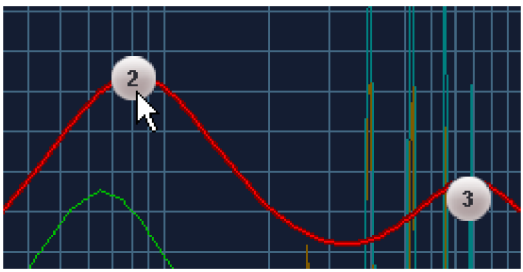


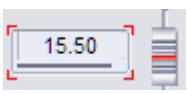
In/Out displays: These meters show the input and output displays in dB.

Reset (meter): Reset the in and out displays.

EQ116 - Editing the reference line

Filter graphic: The band filter parameters gain and frequency may be changed within the graphic by clicking and moving the numbered level points. By turning the mouse wheel you can change the bandwidth (Q) of a band.

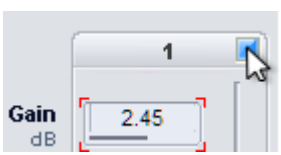




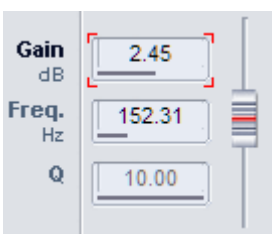
You can change the value of a parameter in the input field by vertical dragging of the mouse or by entering numerical values.

Depending on the parameter you select by clicking in the input field (gain, frequency, or Q factor), the fader to the left of the input fields controls the corresponding parameters. You can also use the mouse wheel to regulate the selected parameter. If you hold the "Shift" key, the values change in smaller increments.

Volume: Use this controller to adjust the master volume if the level rises or sinks too much while filtering.



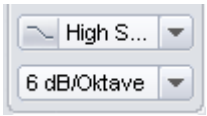
Band on/off: In the header for each band group, there is an on/off switch for turning the corresponding band on or off. The associated level point will disappear from the filter graphic accordingly. The corresponding base point will then disappear in the filter graphic.



Gain: These controllers allow you to raise or lower the filter. Setting the controller to 0 deactivates the filter and thereby uses no CPU power.

Freq. (frequency): The frequency controllers set the cut-off frequency of the individual filters between 10 Hz and 24 kHz. You can freely choose the frequency enables multiple filters to be set to the same frequency giving a greater effect.

Q (bandwidth): Set the bandwidth of the individual filters between 0.10 (very wide bandwidth) and 10 (extremely narrow bandwidth).

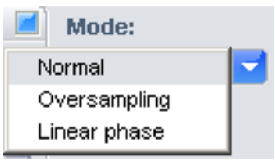


Type: Set the filter type here. These include **normal (= peak)**, **low shelving**, **high shelving**, **high pass**, and **low pass**

Slew rate (gradient): This influences the internal process of a filter band and thereby the slew rate of the high or low pass in the lockout zone. The settings are not equally available to all filter types. For **high pass** and **low pass**, a slew rate may be selected between 6 dB/octave, 12 dB/octave, 24 dB/octave, or 36 dB/octave. For **low shelving** and **high shelving**, select between 6 dB/octave and 12 dB/octave. The filter type "**Normal (= peak)**" is fixed at 12 dB/octave.

EQ116 - Internal operation modes

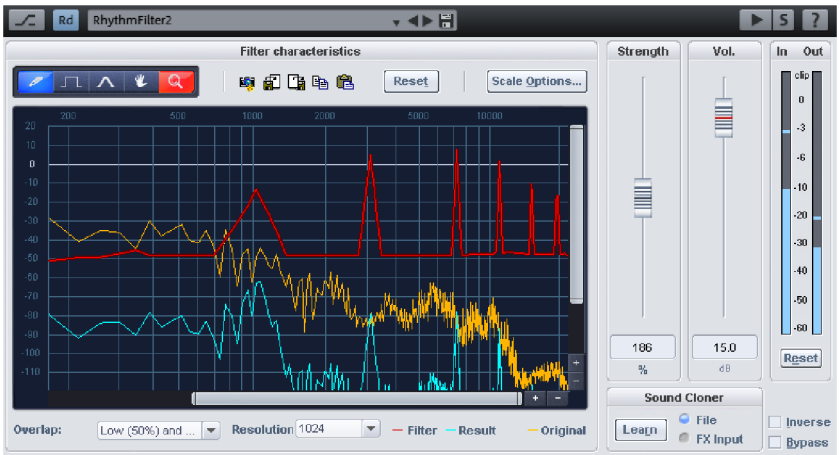
Mode: Set the internal working mode of the EQ116 here.



The following modes are supported:

- **Normal:** Equalizer functions in normal operating mode, which corresponds to a minimum phase EQ. The mode corresponds with the parametric 4-band equalizer found in older versions of Sequoia, and therefore represents a compatibility mode.
- **Oversampling**: This effect works internally compared to the first mode and with a higher rate where the filter formation takes place. Especially for higher frequencies, a more precise frequency response can be reached, and fewer aliasing artifacts arise. The disadvantage is that this mode requires more computing power.
- **Linear phase:** Equalizer works in such a way as to eliminate frequency-dependent phase displacements. This process is completely different than in the first two modes. The linear phase mode sounds more neutral and less colorful, and the influence on the amplitude of the overall signal is weaker because no amplification and extinguishing effects appear as a result of phase offset. However, it does command higher computing power. In addition, for impulse-heavy material with high frequencies a special artefact, called "pre-ringing", can take place.

FFT Filter



The FFT filter (for real time use in the object editor, in the track or master section) allows you to precisely filter linear signals.

Hint: The filter has a bigger drawing area when used as an offline effect ((Menu Effects > Process effects offline).

Edit the red filter curves by drawing with the pencil tools in the graphical display. If you let the object play along, you will immediately hear how the sound of the edited object.

The graphical view shows 3 curves that are integrated in a coordinates system. This includes the vertical axis, which shows the volume in dB, and the horizontal axis, which shows the frequency in Hertz (the pitch).

The **yellow curve** represents the **original frequency response**, and the **blue one** represents the **corrected frequency band** that accompanies the original frequency response after the **red filter curve** is applied.

The following tools are available via the object FFT filter:



Pen for free-hand drawing: This tool allows you to draw the filter curve. Hold the "Shift" key down to draw straight lines.

Pen for quantized drawing: Use this tool to draw stepped filter curves; the quantization steps depend upon the set resolution. Hold the "Shift" key down to draw straight lines.

Bend tool for curving lines: Click with this tool above or below the curve to bend the curve/frequency area. The further away from the line you click with the bend tool, the sharper the curve will be. This effect can be intensified by simultaneously holding down the "Ctrl" key.

Navigation tool: Use the navigation tool to move the visible selection vertically and horizontally while zoomed in.

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

Besides the FFT filter tools, the following buttons are also available:



Snapshot: If this feature is activated, the filter's current input and output spectrum will be preserved in the graphical interface.

Save: This command saves the filter curve as a text file.

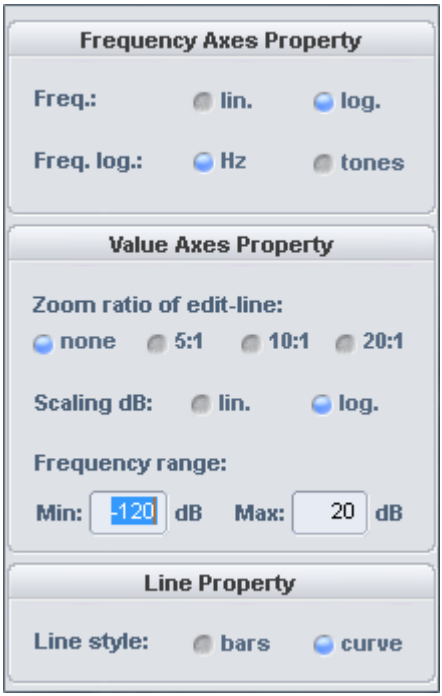
Load: This command loads a filter curve from a text file.

Copy: This command copies the filter curve to the clipboard (keyboard shortcut: Ctrl + C).

Insert: Use this command to insert the filter curve from the clipboard into the chosen FFT filter of the selected object; this replaces the current filter curve in the graphical interface.

Reset: This button resets all curves for the FFT Filter.

Scaling options: Set the graphical display of the curve value here.



Frequency axes property: The frequency can be output as a linear curve or as a logarithmic curve. If the logarithmic display is selected, then the display's x-axis can be output either according to frequency or according to pitch.

Zoom ration of edit line: Here you can stretch the filter curve on the y-axis, either 5:1, 10:1 or 20:1. For the dB values choose either the linear or the logarithmic display. The range of values can be defined freely. Simply enter the desired minimum and maximum values into the corresponding fields.

Line style: Display the curve roughly as a stepped bar or as a soft curve.

Level: This controller compresses or stretches the filter curve. Set the level of adjustment for the transmitted frequency response between 1 and 200%.

Vol.: This controller adjusts the total volume of the filter curve in dB, and thereby the "Out" value of the level display.

FFT filter/Sound Cloner functions

With the Sound Cloner feature you can determine the sound characteristic of a selected object and transfer it to another one. For instance, if individual songs of a music collection are different to others, the sound cloner feature can adapt them to

other pieces. An 80's hit could be treated with the characteristic sound of the late 60s, for example.

First of all, determine the spectral image of the object that should be used as an audio template. Save this as a "clone". The clone may now be applied to the target audio material.

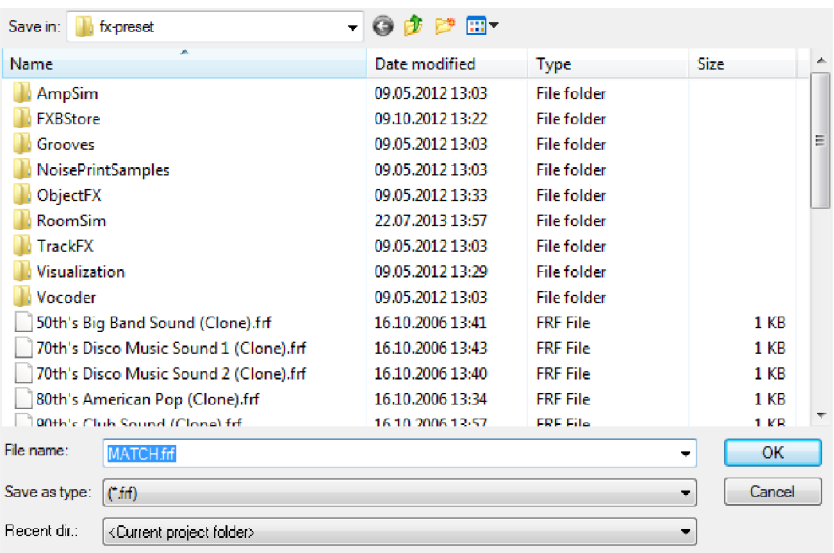
The sound cloner's preset menu also contains a few useful filter settings besides the "clone" presets.

1. The learning process

Load a song with a sound you would like to use it as a reference. Place the play cursor at a location in the song where the elements of the sound to be cloned appear. Don't select the intro or a break. All of the instruments involved should be playing at this point. Press the "Learn" button.

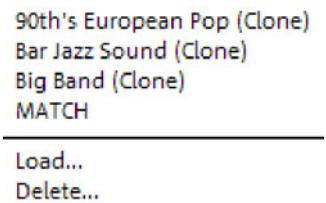


The options "File" or "FX input" specify whether the sound clone should originate from the original wave file or from the object that has been processed with effects. The sound characteristic of the reference song is now created as a fingerprint. Save the sound clone that has been created in the "FX presets" folder.



2. Transfer frequency response

Now select the frequency response of the object that should be adjusted and open the FFT filter. By loading a **match preset from the preset menu**, the sound characteristic will be communicated to the sound at the position of the play cursor.



The filter curve is calculated via the target audio material's spectral analysis, which provides an audio image that is similar to the audio image in the preset. If used as an object effect, the target material to be analyzed is taken from the associated object, in which case the play cursor specifies the position inside the object. The "Level" controller regulates the intensity of the sound transmission.



When loading the sound clone, it's important to place the play cursor at a location in the object where all of the instruments are playing; the sound cloner uses the audio material at the play cursor as the basis for calculation of the filter settings. The selected filter curve will not be transferred directly into the FFT filter; initially, a spectral analysis of the target audio material is executed at the play cursor before the selected object's filter curve is adjusted. By default, 20 seconds will be analyzed, but the length of the area to be analyzed can be adjusted by dragging with the mouse on the range in the arranger.

FFT filter/Sound cloner controls

Learn: Calculates a frequency analysis at the playback position. A dialog opens in which the result can be saved as a special match preset.

File: The sound clone's reference is created using the selected object. This produces a spectral image of the object. Next, the Sound Cloner preset can be saved and then appears at the bottom of the presets list. Real-time effects are ignored in this mode when defining the reference.

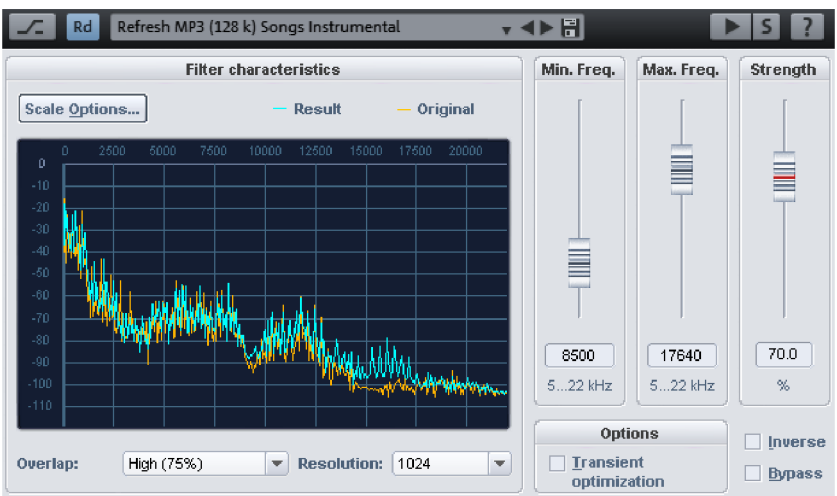
FX input: The input at the effect is used as a reference. Realtime effects looped before the FFT filter which is used as a cloner are accounted for in this mode when defining the reference.



Inverse: Inverts the filter curve. This function inverts the sample data within the selected range along the amplitude axis, i. e. negative values become positive and vice-versa.

Bypass: The algorithm is removed from the signal route. In this way, the unedited signal can be compared to the result of the algorithm.

Brilliance Enhancer

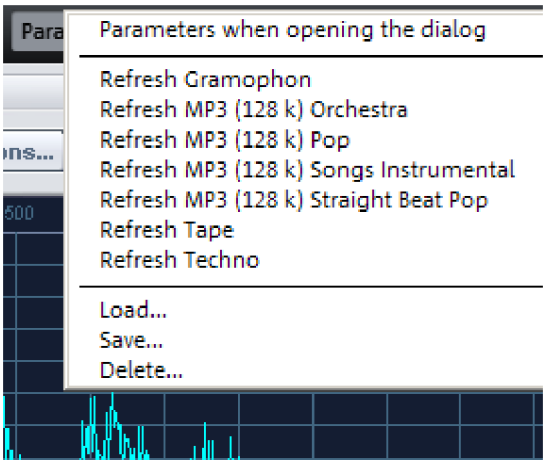


The Brilliance Enhancer is a tool which helps compensate the loss of higher frequencies to provide your audio material with a glossy shine.

This loss often occurs during MP3 compression or tape recording. In contrast to an equalizer, which only raises existing frequencies, the Brilliance Enhancer recalculates higher frequencies from the remaining signal to markedly refresh the sound.

You can use the brilliance enhance in pop or rock as a mastering effect in case the sound's transparency and brilliance need to be increased.

Brilliance Enhancer – Presets



Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.bre**.

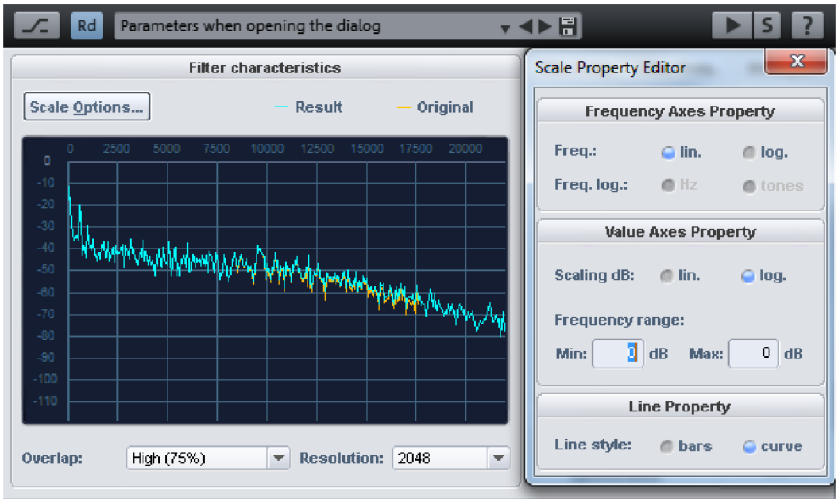
Brilliance Enhancer – fine settings

You will see two curves in the graphic display:

- The orange curve is the original signal
- The blue curve is the edited signal

Scaling options: This button opens the "Scaling options" dialog, which can be used to adjust the signal display to your needs.

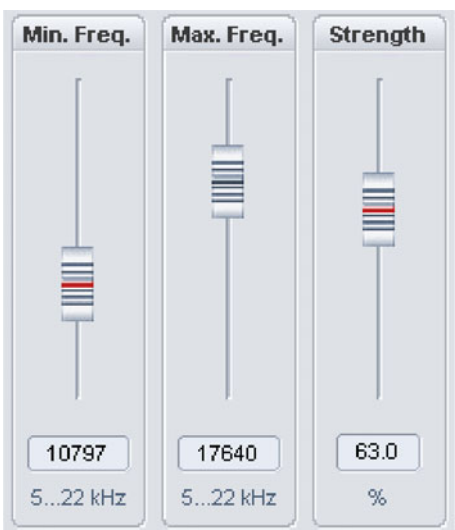
Value/curve display: With the help of input and output spectrum you can decide how the scaling options have an effect on the display during playback or monitoring.



Overlapping: This internal parameter changes the overlapping of the time window for the modulator signal spectrum calculation. Higher values improve the result, but also increase CPU load.

Rate: This parameter sets the internal rate used by the algorithm. You may wish to experiment with this parameter because a higher resolution does not always guarantee the best results.

Brilliance Enhancer – parameters and controls

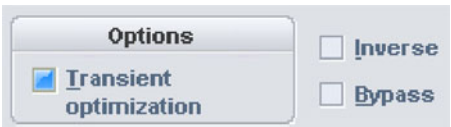


Minimum frequency: Here you can set the lower limit for the frequency band which will be enriched with new overtones.

Maximum frequency: Here you can set the upper limit for the frequency band which will be enriched with new overtones.

Intensity: This parameter is used to set the intensity, with which the new overtones should be introduced into the audio material.

Options



Transient optimization: If this option is activated, transients will be freshened up by the addition of higher frequencies. Older or compressed pop and jazz recordings benefit especially from this setting. The setting is less appropriate for tonal material, as odd overtones may result.

Inverse: In this button is activated, you will hear only the newly-generated part, the overtones being added.

Bypass: The algorithm is removed from the signal route. In this way, the unedited signal can be compared to the result of the algorithm.

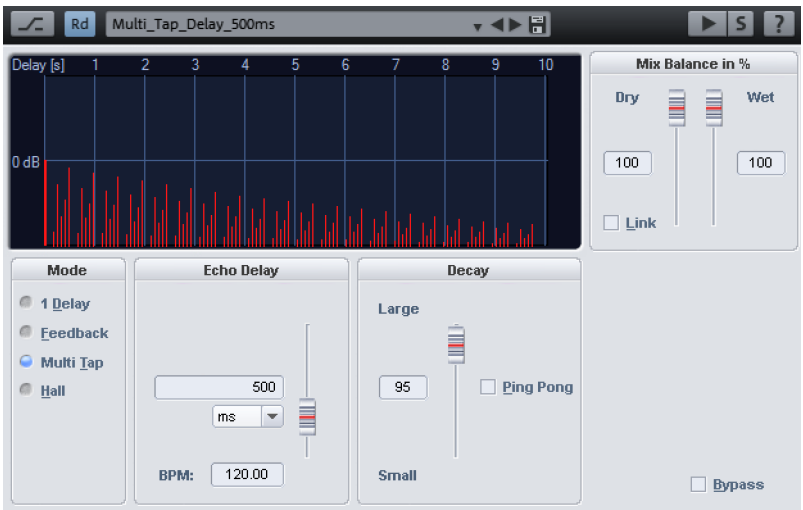
Filtox

More information is available in "Effects -> MAGIX plug-ins (view page 977)".

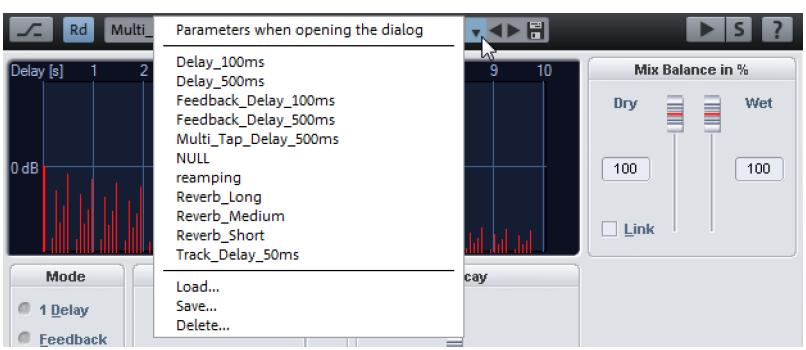
Delay/Reverb

Delay

With this effect, you can embed an echo or reverb into an audio file or object.



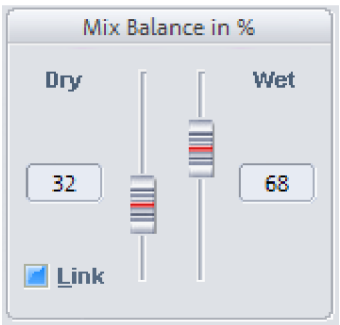
Delay presets



Load, **save**, **delete** presets: Here you can save, load, or delete settings. The default file extension is ***.ech**.

Delay parameters

Mix Balance in %: Here you can enter the ratio of the original signal to the echo/reverb signal in percent. If you switch on the "Link" option, both values always add up to 100%.

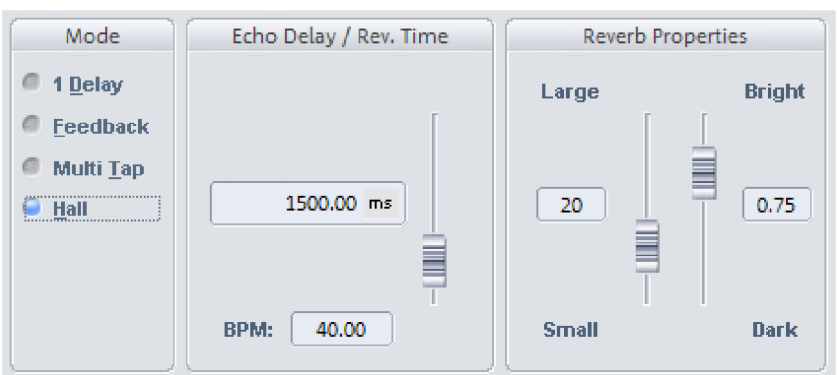


Mode: With mode you can select the basic character of the effect. The following modes are available:

- Individual delay
- Delay with feedback
- Multi tap delay
- Reverb

Echo Delay/Reverb Time: This fader sets the delay between single echoes or the original signal and the first echo in milliseconds. The fader controls the reverb time in "Reverb" mode.

BPM: Enter the current song beat in BPM, which automatically sets the delay value that corresponds to a quarter note in this tempo.



Decay/Reverb Tonality: Here you can set the decay value for the simulation of the size of the room or reverb tonality numerically as well as using faders. A ping-pong

effect is also available for the delay modes. In "Reverb" mode you can set the reverb room between "Small" and "Large" and "Light" and "Dark" using two faders.

Bypass: The algorithm is removed from the signal route.

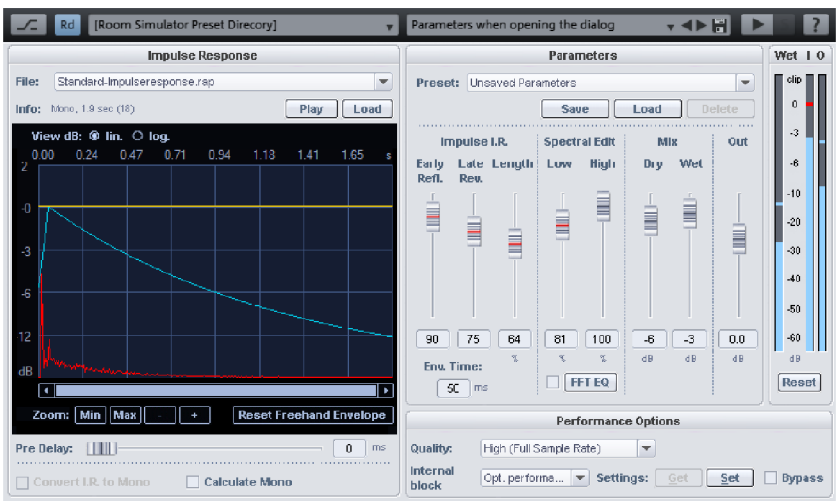
Room Simulator

This function allows you to simulate the reverb qualities of various rooms. The impulse response or reverb time of a room is calculated with a audio material.

A room impulse response is the reverb of a explosive noise, for example like that of a pistol. It contains all the necessary information to simulate room reverb accurately.

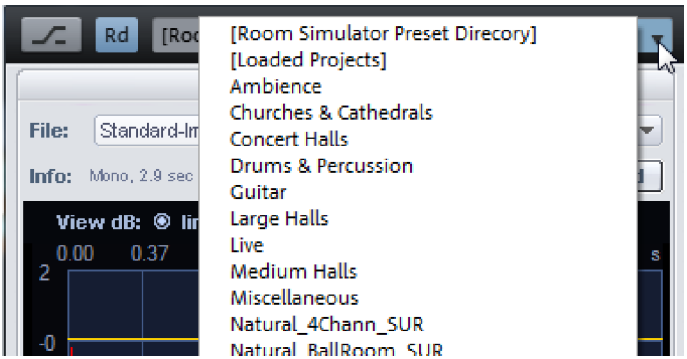
You can manipulate the impulse response using the dialog parameters. In this way you can adjust the reverb properties in a similar fashion to how digital reverb-effect devices do this. In addition, however you also have the ability to set the reverb character by fundamentally defining the impulse response.

A graphic representation of the impulse response and the envelope allows an optical overview of the impulse response manipulation using the Room Simulator parameters dialog. In Sequoia this effect is also available as a realtime effect in object, track, AUX bus, and master channel.



Room simulator – banks and presets

The room simulator offers a range of reverb banks in categories such as "Medium halls" and "Miscellaneous", in which various presets and impulse responses are saved.



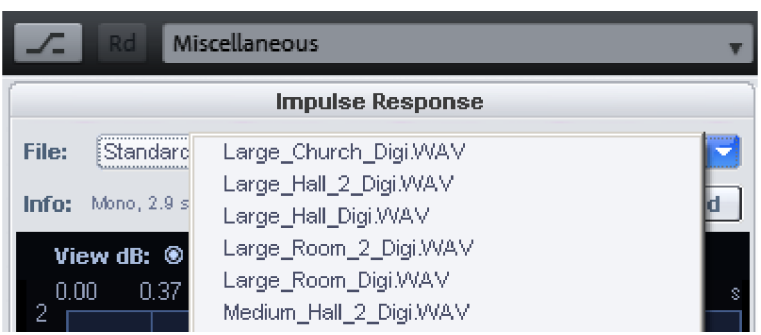
When a preset is loaded, the parameter settings of the dialog field and an impulse response are loaded. The impulse response can be any sample, which is saved on the hard drive as a wav, hdp or rap file.

Included impulse responses have ***.IMR** extensions.



The presets have the extension ***.rms**. Files with the impulse responses of the included presets are found in the "fx-preset > RoomSim" folder.

Presets found in the "RoomSim" folder are organized according to their affiliation with a certain bank in the preset list and are loaded by selecting one of the list entries.



Clicking on "Load" lets you retrieve preset files from other folders or from the room simulator.

Impulse response - controls

File: Select an impulse response. Impulse responses from the "RoomSim" folder and the audio files opened in the program (bank: [Loaded projects]) are available.

Note: If during Sequoia installation you have deactivated the "Install impulse response" option, you will have access only to the opened audio files (Bank: [loaded projects]) as well at the "Standard-impulse-response.rap" from the "Room Simulator preset directory" bank.

Info: Here you can get information about the length of the impulse response and about whether it will play back in mono or stereo.

Play (Impulse response)

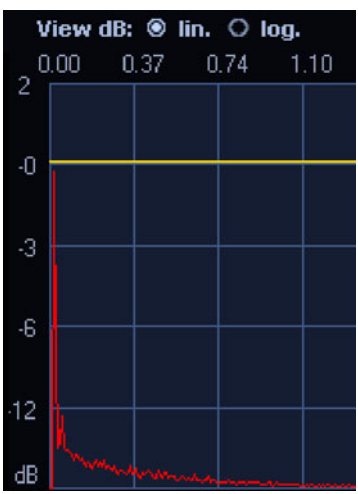
The impulse response will play after pressing this button. Impulse responses from IMR files will be played back only with 16 bits, even if the impulse response has a 32-bit float format.

Load (Impulse response)

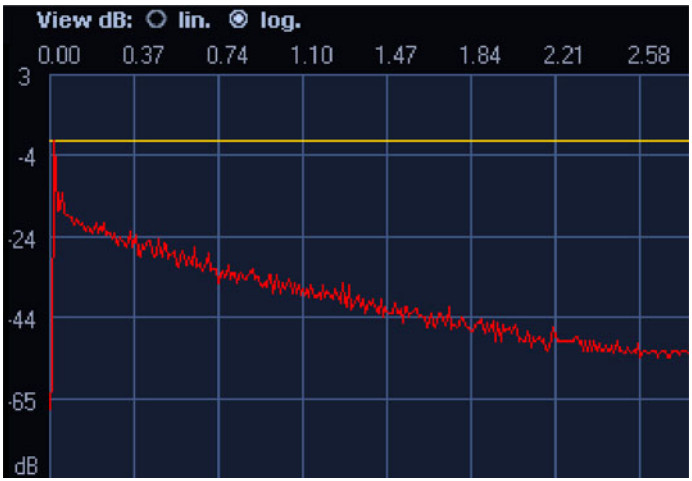
The impulse response can be any sample that is saved to the hard disk as a WAV, HDP, IMR or RAP file. from any folder on the hard disk

Graphic display

dB lin display: The graphic display with the impulse response has linear amplitude scaling. This settings corresponds to the common display of samples.

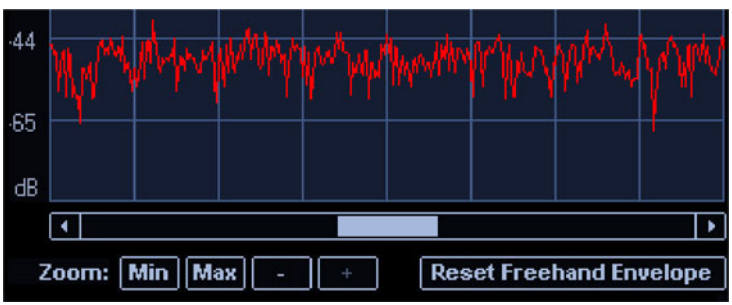


dB log display: Choosing this setting will scale the amplitude display of the impulse response logarithmically. This corresponds to the way the human ear perceives volume changes.

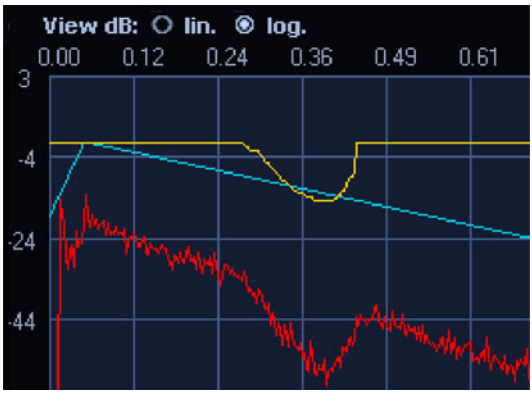


Zoom options (All, In, Out, Max, Position):

This option lets you zoom into the display and select the range to be displayed.



Curves



Red: The red curve represents the impulse response.

Light blue: The light blue display is the two-segment envelope curve to dampen reverb. The front segment is the earlier reflections, and the later segment is the post-reverb.

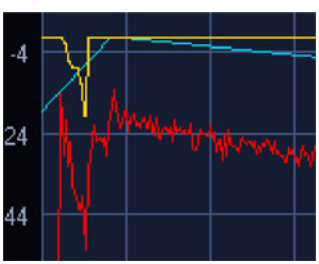
Yellow: The yellow curve is a freehand drawable envelope curve.

Freely definable envelope curve

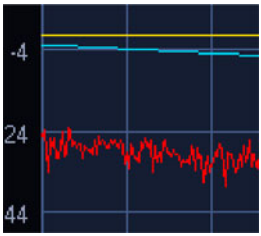
The impulse response scale can be manipulated with a yellow freehand envelope curve. Typically, it is used to dampen or delete early reverberations of the impulse response.

To edit it, click on the yellow curve with the mouse cursor and drag it with the mouse key held down.

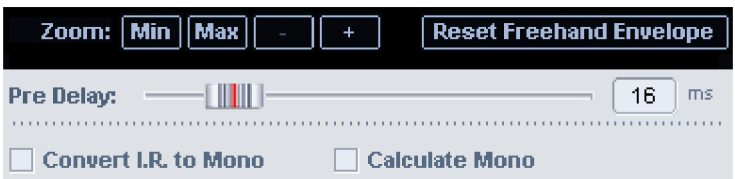
The use of zoom options makes it possible to work precisely.



By clicking on the "**Reset freehand curve**" you can reset the yellow envelope curve.



Pre-delay



The "Pre-delay" function delays the entire impulse response by a time interval variable between 1 and 100 ms.

Convert impulse response into mono

Selecting "Convert I.R. in mono" recalculates stereo impulse responses into mono. Convolution is calculated on two channels as before, in Surround Mode corresponding to the count of the group channels.

Calculate mono

Convolution is only calculated for one channel which reduces the required processing power. The input signal and the impulse response (if stereo) are converted to a mono signal before convolution. For surround application, all channels of the group are added to one mono signal.

Room Simulator - parameters

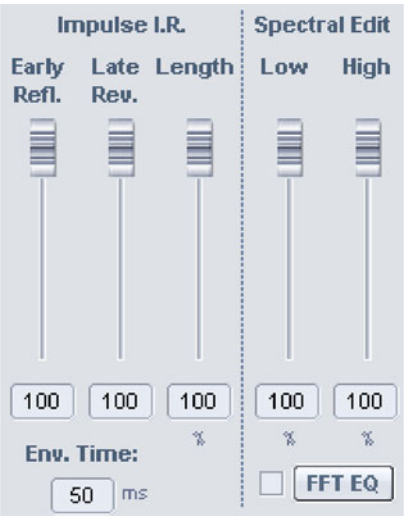
Parameter presets

Select, save, load, and delete presets that consist of parameters for envelope curves, frequency sequences, and the mix here.

The presets are useful for testing various parameter settings without having to change the impulse response.

The parameter presets have the extension *.rsp.

Envelope, EQ



Early reflect: This parameter allows you to lower or suppress early reflections by dampening the first part of the impulse response.

Late Reverb: This parameter lets you decrease or suppress the final part of the impulse response.

Length: This setting can shorten the length of the reverb effect to up to 5% of the original length by shortening the impulse response pattern. Please keep in mind that the reverb can end rather abrupt, which may lead to unnatural decay patterns. You can compensate by applying a lower Late Reverb setting to fade the impulse response with the 2 segment envelope curve.

Env. Time: Here you can set the time length of the first of the two segments of the envelope curve. Using this parameter, you can influence the damping of the earlier reflections.

Spectral Edit: Parameters for editing reverb frequencies

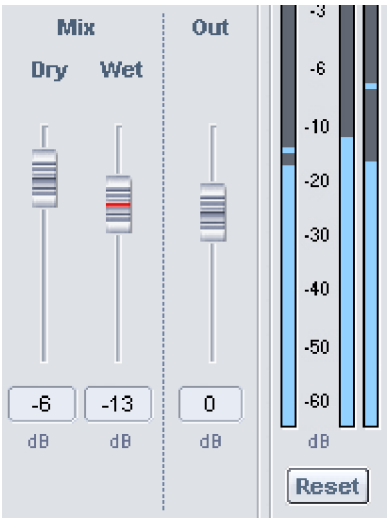
Low: Using this parameter you can adjust the low frequency component of the reverb.

High: Using this parameter, you can adjust the high frequency component of the reverb.

FFT EQ: The reverb component can be edited with an additional **FFT Filter** (view page 890). The graphic display with realtime spectrum of the reverb component in the FFT Filter allows optical control of the frequency characteristics of the room, where the

impulse response was recorded. Undesired resonancies can be quickly opened and removed.

Mix, Volume, Reset



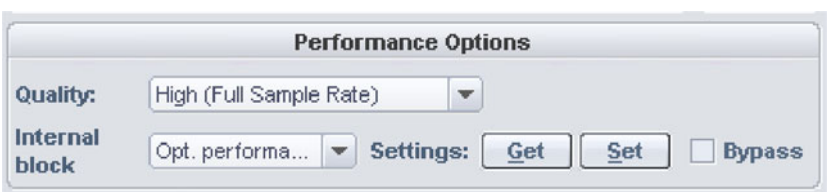
Dry: Here you can set the original signal level in dB.

Wet: Here you can set the reverb level in dB.

Out: Here you can set the output level in dB.

Reset: Here you can reset the held highest levels on the level display.

Room Simulator - Performance Options



Quality: In both "**Normal**" modes **"Normal"** and **"Normal Plus"** room simulation is calculated only with half of the sampling rate. In most cases this will be quite enough, since natural impulse responses and even impulse responses generated by digital reverb devices rarely posses components above 10 kHz (you can check this in the spectral representation of the integrated FFT Filter). (You can check this in the spectral display of the integrated FFT filter). Many older reverb devices work internally with half sampling rate, making calculation of the entire frequency range a waste of CPU capacities.

The difference between the two "**normal**" modes lies only in the quality of resampling that leads to sampling rate reduction. In "Normal plus" mode, the resampling quality is higher. The required calculation performance is slightly raised in this mode.

In the **"high"** mode, the entire frequency scale will be calculated. The CPU load doubles in comparison to the **"normal"** mode.

Internal Block Length: This parameter specifies a block length with which the convolution function can be calculated internally.

Short block lengths raise the count of the required calculation operations, causing CPU strain to increase. Long blocks lead to irregular CPU loads. The parameter has no effect on the calculation results.

The optimum for realtime processing tends to be about **4096** to **32768** samples.

Note: For use in the AUX, track or master without low latency, the setting of 2048 or lower is recommended on faster systems for an acceptable response latency.

For use in low-latency conditions, the value can be reduced to **32** on faster systems.

The value "**optimal latency**" adjusts the internal block length in such a way as to keep the latency as low as possible without influencing the performance too much.

The value "**optimal latency**" adjusts the internal block length in such a way as to keep the performance as high as possible without raising the latency too much.

Hint: It doesn't make sense to set a value smaller than the set ASIO buffer size. If the value corresponds to the ASIO buffer size, the room simulator processing is latency-free (only with "higher quality").

When using the room simulator for offline processing, the block length parameter is raised internally, as processing with a lower latency would in this case result in a useless increase of the necessary CPU operations.

Retrieve, Set: Opens and sets the quality options globally. When you have found a fitting setting, click on "Set". This saves the setting. Now you can continue to experiment. If you want to revert to the saved settings, click on the "Retrieve" button. This restores saved quality options.

Bypass: If you activated "Bypass" you will hear the unprocessed signal.

Room simulation - Tips and tricks

By using the numerous functions for destructive sample/object editing in Sequoia, you can alter the qualities of the impulse response in countless ways by using the Room simulator dialog.

- If you reverse apply the "Effects > Sample manipulation > Reverse" function to the impulse response, you will get a reverse reverb effect.
- By using the Timestretching function the size of the room can be altered without affecting the resonance behavior of the room itself.
- Employ an explosive envelope on any short samples and use the result as an impulse response - a great way of creating exotic reverb effects
- You can also get interesting reverbs by using the impact on a percussion instrument as an impulse response.

Room simulator - problems and solutions

Reverb sounds unnaturally hard.

You can control the impulse response by setting the "Reverb" parameter. The graphic display of the impulse response makes fast optical control possible especially in logarithmic display (dB log).

The signal created contains a strong DC offset.

Remove DC offset with the same function; "Remove DC offset (view page 947)"

Heavy hard disk activity or an error message concerning insufficient memory was received.

The algorithm needs lots of storage space, especially with long impulse response patterns. All background processes should be terminated and all projects that are not needed should be closed.

CPU power usage is too high.

- Check the "Internal block length" parameter setting.
- Set the "Quality" option to "Normal".
- Shorten the impulse response length.

Problems with very long impulse responses.

In realtime application, the impulse response length is limited in principle only by CPU capacity. For offline calculations, the limit is 380.4 seconds (at 44.1 kHz sample rate). If this upper limit is exceeded, the samples behind it are ignored on calculation. Impulse responses of this length are, however, not practical for room simulation. If you would like to experiment around with very long impulse responses (longer than 1

minute), you should note that the algorithm usually requires a lot more RAM. When loading long audio files as an impulse response, a warning will appear noting that calculation will take a lot of time.

eFX_Reverb

Detailed information on eFX_Reverb can be found at "essential FX > Reverb (view page 982)".

eFX_StereoDelay

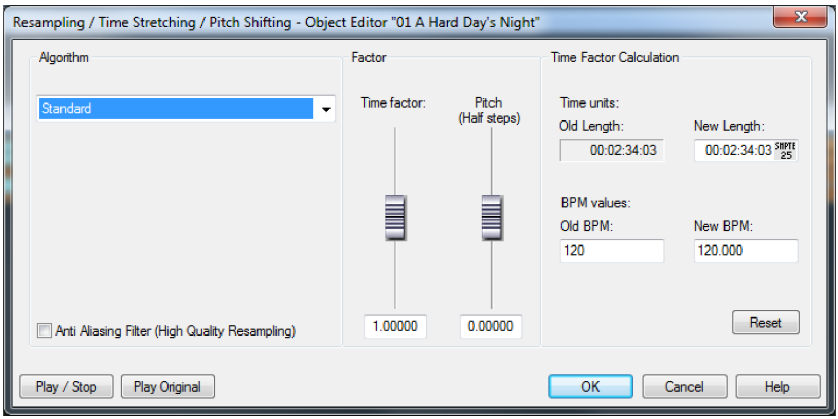
Detailed information about StereoDelay is available in "essentialFX > StereoDelay (view page 984)".

Ecox/VariVerb

More information is available in "Effects -> MAGIX Plug-ins (view page 977)".

Time/Pitch

Resampling / Time Stretching / Pitch Shifting



All algorithms in this dialog use the "time factor" and "pitch (semitones)" as input parameters.

Open the "Time factor calculation" section to determine the required time factor from the original length and original tempo as well as the desired new length/desired new tempo.

Tip: To change an audio file's sample rate, e.g. from 48 kHz to 44.1 kHz, use the function "Effects -> Sample manipulation -> Adjust sample rate (view page 975)".

Resample

Samplers and PCM Synthesizers transpose samples using this procedure. Time factor and pitch are dependent upon each other: the shorter the audio material, the higher the pitch and vice versa. The effect is comparable with changing the playback speed on record players or tape recorders.

Resample is a relatively lossless process. If a pitch change resulting from the change in speed is wanted or acceptable, you can use this algorithm.

Timestretching/Pitchshifting

With all other algorithms, time factor and pitch variation are independently adjustable. These algorithms create, under circumstances, artifacts which you can balance using the Anti-aliasing feature. The preset timestretching/pitchshifting algorithm can be saved via Timestretch/Pitchshift Patcher in Wave project (view page 813) ("Object" menu).

Monophonic voice

This is a special Time Stretching and Pitch Shifting function for vocal solos, speech, or solo instruments. The audio material should not contain background noise – strong reverb may also this method's effectiveness. With suitable audio material, the audio quality will remain high.

The "**Use formant correction**" option preserves formants during Pitch Shifting, i.e. the "Mickey Mouse effect" does not occur. This allows realistic-sounding background choirs to be compiled using one solo voice. The formants, however, may also be moved by +/- 12 semitones in order to achieve desired vocal distortions.

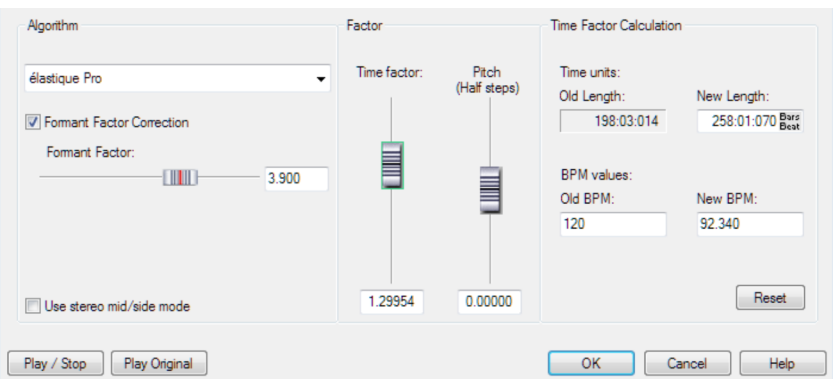
Typical uses of this algorithm include:

- Intonation correction: The note with the imprecise pitch should be cut out as an object so that it may be manipulated independent of the other notes.
- Harmonizer effects: Copy the object containing the singing. If the pitch is altered now, a second voice will be generated.
- Creating background vocals from existing vocal samples
- Time Stretch distortion of a spoken sample, e.g. grandfather's voice.

The "Monophonic voice" algorithm is especially useful for: Speech, single-voice singing, single instruments without overlaps/with low levels of reverb and background noises.

élastique Pro

With this algorithm licensed from zPlane you can achieve impressive results with minimal additional information such as "Time Factor" and "Pitch". As with the „Monophonic Voice“ (view page 913) algorithm, this mode can also be used to preserve or shape formants.



Use stereo M/S mode: This option enables M/S processing. This means that the algorithm internally interprets the adjacent stereo signal as a mid signal in the left channel and a side signal in the right channel.

élastique Efficient

When real-time application of élastique Pro threatens to overload the CPU this algorithm offers a less CPU-intensive alternative. However, if you choose this option, be prepared to make some compromises regarding sound quality.

Elastic Audio (Real-time)

Elastic Audio – general

Elastic Audio is a specialized editor for changing the pitch of audio material. It uses a combination of automatic resampling and pitch-shifting algorithms as well as pitch detection for monophonic material.

Elastic Audio offers the following options:

- Detection of the basic frequency in monophonic audio material.
- Automation of resampling and Pitch Shifting Algorithms. This also includes an algorithm for format-true pitch shifts of monophonic audio material that utilizes results of the pitch detection basic frequency analysis.
- Automatic and manual correction of the basic frequency process in monophonic audio material.
- Automatic and manual correction of the pitch of notes from monophonic audio material.
- Changing melodies in monophonic audio material.
- Manual correction of the basic pitch curve.

Note concerning Time Stretching: Time Stretching cannot be automated, but it can be used statically with Elastic Audio.

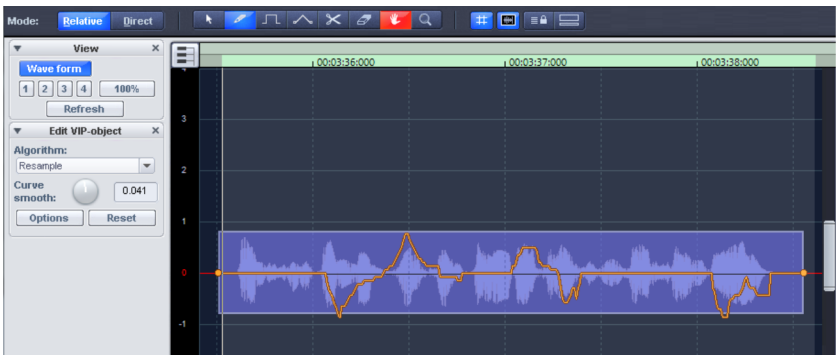
Elastic Audio - editing modes

The Elastic Audio editor may be used in two different operational modes, i.e. **Relative** and **Direct** mode. If Elastic Audio should always open in "Direct" mode, then select the function **Open in mode "direct"** from the Elastic Audio "View" menu.

At the lower part of this menu you can see which controls are available and switch them on and off.

Elastic Audio – "Relative" mode

Edit the pitch curve (automation of the pitch's temporal path) as a relative pitch deviation here. This corresponds roughly with the pitch bend controller for MIDI data.



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

Overview of graphics in "Relative" mode

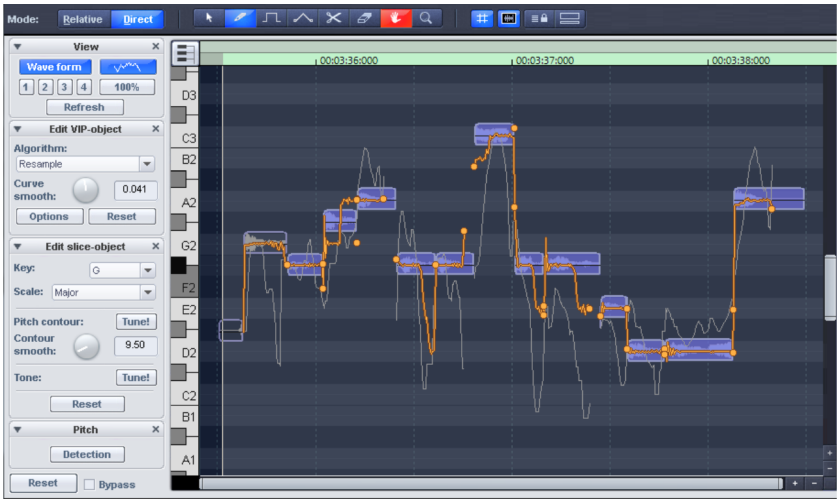
Y-axis: Display of pitch changes in semitone steps (within a range of -24 to 24 semitones).

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

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

Elastic Audio – "Direct" mode

The pitch curve is traced immediately in "Direct" mode, and changes are absolute. To customize the pitch correspondingly, the original pitch of the audio material must be known. A preceding analysis of the audio material's pitch is therefore required for "Direct" mode. This is basically only for tonal, monophonic material like solo vocals, solo instruments, and speech.



Start the analysis function with the "Pitch Detection" button. The analysis may take longer with larger objects. After analysis, the VIP objects are divided into individual slice objects according to the recognized pitches. The medium pitches of a slice object determine its position in the graphic, independent of the set progression of the pitch curve inside the slice object.

Select all of the slices first. Click an empty area in Elastic Audio Editor to deselect all of the slices. The individual or multiple slices may now be selected via the modifier keys "Ctrl" and "Shift".

Two handles are created for the pitch curve on the borders of the slice objects. These handles can be moved in order to produce an increasing or decreasing pitch characteristic, but still keeping the small changes in the basic frequency (vibrato).

Besides the pitch trace feature, this mode also provides you with "Tune!" functions for automatic pitch correction.

Overview of graphics in "Direct" mode

Y-axis: The piano keys on the left side of the window display the pitch assignment in notes. The individual pitches may be deselected by clicking them in this view so that they won't be used in the automatic pitch correction or in quantized drawing. You can get appropriate scale settings by selecting the tonic keynote and the scale in the “Edit slice-object” box.

Orange line: Displays the editable pitch properties.

Gray line: Displays the original pitch properties.

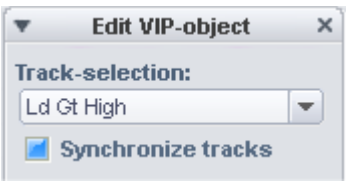
Blue line: Median pitch of a slice object.

Apply Elastic Audio to objects on multiple tracks.

To do this, select the objects you wish to edit in Elastic Audio in the Arranger. The objects can come from different tracks.

The Elastic Audio Editor loads all objects that were previously selected. When objects are loaded into Elastic Audio Editor from several tracks, switch the track displayed in the editor with the option "Edit VIP objects" in the track selection.

With the "Synchronize tracks" function in the Relative mode, you can apply pitch changes of the active object in the track selection to all other loaded objects, even if they are found on different tracks.



Example:

Imagine if you distributed choir recordings over multiple tracks.

1. Select all choir recordings to be edited on various tracks in the Arranger.

2. If you now open Elastic Audio, you can use track selection in Relative mode to pick which recording from which track is displayed, and edit it first.

3. Activate the "Synchronize tracks" option, so that all pitch changes also effect other selected choir recordings on other channels.

Elastic Audio: Control Elements

In the lower section of the "View" menu of Elastic Audio Editor, you can see which control elements are available in each mode and switch them on or off.

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:

Keyboard 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:

Keyboard shortcut:	Numeric keypad 4
	Numeric keypad 5
	Numeric keypad 6

Left-clicking on the zoom button while holding down the Shift key overwrites the last zoom level.

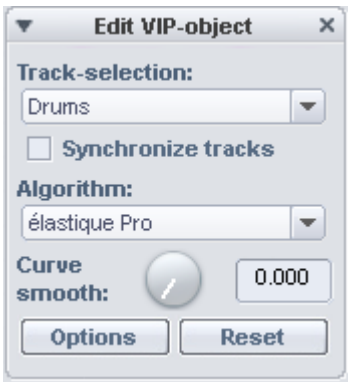
Left-clicking on the button while holding down the Ctrl key opens only the horizontal position of the respective zoom level.

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

> **Note**: The key combination "Ctrl + Alt" + Mouse wheel provides simultaneous horizontal and vertical zooming.

Refresh: This updates the graphical display of the material. Resulting changes to length are adapted. When actively synchronizing the arranger and Elastic Audio Editor (Elastic Audio Editor "View -> Horizontal -> Synchronize VIP") the effects on the VIP are also displayed in the arranger window.

Editing a VIP Object



Track selection: If objects in multiple tracks have been loaded into Elastic Audio Editor, they can be selected here.

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

> Detailed information about these algorithms is available in "Effects -> Time /Pitch -> Resampling/Time stretching/Pitch shifting (view page 912)".

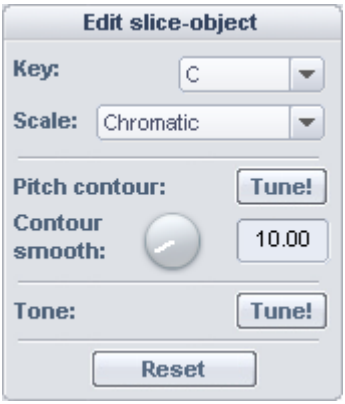
Curve smoothing: This parameter brings about the smoothing of the given pitch curve using a time constant (in ms). When smoothing to a large extent, even jump-like pitch shifts can are changed to a "glissando".

Options: These are advanced settings that depend on the selected algorithms.

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

Options for automatic pitch correction

Edit slice object (direct mode)



Key: This option sets the basic key of the scale. In the chromatic key, this setting is not included in calculation.

Scale: Use this option to select the type of scale, i.e. the sound make-up/mode. Choose from "major", "minor", "harmonic minor", "pentatonic", and "chromatic".

Pitch contour: Tune!: This button quantizes ("levels out") the pitch contour of the selected slices.

Contour smooth: Set the strength of the quantization (lower values quantize hard). This function removes repeated small pitch alternations in natural sound sources, e.g. vibrato ("Cher" effect).

Tone: Quantize the mid pitch of the selected slices. The corresponding sections of the pitch contour of a slice object are moved together and adapted to the mid pitch. Pitch fluctuations within a slice object remain the same.

Note: Automatic correction requires the slice's pitch (pitch -> detection).

The results of the calculation of mid pitches for slices featuring glissandos often do not correspond to the pitches that human hearing assigns to them. In this case, automatic correction of the mid pitches does not give satisfactory results. Removal of the glissando passage by manually changing the slice borders (keyboard shortcut: Ctrl + rubber handle tool) or the splitting of slices can help.

Reset: This command resets the selected slices. The orange curve is superimposed over the gray curve and the slice is reset to the original mid pitch.

Harmonizer

Use the **harmonizer** to add additional voices to the object. Please note that the harmonizer is only available in "Direct" mode via the "Monophonic voice" algorithm. Format correction is required for a satisfactory result which is only possible with this algorithm.

Chord: Select the chord created by the voices if the "Harmonic adjustment" parameter has been selected.

If "Harmonic adjustment" is deselected anyway, the voices are sent so that they are overlaid onto the sounds set in "Edit slice object". The set chord then only serves as a "guideline".

Humanize A "human touch" may be applied by moving the individual voices more or less subtly away from each other.

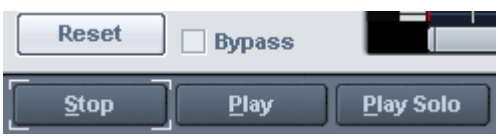
Harmonic adjustment: Activating this field causes voices to run parallel to one another and correspond to the set chord.

Create voices: Use this button to recreate parallel voices.

Pitch (fundamental frequency)

Detection: Use this button to start the basic frequency analysis. In this case, the absolute pitches are displayed graphically in the Elastic Audio Editor. This function is indispensable for additional editing steps you may wish to make in "Direct" mode. For this reason, you should only open the pitch recognition before executing pitch corrections or other editing processes.

Playback control



Reset: All pitch curves for the entire material loaded into the Elastic Audio Editor are reset.

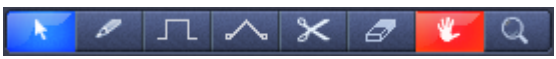
Bypass: Plays the arrangement without editing.

Stop/play: Stops/starts the playback of the arrangement.

Play solo: Only activated objects in the editor are played.

Toolbar

Several tools are available for processing the slices and pitch envelope. Two tools can each be assigned to one of the two mouse buttons. The tool assigned to the left mouse button turns blue; the function assigned to the right key is marked in red. Click with the corresponding mouse button on the desired console button. The "Zoom" tool is an exception: both buttons are assigned automatically.



Selection tool (arrow): Use this tool to move slice objects vertically. This modifies the pitch of slice objects as a whole. Slice objects and curve handles may also be selected, and a slice object lasso can be dragged out. Multiple selections are possible with the "Ctrl" or "Shift" key.

Freehand drawing function: Use the pencil tool to draw the pitch curve freely. If the "Shift" key is pressed, a straight line is drawn from the starting position to the current position of the mouse. When the "Ctrl" key is also pressed, the slice objects are combined during drawing.

Pencil tool for quantized drawing: This is the tool for quantized drawing. Quantizing means that the line is free horizontally, but snaps into semitone steps vertically in "Relative" mode, and to the tones of the scale selected via "tune" in "Direct" mode.

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

Rubber handle tool: This rubber band tool may be used to bend the pitch envelope between two neighboring handles. The range between the curve handles is moved, but the handles themselves remain unchanged. The curve is bent inwards at the handle point. In combination with moving the curve handles on the slice object's boundaries, you can bend the tone envelope and thereby maintain the microtonal structures (vibrato).

Cut: Use this tool to manually split the audio material into slice objects. After automatic pitch recognition has been applied, use the "Cut" tool to split certain passages into individual slices. To recompile slice objects, use the pencil tool to drag along the slice while holding down the "Ctrl" key.

Eraser: The orange curves are set back to their initial values, i.e. the pitch envelope corresponds 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. Left clicking and dragging stretches the range displayed as the zoom area.

Controls for global display options

These buttons have a global effect for the Elastic Audio Editor.



Grid: Use this function to switch the raster on or off.

"Single object" mode: The maximum horizontal zoom level is limited to one object.

"Synchronization" mode for VIP and Editor: Horizontal zoom levels of both windows are synchronized with each other.

"Docked view" mode: When "Docked view" is active, the arranger window is located below the Elastic Audio Editor.

Using Elastic Audio Editor

Elastic Audio: Overview of keyboard commands and configuration of the mouse wheel.

Navigating with the mouse wheel

Horizontal scrolling	Mouse wheel
Vertical scrolling	Ctrl + Shift + mouse wheel
Horizontal zoom	"Ctrl" + mouse wheel
Vertical zoom	Alt + mouse wheel
Zoom horizontally and vertically	Ctrl + Alt + mouse wheel

Keyboard shortcuts

Ctrl + space bar	Play solo/stop
Space bar	Play/stop
Ctrl + A	Select all
A	Refresh view
Ctrl + Z	Undo
Ctrl + 1 - 8	Select tool for the left mouse button
Ctrl + Shift + 1-8	Select tools for the left mouse button.
Shift + Alt + P	Show/hide pitch curve

Ctrl + Alt + T	Show/hide other loaded tracks
Ctrl + cursor up	Direct mode
Ctrl + cursor down	Relative mode
Cursor left	Move play cursor to left
Cursor right	Move play cursor to right
Ctrl + Cursor left	Horizontal zoom in
Ctrl + Cursor right	Horizontal zoom out
Shift + R	Change scale tuning (Frequency for "A" concert pitch).
Shift + A	Show/hide "View"
Shift + O	Show/hide "Edit VIP object"
Shift + F	Show/hide "Pitch"
Shift + S	Show/hide "Edit slice object"
Ctrl + number pad 4, 5, 6	Save zoom snapshot 1, 2, 3
Alt + number pad 4, 5, 6	Load zoom snapshot 1, 2, 3 without vertical zoom
Number pad 4, 5, 6	Load zoom snapshot 1, 2, 3 without vertical zoom

Tone pitch automation including changes to the length (resampling)

- Load objects into "Elastic Audio Editor -> Relative mode".
- Select the "Resampling" algorithm.
- Select the drawing tool.
- Draw the orange automation curve with the drawing tool.

Pitch automation without change in length

Proceed here as in the previously explained resampling section. You can also select a pitch algorithm. Here the length of the VIP objects remains the same.

Pitch correction (intonation correction) with monophonic audio material

- Load the object into the Elastic Audio Editor. Mode -> "Direct"
- Perform pitch detection ("Detection" button in "Direct" mode).
- Select desired slices
- Select suitable algorithm
- Edit pitch with the mouse toolsy

Manual correction of the whole pitch of a pitch slice

- Select the slice with the selection tool (arrow).
- Move the orange line vertically with the mouse tool.

Automatic correction of the whole pitch of a melody slice.

- Select the slices with the selection tool (arrow). Mode -> "Direct"
- Detect pitch
- Select the scale in the "Edit slice object" group and, if necessary, discard the additional tones that are not to be quantized via the keyboard on the left side of the editor by clicking and deselecting.
- Click the Tone "Tune!" button

Automatic correction of the pitch characteristic with quantization

- Select the slices with the selection tool (arrow). Mode -> "Direct"
- Detect pitch
- Select the scale in the "Edit slice object" group and, if necessary, discard the additional tones that are not to be quantized via the keyboard on the left side of the editor by clicking and deselecting.
- Click the Tone: "Tune!" button
- Reduce level of the quantization with the parameter "Contour smooth"

Correct pitch increase or decrease

- Select the slices with the selection tool (arrow). Mode -> "Direct"
- Detect pitch
- Move the orange-colored pitch curve to the edge of the slice object by moving the handles or click and drag the orange line

Create frequency modulations like warblers and vibrato

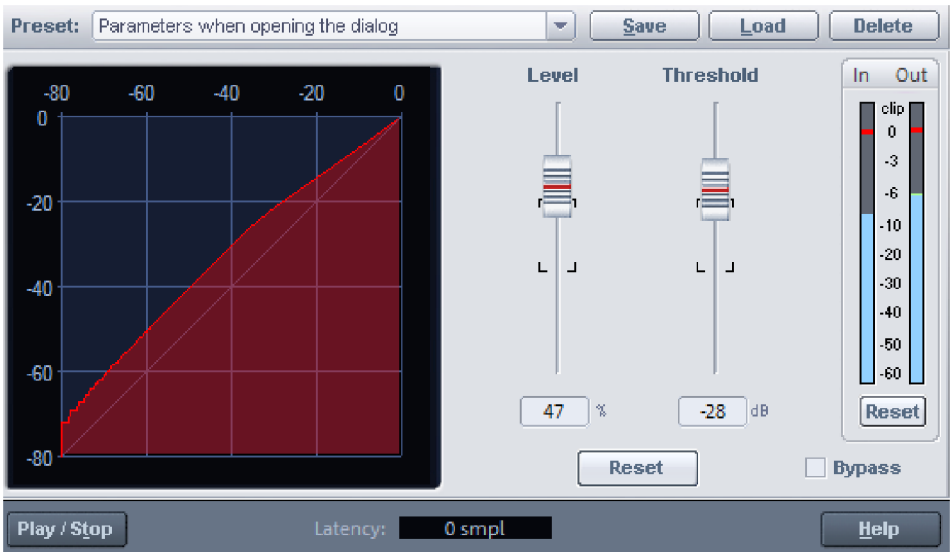
- Select the slices with the selection tool (arrow). Mode -> "Direct"
- Detect pitch
- Select the drawing tool
- Draw pitch modulation

Creating second voices or harmonizing effect

- Create a duplicate of the object on the track below it in the arranger
- Selection of both objects in the VIP
- Open Elastic Audio Editor
- Select the track via "Edit VIP object -> Select track"
- Edit each track individually. Do not use the resampling algorithm to avoid different tempos in the voices.

Distortion

This effect allows you to distort audio material by using a non-linear transfer reference line. The signal becomes louder and overtones are added.



By adjusting the starting point of the distortion (threshold), a soft, analog-sounding distortion (overdrive), or a hard, digital-sounding distortion (threshold at 0 dB). The intensity of the distortion may be set using "Level".

Vandal

Detailed information about Vandal is available in "Vandal (view page 1047)".

BitMachine



Audio material can always be edited into high quality with Sequoia. Nevertheless, there are some situations, for example, a more imperfect lo-fi sound would perfectly suit a drum loop or a synthesizer sound.

Remember, for example, the first hardware samplers from the 80s that usually only ran at 8 or 12-bit rates and at low sample rates. With the BitMachine, changing the sound with such an “antique” device is no problem.

You can use the BitMachine to bring back to life the times when minimalist and scratchy soundchips in home computers were commonplace.

The BitMachine opens up a gateway to “acoustic time travel” where you can encounter bit and sample rate reduction and downstream filters based on analog models.

Furthermore, the effect has a modulation section with which you can control individual parameters using an oscillator (LFO) or the input signal.

We have designed a range of “typical” presets to demonstrate the time travel abilities of the BitMachine. These can be opened at the top right of the interface

The following section describes the details of BitMachine:

"Reduction" section

Bits

This dial controls the resolution of the audio material. Turning the dial to the left results in 16-bit quantization (CD quality). The further it is turned to the right, the lesser the signal dynamic becomes. In extreme cases (1-bit), there are only “on” or “off” states.

At the intermediate levels, you’ll notice an increase in the background noise and a decrease in the dynamics. For example, 8-bit quantization will exhibit dynamics of

only 48 dB. Quieter points in the material sound noisy and very quiet points sound "capped." This effect is amplified the more you turn the dial to the left until it starts crackling or "groaning."

Sample rate

The audio material is "down-calculated" with this dial, i.e. the internal sample rate is reduced. A new separation ratio between old and new rates is created. In relation to this ratio, a sample from the data stream will be "dropped" at the various points.

Note: The two smaller dials from this section are explained under **Modulation**.

"Filter" section

The filter in the BitMachine is a digital model of one of the most well-known filters in music electronics, i.e. the "Chamberlin 2-pole" filter used in old Oberheim synthesizers. These types of filters sound exceptionally musical. They can also be used quite creatively in the BitMachine, but should not be used exclusively to smooth out existing artifacts.

The filter works in the so-called "high-pass" mode, i.e. it lets through deep frequency (or medium) material according to setting, and dampens highs and medium areas.

Freq:

You can specify the cut-off frequency of the filter using "Freq." Filtering starts above this frequency.

Reso:

The signal in the area around the cut-off frequency can be strongly elevated to just below self-oscillation. Sharp, cutting sounds are possible at this level, and the effect becomes even clearer when you vary the cut-off frequency.

Drive:

Both of the individual filters of the connections mentioned above have the ability to overmodulate themselves internally. With the "Drive" dial, you can regulate the amount of overmodulation. The more you turn this dial up, the more the signal is overmodulated. In this case, the parameters of the internal workings of the filter interact with one another. Increasing drive weakens the resonance, but, at the same time, the signal gets more volume, more bass and becomes acoustically fuller.

Note: The two smaller dials from this section are explained under "Modulation."

"Modulation" section

You can automate your effects via the settings in the modulation section.

Here, you'll find the so-called low frequency oscillator (LFO), which resonates with adjustable speed. You can influence the speed and type of resonance.

To influence the resonance, use the two small dials in both the reduction and filter areas. These four dials display modulation targets.

Example: You've left the dial for the sample rate at its default setting. Change the small dial beneath from its middle position to either side. The modulation for the dial value is added to the sample rate: The LFO now controls these parameters proportionately and the sample rate reduction resonates at this modulation.

You can use this technique on other dials as well. You just have to make sure that the main dial isn't turned up to full, because then the modulation wouldn't have any effect. The modulation is always added to the set value.

Example: Turn the small dial beneath the "bits" dial fully to the left (Value: -50) and the one beside it (beneath "sample rate") to the right (+50). You've now assigned a modulation to both parameters with the LFO. They are not changed uniformly, but rather opposite to one another: A negative setting is nothing more than an inversion of the modulation, so you're effectively turning down the control signal.

Waveforms of the modulation section

We've already explained this example with the help of sine oscillation. The LFO can be in:

- Sine form
- Square wave (0 or 1, no intermediate level)
- Random value (an internal randomizer will be queried at the set speed)

Oscillator speed

The LFO speed is specified with the "speed" dial. If the "sync" button is active, then the LFO adapts to the song speed, and the dial locks musical values into place (e.g. ¼ note). Rhythmic paths of the sound distortion are therefore enabled. You can also switch off this synchronization and set the speed manually (in Hz).

Modulation with the "Envelope follower"

In the modulation section you'll find a fourth button, the audio input signal. If this mode is active, then the signal itself can be called upon to extract "modulation

tension"; a so-called "envelope follower" continuously scans the volume of the input signal.

Note: The BitMachine doesn't recognize the type of audio signal automatically. For this reason, you should set the input sensitivity roughly with the "gain" dial. To do this, use the control LED: With accurate detection of the signal dynamics, assigning the four small dials to modulation lows is easier and you can use the full control range.

In envelope mode, the "speed" dial is used to control the response speed of the envelope (the display now switches to milliseconds). Lower times result in a faster response, higher times make the envelope rise (and fall) slower. You should experiment with the signal according to its complexity. The presets provided can only point you in a rough direction.

eFX_Tube Stage

Detailed information on eFX_Tube Stage can be found at "essential FX > Tube Stage (view page 991)".

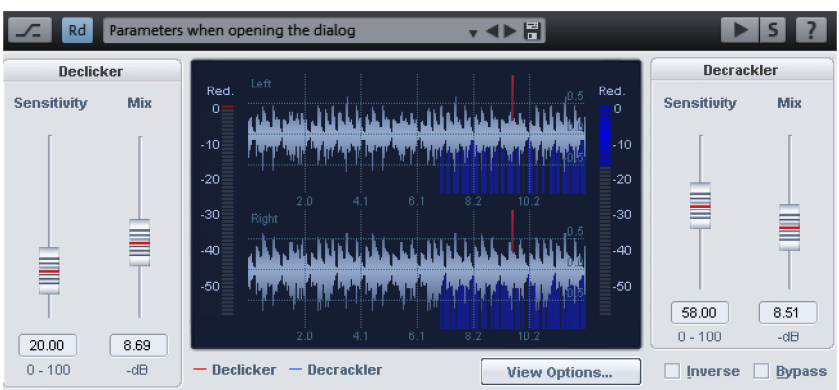
Restoration

DeClicker/DeCrackler

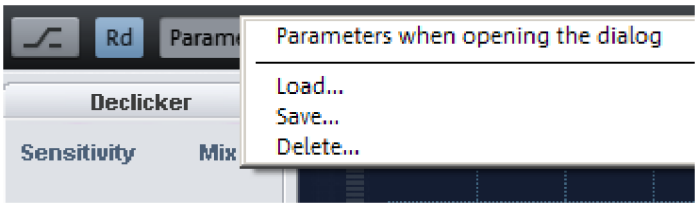
The DeClicker removes crackling and single clicking noises that are typical of scratched records.

The DeCrackler algorithm was developed specifically to remove crackling noises, making it easier to remove crackling from old records.

The DeClicker is on the left side of the dialog, and the DeCrackler is on the right.



DeClicker/DeCrackler - presets



By selecting"**Parameters when dialog is opened**", you will undo all changes made in the dialog since it was opened. By closing the dialog, you will apply all current settings.

Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.dck**.

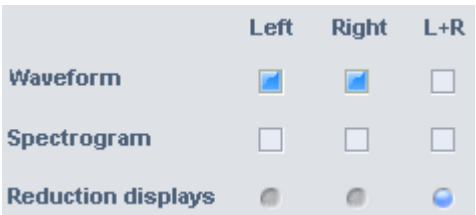
DeClicker/DeCrackler – signal display

The signal display shows you the edited material as a **continuous waveform** or a **spectrogram**. In the display you can see at which positions the DeClicker (red) or the DeCrackler (blue) are active.



The left border of the graphic display features the reduction display for the DeClicker. It shows by how many decibel the detected click has been dampened. The right border features the reduction display for the DeCrackler. This indicates how many decibels the crackling has been reduced.

Display options: This opens a settings dialog to configure the signal display. Select the right and left channel or the middle from both left and right (L+R). You can additionally set the channel that should be visualized by both reduction displays. The settings are applied by closing the dialog.



DeClicker/DeCrackler – parameters and control elements

DeClicker sensitivity: This parameter determines the DeClicker's sensitivity to noise.

DeClicker dampening: This parameter influences the intensity at which the DeClicker engages at the detected areas in the audio material.

DeCrackler sensitivity: Use this parameter to determine the DeCrackler's sensitivity to noise.

DeCrackler dampening: Use this parameter to influence the intensity at which the DeCrackler engages at the detected areas in the audio material.

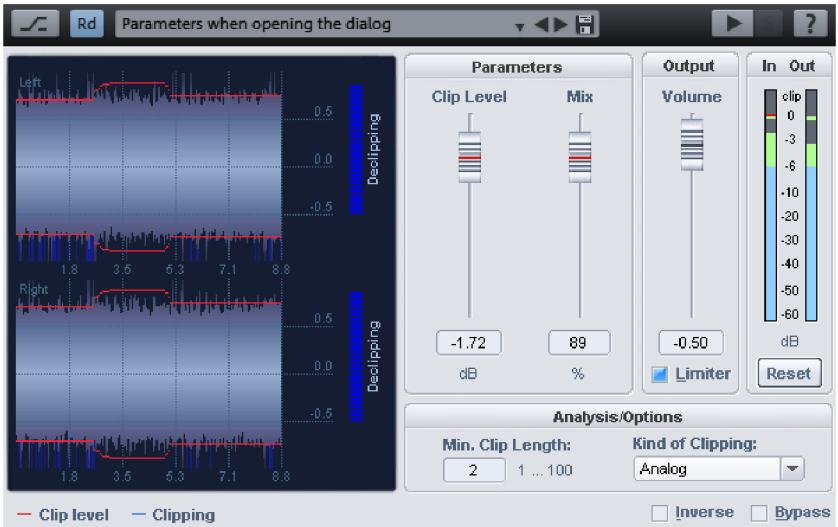
Inverse: If this switch is activated, only the part of the signal that is removed by the algorithm will be audible. If the parameters are set optimally, the complete distortion and just moderate components of the desired signal will be audible. If the correction parameters are too high, parts of the music or speech signal will also be filtered, which will discolor the music.

Bypass: The algorithm is removed from the signal route. This enables the unedited signal to be compared to the result of the algorithm.

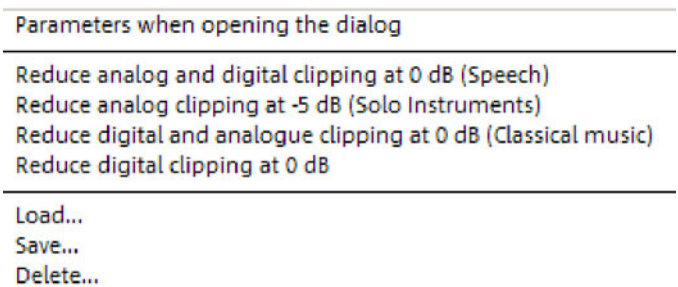
DeClipper

The DeClipper is a tool for removing overmodulation and distortions. Overmodulated passages are recalculated, or interpolated, based on the material immediately surrounding them.

The declipping algorithm is suitable for material featuring clearly audible overmodulation, e.g. distorted piano or vocals.



DeClipper - presets



By selecting"**Parameters when dialog is opened**", you will undo all changes made in the dialog since it was opened. By closing the dialog, you will apply all current settings.

Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.dcp**.

DeClipper – signal display

The signal display shows you the edited material as a continuous waveform. The **Clip level** is shown as a red limiting line. Use the blue markers to see the positions where the de-clipper was used. The meter to the right of the waveform display also shows the intervention of the effect.

DeClipper – parameters and controls

Clip level: Enter the level at which the sample will be considered overmodulated and corrected.

Reduction: Use this parameter to set the intensity of the reduction in % at which the DeClipper alters the audio material.

Volume: This setting makes it possible to lower the output level, because the corrected output will tend to be a bit louder than the input signal as a result of waveform interpolation.

Limiter: This option limits the peaks that exceed the maximum level. Peaks higher than the max may result from interpolation.

Reset: Set the level to its original position here.

Analysis/options

Min. clip length: Set the minimum count of consecutive overmodulated samples for clipping detection.

Type of clipping: Optimizes clipping detection by setting the applicable setting for the material – "Analog", " Digital", "Analog + Digital".

Inverse: If is activated, only the part of the signal that is changed by the algorithm will be audible.

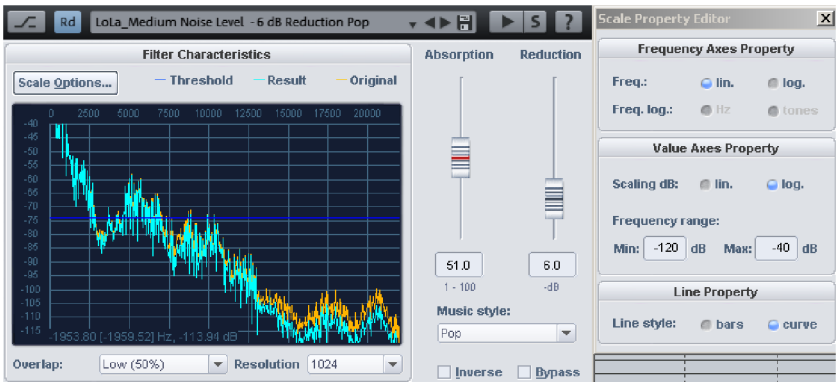
Bypass: The algorithm is removed from the signal route. This allows the unedited signal to be compared to the result of the algorithm.

DeClipper – sample applications

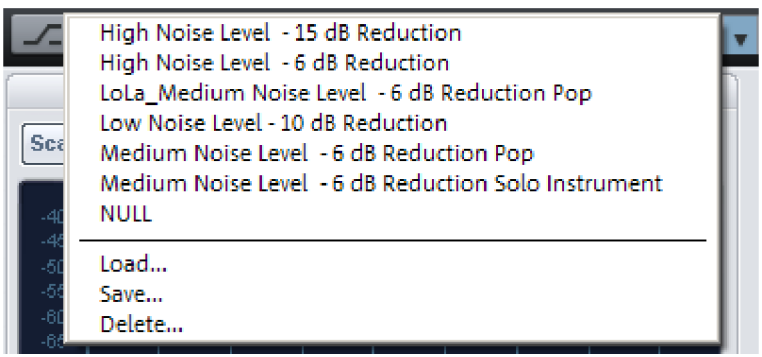
- Some DAT recorders have a protective analog switch, so that the level never reaches the digital mains (0 dB). In this case, a value of -0.5 dB or less is suitable.
- With an input of -6 dB (for example), all samples above half the control level are registered as overmodulated and recalculated. This allows a signal damaged by analog to be repaired.

DeHisser

This command opens the DeHisser for noise removal. The DeHisser eliminates regular, low-level "white" noise that is typically produced by recordings, microphones, pre-amplifiers, or transformers. Unlike during noise reduction, a sample of the distortion is not required.



DeHisser - settings and controls



The **load, save** and **delete** functions are integrated into the preset list, where they are always available. The default file extension is *.deh.

Correction (1-100): With this parameter a threshold value will be set for differentiating the signal from noise. The right setting for this parameter is crucial for good results. In comparison to noise reduction, where this setting will be automatically determined from the noise sample, with the de-hisser, you have to determine the noise level manually in relation to an absolute noise level.

Low settings can lead to incomplete noise removal or increased artifacts. High settings can cause discoloration of the original sample, phase-like distortion or

chirping modulation effects. The louder the noise, the more precisely your should try to set the DeHisser.

Noise reduction (0-30) dB: Set the noise damping here. The highest possible damping value is 30 dB.

In practice, extremely quiet passages, like the slow fading out of one instrument prove to be critical – here the noise level can exceed that of the audio signal.

Limit yourself to one low noise damping (-10 -15 dB) to avoid side effects like artifacts.

Music style: Minimize artifacts by selecting a fitting music style to aid in determining transients in the track and by de-hissing only between them.

Inverse: If this switch is activated, you will hear only the part of the signal that is removed by the algorithm. If the correction parameters are set optimally, only the noise will be audible. If the correction parameters are too high, parts of the music or speech signal will also be filtered, which will discolor the music.

Hint: If "inverse" is selected, the damping parameter will have no influence.

Bypass: The algorithm is removed from the signal route. In this way, the unedited signal can be compared to the result of the algorithm.

Overlapping: High values for overlapping allow the de-hisser to follow the signal quicker. When low values are selected, the algorithm is less quick when adjusting to the signal properties. The settings "Medium" and "High" are useful for most standard cases. "High" is most effective when applying the De-hisser to mixed (multiple instruments) tracks.

Rate: This parameter sets the rate (low, medium, or high) used by the noise reduction algorithm. Experiment with this parameter; a higher rate does not always guarantee the best results.

As an example, Low resolution often works best when applying noise reduction to short percussive sounds and speech. Instrumental music with slower tempos high resolution.

Scaling options

Frequency display Freq lin/log: Linear display of frequency provides a more detailed view of the high end. Logarithmic display of frequency provides a more detailed view of the low end. The logarithmic display corresponds to the human perception of volume. Like on a keyboard of a key instrument, here the graphic spacing of the entire

area is uniform throughout. If you have selected the logarithmic view of the frequencies, you can select between frequency and note representation for the horizontal axis.

Value display dB lin/log: Here you can choose between a logarithmic and linear level scaling. The logarithmic display corresponds to the human perception of volume.

Value range min/max dB: Enter the level range for displaying the real-time spectrum.

Curve depiction: Choose to display the original and the corrected frequency progression as a curve or as a bar diagram.

DeHisser – graphic display description

The vertical axis to the left indicates the level of the spectrum; choose between frequency or notes for the horizontal axis.

The yellow curve displays the original spectrum of the signal. Simultaneously, the first second of the selected audio segment of the active segment will be spectrally analyzed.

The light blue curve represents the spectrum after manipulation by the algorithm.

The dark blue curve represents the threshold of the "correction" parameters.

If the threshold is higher than the level of the spectrum (i.e. the dark blue line is above the yellow curve), the signal for De-hissing will be filtered out at these frequencies.

DeHisser - artifacts

The DeHisser was created especially for use with mastering in order to remove uniform and low-level hissing, which allows music and speech signals remain as pure as possible.

The algorithm is only partially suited to removing sound distortions featuring levels that reach or exceed the original signal. A metallic chirping or twittering sound may appear, the so-called artifact noise. In these cases, use the DeNoiser (view page 939) algorithm to remove noise and distortions.

DeHisser - the best setting

1. Search your audio material for a critical segment to preview. Critical segments are the softest parts of the music or speech, where the noise has a comparable level to that of the signal.

2. Set the damping parameter to the highest value (-30 dB).

3. Slowly increase the value of the correction setting.

There are four stages in this process:

Stage 1: If the value is very low, no noise will be removed.

Stage 2: The noise is partially removed. Depending on the level of the noise signal, a small number of artifacts may be introduced.

Stage 3: The noise is completely removed.

Stage 4: If the value is very high, not only will the noise be removed, but also a part of the signal. Press "Inverse" to hear this. The audio material will lose its brilliance and sound duller.

Optimal settings are usually available in **Stage 3**, where the noise signal will already be removed, but the audio signal is played back unaltered.

Tip: When monitoring the changes, it often helps to use a high monitoring level on the mixing board amplifier. It is recommended to use a system with a high signal-to-noise ratio. Monitoring using headphones is also helpful.

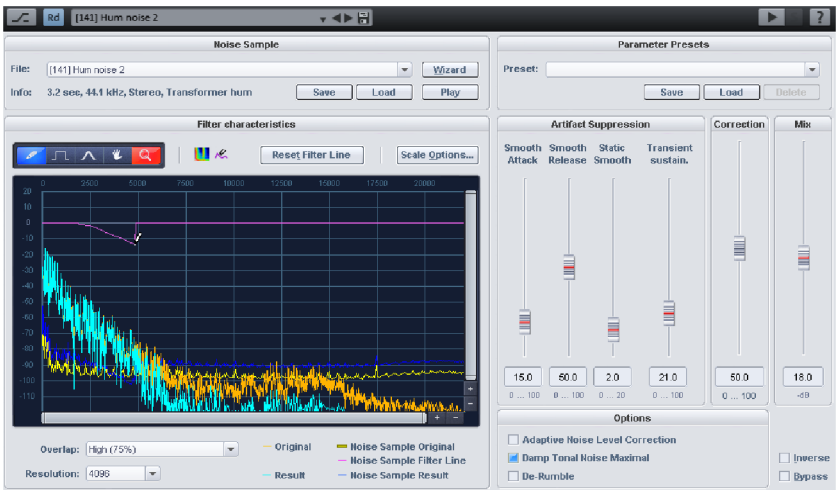
4. If the noise signal cannot be removed without discoloring the music or speech, reduce the damping parameter value until a compromise is reached between noise reduction and an acceptable discoloration of the audio material.

Tip: In the case of very high noise levels, try first to remove the noise from single tracks. Dehissing the complete mix requires more careful manipulation.

To remove humming noise, please use noise reduction (DeNoiser) (view page 939).

DeNoiser – remove unwanted noise

Use the noise removal function to free your wave projects/objects of unwanted noise without any noticeable reduction of the sample's range. The algorithm requires a noise sample beforehand to do this. The function works especially well to remove consistent, long-lasting noise like ventilation noise, artifacts from low-quality sound cards, tape deck noise, or feedback. Although the algorithm has not been specifically developed to remove clicking crackling, it is possible to get good results using it to remove this type of noise from old records.



DeNoiser – quick start

1. Select an area in the selected object or in a wave project of a virtual project that only features the noise.
2. Copy this area using the "Effects -> Restoration -> Get noise sample" into the "Noise sample" project. Especially in the case of hissing, better results are usually achieved by the algorithm the longer the noise samples are. If the length of the noise sample is longer than one minute, then there will only be a marginal improvement to the sound.
3. Select the range of the original sample in the wave project where de-hissing should be applied or select the corresponding object in the VIP.
4. Open the De-noiser via "Effects -> Restoration -> DeNoiser".
5. If the sample selection list for the noise sample (under "Noise Sample -> File") is not already set to "Noise Print", rearrange the list entry accordingly.
6. Press the "Play/stop" button for the real-time preview function. Once you are satisfied with the result, press "OK". Otherwise, change the parameters.

DeNoiser - Noise Sample

File: Select a group of noise tests in the selection list. The noise samples displayed in the list are located in the "Noise sample" directory or are temporarily available noise

prints. If the effect reference file is an automatically created copy of the original noise sample, the reference file name appears at the end of the original name.

Info: Provides information about length and type of the set noise sample.

Save: This button saves the currently set noise sample as a file for later use.

Load: This buttons opens a file selection dialog to import a wave file as a noise file.

Assistant: Opens the **"Noise print assistant"** dialog to extract a noise sample. Learn more more about this assistant below (view page 945).

Play: This command plays back the recently set noise sample.

DeNoiser – parameter presets

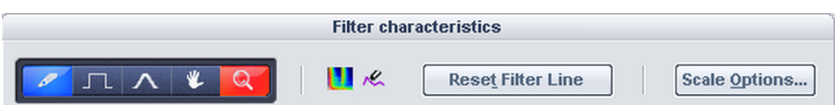
Load, **save**, **delete** presets: Save, load, or delete settings here. The noise sample is not included. The default file extension is ***.nrp**.

DeNoiser - filter settings

Pencil function: Drawing changes in the graphical display using the mouse tool edits the purple curve. This curve filters the noise sample.

There are 5 tools to help draw and navigate in the signal display:

- Freehand drawing with pen tools: Hold down the "Shift" key to draw straight lines.
- Drawing pencil for quantized drawing in predetermined decibel steps. The dB steps depend on magnification of the filter curve in the value display. Hold down "Shift" to draw straight lines.
- Reshape tool for bending curves: Through repeated clicking, the curve bends away from the mouse tool. The further the mouse tool is from the filter line, the wider the bend will be. If "Ctrl" is also held down, this effect will be even stronger.
- Navigation tool: Use this tool to navigate the coordinates system in the range while at a zoom stage. If you have not zoomed into the display, the navigation tool, the graphical display will not be affected.
- Zoom tool: Click with the left mouse button to zoom into the display; zoom out with the right mouse button.



In addition to the drawing tools, two additional buttons are available to you.

Activate spectral view: By clicking this button, **the signal will appear as a spectrum**. The areas of the spectrogram influenced by the De-noiser will appear red.

Activate direct mode for valuation curve: The second tool activates **"Direct" mode for drawing freehand curves**. The settings will be recalculated into a relative filter curve in the background.

Reset filter curve: This function sets the filter curve to its default settings.

Scale options: This button opens the "Scale options" dialog to adjust the signal display and the filter curve's value range as required.

Signal display: The graphic displays the **original spectrum** of the distortion **(yellow curve)** and the **corrected spectrum** which the algorithm uses internally to remove distortions **(blue curve)**. The level of the corrected spectrum is set via the **correction** fader. The curve can be smoothed via the **static smoothing** parameter. If this effect is active (for example during playback or monitoring), you can monitor the input and the output spectrum.

The **violet line** provides **filtering for the noise spectrum** (see above). Filtering effects the display of the corrected spectrum (blue line).

Overlapping: The exactness of the algorithm may be selected here in several steps. Processing times increase with increasing overlapping as the quality of the results improves.

Resolution: This parameter sets the rate (low, medium, or high) used by the noise reduction algorithm. The higher the resolution, the longer the algorithm will take to process. Experiment with this parameter, since a higher rate does not always guarantee the best results. For example, low resolution often works best for noise reduction applied to short percussive sounds and speech. Low-frequency sounds require adequate resolution in the low-frequency range; this may be achieved by setting this parameter to at least 4096.

DeNoiser - artifacts

When editing sound distortions with levels that reach or exceed that of the original signal, the algorithm may leave behind a metallic chirping or twittering sound, the so-called "artifact" noise. Its level is much lower than that of the original distortion, usually at around -20 dB, but due to its synthetic origins, the ear reacts sensitively to it.

DeNoiser – parameters

Artifact suppression

These settings suppress artifacts as they occur in broadband distortions of higher levels. Note that when higher values are selected, the quality of playback result may decrease.

Smooth attack: This parameter controls the attack during noise reduction. If delay is high, artifacts will be suppressed more effectively. The time ratio or impulse power of the audio material may decrease, however. In case of speech or singing, high values will not always produce the best results. This method of artifact suppression is recommended for orchestral instrument.

Smooth release: Use this parameter reduce an overly strong reduction of the release phase. Artifacts may reappear when higher values are used, however. Look for an acceptable compromise between artifact suppression and impulse response.

Static smoothing: This function smoothes the corrected noise spectrum used internally by the algorithm to remove noise. Smoothing of the blue curve may be visually confirmed by looking at the graphic. The artifacts are therefore reduced. In speech, vocals or pop music, higher values will usually provide good results. For orchestral instruments, high value settings of this parameter may cause unpleasant roughening of the sound

Consolidate transients: Use this parameter to influence the algorithm so that the distortion removal works more contained with transients. For example, in case of noisy jazz or pop recordings, this results in a clear improvement.

Correction

This parameter raises or lowers the noise spectrum (dark blue curve). Low settings may cause incomplete noise removal or increased artifacts. High settings can cause discoloration of the original sample, phase-like distortion, or to chirping modulation effects. The occurrence of these effects is dependent on the type of noise as well as the consistency of the original material. A higher noise level does not necessarily require a higher correction level.

Damping

Set noise damping in dB by entering a value between 0 dB and -40 dB.

In many cases, it is desirable not to remove noise completely: for example, when working with gramophone recordings, it can be desirable to leave some of that "gramophone feeling" in. Background noise from "on location" reports does not need to be removed completely. When noise is not completely eliminated, the occurrence

of artifacts or discoloration is reduced. Think of the process in terms of reduction, not 100% elimination.

Options

Adaptive correction: Activate this switch to achieve time-adjustable, automatic customization of the value for the "Correction" parameter. While the DeNoiser is active, you can the monitor the constant adjustment of the "Correction" fader in the input field. "Adaptive correction" is useful for distortions with variable noise levels.

Maximize tonal distortion dampen: Removes tonal distortions such as humming or camera sounds; activate this button to remove this unwanted material. The "Dampen" parameter only affects the non-tonal portions of the signal (hissing). This may produce better results, since the dampening of tonal signal portions creates fewer artifacts than the dampening of non-tonal distortions. The resolution value should amount to at least 4096.

De-rumbler: Frequencies below 40 Hz will be considerably dampened. This way impact noise like footsteps or rumbling on records can be removed.

Inverse: If this is activated, only the part of the signal that is removed by the algorithm will be audible. If the parameters are set optimally, you will hear the complete distortion and just moderate components of the desired signal. If the parameters are not set correctly, larger components of the music or spoken signal will be filtered, which may cause discoloration of the sound. However, the part of the artifact resulting from the remaining components of the distortions is hardly decipherable by monitoring the inverse signal.

Bypass: The algorithm is removed from the signal route.

DeNoiser - the best setting

To find optimal settings for noise removal in the material, proceed as follows:

1. Search for a good "**Correction**" setting. The distortion sound should no longer be audible. If artifacts occur, they shouldn't be suppressed by setting a very high correction value. The result may sound lackluster.
2. To increase artifact suppression, increase the values for **smoothing** and/or **static smoothing**. Which parameter is most useful depends on the nature of the audio material from which the distortion is to be removed.
3. Try to slightly **reduce** the **correction**. Artifacts will increase in this case. Now raise the "**Smooth attack**" parameter and/or "**Static filter smooth**". Use the slider to improve your results.
4. A high degree of **overlapping** will increase quality.
5. Drag the **"Preserve transients"** parameter as far out as possible until noise increases and the distortion becomes audible at the transients (with hissing, this

effect is recognizable by the type of hissing modulation). Reduce the level again slightly.

6. For difficult cases, we recommend the **freehand draw filter curve for the noise sample**. Problematic areas may be **influenced by increasing or decreasing the distortion spectrum**.
7. If there is no audible improvement, try correcting the noise instead of completely removing it. This may be attempted by reducing the value for mix **dampening**.
8. Whether or not artifacts are audible also depends on the monitor volume and the frequency response in the playback. If you know how your work will be reproduced (cinema theater, on TV, radio, etc.), you should work under corresponding monitoring conditions.

DeNoiser - Problems and Solutions

Problem: The result is silence

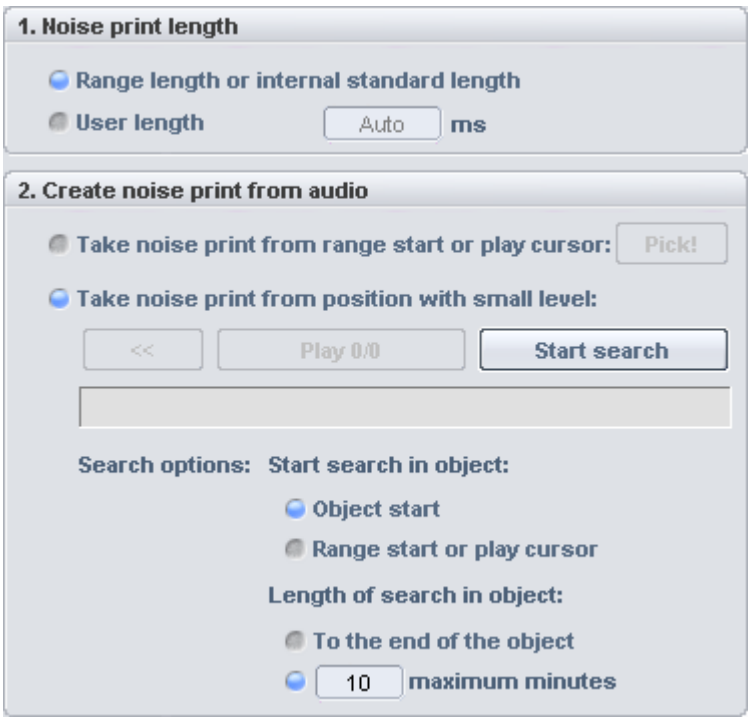
Solution: It is likely that the noise sample inadvertently contains the original signal. Use another noise sample.

Problem: Strong distortion, loud tweeting, strong damping of the used signal

Solution: It is likely that the sound sample contains sections from the original signal or from another sample. Use another noise sample.

DeNoiser - noise print assistant

The noise print assistant offers (in addition to the "Get noise sample" menu) two options to create a noise sample, whereby the "Pick" method of the "Get noise sample" function may be the better approach.



Proceed as follows to create a noise sample using the noise print assistant:

1. Determine noise print length

Range length or internal standard value: If a range has been stretched, a range length will be used. Ensure that the range is not longer than necessary, since every created noise sample instance in real time will be added to the project. If, conversely, no range is selected, an internal length is used, which will normally deliver useful results.

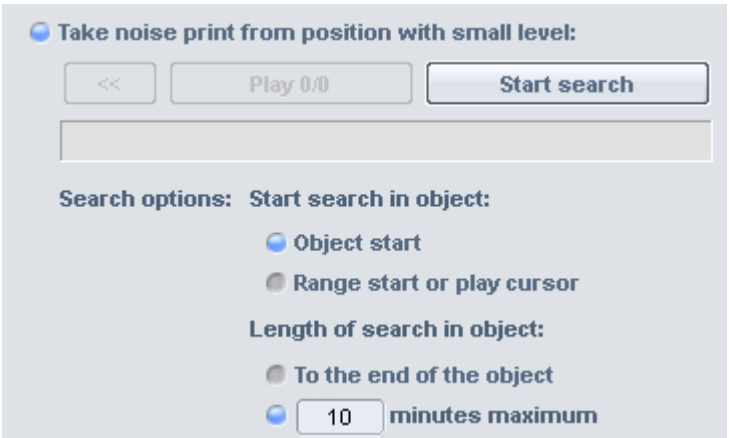
Target length: Set an explicit length for the noise sample.

2. Extract noise print from the audio material

Take noise print from range start or play cursor: The noise sample will be taken from the selected range, if available; otherwise, the standard or user defined length is used, starting from the play cursor position.

Take noise print from position with small level: This search function analyzes the audio object for soft areas. Set the search area above the search options and click the "**Start search**" button. Cancel the search by pressing "Esc".

After at least one quiet range has been located, select it using **<<** and **>>** and listen to it using "**Play...**"



Search options:

Start the search in object – object type: The search area starts at object type

Start the search in object – range start or the play cursor: Search area starts at the start of the range if you have defined a range beforehand. Otherwise, the search will begin at the play cursor.

Length of search in the object – until object end: The search for soft passages stretches until object end.

Search length in object – x minutes maximum: Enter the search length in minutes here.

3. Exit the assistant with "OK"

Get Noise Sample

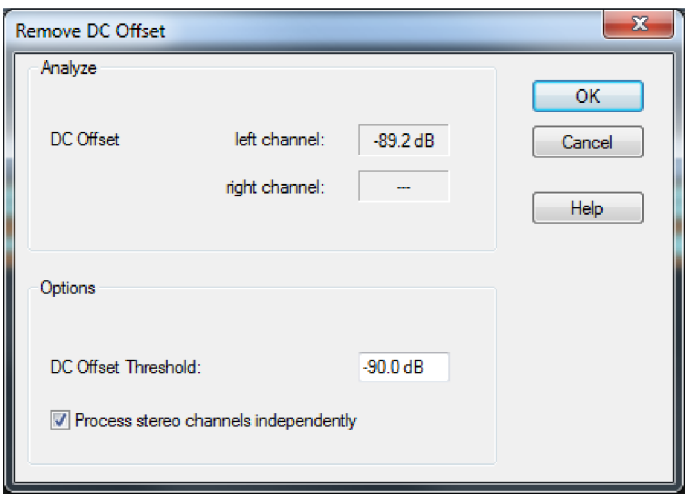
With this command you can create a noise sample. To do so, you must select an area in a Wave project or an object in a virtual project, which contains only the noise to be removed. The noise sample offers noise reduction a preset to aid in noise removal.

eFX_DeEsser

Detailed information on eFX_DeEsser can be found at "essential FX > DeEsser (view page 993)".

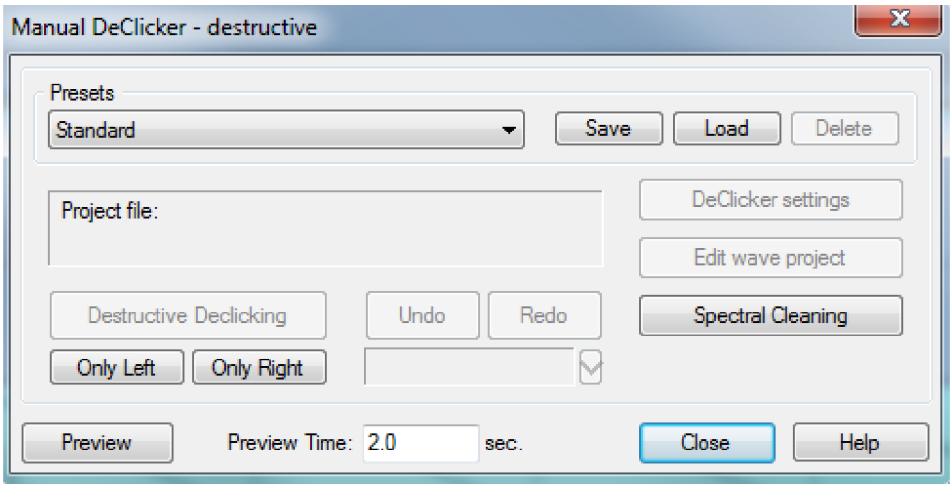
Remove DC offset (offline)

This function removes the DC offset of an audio file/object. This is useful if the sound card overlays a recording of your sample with a constant DC offset, which may cause clicks during cutting or editing.



Options: Here you can enter a minimum DC offset threshold, which would indicate where DC offset removal will kick in. If you unclick the corresponding check, stereo channels will be edited together. This serves to save computational time, especially when working with long files.

Manual DeClicker...(offline)



This dialog field enables manual offline DeClicking of individual sections. For this to work the separately available ScratchFreePro plug-in from Algorithmix has to be installed. For more information on the plug-in, please go to www.algorithmix.com.

The dialog contains a button for access to DeClicker settings, a button for direct audio editing mode and Spectral Cleaning.

The settings can be loaded and saved. Self-created settings, that are saved in the fx_presets folder, are shown in the preset list.

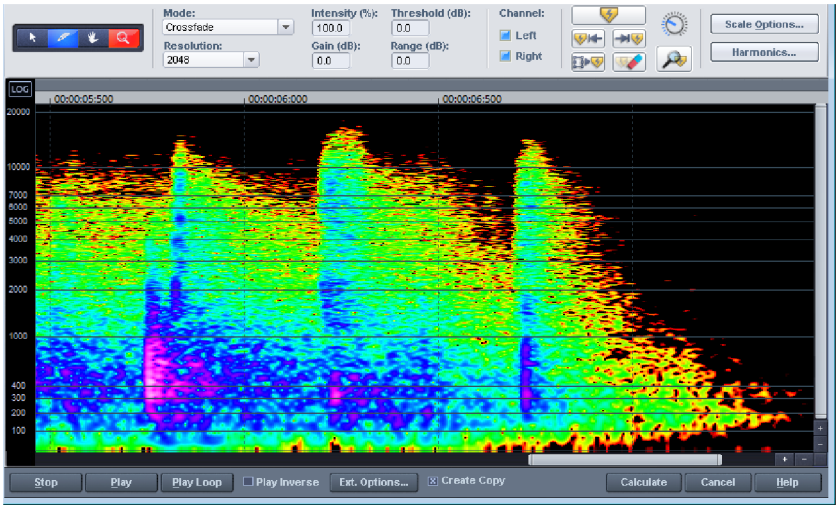
Shortcut: Ctrl + Shift + D

Spectral Cleaning (Offline)

Spectral Cleaning – the basics

With Spectral Cleaning you can remove distortions (like coughing, whistling, or singular claps) from a recording without influencing the wanted signal.

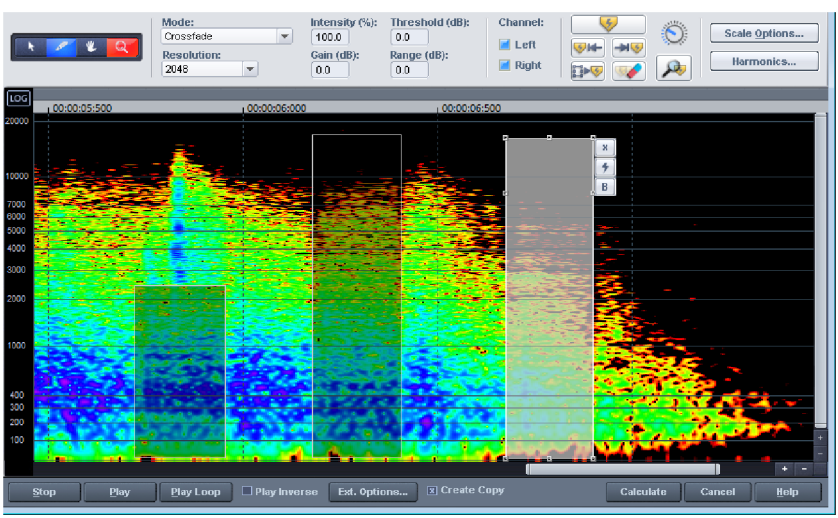
The graphical interpretation of the music can be seen in the Spectral Cleaning Editor via a spectrogram. This displays the frequency proportions in a time curve. The volume of frequencies is visualized via a color code or via its brightness.



Audible distortion noises louder than the wanted signal are usually limited to a certain frequency spectrum. They are highlighted with colors in the Spectrogram. So you can easily select the interference with the mouse and remove it.

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.

In Spectral Cleaning you can highlight interferences with the drawing tool before removing them. So that there is no audible gap, removed components of the original frequency spectrum from the wanted signal that surrounds the distortion are recalculated into the recording.



The Spectral Cleaning is useful for removing all short, impulsive sounds in the music like popping, coughing, or even short dropouts. For permanent noises like humming, etc. use the DeHisser (view page 935) or the DeNoiser (view page 939).

Spectral Cleaning – operation

Select the object and set the play cursor to the rough position of the distortion. Next, open the Spectral Cleaning Editor via the "Offline effects" menu, or via the object context menu under "Effects (offline)".

Select the drawing tool. The mouse pointer will turn into a pen icon. Highlight the nose by drawing a rectangle around it using the mouse. You can also highlight and remove multiple interferences in the editing window.

If you move a noise selection with the selection/draw tool, then you can **lock out vertical movement on the frequency axis by holding down "Alt"**. Holding down the **"Shift" key also locks out horizontal movement** on the timeline.

"Ctrl + X", "Ctrl + C", "Ctrl + V" may be used to cut, copy, and insert selections, respectively. This only copies the settings of the rectangle, and not any audio material.

If you set the cursor in front of the selected distortions and press "Play", you will hear the result of the noise removal in the preview function. You can modify the frame around the distortion selection using handles to optimize the signal.

Three small buttons visible at the position of your selection.



Delete (x symbol): Remove the selection by clicking this button.

Show distortion (lightning symbol): After marking the noise signal with the drawing tool, the result of the noise removal will normally be displayed in the selection. The "Show original" button displays the original signal for comparison purposes.

Bypass (B): Once you have drawn in the selection, you can already monitor the results of noise removal during playback. If, however, you do press the "Bypass button, you will continue to hear the original signal.

Sometimes it's necessary to modify the effects parameters (see "Spectral Cleaning - Edit modes"). If you're happy with the result, press the "Calculate" button.

Spectral Cleaning – tool bar

You can assign various tools to the left and right mouse buttons, by clicking on the corresponding tool. The tool for the left mouse button is displayed in blue and the right mouse button in red.

Selection tool: Use this tool to modify the existing selection window.



The mouse pointer will turn into a double ended arrow when you hover over the edges of the selection rectangle, you can drag the edges and adjust the selection size.

If you are using the selection tool inside a frame the mouse will turn in to a crosshair. Now you can position the entire selection frame in the Spectral Cleaning window however you like.

Drawing tool: This tool allows you to change the selection size (just like with the selection tool) and position the selection window anywhere.



You may also drag out any number of selection windows with this tool to remove noise.

Note: Double clicking on the selection applies vertically it to the entire frequency range.

Navigation tool: This is useful when you are working with a high zoom factor and can't see the entire object contents in the Spectral Cleaning window.



With this tool you can move the window section displayed by holding the left mouse button down.

Zoom tool: After selecting this tool you can left click to zoom into the window.



By clicking and dragging with the left mouse button held down, you can stretch the zoom area in the Spectral Cleaning window. With a right-click you can zoom out of a selection.

Spectral Cleaning - Editing modes

With the "Mode" setting you can specify how you want to remove the distortion. All marked interference will be edited with the selected mode independent of the "Level" fader.

The following modes are available to you:

Crossfade: This mode replaces the noise with the signal from the areas directly bordering the selection.

Crossfade (hard): This mode behaves like "Crossfade" but has a more powerful effect. In this way you can eliminate particularly noisy interference more effectively. It is especially important to mark the interference precisely when doing this.

Crossfade (from left): In this mode the noise is replaced by the audio signal located directly to the left of the selection.

Crossfade (from right): In this mode the noise is replaced by the audio signal located directly to the right of the selection.

Gap: This mode is especially useful for very short dropouts.

Dampen: This mode dampens the selected noise. Surrounding audio material will not be included. In this way no artifacts will be created, but there may be unexpected silence or even a drop-out. Use this mode if a distortion is present in a very narrow frequency range or if the surrounding noises are very quiet.

Fade In: This mode is similar to the "Dampen" mode. However, here the signal is faded in instead of uniformly dampened. This means that frequency-selective fade-ins can be created.

Fade Out:: In this mode the selected noise will be faded out across the selected area on the timeline. In some cases you can use the „Hide“ option, e. g. for noise at the end of a track.

Level: Use this control to specify the degree of change. Moving the fader to the far right corresponds to a 100% level of change.

Note: The "Edit selection" and "Level" settings can be controlled for each selected area individually, or as a multiple selection using "Ctrl".

Resolution: Determines the resolution of the applied FFT (Fast Fourier Transformation). Lower values often produce a more effective removal of the disturbance, but result in more tonal artifacts. They are also suitable for short interferences with powerful transients, such as clicks. Higher values are recommended for longer noises such as coughs etc. For optimal results first try editing an area with a low resolution, remove a transient part and then eliminate the rest of it with a higher resolution.

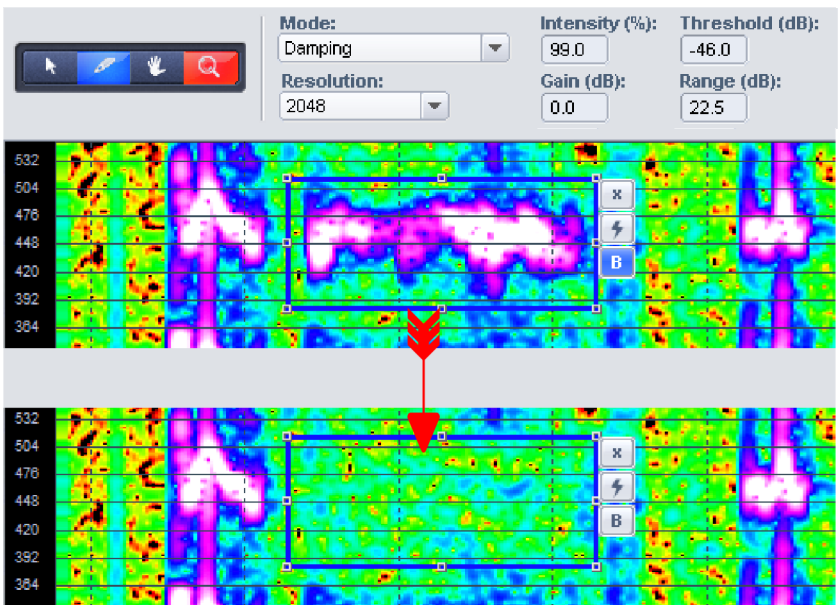
Note: The setting "Resolution" applies to all edits and cannot be customized for each selection.

Gain (not in Samplitude Pro X2 Suite): This can be used to boost the level to compensate for a drop in volume caused by interpolation. This is particularly suitable for removing gaps when editing multitrack recordings.

Threshold Value (not in Samplitude Pro X3 Suite): This setting can be used to limit the interpolation to a specific amplitude of a signal in the spectrum. This means certain notes can be removed from the spectrum without influencing background noises.

Note: This function will only be effective with a set area and should be used in the editing mode "Dampen" where possible.

Range (not in Samplitude Pro X5 Suite): This range determines the volume window in which interpolation of the set threshold can be performed. Quieter or louder signal components outside the area will not be affected.



Channel left/right: These buttons show you the corresponding channel in the spectrogram. If the interference only affects one channel, you can switch off the other channels leaving them unaffected.



Track Selection (not in Samplitude Pro X5 Suite): If there are objects from different tracks selected, you can specify here for which track the calculation should be performed.

Notice: To select more than one object, the objects have to be of equal length and start at the same time.

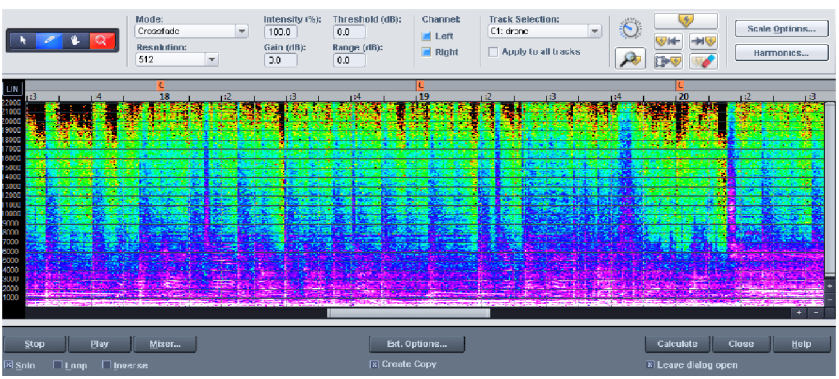
Apply to All Tracks (not in Samplitude Pro X5 Suite): With this option you define, if the calculation should be transferred to the selected single track, to all tracks of a previeously defined group of tracks or to all tracks.

Spectral Cleaning - Click-Marker

To highlight special points, quickly jump between or delete markers in the spectrogram, use the so called "Click marker" buttons in Spectral cleaning interface.

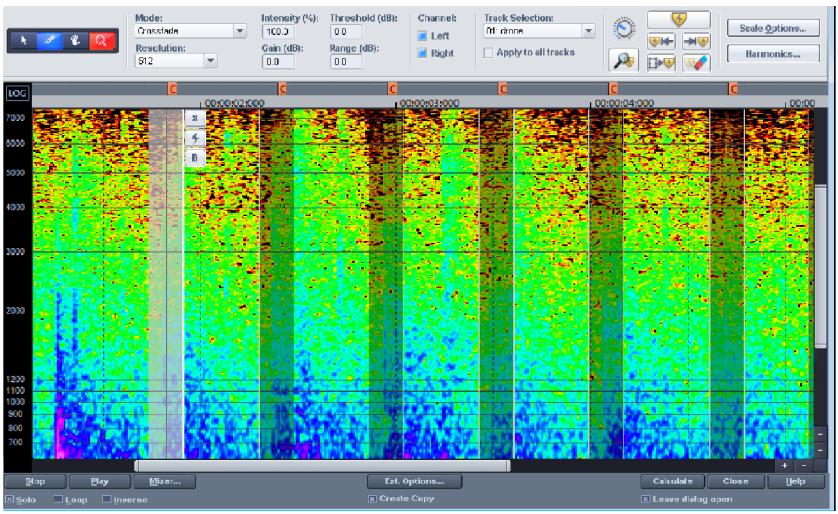


You can set and delete click markers. Created click markers are marked with a "C" in the editing window's marker bar.



With "Jump to next marker" (Shortcut: +) or "jump to previous marker" (Shortcut: -) you can navigate quickly between the markers in the Spectrogram. To move among the click markers only and not the standard markers, hold down "Ctrl".

By pressing the "Apply range selection to all click markers" button, you copy the current selection frame to all positions where there is a click marker. This allows you to use the settings of the current markings on all click marker positions. This is particularly well suited for removing interferences.



Not in Samplitude Pro X5 Suite: You can use the button „Automatic Search for Click Markers“ to search through the entire project for clicks and mark them so they can be edited together. With the corresponding knob you can set the sensitivity of the search, i.e. the higher the level the more clicks will be found.



Note: Select suspicious areas in the arranger before opening the Spectral Cleaning dialog. To transform a standard/beat/tempo marker into a click marker in the spectrogram, click the corresponding marker in the Spectral Cleaning marker bar. Next click on the button "Set Current Marker". Now you can click again on the marker position to display the created click marker.

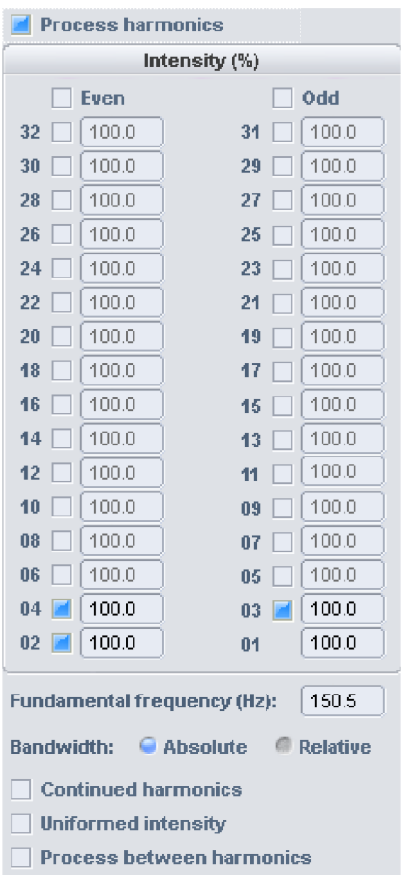
Harmonics

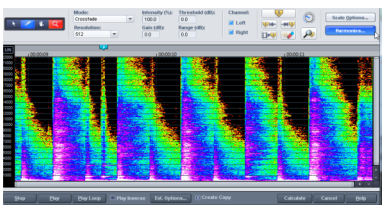
(Not in Samplitude Pro X5 Suite)

You can remove individual noises, e. g. 50 Hz buzzing, as well as harmonic overtones from a spectrum using this function. To do this an individual sound is selected in the editing window with the smallest possible edge. Once the calculation is activated for harmonics, the corresponding harmonics will be automatically provided with an editing rectangle in the editing window above the selected root.

Changing the root always leads to a change in the harmonics. In the settings dialog you can determine whether or not harmonics should be observed during calculation and if so, which ones in particular.

Using the option "**Continue harmonics**" it is possible to subtract out those harmonics that are above 32.





Edit harmonics: If active, the settings of all dialogs for calculating harmonics are included in the calculation.

Level (%): Customized selection of which harmonics are to be observed during calculation. The "Level" which is used can be determined for each individual harmonic. You can adjust the settings for the root (1 harmonic) to up to 32 harmonics. In addition you can use the "Harmonic continue" option to include harmonics in the calculation.

Even/Odd: Selection of even or odd numbered harmonics.

Fundamental frequency: This displays the fundamental frequency of the bottom most selection in the editing window and can be changed by entering a value or by moving it in the editing window.

Absolute/relative bandwidth: In the "Absolute bandwidth" the height and therefore frequency range of the harmonics selection rectangle is the same as the selected root. The frequency range increases in size with each harmonic in relation to the harmonic's frequency if the bandwidth is set to "Relative". This is so that relevant acoustic ranges can be achieved properly.

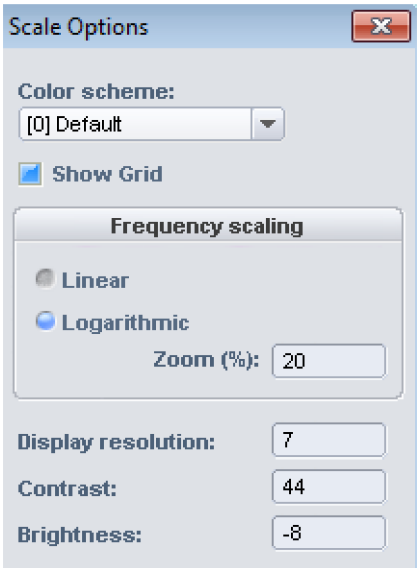
Continue harmonics: With this option any harmonics above 32 harmonics are included in the calculation with a linear dropping curve in order to achieve a thorough removal of all possible harmonics.

Uniform level: This means that the level set for 1 harmonic is also used for all other harmonics.

Edit between harmonics: If this option is activated, Spectral Cleaning will edit the sections between the selected harmonics. If you deactivate one or several harmonics, your sections will be edited up to the next active harmonics.

Scale Options

In this dialog you can adjust the settings that deal with color and content display.



Color scheme: Here you can choose a predefined color scheme which is used to display the spectrum,

Show grid: Displays a time and frequency grid in the spectrogram.

Frequency Scaling (not in Samplitude Pro X5 Suite): Here you can switch the display of the spectrum from linear to logarithmic. This is recommend for editing low frequency ranges. So that low frequencies are triggered higher up yet still display the entire frequency spectrum. The logarithmic ordering can be used to adjust the logarithmic display to meet specific requirements more precisely.

Note: Frequency scaling can be switched between linear and logarithmic using a button in the editing window.

Display Resolution (not in Samplitude Pro X5 Suite): This setting allows you to display the spectrogram in another resolution in order to achieve either a better spectral or chronological resolution for the range of interference.

Note: Calculation happens independently of the selected display resolution and is performed with the set resolution in the spectral cleaning dialog.

Contrast, Brightness (not in Samplitude Pro X5 Suite): Adjusts the contrast and brightness of the spectral display, e. g. to make very quite audio material visible.

Spectral Cleaning - Playback section

Stop: Stops playback (Keyboard shortcut: Space Bar)

Play: Plays from the playback marker position (Keyboard shortcut: Space Bar)

Solo: Plays the edited track solo (Keyboard shortcut: S)

Play loop: Repeats playback within a selected range (Keyboard shortcut: L)

Play Inverse: Plays back the differential signal, i.e. the removed sounds (Keyboard shortcut: I)

Mixer (not in Samplitude Pro X5 Suite): Opens the mixer (Keyboard shortcut: M)

Advanced options: Opens the settings dialog for destructive effect calculation

Create copy: Creates a copy of the current file during calculation

Calculate: Executes calculations and closes the dialog (Keyboard shortcut: Enter)

Undo (not in Samplitude Pro X5 Suite): Undo slice edits.

Redo (not in Samplitude Pro X5 Suite): Redo slice edits.

Leave Dialog Open (not in Samplitude Pro X5 Suite): The dialog remains open after the calculation is complete

Close: Closes the dialog window

Help: Opens the "Help" file.

Spectral Cleaning - Shortcuts

(Not in Samplitude Pro X5 Suite)

Logarithmic mode on/off	Ctrl + L
Increase zoom for logarithmic mode	Ctrl + +
Decrease zoom for logarithmic mode	Ctrl + -
Increase display resolution	Ctrl + F3
Decrease display resolution	Ctrl + F2
Increase contrast	ALT + F3
Reduce contrast	ALT + F2
Increase brightness	Shift + F3
Decrease brightness	Shift + F2
Activate/deactivate harmonic mode	Ctrl + H
Switch between relative and absolute bandwidth in harmonic mode	ALT + H

Navigation and playback

Play	Space
Sets the play cursor left	Left arrow
Sets the play cursor right	Right arrow
Sets the play cursor to object end	End
Plays as loop	Ctrl + Spacebar
Plays with 7s pre-roll time in front of play cursor	NUM_1
Plays with 3s pre-roll time in front of play cursor	NUM_2
Plays with 0.3s pre-roll time in front of play cursor	NUM_3
Loads entire zoom range (X and Y axis)	NUM_0

Loads user-defined zoom range (X and Y axis)	NUM_4 to NUM_9
Loads user-defined zoom range (X axis only)	STRG + NUM_0, STRG + NUM_4 bis STRG + NUM_9
Loads user-defined zoom range (Y axis only)	ALT + NUM_0, ALT + NUM_4 bis ALT + NUM_9
Saves user-defined zoom range (X and Y axis)	STRG+ ALT + NUM_4 to CTRL + ALT + NUM_9
Loads user-defined zoom range (X axis only)	1-5
Loads user-defined zoom range including FFT block size	6-9
Saves user-defined zoom range (X axis only)	CTRL + ALT + 1 to CTRL + ALT + 5
Saves user-defined zoom range including FFT block size	CTRL + ALT + 6 to CTRL + ALT + 9
Saves only user-defined FFT block size	CTRL + ALT + SHIFT + 6 to CTRL + ALT + SHIFT + 9

Edit selection

Move selection horizontally only	ALT + LCLK

Move selection vertically only	SHIFT + LCLK
Double-clicking on the selection applies it to the entire frequency range.	DBLCLK
Copy selection (settings only)	CTRL + C
Cut selection (settings only)	CTRL + X
Add selection (settings only)	CTRL + V
Multiple selection	CTRL + LCLK

Click marker

Create Marker	SHIFT + C
Jump to next marker (all markers)	NUM_+
Jump to previous marker (all markers)	NUM_-
Jump to next marker (without click marker)	CTRL + NUM_+
Jump to previous marker (without click marker)	CTRL + NUM_-

Edit

Restores standard value to an edit field	ALT + LCLK

More

Calculates	ENTER/RETURN
Solo on/off	S
Loop on/off	L
Inverse on/off	I
Opens mixer	M

Spectral Editor – Algorithmix reNOVAtor (Offline)

Access the "Algorithmix reNOVAtor" via Sequoia here and apply it to objects.

Stereo/Phase

Switch Channels

Use this function to swap the right and left channel of stereo samples in order to correct accidentally swapped channels when recording.

This function may be reversed, e.g. if you don't re-select the range, opening it again will restore the original material.

Multiband Stereo Enhancer...

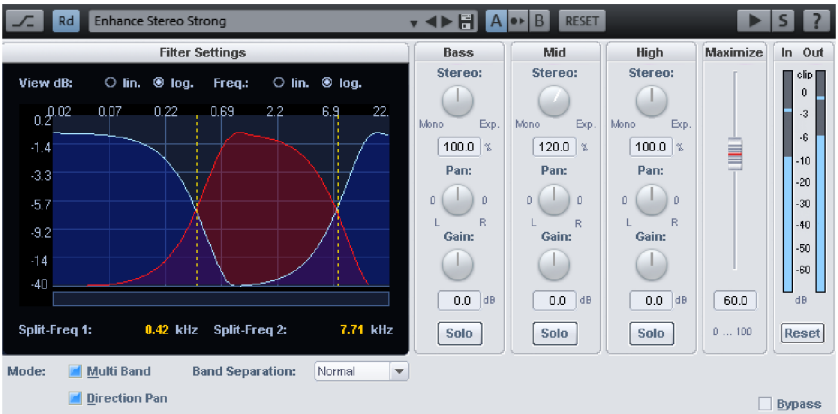
The multiband stereo enhancer allows you to perform detailed modifications and corrections to the stereo image in three independent frequency bands.

Thanks to the FIR complement filter technique no discoloration of the frequency response is introduced.

The settings for the stereo image in multiple bands possesses numerous important advantages compared to a standard resolution without frequency divisions. Enlarging the basis width in the mid range only can hinder undesired appearance of noises such as "strained highs" and "blurred bass" resulting from cancellation, for example. Reduction of mono-compatibility during enlargement of the bass width by limiting the increase of bass width is also limited to a single band.

Important uses for Multiband Stereo Enhancer applications are:

- Reduction or expansion of the stereo base width.
- More powerful bass by reducing the base width in the bass range.
- Control and correction of problems in the stereo image of a completed mix.
- Moving the mono section of a stereo recording within the panorama (direction mix). This way, centered vocal recordings may be moved to the left or right of the stereo balance later.
- Damping or removing mono signals in the mid-frequency range in order to create room in tracks or drum loops. This allows the addition of further instruments or vocals.

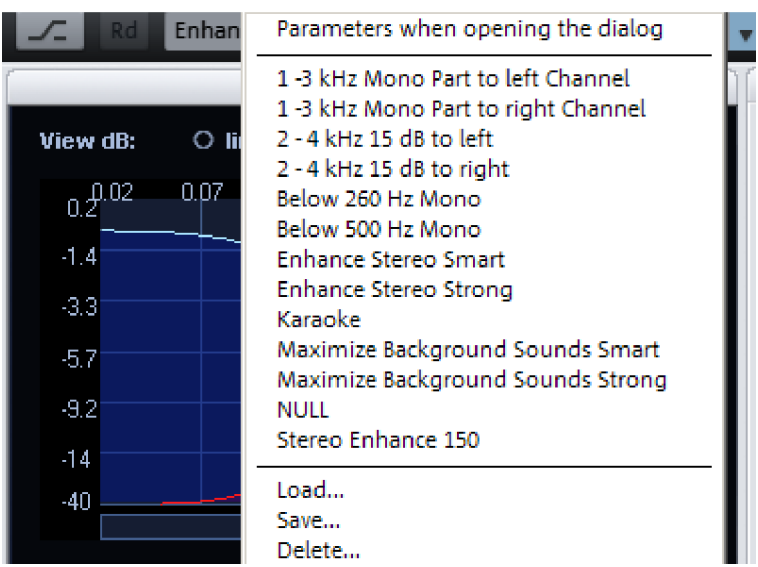


Overview

Filter presets are found at the top left dialog window.

The graphic displays approximately the frequency response of individual bands. The left axis label displays damping in -dB, the top axis label - frequencies in kHz.

General controls



By selecting"**Parameters when opening the dialog**", you will undo all changes made in the dialog since it was opened. By closing the dialog, you will apply all current settings.

Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is: *.ste

Bass, mids, and highs: You can set three frequency ranges for stereo, panorama, and gain (see below).

Maximize: Use this controller to compress the side-chain signal, which also increases the stereo transparency without influencing the mono compatibility. In "Multiband" mode, this setting affects the middle bands.

Solo mode: In solo mode, you can listed to frequency bands individually. This greatly simplifies the process of changing filter parameter settings. As an example, you can isolate a specific frequency range of a mix, and change the aspect of the stereo image for the range.

Multiband mode: The algorithm works with three frequency bands if this option is checked. The "Multiband" mode is not automatically active when using the enhancer in the mixer master section. This reduces the initial load on the CPU when selected in the master section.

Direction pan mode: This switches between two panorama control modes:

If the option is checked, only the mono portion (the center signal) is considered when changing the panorama setting. The panorama control functions as a directional mix controller. A centered vocals recording can thereby later be moved to the left or right of the stereo balance. The difference containing the sound sources allocated outside of the centre, remain unchanged.

In the inactive state, the panorama controls work as usual. The complete stereo signal (mono portion and the difference) is altered.

Phasecorrel. (Phase correlator): This open the phase correlator. This option is very useful when Solo mode is switched on: The base width and panorama setting for each frequency band can be checked visually.

Parameters for stereo manipulation

The following is available for each frequency band (bass, mids, highs):

Base width controller: Set the base width between 0 and 200. 0 indicates "Mono", 100 corresponds with unchanged base width (Stereo), and a value of 200 produces the maximum base width (difference signal).

Depending on the correlation between left and right, the level may increase when reducing the base width. In more extreme cases, this may cause maximum

correlation, which occurs when the left and right channels are identical. When the base width is set to zero (mono), a level increase of 3 dB will occur.

Raising the bandwidth (values over 100) diminishes the mono compatibility. If the base width is reduced, mono compatibility will be maintained.

Panorama controller: Adjust the individual channels of the panorama here. Dampening for left and right is displayed above the controller in dB.

If the direction pan mode is active, the controllers operate as direction mixers. In this case, only the mono component (mid signal) will be included.

Filter setting parameters

Separation frequencies: Use these two sliders ("1" and "2") to adjust the separation frequencies of the three filter bands. The values are displayed in kHz (these correspond with the intersection of the neighboring frequency curves).

Band separation: This parameter influences different filter properties to increase precision. Select between "Low", "Normal", and "High".

Increasing the band separation setting has the following effect:

- The edge steepness of the filter curves increases and the transition range between two bands decreases.
- Dampening in the cut-off range increases (low setting: approx. 25-35 dB, normal: approx. 35-45 dB, high approx. 55-75 dB).
- The ripple of the frequency range of the bands decreases. However, this is not a problem, since the ripples of the individual filter bands compensate each other after combination, thanks to the complementary filter technology. In any case, the output signal will not contain ripples.

Multiband Stereo Enhancer in the master area of the mixer:

Access the stereo enhancer in the master area of the mixer by pressing the "**StEn**" button, which is located above the master faders. Right-click this button to open the stereo enhancer dialog window. The knob in the mixer is connected to the base width controller of the mid band.

If Multiband Mode is not active in the dialog, the changes affect the entire signal. In this case the knob functions as a standard base width control.

Stereo Enhancer - Tips and Tricks

General info concerning base width settings and standard use of the multiband stereo enhancer

- The human ear is generally incapable of localizing frequencies below 300 Hz. Stereo effects in the bass range can be disruptive as different delay levels often result in cancellation, which becomes audible in spongy and squishy playback of the basses.
- The treble range may be important for directional ; however, spreading the base width often causes annoying side effects in this case.
- For these reasons, the base width is usually enlarged or the direction of a mono source is changed in a middle band.
- In the bass area, the base width is set to mono by default. Enlarging the base width is only useful for special applications.
- The treble band is maintained by the following standard applications and is not changed.

Increasing the stereo base width

For this task, set the basic width controller of the middle band to a higher value (between 101 and 200).

More powerful bass by reducing the base width in the bass range

Set the base width controller of the bass band to mono. The setting of the lower separation frequency is also important for this task. Typical settings for this application are between 300 and 600 Hz. If the separation frequencies are higher, narrowing in the stereograph will already be audible, depending on the properties of the audio material.

Checking and correcting a completed mix in the stereograph

To check and possibly correct the base width and the panorama setting in various frequency ranges, proceed for all three bands as follows:

1. Activate "Solo" mode.
2. Open the phase correlator during real-time preview.
3. The base width may be estimated according to the amplitude of the display. The horizontal amplitude increases in relation to the vertical amplitude as the base width increases. The panorama setting may be verified by the gradient of the display. The display tilts to the side of the stereo channel with the higher level.
4. If necessary, correct each channel with the panorama and base width controllers.

Positioning a mono source in the panorama (mix direction)

The first step is to "filter out" the mono source (for example, vocals) so that the remaining part of the mix remains unaffected.

1. Activate "Solo" mode in this case as well and select the solo switch for the mid band.
2. Change both separation frequencies so that they just about pick up the source.

3. Now activate "Direction pan" mode so that the panorama controller acts as a direction mixer.
4. With the pan controller of the middle band, you may now position the mono source. The differential signal is retained.
5. Deactivate "Solo" mode to hear the effect of this frequency-selective directional mix on the overall signal.

Dampening or removing the mono signal from the middle frequency range

For instance, if you want to create space for adding vocals or solo instruments in the middle range of playbacks and drum loops, cancel out or dampen the original mid signal in the mid range by enlarging the base width.

As described above, attempt to "filter" out the mid signal well in "Solo" mode. The base width should then be set to the highest value (200) in the mid band.

Invert Phase

This function inverts the sample data within the selected range along the amplitude axis, i.e. the phase is inverted. Negative values are reversed to become positive values.

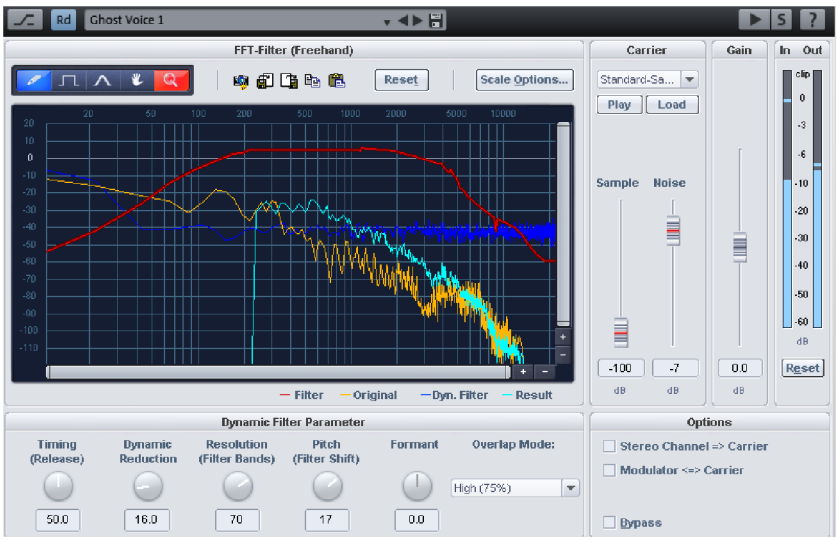
This way enables customization of samples with differing phase lengths. The function may be reversed and applied using both channels, just the left, or just the right in wave projects.

eFX_Tremolo Pan

Detailed information on eFX_Tube Stage can be found at "essential FX > Tremolo Pan (view page 994)".

Modulation/Special

Vocoder



The vocoder sends a carrier signal (e.g. a string instrument or a synth chord) through a modulator (e.g. speech or a song, or even drum loops) to make it seem as if the string instrument is speaking or singing.

This is achieved by transmitting the modulator's frequency characteristics to the carrier. The modulator signal is divided into a number of frequency bands and apportioned positions at regular intervals in the respective frequency bands. These measurements control a filter for the carrier that correspond to the same frequency bands.

Strictly speaking, a vocoder has two inputs and an output. Since Sequoia effects normally only feature one input, the carrier signal will be extracted within the effect to make it mixable with the white noise and any wave file. The carrier and modulator may also be swapped in order to use the vocoder input as the carrier. The vocoder also features a real-time FFT filter for editing the vocoder signal.

Filter curves

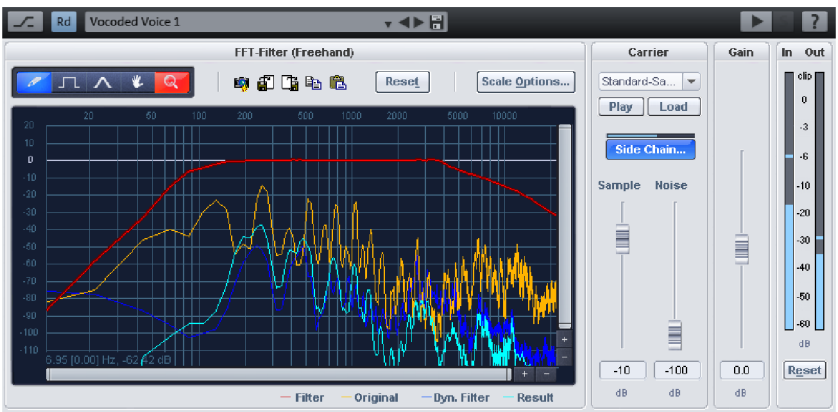
The yellow line corresponds to the frequency progression of the modulator, and the light blue line represents the carrier signal. The red line is a freely-adjustable FFT filter. The frequency progression may be drawn to optimize the vocoder results. The dark blue line is the resulting filter curve of the vocoder.

Reset: This resets the red filter curve to its initial state.

Carrier signal

Sample selection list: The desired carrier samples may be selected here. All open wave projects appear in the selection list as well as some special carrier samples that are loaded from the "Vocoder" directory. Preferred carrier samples consist of material with even frequencies e.g. orchestral chords, broad synthesizer sweeps, sound of the wind, etc.

Sidechain: The sidechaining option is now available for the vocoder if the effect is used **as a track or master effect** and the track isn't the Surround master.



Sample (dB): Use this fader to adjust the proportion of the carrier sample.

Noise (dB): This controller mixes white noise into the carrier. This is especially useful if the carrier material can't be modulated well or sounds too uneven. Whispering voices may also be produced in this fashion.

Volume (dB): Adjusts the vocoder output level.

Vocoder - options

Modulator <=> carrier: Swaps modulator and carrier signals. This is particularly useful if the "stereo channel as carrier" option is used.

Stereo channel as carrier: If this option is active, the sample from the selection list will no longer be used as a carrier signal, but an input signal channel will be used instead. The other channel will continue as the modulator.

This results in a more accurate synchronization of the carrier and the modulation signal, which is independent from play start.

Bypass: The "Bypass" button plays the original signal.

Dynamic filter

Response time (release): Affects the speed of the dynamic filter adjustment to the modulator spectrum. As the release value increases, the Vocoder follows the modulator slower and slower, and the sound changes sound softer and feature more reverb in the carrier. To improve the clarity of spoken words, this parameter should be set to a low value.

Dynamic (reduction): This parameter affects the dynamics of the modulator signal for reducing the modulation depth of the dynamic filter.

This prevents two often undesired side effects of modulation: on the one hand, the volume change of the modulator signal is added to the output signal in a slightly more moderate form, which may improve the power of the Vocoder voice. On the other hand, the low-level portions of the modulator signal are ignored in order to prevent modulation of the carrier by breathing or noise.

Instead of using dynamic reduction within the Vocoder, you can adjust the modulator signal dynamics (or the resulting signals) with the dynamic tools integrated in Sequoia.

Resolution (filter bands): Dynamic filter resolution (approximately dependant upon the number of filter bands). The best results are achieved with medium to high rates.

Pitch (filter shift): The dynamic filter of the Vocoder shifts the frequency up or down to create certain pitch effects. For the clearest speech results, it is recommended not to change this parameter.

Formant: Stretches the dynamic filter curves to manipulate the formants. This changes the characteristics of the Vocoder voice.

Overlapping: This internal parameter changes the overlapping of the time window for the modulator signal spectrum calculation.

On the "low" setting, the Vocoder sounds softer and more melodic, but some of the clarity of speech may be lost.

eFX_Chorus/Flanger

Detailed information on eFX_Chorus/Flanger can be found at "essential FX > Chorus/Flanger (view page 979)".

eFX_Phaser

Detailed information on eFX_Phaser can be found at "essential FX > Phaser (view page 980)".

eFX_Vocal Strip

Detailed information on eFX_Vocal Strip can be found at "essential FX > Vocal Strip (view page 995)".

Corvex

More information is available in "Effects -> MAGIX plug-ins (view page 977)".

Backward

With this function, the data of the sample in the wave project within the selected area/the selected object in the virtual project along the time axis is reversed so that it can be played from back to front.

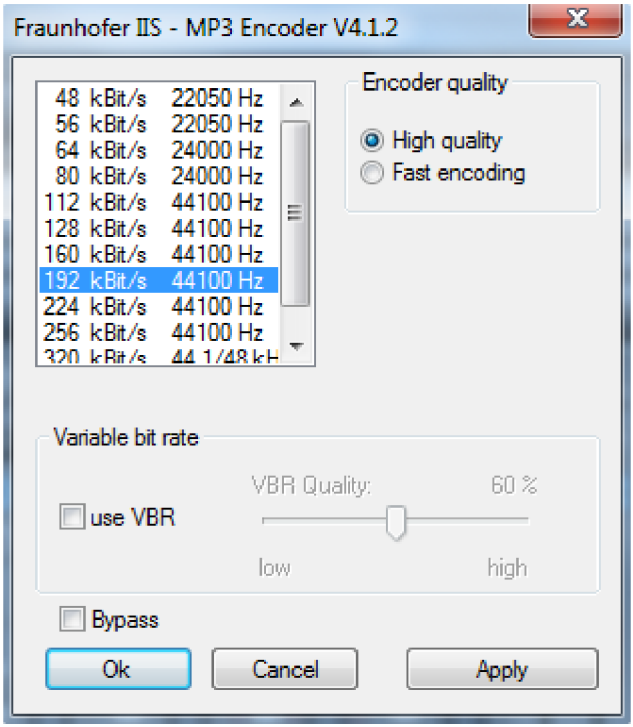
Encoder Preview

MP3 Preview in the Mixer Section > Master Plug-ins

MP3 Preview Plug-in: In the master plug-ins section of the mixer there is a preview function available which can be used to prepare MP3s or AAC files for export. This makes it possible to hear in realtime what the generated MP3 file will sound like after encoding.

During the mastering process you have the option of taking the idiosyncrasies of the encoder into account and adjusting the quality relative to the requirements.

You can change the export settings "Bit Rate" and "Encoder Quality" directly in the dialog window.

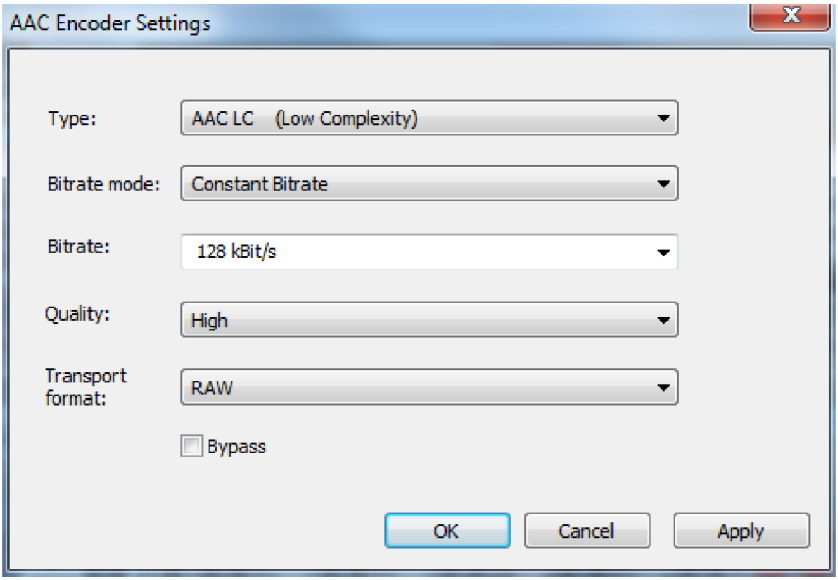


AAC Preview in the Mixer Section > Master Plug-ins

AAC Preview Plug-in: In the master plug-ins section of the mixer there is a preview function available which can be used to prepare AAC files for export. This makes it possible to hear in realtime what the generated AAC file will sound like after encoding.

During the mastering process you have the option of taking the idiosyncrasies of the encoder into account and adjusting the quality relative to the requirements (e. g. „Mastered for iTunes").

You can change the export settings "Type", "Bit Rate Mode", "Bit Rate", "Quality" and "Transport Format" directly in the dialog window.



MIDI Velocity Dynamics

MIDI Velocity Dynamics Settings...

Detailed information about MIDI velocity dynamics settings is available in "MIDI editors -> Velocity dynamics MIDI functions (view page 309)".

Apply MIDI Velocity Dynamics

This command applies the current MIDI velocity dynamics settings to MIDI objects.

Set MIDI Velocity to Fixed Value

This command applies the current MIDI velocity dynamics settings next to the "Set value" button to selected MIDI objects.

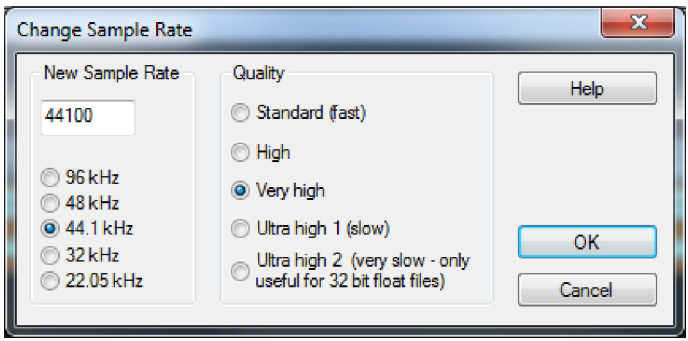
Randomize MIDI Velocity

This command applies randomly varied MIDI velocity dynamics to selected MIDI objects.

Sample manipulation

Change Sample Rate... (Offline)

Use this function to change the sample rate of a wave project.



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

If, however, you reduce the sample rate, a loss in harmonics quality occurs. When halving the sample rate of a 44.1 kHz sample to 22.05 kHz, the frequency response of the resulting signal is limited to 11.025 kHz. The frequency response always corresponds to half of the sample rate.

By clicking on "OK", you may enter the file name for the new project.

Quality

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

Standard: At this quality level, a faster interpolation algorithm is used, which is also applied during real-time 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: In this case, 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 use of highest quality settings even in realtime use (object resampling) that were previously only available for offline processing in predecessor 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 32Bit float files because the maximum possible artifacts from aliasing are below 96 dB.

Note: You can carry out the sample rate adjustment while recording and playing. The quality of the resampling used in this case can be set in "File > Program Preferences > Resampling Quality Options... (view page 707)".

Backward

With this function, the data of the sample in the wave project within the selected area/the selected object in the virtual project along the time axis is reversed so that it can be played from back to front.

Makes Loops (offline)

With this function you can call up a complex algorithm for loop editing in wave projects.

First, select a range that roughly shows the loop positions. If you have selected "Loop" in the transport window and move or edit the range on playback, you'll be able to find some good loop points. Select the "Range menu > Split range" view. Here, the sample is displayed in three sections.

"Build Physical Loop" allows the borders of the range top be set to the precise crossover point of the sample in order to get loops without any crackling. Moreover a crossfade of the audio material at the loop end is performed with the audio material from the loop beginning, to ensure a smooth transition.

If a marker is placed before the highlighted range, the range between the marker and range start will be used for the crossfade. Therefore you can also manipulate the length of the crossfade, like when sampling. If you move the marker close to the start of the range, you will get short crossfades, if the marker is placed far away from the start of the range, long crossfades will be the result.

For a crossfade you should bear in mind that the distance between the marker and the range start cannot exceed the length of the range itself.

Plug-ins...

You can find more information about VST plug-ins in "Mixer > Effect Routing / Plug-ins Dialog -> Effect Sequence / Plug-ins (view page 388)".

Common controls

Parameter smoothing/ Controller knobs

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

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

Essential FX Suite

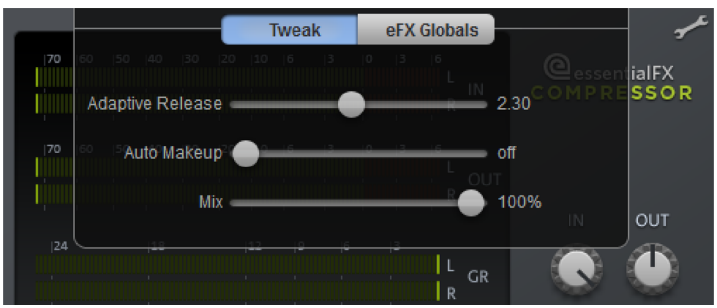
The essentialFX Plug-in Suite offers high-quality audio tools based on state-of-the-art algorithms. They have been designed with special emphasis on a clean and simple display of the individual parameters. The uniform control concept combined with low CPU resource demands makes this effects suite the perfect tool for a wide range of studio tasks.

essentialFX Settings

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

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

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



eFX Globals

These settings apply to all Essential FX.

Metering: Here you can set the meter display.

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

Mouse: Here you can set the display reaction to mouse movements in the essentialFX graphic display.

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

Chorus/Flanger

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

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



Chorus flanger parameters

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

mode:

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

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

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

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

Tweak

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

Phaser

The phaser is often mistaken for a flanger due to its typically sharper and cutting effect. In any case, the pitch is not modulated. Instead, the modulation process burrows multiple notches into the frequency response, somewhat like a comb filter. Just like an airplane taking off, the phaser functions with a similar jet effect. It is suitable for enduring signals like synth surfaces or for producing sound designs to create atmosphere or distortion effects.



Phaser parameters

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

Mode: The selection includes a number of filter stages. At **4 stages**/**8 stages**, a more plastic effect is achieved, and more complex patterns are reached at **16 stages**. Please note that the more stages are involved, the more computing time will be needed.

- **Rate:** Speed of filter modulation. The essential effect is the same for both chorus and flanger.
- **Depth:** Similar to chorus/flanger, whereby it's the filter notches that are addressed, and not the pitch modulation.
- **phase:** This fader moves the right channel's oscillator phasing relative to the left, wave is put back to the right. So that the Tremolo effect drifts apart in the stereo field with ever increasing values. At 180° both oscillators work in reverse, therefore the stereo effect is at its strongest.
- **Feedback:** The feedback portion produces a more drastic effect in this case. Similarly to the chorus/flanger, co-phasal or opposite-phase feedback is possible.
- **Mix:** Regulates the mix ratio of the original signal and the delayed portion.

Tweak

- **Center Freq:** Sets the mid-frequencies of the phaser. The filter modulation acts on these frequencies.

Reverb

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

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



Reverb parameters

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

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

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

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

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

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

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

Tweak

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

Stereo Delay

The stereo delay is a simple too for typical bread and butter delay effects. The stereo delay offers the “analog algorithm,” which features the sound of echo devices of yesteryear.



Stereo delay parameters

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

Mode: This selects between the essential algorithms.

Digital: Normal, transparent delay

Analog tape: Analog tape delay simulation. In this mode, a band echo is simulated with a typical compression and saturation behavior, including phase shifts with high feedback settings.

Analog BBD: Simulation of a bucket brigade delay (BBD, bucket brigade delay). These devices, which originate from the pre-digital era, used analog building blocks for storage. The signal was held for a short time in a relatively simple circuit and then moved on to the next. This “bucket brigade” principle created a longer signal delay. BBDs create different delay times by varying the system beat (clock), meaning that for short delays the beat is faster, and for longer it is slower.

- **Tempo sync:** If this button is active, the plug-in is directed at the host/sequencer tempo. In this mode, changes are made to the delay period via the L/R delay using the musical snap grid (e. g. 1/4 note).
- **Delay L/Delay R:** Specify the delay period for the left and right channels here.

- **Damping:** This specifies the cut-off frequency at which the highs are dampened during the delay. This is useful for making the delays reverberate more naturally or for creating special effects (reggae/dub-style effects).
- **Feedback:** This parameter regulates the internal amplitude of the delayed signal that is fed back to the input. In "Digital" mode, this process is completely transparent; in "Analog," on the other hand, higher values, a very loud input signal, or the sum of these will make the use of dynamics compression audible. In both modes, the nullification of the feedback parameter is in the center of the fader. To the right, the plug-in works in "Dual delay" mode (both sides work independently), and to the left, "Ping pong" mode will be activated (the delayed signal alternates between the left and right sides).
- **Mix:** Regulates the mix ratio of the original signal and the delayed portion.

Tweak

- **Feedback Low Cut:** This control sets the filter frequency of a high-pass filter for the "Feedback." Signal components below this frequency will be filtered out.
- **Tape / BBD Noise:** Here you can control system noise for both analog modes. Especially if you work with high feedback rates, noise can create quick or stable oscillation, making the effect even more authentic.
- **BBD Stages:** Here you can determine the count of memory cells. For very long delays, chips with many memory cells are used. This explains why bucket brigade echoes with long delays sound so muffled and dirty. Please keep in mind that the shorter the delay, the faster is the "Virtual" system beat. For performance reasons, the system beat is limited. The limit is displayed below the delay control (with deactivated "tempo sync") as soon as it is reached, e. g. 46 ms **(min)**.
- **BBD Compander:** Here you can simulate Compander settings. Due to per-cell loss, BBDs have low system dynamics. For this reason, some have an integrated compander (compression at the input, counter-expansion at the output). Strong compander settings interact noticeably with the input signal, especially with high feedback, because the input effects the expansion ration at the output, even when there is no signal at the input.
- **BBD Clock Drift:** This parameter can add drift (during audible jitter effects) by slightly varying the system beat of BBD cells. It works similarly to an LFO, but is randomized.

Compressor

This plug-in is a simple but effective tool for reducing the dynamics of a signal. Percussion tracks may be modeled more compactly and with more pumping, vocals may be integrated into the mix better, or the entire sum or group or signals may be made denser.

In contrast to some other plug-ins, this type works the compressor with a comparatively soft curve and an adaptable regulation process, it therefore compresses completely musically. Additionally a separate input (external sidechain signal) can be defined as the regulation source.



Compressor Parameter

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

Internal sidechain corresponds with conventional methods; this selects the actual input signal to control the compression process.

If another track in the project triggers the compression, select **External sidechain** here. This will use a second stereo track routed to the plug-in. Make the regular adjustments in the arrangement to route the sidechain signals.

Note: Sidechaining is not available in some host programs.

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

- **SC filter:** This fader determines the application/middle frequencies, that are filtered with the control signal for level detection. The filter works selectively as a high-pass, band pass or low-pass. In many cases evaluating the detection is a good for optimizing the leveling process. So, 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 via mids and highs.
- Using the filter type switch the properties of the detector circle can be drastically changed. The speaker symbol is there to preview changes.
- **Threshold:** This sets the response threshold for regulation. For example, -20 dB indicates that the input/sidechain signal is only compressed once it reaches -20 dB; 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 -25dB. This so-called soft-knee characteristic ensures a soft, musical compression process.
- **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 of group or sum signals, while 50:1 produces a very hard limiting that cannot be made inaudible by even the most transparent compressor.
- **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 another area. Note that due to the adaptable plug-in leveling actual times may differ. This semi-automatic process favors quick adjustment, without leading to the often dreaded artifacts (rough sound image with rushed regulation, overly low/inefficient compression with long time constants).

Tweak

- **Adaptive release:** With increasing values, the compressor works increasingly with "Adaptive release". This means that the longer and harder the compressor intercedes upstream in the signal path, the longer is the resulting release phase.
- **Auto makeup gain**: Normally, you have to continuously adjust level reduction to generate "compression" at the same maximum level. This is done by activating auto makeup gain. The volume difference expected from the set working parameters is determined and applied as an output factor after master regulation. If you prefer to adjust the "classic" level reduction and amplification manually, you can deactivate this function.
- **Mix:** Addition of the original signals is done to get transients and spectral balance form the source. A "mixed" signal is particularly discreet, more transparent, and less "squishy" with vocals, whereby the compressed part usually has a higher level reduction than without adding the original.

Gate

The Gate is operated in much the same way as the eFX_Compressor. In particular, the side-chain function, and selecting various filter types can be also be applied here.



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

Gate parameter

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

sidechain int/ext: As with eFX_Compressor an external control input can be used for triggering.

Note: Sidechaining is not available in some host programs.

soft knee: Normally the Gate has a hard characteristic i. e. below the threshold this signal is markedly, 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 and, so that transitions can be made to be more soft and undetectable. This is particularly recommendable for work with acoustic instruments such as drums where the signal levels can fluctuate a lot.

hold: This three-way switch controls how long the gate process waits after passing through the attack phase before subsequently passing through the Release phase. The

Hold Function is useful for signals that need a rather short release time, but shouldn't be flattened out.

- **sc filter:** The control corresponds to the essentialFX Compressor.
- **threshold:** This fader defines the threshold beneath which the gate should be initiated.
- **range:** Here you can set the Gating strength. Right extreme means that the signal beneath the threshold is cut out completely. Subtly attenuating the signal can remove background noises or breathing from vocal tracks. Where necessary the process can be attenuated even further by switching on the soft knee mode.
- **attack:** Regulates the attack time from the closed gate to the point at which signal is let through again.
- **release:** Sets the time, that the gate requires to go from a normal state to gating.

Tweak

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

Limiter

This plug-in 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. You can also use a separate input (external sidechain signal) to define the source for the regulation.

Limiter Parameters

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

- **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.

Tweak

- **Attack:** This sets the time frame between the point when the signal exceeds the threshold and the maximum extent of the effect.
- **Adaptive Release:** With increasing values, the limiter works increasingly with "Adaptive Release". This means that the release phase gets longer according to how long and hard the limiter has been applied to the signal processing.
- **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.
- **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 function.
- **Input Low-Cut:** You can use the fader to set the filter frequency of a high-pass filter for the input signal. Signal components below this frequency will be filtered out.
- **SC Low-Cut:** This sets the filter frequency of a high-pass filter for the sidechain signal. Signal components below this frequency will be filtered out.
- **Clip Shape:** The left velocity of the fader (soft) corresponds to the oscillator sine wave. The oscillator signal is gradually morphed to the right in the direction of the rectangular wave. This makes the effect more intense, i.e. "harder".

Tube Stage

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



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

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

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

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

Both of the available tube stages can be globally switched in the A or AB operation. In the A-circuit the entire signal runs through, as is common with simple tube preamps, through a tube. However, since these can only edit a half-wave, the working point of the circuit is set approximately in the middle of the reference line. However neither

halve will ever be treated equally, so that positive displacements are amplified differently to negative displacements. (so-called Class A operation). Thereby odd and even harmonics are created. The typical 'warm' tube sound to a large extent dates back to the characteristic overtone spectrum.

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

Tube Stage Parameter

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

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

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

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

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

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

DeEsser

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



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

DeEsser Parameters

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

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

Tweak

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

Tremolo Pan

This plug-in is on one hand a Tremolo, that creates typical textures for synths or vintage guitars. On the other hand you can create interesting, rhythmic stereo effects with auto panning.



Tremolo Pan Parameter

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

rate: Sets the speed of the internal oscillator in Hz or note value (see "tempo sync").

shape: The left velocity of the fader corresponds to an oscillator's sine wave. The oscillator signal is gradually morphed to the right in the direction of the rectangular wave. This makes the effect sound intense, "harder".

phase: This fader moves the right channel's oscillator phasing relative to the left, wave is put back to the right. So that the Tremolo effect drifts apart in the stereo field with ever increasing values. At 180° both oscillators work in reverse, therefore the stereo effect is at its strongest.

depth: This determines the strength that the oscillator signal has on the Tremolo/Panning effects.

tempo sync: when the switch is activated the "rate" fader snaps to values on the musical grid.

Vocal Strip

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



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

Vocal Strip Parameter

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

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

Tweak

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

Note: Detailed information about compressor-specific settings can be found under "EssentialFX > Compressor > Compressor Parameter (view page 986)."

Vintage FX: CORVEX – Chorus/Flanger



When referring to the chorus and the flanger effects in CORVEX, we are not dealing with just one of the two effects, but also the variations between them that go beyond the standard concept.

A chorus generates the typical "floating" sound that is commonly used for guitar sounds or synthesizer pads. You can add acoustic "depth" to an instrument and make it sound "fuller" or even create the illusion that several of these instruments are playing simultaneously.

The chorus sound is created with the "Doppler effect". You've probably noticed this phenomenon in everyday life when the siren of an approaching ambulance sounds more high-pitched the closer it gets and then sounds more low-pitched as it drives further away. This effect is a result of the speed of the sound which first increases and then decreases, thus also changing the sound pitch. The wave length also changes relative to the frequency. If there were a second siren at your location, an oscillation would develop between both sounds (just like when two instruments are out of tune).

Chorus also splits the signal into at least two components: a direct sound and an effects portion and there are multiple effects components in CORVEX.

The Doppler effect is created by a short signal delay from the effect. For most devices, this delay is within the range of 10-30 ms (as in this one) This means that it is short enough that it won't be perceived as an echo. The times would also be similarly short if you duplicated a guitar track, for instance. A short delay in the mix with the direct signal already sounds "doubled", but this is not authentic. This is where the above-mentioned "out-of-tune" effect comes in: The pitch of the effect signal is slightly modulated by gently "drifting" forward and backward in the delay curve. This creates a modulation with a liveliness that is influenced by the speed of this "drift".

The "Flange" effect is similar to that of the chorus, but it has a different technical and historical background. It resulted by chance: Someone (various sources say John Lennon) slowed down one of two running interconnected tape machines in a studio with his hand. The result was a rather brief delay of the second signal compared to the first, brought about cancellations within the frequency spectrum, leading to a comb filter effect (the sum of both signals creates "peaks" and "dips" in the spectrum that look familiar to the teeth of a comb).

Flanging is basically a chorus effect with a shorter delay time (less than 10 ms). Releasing or doubling the signals is not the main focus in this case. Here we are interested in a more creative distortion of the frequency response.

A complete flange effect will definitely require feedback: the flange portion is returned to the input to increase the effect. People often refer to this as the "jet effect" since it resembles a jet on take-off.

CORVEX – delay/modulation section



drift: This parameter introduces drift. It works similarly to an LFO, but is randomized.

time: Here you can specify the delay time and fundamentally set if you want to have a chorus or flanger sound. With the flanger, delay times are usually between 1 – 10 ms, as, with these times, the typical comb filter artifacts usually lie within the audible frequency area (Frequency = 1 / delay in ms). For a chorus, values of 20 - 40 ms are normal.

Wars of attrition are fought about what the actual correct delay times are. However, we don't advise you to strictly conform to such standards. Just let your ear decide.

voices: Use this to set how many internal voices the effect should contain. Two to max. eight delay units can be activated here. With more than two voices the sound becomes fuller and fatter. Uneven voices (1, 3, 5, 7) are assigned to the left channel, even voices (2, 4, 6, 8) to the right. The subsequent elements like the filter and diffusion unit also count as part of these voices/delay units. For this reason, you should note that an increase in the number of voices leads to increased CPU strain.

Span: The time of each delay unit activated by 'voices' can be shifted using this controller. Example: With the 'time' potentiometer you can set 10 ms and select four voices. "span" at 50% would mean that voice 2 would be delayed by 15ms, voice 3 by 20ms and voice 4 by 25ms. For example, you can break up the resonance by increasing the "span" value at high feedback rates or break apart the sound field at large stereo width.

The modulation depth: The depth or intensity can be adjusted here using the small fader below "time". Here, you'll see one of the most important properties of these three effects: the small fader on the lower row on the front plate is seen as having the same value. They are basically modulation targets. If, like in this case, the modulation target of the LFO is the "time" fader, you have set the depth of the pitch shift.

At max, the amplitude is at its largest, in the minimum position the effect remains static.

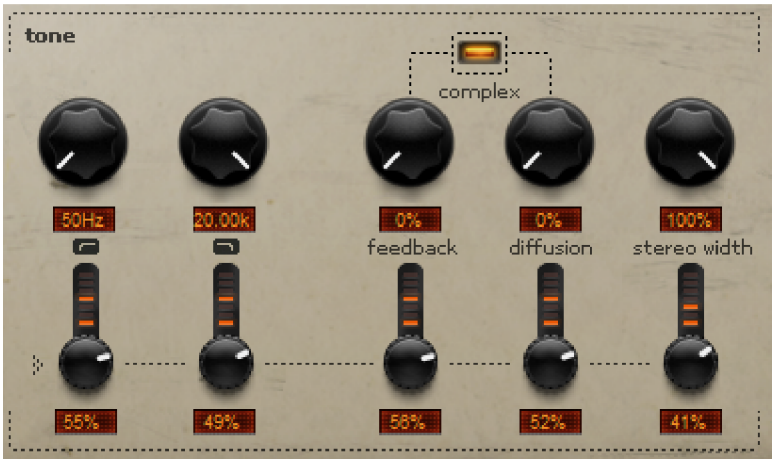


Modulation speed: Slow times create a quiet beat; high speeds sound like vibrato or, in extreme cases, like it's under water. The type of modulation can be selected using the buttons.

The **sync** button snaps the speed to the song tempo. Alternately you can set the modulation tempo in Hz manually.

Waveform selection: sine, square (sqre) and random (rand) are available. The sine wave is suitable for rhythmic, quiet sounds. On the other hand, a square wave sounds quite drastic and very rhythmic. Random mode is a good partner for ambient sounds or sound landscapes, the progressions are not foreseeable and can add an interesting accent to the sound.

CORVEX – Tone/Filter section



The **low and high cut filters** limit the signal within each voice/delay unit.

feedback: Here you can set how strongly the signal in each unit is fed back to itself. At high values you can get a typical, cutting flanger sound. What makes CORVEX stand out from many other digital devices is that high feedback rates do not result in overmodulation. In each delay unit, the signal is fed at a saturation level which brings about a soft, analog-sounding restriction.

diffusion: This is one of the most interesting parameters of CORVEX and rather unusual for this effect range. Normally, larger delays are audible as discreet echoes. Using "diffusion" softens the signal in each voice, scattering its shape. In extreme cases and with sufficiently high "time" values, even reverb-like sounds are produced. This enables a small room to be replicated using "time", "voices" & "span", and "diffusion" can be applied to simulate the natural properties of scattering signals over surfaces. With high feedback values, simply modulate the pitch a little (the small controller below "time") to break up the creation of comb artifacts resulting from static repetitions.

stereo width: Controls stereo width. You can widen the audio field using this parameter.

complex: Usually each voice pairing functions in "PingPong" mode, that is, the left channel is thrown back to the right and vice-versa. In "Complex" mode, the feedback & diffusion parameters are combined, which produces a quite chaotic sound. In "Complex" mode, each one of the 8 possible voices affects every other one, i. e. echo repetitions ensure that diffusion is even faster. Here, at the maximum number of voices, drastic spatial sounds are possible. For this reason, CORVEX manages to eclipse some specialized reverb effects...

CORVEX – modulation depth intensity setting

As mentioned above, this relates to an **intensity setting of the modulation depth**: Each of these faders specifies how much LFO affects the relevant parameter above it.

The following applies:

- The envelope of the LFO affects the "time" parameter directly, meaning that lighting of the round LED in the modulation section results in an increase to the delay times.
- The LFO also affects all other parameters to the right of the middle in a similarly direct fashion. To the left on the other hand, the inverse value of the modulation envelope is taken. If, for example, you have selected "sine" as the waveform, the inverse curve shape will correspond to a sine wave which is mirrored along the X-axis. In practice, this results in a temporal shift of modulation by exactly half of one period length.

Basically, the modulation values combine with the settings of the main pot. For example, to get a rhythmic stereo width introduction, set the stereo width control to 0%. Switch on "sync", adjust the "speed" to "1/4", and set the small fader for the modulation target (stereo width) to the right. This way the stereo picture plays at maximum every fourth note. Now turn the small pot below "stereo width" to the left: the stereo picture now plays on the off beats. This is where the inverse envelope is effective and the mentioned temporal shift is kept intact.

Use the exact same method to modify the other modulation targets and get sounds out of CORVEX that a normal chorus or flanger wouldn't know. We've created a few presets that make heavy use of modulation which can be used to quickly get things right...

Vintage FX: ECOX - Echo/Delay

This delay provides a creative playing style that is different from common delay effects.



The delay times can be changed during playback without creating scratchy digital artifacts. Instead, the times are faded out softly, similar to the old tape echo machines or bucket-brigade circuits that used tape speed to change the delay.

ECOX can reproduce this type of sound relatively easily including tracking fluctuations and loss of highs that always occur during each repetition (feedback).

Like the CORVEX, the internal feedback features a dual-band filter (low and high-cut) for creating dark, light, or mid repetitions (depending on the settings).

A special characteristic of ECOX is that the delay cannot be digitally distorted. Even in an endless "looped" repetition, the signal is not allowed to distort too much. Instead it is compressed by an increasingly slighter degree and distorted like a tape.

ECOX - delay time



drift: This parameter introduces drift. It works similarly to an LFO, but is randomized.

left + right delay time: The delay times can be adjusted separately for left and right. Use the "synced" settings to select the note value which the controller should snap to. The same note values as with CORVEX can be chosen. You can also leave out "sync" and specify free delay times in milliseconds.

The **sync** button beneath it affects the display of the delay time of the left and right channel in the bar grid.

Link button (lock symbol): Press this button to change both channels simultaneously with a delay knob. The link function also has an effect on the two modulation depth faders of this section.

Modulation depth left/right: These two knobs control the modulation depth and influence the pitch of the resulting delays.

ECOX - Modulation Section



The same features may be selected as with CORVEX. The same conditions apply as those mentioned above to change the pitch modulation.

ECOX - feedback path



Filter section, feedback, diffusion, stereo width: see CORVEX - tone/filter section (view page 999).

More information about about modulation depth can be found under CORVEX – intensity settings for modulation depth (view page 1000).

Vintage FX: FILTOX - Multimode-Filter



Much like CORVEX and ECOX, FILTOX is also a "modulation effect". These settings have to do with frequency response bending. Modulation source controls **two filter units: filter left/filter right**.

Possible areas of application are synthesizer sounds (filter sweeps on pads) or creative distortions in drum loops (e. g. for variations, fills, etc). With guitars you can create typical 'wah' effects either by tempo modulation or in a special mode with modulation via the signal envelope curve.

The 'core' of FILTOX is a stereo multimode filter based on an analog model (Chamberlin 2-pole filter), which is known as an Oberheim filter. With FILTOX two of these modules are cascaded per channel to achieve a switchable 24dB roll-off.

Our digital model of the filter is designed so that it delivers the typical "analog" sound character, but it's also useful if you want to apply internal overmodulation. In this case, interaction occurs between the cut-off frequency and resonance, which makes the sound seem "undigital".

FILTOX – Parameters

The following filter types are available:

- Low-pass
- Band-pass
- Notch
- High-pass



Usually, these filter types are designed so that they are able to be toggled. However, a "state-variable" network is used with the Chamberlin filter, which means that all filter types can be taken from "taps" simultaneously. So why leave these taps static when they can be processed dynamically...?

The actual filter circuit provides the following parameters:



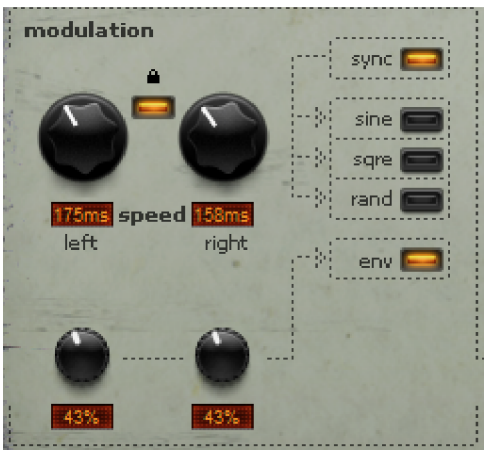
- Cut-off frequency (**freq**)
- Resonance (**reso**)
- Internal saturation (**drive**)



The filter can be set up separately or together with the link button.

You can set up the cut-off frequency or the resonance for each channel as modulation targets. Just like with CORVEX and ECOX, the small knobs on the lower row specify the modulation level.

For the modulation source, in general the same applies as with the previously mentioned effects. In addition, FILTOX provides the modulation of the filter section via an envelope follower. In this case, the input signal itself serves as the modulator. This makes it possible to produce well-known "auto wah" sounds.



To use this envelope follower mode, proceed as follows:

- In the modulation section, press the "**env**" button.
- The two small "**gain**" controllers on the bottom help customize the input level for setting the envelope. The LEDs at the top no longer show the LFO speed, but rather indicate the set sensitivity visually.

- Usually, the sensitivity is controlled so that signal peaks activate the LEDs more brightly. If gain is set too low, you'd have to set the knobs unnecessarily high for the modulation targets. If input levels are too high, the behavior will not be noticeable. Additionally, the problem of overmodulation arises when the detector circuitry of the envelope follower leads to inaccurate tracking as a result of saturation. This tracking is also influenced by the following parameters:
- "**Speed left & right**" controller: These set the LFO speed. In "Envelope" mode, these may be used to set the attack & release of the envelope for each channel. This means that minimum speed settings can cause a quick increase of the envelope to produce quick modulation access. Since attack & release are coupled settings, quick attacks correspond to short release values. This way, the

control voltage for the filter drops more quickly than at middle or slower settings.

- You should adapt the speed as precisely as possible to the signal. Times that are too short can cause errors in tracking and fluctuations, while times that are too long will miss short signal peaks. Internally, however, the detector works semi-automatically for release time, so that the setting is less critical than with a purely manual method.
- The lock symbol in the modulation section has a special meaning in envelope mode: once active, both channels are linked together for detection so that panning effects may be based on different controller settings, and not because of a stereophonic input signal.

Analogue Modelling Suite: AM-Track



"AM-Track" is a combination of an analog compressor and a tape simulator in a single device. The plug-in was developed for specifically editing and enhancing input signals (instruments, vocals). The audio signal may be enhanced and made more lively with a combination of compression and tape saturation in order to assert itself in the entire mix.

An explanation of the compressor section of the AM-Track, its special advantages compared to "conventional" software compressors, and the available parameters follows. We will also explain the features of tape simulation.

Compressor Section

Two completely different compressors work in AM-Track, each with their own independent control and sound methods.

You may be wondering why we mention sound when talking about a compressor, since compressors merely relate to control actions. This isn't as simple as the idea of “making loud quiet.”

Various designs, algorithms, and topologies for solving the actual problem (dynamic reduction), which all have their own unique character, have come from the history of analog and digital signal processing. For example, pre-filtering in the detector circle

and the type of detection have a large influence on the audio results. Plenty of hardware compressors have the same established VCAs (voltage controlled amplifier), but they all sound different and influence a signal, an entire production (or even a genre) with their "signature sound." We intend to provide you with acoustic variation via these dynamic tools in the digital world.

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

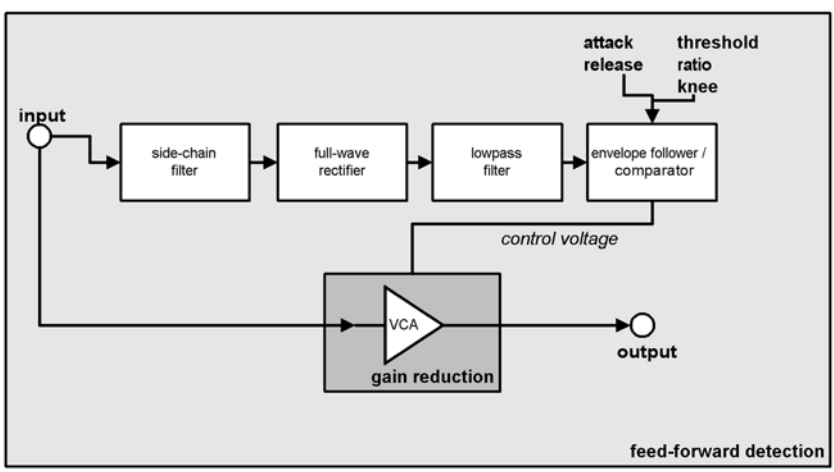
VCA Mode



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

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

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



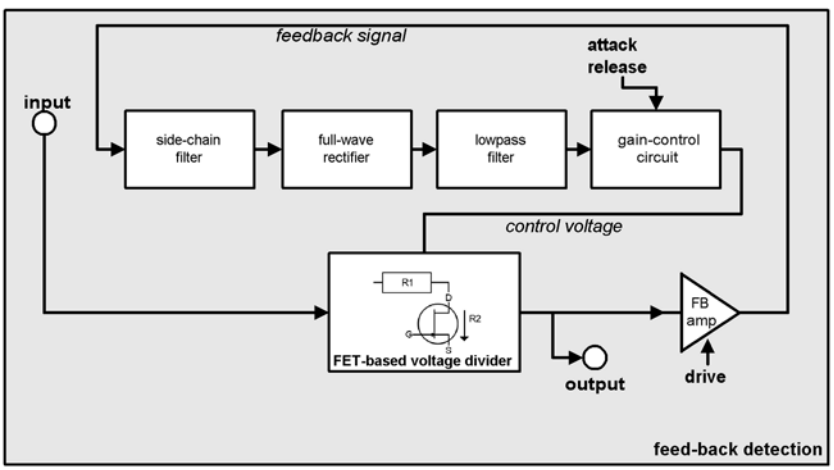
In contrast to this, there is a "feedback" method which provides a second compressor mode.

VINTAGE mode



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

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



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

Because of the feedback topology, the maximum gain reduction is usually lower than VCA devices with forwards detection, usually 20 dB. This way, there is almost always a level matching amplifier in the feedback loop. The “drive” controller of the AM-Track regulates the feedback amplification here. This can be so high that the detector may become saturated by a loud input signal, resulting in signal peaks being

swallowed up. Simultaneously, the setting becomes more intense as quieter signals also start reaching the threshold. You can creatively implement this according to the situation to create complex signal compression which doesn't much sound like dynamic compression due to the transients that slip through and release at high “drive” levels.

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

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

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

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

Compression Parameters

VCA Mode

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

- **Threshold:** The threshold above which dynamic reduction begins.
- Check the threshold display if necessary (**thr**): If the input signal reaches the set threshold, the blue dash will move around the arrow symbols. If this dash moves upwards, the threshold is below the average level and compression is active. Inversely, if the dash moves below the marking, the input signal becomes too quiet to be able to reach the threshold; compression will be applied.
- **Ratio**: This ratio (1:n) specifies by which factor the signal should be reduced once the threshold has been reached. For example, if the threshold is set to -20 dB and the ratio to 1:4, an input signal of -10 dB will only be amplified by 2.5 dB (10 dB : 4 = 2.5 dB).

- **Attack**: This is the response time, e.g. how long the project takes to execute the required level reduction. Short attack times intercept level peaks, and longer ones let them through unimpeded (compression only starts past this value).
- **Release**: This is the time allocated to the circuit to reach the normal amplification factor.

Note on attack & release: In general, short attacks are used for moderate compression and making the transient response softer; longer times retain the "bite" of a specific instrument at larger compression rates or make the sound a bit snappier. With more difficult sources, like a very dynamic vocal track (ballads), for example, you can use a longer attack so that the project runs more smoothly and quietly; the release time may be trimmed audibly to match the pauses or the song speed.

Shorter release time may be used for modern, aggressive "close up" vocals, e.g. when breathing sounds are an important stylistic device and the voice should sound very full and compact.

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

Vintage Mode

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

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

Compression Expert Settings

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



- **Look ahead**: AM-Track is always ahead of the signal. You can specify how many milliseconds you want to "look ahead." The audio signal path is delayed according to the signal route so that the detection circuit is fed first with the input signal (so-called "Look-ahead delay"). You can increase the attack time and still avoid fast peaks. The latency compensation in the host program ensures that other tracks in the project are adjusted and that no time delay occurs. For percussive signals, you can even set the delay all the way to "0."
- **Detector hp filter**: This high-cut filter is positioned before the two compressors' detection circuit. You can use it to specifically exclude basses and mids from these rules. Complex signals with bass and hi information like a subgroup or complete mixdown produce fewer "Pumping" artifacts. This is because low-frequency signals feature the most power and therefore always trigger regulation and modulate other frequency ranges in the volume
- **Auto makeup gain**: Normally, you have to continuously adjust level reduction to generate "compression" at the same maximum level. This is done by activating auto makeup gain. The volume difference expected from the set working parameters is determined and applied as an output factor after master regulation. If you prefer to adjust the "classic" level reduction and amplification manually, you can deactivate this function.
- **Adaptive release**: This is "semi-automatic," i.e. you can roughly adjust the release time, and AM-Track reduces it according to the current signal power from "a little (1%)" to "Considerably slower (100%)." In "Vintage" mode, this regulation method is particularly intense, since it affects the feedback loop process. For instance, if you are editing vocal tracks or dense, complex material, it can sound "calmer" or more "musical" if adaptive release is activated.
- **Capacity**: Adjusting the "capacity" controller sets the time response of the "adaptive release." The greater the capacity, the more sluggish the release adjustment. You can therefore influence larger parts of the compensation response. For instance, if you want to use vocals that have been "moved forward," you should use a short release time (maybe 80-100 ms) and a greater value for semi-automatic (e.g. 80). Vice versa, you can reduce automatic feed by switching the relation (smaller capacity, generally greater release time).

- **Comp mix**: Parallel compression is a popular “studio trick,” particularly with complex material. Adding the original signal retains the transients and spectral balance of the source. You can add compression by turning the mix controller. A mixed signal is particularly discreet, more transparent, and less “squishy” with vocals, whereby the compressed portion usually has a higher level reduction than without adding the original.

Tape Section

The tape simulation in AM-Track comes after the compression section and offers you the opportunity to give your recordings an "analog touch" by reproducing typical aspects of a tape recording.

What also happens in this case is that the magnetic storage space of the tape becomes exhausted and the signal more distorted when the recording level is increased. There are other factors as well however, e.g. pre and de-emphasis. Since saving on tape happens depending on the frequency, pre-emphasis makes sure the dynamic area is used to its full extent, according to a normed characteristic curve (e.g. NAB, EBU). This is because the signal cannot simply be transferred unfiltered at high levels onto the media via the recording head. The pre-filter creates a characteristic harmonic spectrum for possibly intended overmodulation by the user, which then changes according to the level of saturation.

With the de-emphasis circuit, the pre-filter may be undone with an inverse characteristics curve on playback. However, distortion caused by tape saturation might shift the spectral balance.

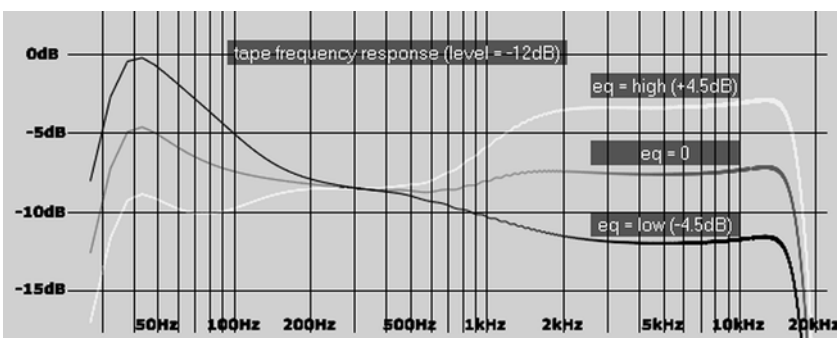
As a result of the enormous complexity of the process of magnetic recording, additional factors also influence the acoustic result, like, e.g. pre-magnetization (bias). This relates to high-frequency voltage swing (usually in sine form at 150 - 200 kHz) which is then applied to the tape via the erase head before the same part passes by the recording head a few centimeters later. This erasure current ensures the equal orientation of the magnetic particles and keeps the hysterisis loop intact, which is important for the magnetization process, and functions qualitatively. Low currents do have a brighter sound picture as a result, but magnetization is inefficient and the maximum recording level is therefore quite low. On the other hand, a bias current which is too high is associated with losses in highs, but does allow for higher levels with less distortion.

In addition, memory effects when recording tapes are responsible for a part of the characteristic sound, since time-dependent factors play a role in the feeding of the tape reel along the heads, e.g. mutual inductance and self-erasure.

Regarding the simulation of these processes, we have concentrated on some aspects of the "real world" and have created a virtual "machine" in AM-Track, which permits the following interventions by using the controllers on the interface:



- **Level**: Sets the input level. This determines when the "tape" is saturated and how strong the effect of the coloring/soiling is. You simultaneously gain more loudness.
- **EQ low/hi**: Adjusts the frequency response (spectral balance controller). You can choose whether you would like to have an output signal of high bass level or whether it should have more highs. This alters the pre-emphasis at the "recording end" as well as playback equalization. In "Expert" mode, you can vary the employed frequencies for lows and highs for distortion and equalization. However, note that the frequency response for the simulation will not be neutral, even if the "EQ Low/Hi" controller is neutral. There will always be some slight frequency-selective amplification.
- **Bias**: Moving the operation point (bias adjust). Increasing the bias will result in a higher "recording level", but tape saturation will occur sooner. You will also increase the above mentioned loss effects, and the result will be a (dynamic) reduction of the highs. Turning the bias controller to the left results in the opposite effect: the highs are not reduced, but the signal level stays lower.
- **Tape mix** (in the "expert" section): Anything that applies to a parallel compression can also be applied to the tape section. Non-edited transients in particular are responsible for perceiving "speed", "liveliness" and "freedom", but may be lost in case of too much saturation/overdrive. Mixing in the original slightly enables you regulate the band quite far while still retaining the so-called attributes.



Tips for Applying Tape Simulation

The harmonics generated during saturation can quickly cause a state of "acoustic fatigue" for the listener, particularly for high-frequency rich material and/or a frequency shift at the benefit of the highs. A direct 1:1 comparison with the tape section switched off reveals slight differences. Subtle settings are generally sufficient for complex signals in order to create a slightly "analog touch".

Use the tape simulation as a "peak stop" if you use one of the two compressors to compress the signals: Transients that the compressor allows through (e.g. when using longer attack times) may be gently blocked by "taping" the signal afterwards.

AM-Track also contains a soft clip circuit at the output to block overmodulation. The audio result is certainly not the same as tape saturation. In practice, the soft clip function stays switched on to provide an emergency brake, especially if the plug-in is in the last station of the master slot. The soft clipper limits the signal with a soft characteristic curve at -0.1 dBFS. It's possible to switch off the function if you can predict good level responses or if the following stages in the mixer/master chain will not have a problem with higher levels.

Analogue Modelling Suite: AM-Pulse



AM-Pulse is a "transient modeler", i.e. a creative tool for editing envelope and sustain processes (attack and sustain) on percussive or dynamic signals.

The oscillation of the signal in particular is an important factor in acoustic perception. So-called "transients" are extremely short oscillations in the make-up of a sound that enable the human ear to precisely differentiate sounds from one another (and to localize where they come from).

Besides transient modeling, this effect also provides the option to polish the signal or to distort it completely. For this to work in AM-Pulse, individual steps have been set up as "virtual hardware" and designed using elements of models, and components have been "borrowed" from the analog world in order to achieve smooth, organic operation and character.

Possible instances when AM-Pulse can be used:

- Drums (for example, kick, snare, toms, subgroup signal): Increasing or decreasing the attack leads to a harsher or softer sound respectively, and increasing or decreasing the sustain changes spatial information for overheads or ambient tracks.
- Acoustic or electric guitars and bass: Accentuate or dampen the attack, e.g. for playing like a plectrum.
- Balance volume fluctuations and explosive sounds for sound recordings
- Reduce background noise

Working Method of the Transient Section

At first glance, it seems as if AM-Pulse has a dynamic compressor working "beneath the hood". However, this is not the case. In comparison to conventional compressors, AM-Pulse is level-independent. For example, if you reduce the attack rate of a snare drum by 6 dB, this is independent of how loud it was played. However, you have to determine a threshold as if for a compressor. No editing takes place below this threshold.

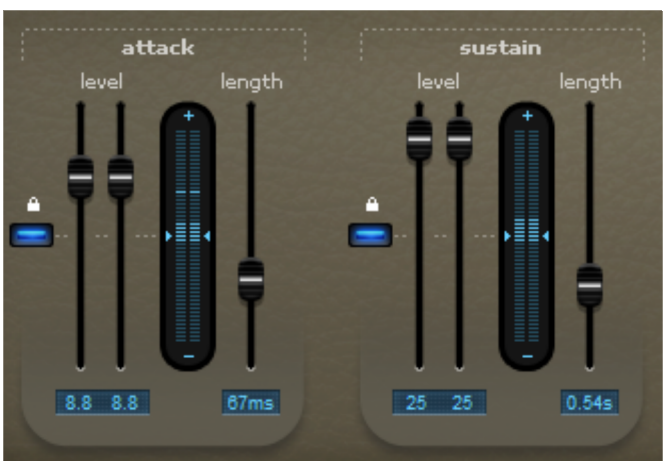
The possibilities mentioned for using AM-Pulse are based on the principle of "transient recognition".

AM-Pulse continuously analyses the input signal by means of so-called "envelope followers" that sample the time-based signal. At the attack and sustain phases, there are multiple envelope followers working which possess different attack and drop-out times. Attack and sustain can be safely recognized by means of continuous comparison measurements.

Internally, the AM-Pulse uses virtual VCAs ("voltage controlled amplifiers") that generate a control voltage from the resulting envelopes. You can use the fader to apply its value to the attack and decay rate. This voltage may be applied via sliders. This multiplies the input signal via the control voltage (controller in positive direction) or divides the signal by the voltage (negative direction.

Based on the principle of transient detection, the dynamic signals described above are especially well-suited for editing. The more percussive the audio signal, the cleaner and more predictable the control.

Transient Section Parameters



There are three faders on the front plate for processing attack and sustain:

- **Level** (left channel): An increase takes place above the line; moving the fader in the opposite direction reduces the level. The signal remains unaffected in the centre position.
- **Level** (right channel): As above. Normally, both channel faders are coupled together by default (as are the internal "detectors"). You may edit the channels separately by depressing the button with the padlock symbol. The control voltages for both channels are then determined separately.

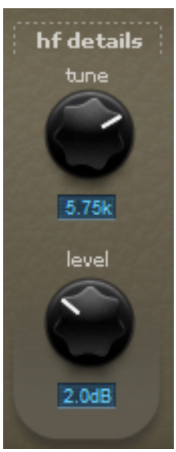
- **Length**: This determines how long the signal is scanned and maintained in the respective section. A lower value results only in a short amplification or reduction which sounds quite electronic. Longer times mostly sound more homogenous.

Note: High "length" values in the "attack" phase are especially in danger of being affected too strongly. The more complex the initial material, the more carefully "length" should be adjusted.

Saturation and HF Details Section

As a creative tool, AM-Pulse offers an adjustable saturation level and a "high-frequency details" module.

Both sections are located within the circuit after transient processing. Modeled sounds can thereby be processed additionally. There are many ways to combine these three sections to create unique sounds. We will describe a few of these scenarios below.



For **hf details**, exciter circuitry may be used to add artificial harmonics to signals at the set frequency.



The **saturation** section works similar to a tube pre-amp. This features the typical "tube-like" volume compression behavior and "drives" the signal into saturation at high levels. The "saturation" controller mainly determines the input gain of the tube stage. When turned fully to the left, no saturation occurs and this section remains inactive.

For the amplification of the signal, odd and even harmonics are created. The output signal first becomes louder and "richer". It becomes rougher and "dirtier" as the amplification increases, and finally sounds drastically distorted at an appropriately high input level.

The saturation of the signal is frequency-selective in this case: how much has to be pre and post filtered depends on the setting of the "saturation" fader. The more "saturation" the higher the filtering and sound modification. The internal degree of distortion is dependent on the setting of the level fader.

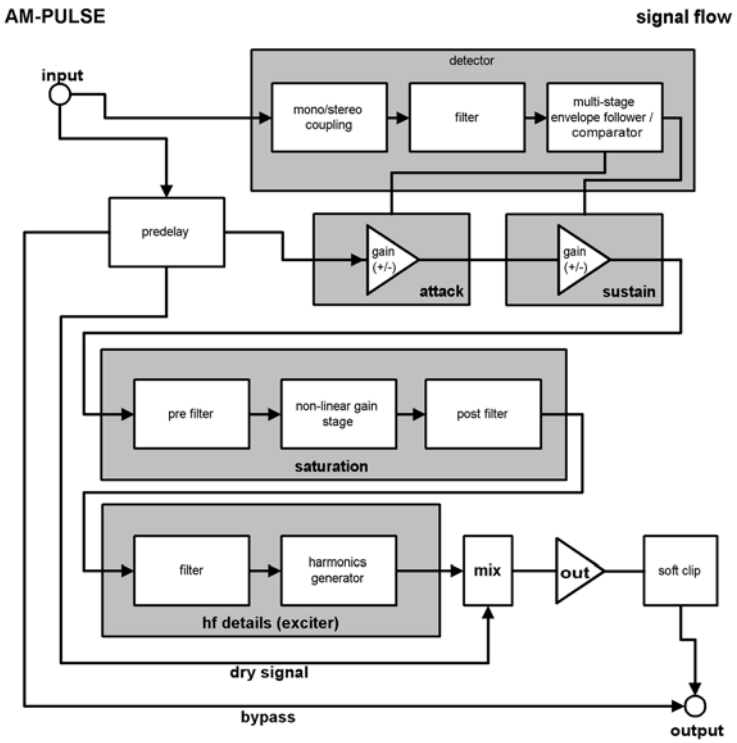
Use the **mix** fader to set how much of the processed signal is in the entire signal.

Creative Tips for Working with AM-Pulse

We have added a few presets to AM-Pulse for selection from the preset menu of the console. Several of these are ideal starting points for the following experiments:

- Attack & sustain are often contrary processes. For example, if you would like to add more "bite" to electronic drums by accentuating the attack, experiment with reducing the sustain level. Both sections then often require less "drive" and the sound generally has far calmer effects than only one of the two sections, but at double the level.
- To increase loudness, sounds with a lot of attack but a low average level may only be "tamed" with the saturation level. A few "saturation" dBs are usually sufficient.
- Try to simulate several amplifier types: combine the transients and saturation section. With a slight reduction in attack (about 2-3 dB) and a slight saturation (maybe 6 dB), you can achieve results featuring soft signal edges, slightly increased loudness, and a bit more "life" in the signal. You could possibly add 1 - 2 dB "HF details". The result would then come very close to a tube amplifier.
- If you would prefer slightly more filtering from the saturation section (the above-mentioned "loudness") but less "crunch", move the "input" fader down slightly and increase the "saturation" controller correspondingly. Keep an eye on the peak meter in this case, too. Subtle changes already have a large influence on the overall impulse behavior of the signal and slight colorations can even be heard in the A/B comparison.
- Set the attack and sustain of the left and right channels to opposite settings so that the signal "moves" in the stereo field according to its own transient data. This "auto panning" function even works with mono sources and is ideal for creating new and unique loops.
- Drastic changes can be smoothed with the mix fader. Among other things, entirely new sounds can also be created. The result usually sounds different than a simply softened setting of the fader at 100% mix. For example, you can use sustain with acoustic drums to break out of the room, but keep the integrity of the recording with targeted mixing of the original.

Much like AM-Track, AM-Pulse contains a soft clip circuit at the output for making sure overmodulation doesn't occur. The method mentioned with AM-Track also applies here.



Analogue Modelling Suite: AM-Phibia



AM-Phibia is a tube amplifier/channel strip. It combines an optical compressor with a pre and post filter unit. By selecting the filter presets the suitable setting can be chosen depending on the input signal. While interacting with the compressor section am|phibia can be used as a vocals pre-amp, tube guitar amp or simply for creating a warm sound when mixing or mastering.

Typical examples AM-Phibia use:

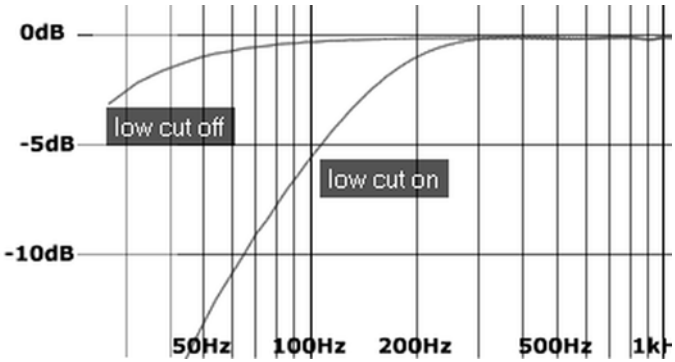
- an optical compressor (models classic LDR-based light sources)
- a valve pre-amp
- a sound pre-processor with corresponding EQ and compressor settings
- valve guitar modeler with pre-amp, sound processor and case simulation
- a signal processor for warm sound synthesis with all selectable valve settings
- Audio "mincer"
- Audio refiner
- Valve exciter

Many application options can be selected from a comprehensive preset list.

AM-Phibia - Signal Flow and Stages

AM-Phibia – 1) Reverse phase, input pad, low-cut





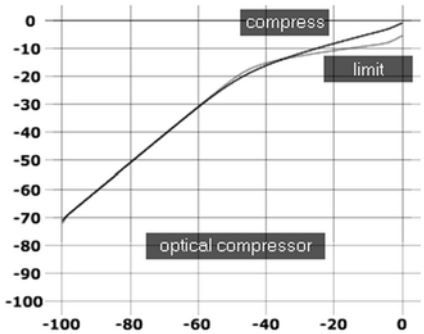
AM-Phibia – 2) optical compressor



The compressor of AM-Phibia is one of the most gentle compressors available on the market and is ideal for difficult signals such as vocals. The fundamental circuit is an emulation of a classic opto-coupling design featuring a so-called feedback circuit.

For this principle a light source (LED, light bulb, or electro-luminescent panel) is supplied the output signal. The emitted light is beamed onto a light-dependent resistor and changes to the resistance result in a change in the amplification factor of the entire circuit (controllable voltage divider). The light source is fed from the output instead of the input due to historical, practical reasons: the circuit gently oscillates to the signal and is very stable without additional settings, and operates very musically so that in hardware reality, this type of device can be built with a minimum amount of required components (which improves sound quality). Due to the overall lag of the opto-coupling system, optical compressors have a relatively sluggish release reaction. This is accompanied by the so-called memory effect of the opto-coupler.A longer exposure to light also means longer return times. Due to the feedback control, the control times and compression ratio also highly depend on the control times, and the compression ratio strongly depends on the input signal.

Specify whether you want to tap the feedback signal after the gain stage or the second filter. Frequency-dependent compression can also be achieved in combination with the pre-filter stage.



The compressor may also be switched between compression and limiter.

Note that the ratio of program regulation depends on the program.

AM-Phibia – 3) prefilter



Different types of EQ are available. Each is treated as an independently structured circuit.

- active A/B

This is an emulation of an analog filter module that operates via positive and negative feedback loops. In this case, A and B differ from one another in terms of the input frequencies of the bands. B is optimized for use with speech or singing.

- passive A/B

This circuit corresponds with the classic Baxandall network and is comparable to stereo systems and some guitar amps. Type A and type B differ slightly; the bass and highs are spaced out further spectrally for type A, and the mid range is more extensive for type B. Variant A is intended for general applications, filter B is optimized for vocals. The mid range, which is also available in this case, is not available in a Baxandall network and has been made possible in the existing circuitry as cascading low and high-pass sections, much like in common passive equalizers. The interesting thing about this setup is the effect the individual steps have on the entire phase response, which contributes to the unique sound of this circuitry.

- guitar passive

Classic switching such as on Marshall/Fender amps. These parameters are highly dependent on each other (more highs = less bass; mids are also influenced). Just like general "passive" switches, complex phases are also present in this case, a result of

variable mixing of individual branches in the filter network and the typical charm of classical design.

- guitar active

A convertible partner for a typically "American" high gain sound. The parameters do not influence each other as much as the passive variant, although a slight amount of internal feedback is applied to give the circuit a bit more power. Of course, this also influences the phase response, which also provides its own character.

- bass passive

A full-tube Peavey bass pre-amp was the source of inspiration for this EQ (T. B. Raxx, similar to Alpha), with a circuit that resembles various Marshal/Fender styles. In this case however, the bass branch does not depend as strongly on the other settings, but a high level of interactivity is present. If the mids are "weakened", the sound turns quite "hollow", which is ideal for slap basses.

- bass active

Similar to "guitar active". In this case however, the corner frequencies are designed for bass guitars.

AM-Phibia – 4) gain



The heart of AM-Phibia is the gain level, an emulation of class A tube circuitry. Simply set the gain setting, and the "magic" happens inside; the frequency response changes, resulting in the right limit containing fewer highs.

A small amount of make-up gain is used to automatically compensate the relatively high amplification factor, and typical valve-like distortions and harmonics occur at the upper end without becoming too loud. Normally, high-gain circuit simulations feature a weak spot at this high a degree of non-linearity, i.e. so-called "aliasing". Harmonics created by distortions ideally exceed the audible range. At a sample rate of e.g. 44 kHz, unpleasant artifacts can occur due to signal mirroring of the Nyquist frequency (22.5 kHz) in the audible range (without harmonic relation). This is particularly annoying in the case of distorted guitar sounds, e.g. if you "bend" at the 24th fret when playing solo and aliasing becomes apparent via bending in the opposite direction. For this reason, we always leave the internal sample rate of the gain stage between 176 and 192 kHz to minimize artifacts.

AM-Phibia – 5) postfilter



This section only provides a low and high parameters; the cutoff frequency point of both may be adjusted, however (see "Expert view" below).

- active

The same parameters as the pre-filter stage, except that the mid band is missing.

- passive cut

Baxandall (cut) network similar to the pre-filter, but without mids.

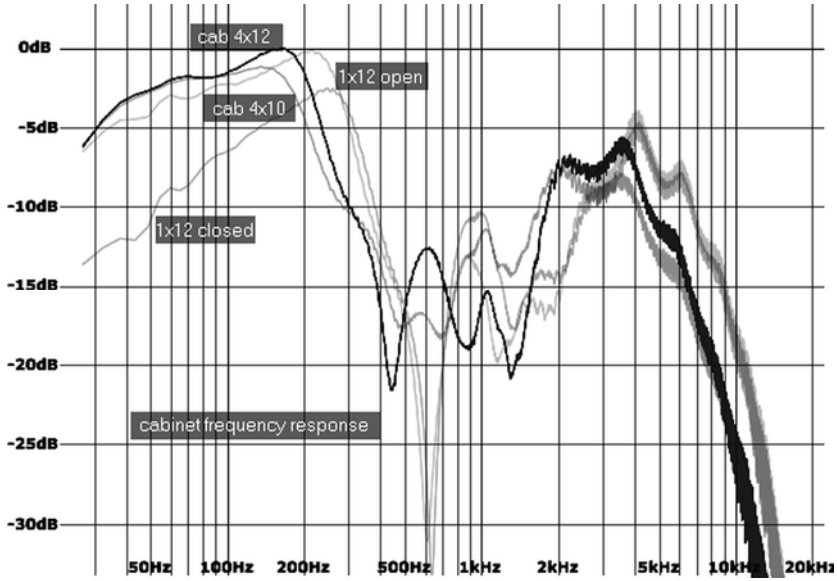
- exciter

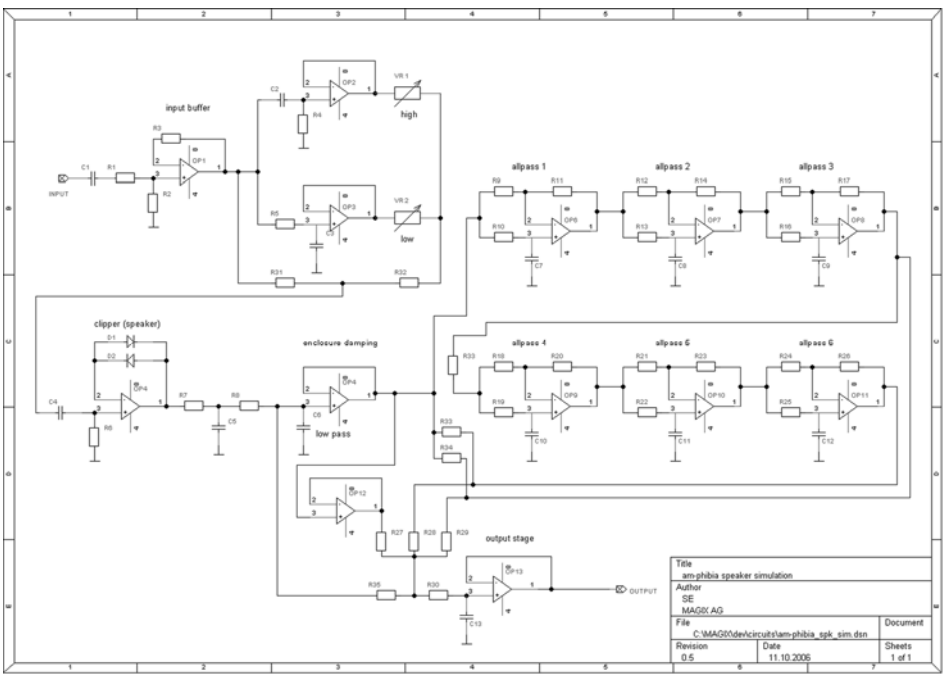
Amplifies the low and high frequency ranges via frequency-dependent saturation. The control elements mix the saturated part with the original, which is common with traditional exciters. Due to the additional harmonics created during saturation, the exciter circuit can sound very different than a conventional EQ in case important spectral ranges are not present. The bass range may also seem fuller due to the switching applied. Despite the immense sound possibilities, use the exciter carefully, since the ear is fatigued by listening to this type of signal and due to the high energy density.

- cab simulations

These sound control networks are a departure from "official" EQ settings. The speaker cabinet simulations imitate the sound of classic speaker/case combinations familiar from conventional collections of guitarists and bass players. In line with the "analog by design theory" of AM-Phibia, we've also focused on switch-heavy interaction in this case. AM-Phibia works with an armada of non-linear amplifier stages, filter circuits for frequency-response changes, and complex phase-shifting processes. To begin with, we examined the sound of the speaker and how it becomes distorted depending on the levels, and we also studied its characteristic frequency responses in relation to its environment.This speaker emits direct sound from the front, while the back of the cone swings in the opposite direction so that the waves enter the cabinet phase-inverted. More sound-altering characteristics occur here: the sound is directed at the enclosure, partially absorbed (e.g. by insulation material), partially reflected, and mixed with the direct sound. Resonance effects such as bass reflex openings, static waves, return effects on the speaker, and so on, are also apparent here. We also added properties of typical models to our speakers such as bass reflex tubes in the signal transit time, which achieves a "typical" sound that would not be achievable

with impulse responses alone. An important aspect in this case is the distortion created by the speaker at higher levels.





AM-Phibia – 6) volume



Volume is compensated using a valve circuit that adds atmosphere. This circuit ensures that the output signal does not exceed 0 dB.

AM-Phibia - Expert View



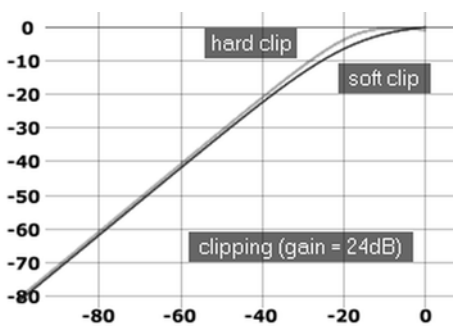
Opening this view mode gives you even more opportunities for editing. Some decisive sound parameters of the device are accessible directly from here:

- opto mem

This changes how the optical coupling device addresses the compressor and thus the related memory effect. You can use the "opto-mem" controller to work directly with the parameters for the system latency and influence the degree of program dependency. Transients generally have less influence on controlling than longer signals. If set to minimum, the circuit will recover relatively fast. If set higher, the release time also increases according to the duration of a loud signal.

- clipping

This function controls the behavior of all tubes in AM-Phibia. Soft clipping (0) generates soft sound characteristics in overdrive. Hard clipping (100) creates a more global volume, although it may be high-pitched if handled incorrectly. If this setting is used with guitars, the sound of the entire device may change drastically, particularly in combination with distortion sounds.



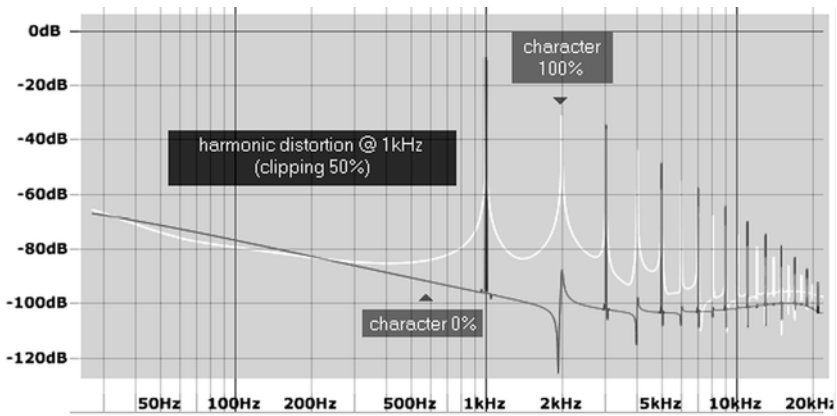
Guitar players can set this function correctly by using the volume controller of the guitar to control the distortion.

You will thereby most likely find the optimal setting at the point where AM-Phibia harmonizes best with the input signal.

With different sound sources, these experiments are less difficult and are worth trying out.

- character

This controller works together with clipping and the gain button. Sometimes, however, this effect is very subtle, meaning that an immediate effect is not audible with all effects. If AM-Phibia is used as a guitar amp simulator, the controller can determine most of the sound. "Character" only works on the tube bias (control voltage for regulating the electron flow within a tube). Capacity changes are also influenced by this reaction, since tubes feature a memory effect. This way you can observe a slight bass increase and more constant harmonies.



A higher "character" value with loud and high bass signals can easily add "oomph", although there is a risk of the sound being rougher in lower ranges. Discreet use of moderate gain settings can sound similar to "transformer-linked" devices.

- pre MF freq

This controller allows you to change the mid frequency of the pre-filter section. The available range is dependent on the underlying model

- post LF/HF

These controls change the corner frequencies of all controls of the post-filter section (basses and highs).

Analogue Modelling Suite: AM-Munition



AM-Munition is a compressor/limiter for spicing up the mix and giving it more kick. This mastering and dynamics tool makes the mix not only louder and adds definition, but can also be used as a limiter. In the process, transients are conserved and a compact, but nonetheless loud signal is delivered. This occurs via a relatively slow serial compression and a soft clipping for the stray level peaks; AM-Munition is sidechain-capable and can also be used for M/S processing.

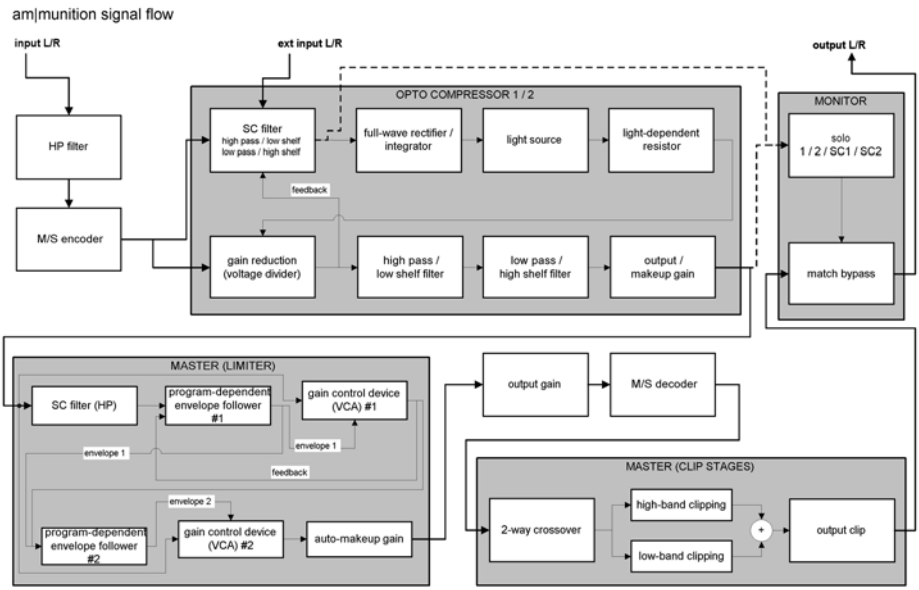
The **AM-Munition** plug-in is an extremely versatile, dynamic tool for editing groups or signal sums, especially in the area of mastering. It has separate units like compression, filtering, sidechain, limiter, and clipper. All modules and parameters are optimized to perform their fundamental functions without any compromises: effective enrichment of the program material without causing bothersome artifacts, a high maximum volume, and an "analog" behavior with an individual sound signature. This follows the maxim of allowing volume without distorting important transients. This is especially a problem with excessive use of the popular brickwall limiter. In fact, many digital limiters drastically increase the average level, but only at the cost of vitality of the sound, usually resulting in a fatigued audio effect. In drastic cases, snare hits are perceived as noise or hissing.

AM-Munition attempts to maintain the attack during increased compression of material. In this manner, the plug-in also functions as a trick: the limiter level is slower than it would be otherwise. Those transients which occur will be blocked in a multi-step soft clipping unit upon output. This clipper works frequency-oriented to enable a relatively high signal level already "solo", without sounding unpleasantly distorted. With the proper settings the combination of limiting and clipping results in a livelier sound with more punch and character.

AM-Munition appears inconvenient and complex compared to other traditional dynamics tools at first. However, with its combination of available steps, modes, and

parameters, the plug-in offers creative flexibility and provides a powerful tool in the hands of ambitious sound engineers.

Overview of AM-Munition



AM-Munition is composed of the following sections:

Mode: The basic functionality of the plug-in is specified here. The normal mode is "stereo", meaning the input signal is entered on 2 channels (left and right) to the sidechain and thereby to the control circuit. In M/S mode, however, encoding the stereo signal will be carried out in parts as "center" (left and right) and "side" (left-right, differential signal). After this, both parts are sent to the sidechain. The functionality of M/S mode is exceptionally well suited to compression of material from the pop and dance genres, where bass sources are commonly positioned in the center of the stereo field. Normal compression in stereo mode will always produce modulation or "pumping", since the bass drum or the bass line reduces the total volume with its own energy. However, the division of the signal into center and side information merely makes a stronger modulation in a different range of the stereo field more noticeable.

OPTICAL COMPRESSOR: The heart of AM-Munition is made up of two compressor units which are indicated by "1" and "2". Depending on the selected mode, you can edit the left and right or the center and side signals. The compressors are based upon a very detail-oriented modeling of an optoelectrical circuit. The principle in this case is very simple: the input signal (Side Chain) is used to control a light source. The simulated light hits a photo resistor (LDR, light dependent resistor), which produces changes in resistance depending upon changes in brightness. These in turn directly

affect the input resistance of the compressor switching. One could say that the signal source is its own volume potentiometer. In this case, the important factors are system capacity (light source, its controls, and the applied photo resistance) and memory effects.Amazingly, circuits based on this principle sound very musical and open, and they are well suited for the compression of program material, provided the actions are not supposed to happen too quickly.

Besides controlling the light source to achieve a specific reference line and envelope curve, AM-Munition compressors enable the ratio of the input signal to edited signal to be mixed. This parallel compression offers far-reaching possibilities for the subtle compression of signals. Because a part of the control circuit is bypassed, important transients of the signal spectral information are maintained. The compressed portion can often be output in depth simultaneously.

Both optical compressor units offer the possibility to restrict the frequency response to a particular output. In this case, a combination of high and low pass as well as shelving filters come into play. For example, filtering is useful for editing the side signals of basses and lower mids in M/S mode (keyword: vinyl cutting) or for restricting the cymbals of the percussion to the edge of the stereo field.

SIDECHAIN: As mentioned earlier, the sidechain signal is responsible for what the compressor sees. The sidechain value is usually a smoothed mid value of the input or a peak. By editing the frequency response of this signal, the relationship of the entire compressor circuit changes rapidly. For example, basses which are filtered out in the sidechain can maintain controllers against unwanted pump artifacts, and in the ideal case this also increases the volume in bass heavy pieces.

A mix of output signal at the plug-in and an external source offers additional possibilities for sound production, especially during a mastering session when single groups are being edited.

MASTER: The master section features one of the most important circuits of the plug-in for sound compression and volume enhancement: the limiter. As is the case with the input, this is combined at the output with a soft clip stage. This makes a fast reaction time or even a look-ahead unnecessary. At the same time, a skilled choice of limiter thresholds produces an airy, natural, fundamental sound, even under difficult conditions and high level reduction. Similar to the sidechain filter of the regular compressors, the limiter also provides the possibility to remove low frequency signals from the controllers.

The limiter switch in AM-Munition works in two stages (see switch image). The first stage has the primary task of extracting an envelope from the input signal. This is transferred to the second stage in place of the actual signal, which in turn forms its own envelope. Both stages work with different parameters and provide different perceptions of the signal. Furthermore, the first step may be switched between forwards and reverse detection. This cooperation between the stages makes a very

exact temporal image of the amplification process possible, especially in difficult situations featuring bass-rich signals, which should normally be compensated for, but produce disturbing inter-modulation noise with regular limiters.

The limiter circuit connects the clipping stage, which is composed of two serial units. The first forms a dual-band clipper together with a 2-way frequency-separating filter. The advantage of separated clipping of basses and highs (mids, too) is a higher achievable loudness with fewer perceivable disturbances in comparison to broadband disturbances. This sort of disturbing intermodulation and roughening of the complete sound image may be reduced, especially with clearly-defined bass material.

The applied frequency filter is modeled on the Linkwitz-Riley filter (4th order, 24dB/Octave, Butterworth characteristic), which is often found in speaker construction. This produces an adequately sharp separation of bands and ensures an exact phase length as the bands are combined at the output. The complete phase shift of the 2-band clipping stage amounts to 360 degrees. This should be monitored when AM-Munition is used parallel to other tracks.

Clipping at this stage and the broadband which follows at the output has an adjustable intensity (see "Config" page below).

The second clipping stage (output end) provides the output's safeguard, and this may be used "solo" for creative intervention.

The clipping at both stages may be set without stages from hard to soft overmodulation. This makes a complete series of "analog" sounding combinations possible.

MONITOR section: The individual compressors (1 and 2) or those of the sidechain may be set to the sum for selective monitoring ("solo"). This makes it possible to adjust the level of signals being edited so that in active "match" state the volume complies with the volume of the unedited bypass signal. This will produce a psycho-acoustic phenomenon: louder signals sound better and can be realistically and easily checked to see if the quality of signal being edited is still in tact apart from volume increase.

Metering: The center of this plug-in interface features several level displays. Among those are two VUs, two peak meters, and two stereo meters.

Both of the outer VU meters display the effects of the main compressors when set to the "reduction" position. The centrally placed displays visualize the value of the reduction caused by the master limiter. When set to "output", all four level displays show the output; the VU meters provide the present RMS value (root/mean/square, average level), while the peak displays measure peaks.

A 0 dB reference value may be set for the RMS value. This is always useful if you would like to have a clear view of the average level. For example, to produce a song or an album with an RMS of -11 dBFS (a rather conservative value when one considers current chart productions, but lower headroom always has a destructive effect). In this case, you would set the "RMS 0 dB ref" controller to "-11 dbFS". If you place the metering switch of the output level to "output" for monitoring, the VU meters will show the displacement of -11 dB. If you read -4, then the RMS level is actually at -15 dB.

Both of the stereo meters are very useful for determining the range of the finally decoded stereo image and for comparing it to the unedited input signal in M/S mode.

AM-Munition Parameters

MODE

M/S/stereo: Selects functionality; 2-channel stereo or middle/side encoded.

Link: At 0%, both portions of the signal will be treated discreetly by the sidechain (L/R or M/S). 100% complies with that of a mono summation of the detector signal. Middle and high values are sensible for complex signals, which sometimes show deflection to the side of the stereo field.

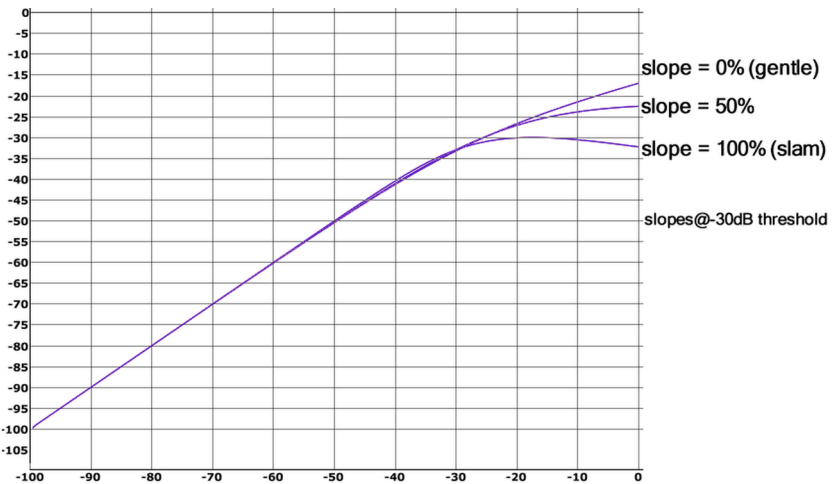
The settings of this controller affect compressor 1 and 2 and the limiter in the master section.

OPTICAL COMPRESSOR

Threshold: This controller defines the threshold at which compression occurs. From a technical standpoint, you control (in contrast to the fader) the signal strength used to activate the virtual light source with this value. The exact trigger point of the threshold is dependent upon the selected compression reference point and internal parameters. For more exact control of this threshold, the VU meters can be set to "reduction".

Output: This parameter is useful for gain increase at high level reduction. We recommend setting the output so that the resulting level matches that of the disabled compressor level.

Slope: This parameter is like the "ratio" when compared to a regular compressor. This is not, however, completely correct. In this case, the compression reference point is influenced so that the result is a softer reference point for lower values and higher values cause limiting or over-compression. Set at maximum, the reference point of lower threshold/higher input level even goes a little lower, instead of remaining at a limiter plateau. This is one of AM-Munition's specialties and is well suited for compression of percussion subgroups, e.g. when room features are supposed to stand out.



The slope controller does not change the reference point itself, but rather the way that transients are processed. Set to the minimum position, the compressors will operate in "feedback" mode, i.e. the input of the detector switch will be fed from the compressor output. This controller works fairly softly and transparently and gauges the signal gently, since existing deflections from the envelope are also included. The resulting reference line is therefore always quite mild.

The more the slope fader is set to the right, the more portions of the signal will be drawn into the compressor input. In this way, the controller combines "feed-forward" and "feedback". The leads to a higher weighting of the transients to produce an aggressive reference line.

We generally recommend a slope setting in the lower third of the fader for a less conspicuous transparent compression.

Response: The switching principle employed here provides musically sensible control times via selection of the "light source" and "photo resistance" system components. The combination of these components determines the shortest possible tune-in and tune-out times. With the response controller, the transient time may be extended to a wide range, since it's as if the image processing time is extended. A variation on this parameter will certainly have an influence on the complete functionality of the photo

resistor, especially on "memory" effects. In this case, the current energy level of the program material decides the actual transient time of the compressor.

Comp mix: Input responsive parallel compression is possible with this controller. This is best adjusted to produce the highest sound transparency together with high level reduction. Any possible compression artifacts are skillfully masked to bypass portions of the signal, and important signal peaks and tuning-in processes of instruments can occur unhindered.

Filter (high-pass/low-pass or shelving filter): The filter switch on the output of the compressor unit sets a higher or lower frequency spectrum limit on the signal. For instance, this is useful in "M/S" mode, for filtering basses in the side signal, which otherwise unnecessarily reduce signal volume and would even lead to problems with phasing of stereo sources during vinyl mastering.

Both filters have gated characteristics (high or low-pass) on their outer ends, and these change by altering the cut-off frequency increasingly in a shelving curve with maximum 12 dB reduction. This results in relatively soft filtering that is ideal for basic corrections to the spectral balance.

Controls (key symbol): Connects corresponding controllers of both compressors and is useful in "Stereo" mode. Separation in "M/S" mode should be deactivated when settings on the filters or output controller are adjusted, e.g. to filter out bass signals from the side signal or to set the width of the bass

SIDECHAIN

Level: This controller allows fine adjustments to be made to the external signal. An adjacent level may be monitored with the peak meter.

Source: Adjusts the mix ratio between the program and an external signal. This mix forms the sidechain signal evaluated by the compression circuit.

Note: To enable external sidechaining, please ensure that "Sidechain input" in the menu bar of the plug-in window is activated and that the relevant source is routed to this additional bus (stereo).

Sidechain filter: Describes the functionality of both controllers for the filter switch in relation to the filter network in the compressors. By limiting this the frequency field, the ranges may be excluded from detection or weighted less.

MASTER

Output: AM-Munition's final output level. Make sure that the soft-clipping switch is located at the final stage of the output. The function is to block signal peaks passed on by the limiter.

We mentioned at the beginning that the philosophy of AM-Munition's sound signature is based in part on the division of work between limiters and clippers. For high signal volumes, the qualitative results are dependent on threshold and output values. A high signal level at the output will inevitably be bound with a high portion of clipped signal peaks. As the case may be, please use the "clip" display to monitor this. This displays the difference between clipped and unclipped signals. The higher the display level, the more clipping results.

Threshold: This parameter defines the threshold for the limiter on the one hand, and also increases the output level after successful level reduction ("auto-makeup gain"). This corresponds with the conventional operation of brickwall limiters, since only a few parameters are needed for quick, targeted loudness.

Release: Sets the return time of the limiter until the controller achieves the output point after falling below the threshold. Note that some kind of program automation monitors the signal independent of the set value. For this reason, you will not find any exact controller times (e.g. milliseconds). The value entered could be understood roughly as the circuit direction. Lower values provide a very fast control response time and weight the program automation quite lightly, while higher values produce increased interaction between both of the limiter's detector stages and participating envelopes.

Sidechain filter (high-pass symbol): For difficult tasks during the compression of material, for instance with pop or dance productions, it's often useful to reduce bass segments or remove them completely. In this way, pumping produced by the limiters can be avoided. The sidechain filter of the limiter can be increased in two steps for this. In this case, always ensure that the bass range is not subject to limiting like the remaining frequency range; the clipping stage needs to intervene and limit the signal as required.

MONITOR

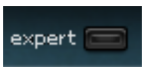
Solo: By pressing buttons 1 or 2, one or both of the compressor outputs will be sent to the monitor line. In this way, you can bypass the master section on the one hand, and watch over the effects of the compressors on the other. This can be very important when working in M/S mode.

The sidechain may be monitored via "SC1" or "SC2" as required.

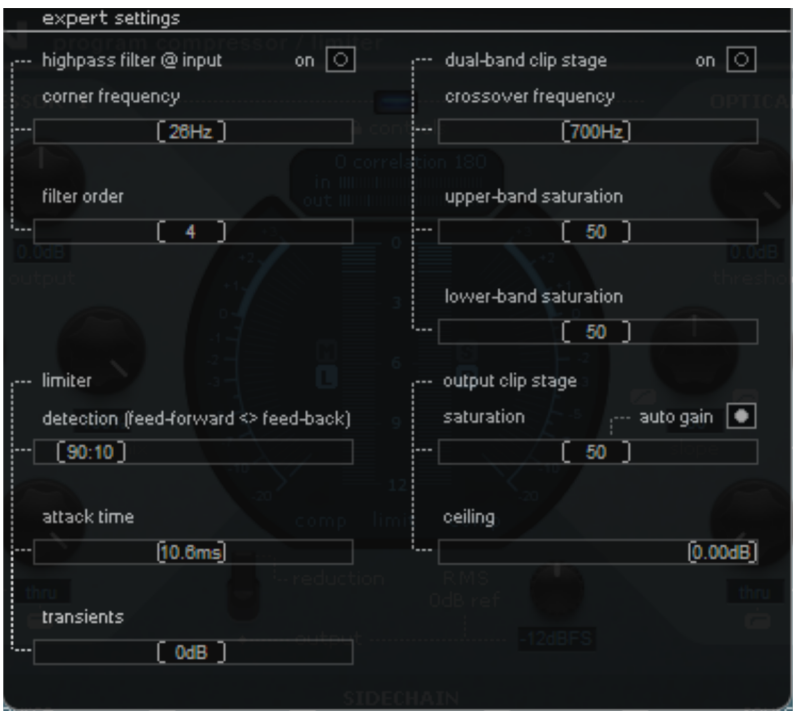
Match bypass: A tool for increasing volume is useful for audio projects, since things usually sound better when they are louder. In order to properly determine whether your material has retained its own quality after these processing stages, it is important to carry out a volume comparison with the unedited bypass portion. The "match" button is provided for this purpose. This works together with the corresponding controller as a signal dampener.

Since the match function lies at the final position of the signal path, you can obviously also use it to remove audio material with a lower level than that defined by the output clipper as 0 dbFS via the plug-in.

AM-Munition CONFIG



Click the **expert** button to access additional AM-Munition parameters.



The regular compressor view will be hidden and a separate display will appear. This alternative view shows a series of additional **preset-based options**:

Highpass filter @ input: At the plug-in's input, there is a high-pass filter with variable cut-off frequency and edge steepness. Especially for mastering applications, this can take over the tasks of "cleaning up" the signal before additional processing and

filtering out low-frequency sections which aren't required or could possibly cause overmodulation/reduction of the signal's maximum loudness.

The filter applied features Butterworth characteristics (q=0.707) and may be switched from 2nd order to 4th, 6th, or 8th order.

For exact filter settings, we recommend working with subwoofers in good spatial-acoustic conditions.

The following options involve **fine settings in the master section**:

Limiter

Detection (feed forward <> feed back: Limiter switching may be set without stages, whether the signal envelopes are to be gained directly from the input signal (forward), or whether they are drawn from existing amplitude process generated control voltage (feed back). The latter approach often creates a calmer, more balanced sound image and weights transients higher than slower signal sections. The resulting threshold of the limiter will be simultaneously rounded off a little more (soft-knee). This usually causes a slightly reduced total volume. Generally, the sound image in "forward mode" is a little more direct and gripping. Sometimes, however, the effects are quite subtle, and in some circumstances, these are only perceivable when levels are reduced substantially.

Attack time: Normally, a limiter should feature an extremely fast response time. For "look ahead" circuits that work with delay in the signal path, the attack time may be relatively long versus the delay, supplying the limiter with enough room to work correctly. With a "zero-latency" design like ours, on the other hand, it basically has to be assumed that the limiter does not get any time at all to adjust. Theoretically, it has to be infinitely fast.

AM-Munition utilizes the cooperation of limiting and clipping. The clipper blocks that at the output which the limiter lets through to signal's transient response. Sometimes, the attack time can be accordingly generous. Lower values of less than 5 ms are available if small level changes need to be smoothed out. Larger values, for instance percussion, may be placed at the forefront of the total signal in contrast to complete instruments or groups. For example, with an adequately long attack time, a snare can maintain its punch in a complex signal and the complete sound can be kept loud and compact. Conventional applications include harder rock or metal productions (a clipper at the output is a better partner than a brickwall limiter anyway, since a limiter tends to consume important transients). The combination applied in AM-Munition, on the other hand, is able to transform amplitude energy into spectral energy.

Dual-band clip stage

Crossover frequency: This parameter defines the separation frequency at which the signal separates into a bass or higher range.

Upper-band/lower-band saturation: When this controller is set to the minimum setting, a hard clipping sound will result as full overdrive is achieved. A high degree of additional overtones/harmonics will appear at the clipping threshold. Below this level, the sound image will remain completely neutral.

Larger values of this parameter produce softer clipping or saturation of the signal. Initially, this highlights only even numbered, lower-order harmonics (mainly so-called "k3" harmonics). For example, a sine pattern of 1 kHz is added to an overtone of 3 kHz and lower amplitude. At 100% saturation, the overtone spectrum is quite similar to that of analog reel-to-reel machines, if driven via saturation of the tape as with full overdrive at the limit.

Output clip stage

This also enables the balance between hard clipping and softer saturation to be fine tuned.

Auto gain: This function increases the total level prior to the output by 3 dB, depending on the clipping settings. This is influenced by the softer saturation characteristic, the maximum of which is normally already reached prior to complete overdrive, although in this case, this value is raised by this amount to provide the maximum amount of headroom. Without "auto gain", loudness will increase as a result of the changed overtone spectrum if the absolute level remains the same.

If "auto gain" is active, this level increase will be avoided and consistent loudness is achieved independent of the saturation curve. In this case, the input of the clipper is lowered by the expected level gain at the output to reproduce a "consistent ratio". This is especially useful for the dual-band clipper, since duplicate saturation of the signal without damping may sound overdriven.

The "auto gain" function therefore ensures a certain degree of room in case additional overtones are generated.

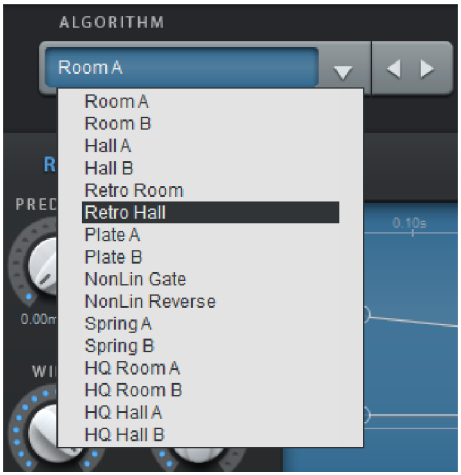
VariVerb II



VariVerb II (short for: VARIable reVERB) is a high-quality reverb plug-in that runs on an algorithmic basis. It provides a series of rooms, halls, reverb plates, spring reverbs and so-called non-linear effects, which can be edited comprehensively and easily.

"Algorithmic" refers to the reverb impression not being created using impulse responses, as is often the case nowadays. Instead, the calculation is based on mathematic models just like those in classic reverb devices. With VariVerb II these algorithms can be chosen from a list, each specialized for a specific replication or a specific purpose and given its own set of parameters. In contrast, technology based on convolution is only able to interpret a static copy of a process, i.e. it doesn't know the room. In fact, realistic reverberation is associated with dynamic, non-linear, high-grade, interactive processes. Not only do these conditions make sure that the artificial reverb method is constantly updated, it also underlines the effect it has as a tool of artistic expression. Let's call this audio aesthetics.

VariVerb II – algorithms

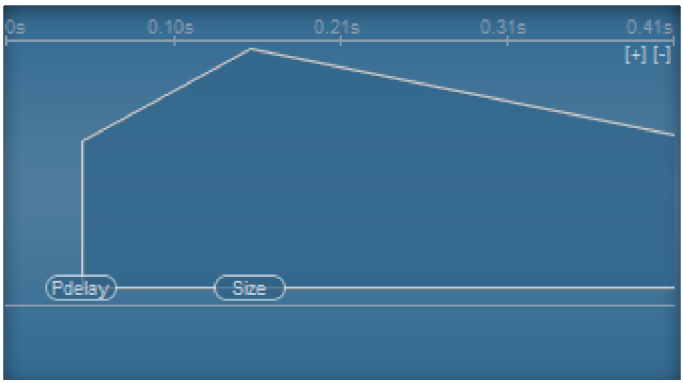


The algorithms in VariVerb II pro specialized in a specific type of "room creation" can be selected from the list on display:

- **Room A:** Small to medium-sized room, dampening of the low frequencies, immediate response, high density/diffusion
- **Room B:** Room similar to A, less dampening, slightly different reflection pattern, responding a bit slower
- **Reverb A:** Larger hall, slower reverb buildup, less density than the "rooms"
- **Reverb B:** Similar to reverb A but later response of the reflection, slightly "colder" sound
- **Retro room:** Classic room effect in the style of well-known hardware reverb devices from the 80s/90s. More artificial algorithms than the previous methods. The vintage/retro character takes center stage here, definitely an "effect" sound.
- **Retro hall:** Type of creation like retro room, but this is a classic hall effect with the corresponding reflection pattern. Thick, cloud-like reverberation tail.
- **Plate A:** Classic reverb plate, high diffusion, quite "dark" and "heavy", slight panorama effects
- **Plate B:** Reverb plate similar to A but with lighter character, faster response
- **NonLin Gate:** Non-linear reverb, no sustain but abrupt end instead (classic "gated" reverb effect)
- **NonLin Reverse:** Reverb with reversed amplitude, sound impression played such as "backwards"
- **Spring A:** Model of a reverb spiral with two linked spring systems, typical echo effects and excitation noise
- **Spring B:** Similar to spring A; however, the vibration is "softer"
- **HQ room A/B & hall A/B:** Extremely high quality and very complex simulation, which, however, is associated with high processor strain. Nevertheless, or maybe because of it, it's the definitive main hall algorithm. Very plastic, tangible room impression with free positioning of the sound source and the listening location. Here, the A and B models differ slightly from one another like in the regular room and hall algorithms, i. e. HQ room A provides somewhat different dimensions and different composition than B. With the HQ halls, hall B communicates with A later and sounds slightly more complex.

The individual models and available parameters are described further below.

VariVerb II - overview of details



The **Display** is the core of VariVerb II, visualizing the available parameters, their current values, and graphically presenting the reverberated envelope.

The preset list is located above the device interface in the **console**, where you will find the typical settings for various everyday studio work tasks.



For general AUX routing there is a separate category featuring all models available in a standard setting. This section can be used to design reverb (if you want to start from scratch). The mixing ratio is preset to 100%.



DRY/WET: The controller for the mix ratio is located on the right of the main interface (direct signal and reverb portion).



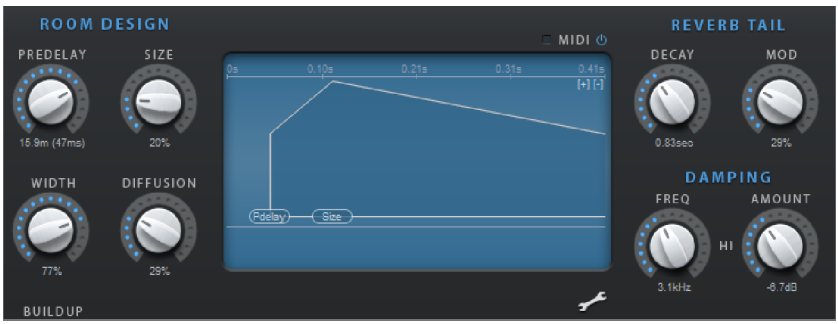
EQ LO / HI: Allows you to "pre-filter" the signal before adding reverb. The filters have shelving characteristics. The trigger frequencies vary according to the reverb algorithm.

Further to the left you will find the "**Mute**" switch.

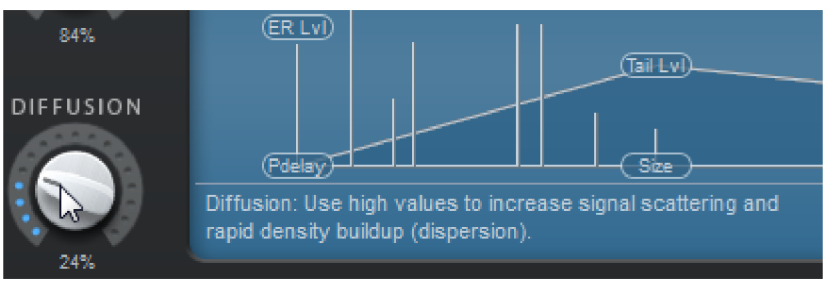


You can use these to temporarily switch off the input signal during longer reverb decay periods, i.e. if you want to judge the resonating reverb precisely.

Next to the display you'll find controllers for reverb parameters such as room size and decay.



Parameters will appear as a "tooltip" on the display if you hover the mouse pointer over a controller.



The following documentation focuses on available algorithmic models and individual parameters. Some apply to several models, in which case they are only referred to once.

Room & Hall

The four available room models (room A/B & hall A/B) are quite complex. Unlike with the regular digital process, with VariVerb II the reverberation is not created by simple echo summation. Such simple systems do not account for the fact that the echo structure becomes denser as the sustain time in real rooms increases. VariVerb II uses a network of reflections that resemble natural processes.

Here you will find the following options:

ROOM DESIGN:

- **PREDELAY:** Determines the time by which the actual reverb process is delayed. This parameter is ideal for fine tuning in order to convey a feeling of "distance" or "isolation".
- **SIZE:** Determines the room size. Moving the slider to the leftmost position means the smallest size, moving it to the right extends the reflection times. Smaller "Size" settings also reduce the distance between the individual reflections. Resonance may also develop due to the short temporal distance between them. Increasing the size of the room creates more space, but this also means that the so-called "modal density" decreases, i.e. more time will pass before the impression of a confined room develops.
- **WIDTH:** Specifies the width of the stereo effect. If set to the left, the reverb is monaural.
- **DIFFUSION:** With this parameter you can simulate diffusing at irregular walls and objects. The higher the value is set, the softer and denser will the reverb appear.
- **ABSORPTION:** VariVerb II calculates 16 initial reflections (compiled to 8 in the display). This controller lets you specify how far the last reflections are weighted. A lower value results in a more "lively" sound impression, although this may also have a more intrusive or unsettled effect. A higher degree of absorption means a smaller amplitude as well as damping of highs.
- **ER:TAIL:** The mixing ratio of the first reflections (early reflections) and subsequent reverberation. These reverberations are decisive for perceiving the room size. Mixing them into the actual reverb improves the possibility of locating voices or instruments. Missing reflections at the start often result in "spongy" sound characteristics without orientation in "room" and "depth".

REVERB TAIL:

- **DECAY**: The reverberation time. With this controller you can define how far the echo will be absorbed, that is, the time for the reverb to die away. You will then only hear the first reflection. Turning the knob to the right minimizes damping and results in long, sustained reverberation.
- **MOD:** This parameter regulates modulation, which introduces a slow variation of delay times.

DAMPING:

- **FREQ HI:** The frequency at which highs will be dampened.
- **AMOUNT HI:** Allows you to influence the frequency-dependent absorption of the reverb. A higher value dampens the highs and upper mids similar to the absorption effect of the air and in particular the material composition of walls.

Plate

A real plate reverb consists of a large metal plate (often 0.5 to 1m² thick or more), that is put into motion by a magnet and coil system (similar to a loudspeaker). The plate is normally hung on springs so that it is able to oscillate freely. Various types of damping are possible to reduce the reverb time. The reverb plate features so-called "taps" at various locations. These are sound pickups similar to those on a guitar. Several of these taps are combined to make up a full signal. Reverb plates are usually mono, i.e. a stereo signal is added to the plate as a sum. A (pseudo-)stereophonic signal is created by combining taps and their position on the plate.

Real reverb plates are only rarely used nowadays and have been almost entirely replaced by software simulation. They continue to be highly popular due to the very dense sound (high diffusion) and inaudible discreet echoes. They are therefore ideal for percussive material. For vocals, plate reverb generates a flattering luxurious effect. The slightly "metallic" resonance of a plate can also be used to generate a vintage effect.

Parameters:

- **PREDELAY**: See room/reverb
- **SIZE**: Size of the virtual reverb plate
- **WIDTH**: See room/reverb
- **DIFFUSION:** See room/reverb
- **BUILDUP:** Low values result in a fast reverb increase. If the value is greater, the activation phase increases
- **DECAY**: Reverberation time, see room/reverb
- **MOD:** See room/reverb
- **FREQ HI:** See room/reverb
- **AMOUNT HI:** See room/reverb

Retro room / Retro hall

As already mentioned, in addition to realistic generators, VariVerb II also features vintage-like algorithms used by many well-known manufacturers to this day. The used processes already have the highest-possible "dispersion" at the start of the reverb phase and generate a dense, wide sound. This is anything but realistic, since discreet reflections are mainly perceived as flat echoes. This sound is usually ideal for breakthrough power and foreground effects. In the past, various manufacturers have attempted to tackle very brief repeat loops and static patterns resulting from insufficient memory. An interference occurs in this case that also adds charm to this algorithm.

Parameters:

- **PREDELAY**: See room/reverb
- **SIZE**: Size of the virtual reverb plate
- **WIDTH**: See room/reverb
- **DIFFUSION:** See room/reverb
- **ER:Tail:** See room/reverb
- **DECAY**: Reverberation time, see room/reverb
- **MOD:** See room/reverb
- **FREQ HI:** See room/reverb
- **AMOUNT HI:** See room/reverb

NonLin Gate / NonLin Reverse

These (non-linear) models are the only ones without a real counterpart in VariVerb Pro. Non-linear reverb is based on a sequence of individual delay blocks (so-called bursts). By weighting these blocks, you can create envelopes. Two of these typical envelopes are "nonlin gate" and "nonlin reverse". These may be set as follows:

- **PREDELAY**: See room/reverb
- **SIZE:** The size of the individual "bursts" i. e. the length of the total reverb summed up.
- **WIDTH**: See room/reverb
- **DIFFUSION:** Smoothens the reverb effect. The greater the value, the more echo blocks are blurred and the more reverb occurs.

Please note that no decay parameter exists for these models, i.e. there is no "decay".

Spring

You may remember reverb spirals from guitar and keyboard amplifiers. At the bottom of these amps is a unit consisting of two to four spirals mounted on a vibration-free carriage. Like the reverb plate, it uses systems for transforming the electric signal into a mechanical one and the other way around. There are different designs and sizes of spring reverb; however, they all have the same typical sound: the splash-like "bloing" caused by movement of the springs. When the reverb dies off, the basic pitch of the spring(s) can usually be heard quite clearly. Furthermore, the frequency range is considerably limited due to the losses in the spirals and used pick-up/transmitter. Nevertheless, or maybe precisely for this reason, their sound is highly unique. Some music styles such as dub & reggae would hardly be possible without reverb.

The spring reverb is implemented in VariVerb II as a digital effect based on so-called "physical modeling" algorithms. Based on a mass-spring system, the effect operates similar to the physical model of a swinging string.

Parameters:

- **PREDELAY**: See room/reverb
- **SIZE**: Size of the spring system, i.e. length of the springs. A smaller value results in a very short response time and short echoes, greater values slightly smudge the typical oscillation sound of the spring. The length of echoes is increased.
- **WIDTH**: See room/reverb
- **SATURATION**: The spring models also include the processes when "outputting" the signal and "receiving" it via the magnet/coil system. Transfer of electric energy into mechanical power and vice versa is non-linear so that harmonic distortions may occur if the input level is high. The saturation parameter more or less increases the output volume and switches the system into saturation. Depending on the material, a very interesting organic effect occurs which highlights the vintage character.
- **DECAY**: Reverberation time, see room/reverb
- **MOD:** See room/reverb
- **FREQ HI:** See room/reverb
- **AMOUNT HI:** See room/reverb

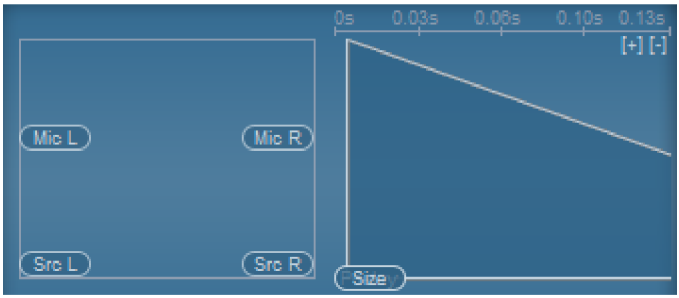
HQ (high-quality) models (Room A/B & Hall A/B)

The above-described room and reverb algorithms are of high quality, very versatile and require particularly low CPU power. If you want even more realistic and higher quality spaces and the load on your CPU isn't as important, take a look at the HQ modules. These simulate a room with rare plasticity and naturalness, almost as if they were based on impulse responses. But they are not; VariVerb II features an armada of delays and filter stages. These may require extra processing time, but they are still very useful.

In comparison to other modules, these algorithms cast a far closer meshed net over the virtual room. The results are natural, fast diffusion of the signal, high complexity of the reverb signal without echo pattern, and the possibility to freely position the two virtual microphones.

If the HQ rooms are opened in "Expert" mode, you will notice that editing for the first reflections is missing. The reason: in normal modules, the artistic and sound manipulating effect is in the foreground. A natural room, however, does not differentiate between early and late reflections. These migrate in time. The proportion of first reflections depends on the position of the audience.

In "HQ" mode, a reflection pattern and entire sound impression including stereo positioning may be edited via a top view of the selected room. The usual section shifts along with the reverb apron in the graphic display.



This includes freely positionable handles for sound sources ("Src L" and "Src R") as well as two microphones ("Mic L" and "Mic R"). For instance, if you increase the distance between the source and the microphone, you will hear the signal move away from you; as the delays increase, the sound characteristics become more "diffused" and overall more complex. At the same time, the previously harsher early reflections will disappear. This tool contains a lot of potential, and it may also be automated.



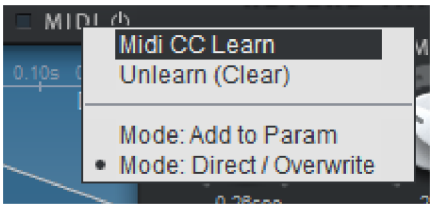
Plus, in "HQ" mode, you can also influence the dampening of highs as well as the reverberation time of the bass section. When interacting with the high band, realistic simulations are possible, e.g. wall surfaces. For instance, the wooden panels on the walls always reflect bass less than concrete walls (also due to possible gaps), which may also resonate and absorb the force of long-wave bass frequencies.

The same also applies to the furnishings of a room. Furniture, shelves, etc. are traps for bass.

HQ room/Hall model parameters:

- **PREDELAY**: See room/reverb
- **SIZE**: Size of the virtual reverb plate
- **WIDTH**: See room/reverb
- **DIFFUSION:** See room/reverb
- **DECAY**: Reverberation time, see room/reverb
- **MOD:** See room/reverb
- **FREQ HI:** See room/reverb
- **AMOUNT HI:** See room/reverb
- **FREQ LO:** Bass dampening frequency
- **AMOUNT LO:** Bass range dampening level.

MIDI CC Setup



If you activate one of the buttons in order to save, any following external controller movements will be used to generate the MIDI CC number. This enables source destination pairs to be saved one after the other.



If all parameters have been set, save the setup by pressing the "Save setup" button. If you leave the MIDI CC view before doing this, the previous settings will be restored.

This controller setup may be regarded as global, i. e. the settings are independent of the selected presets and the current project.

3D Reverb

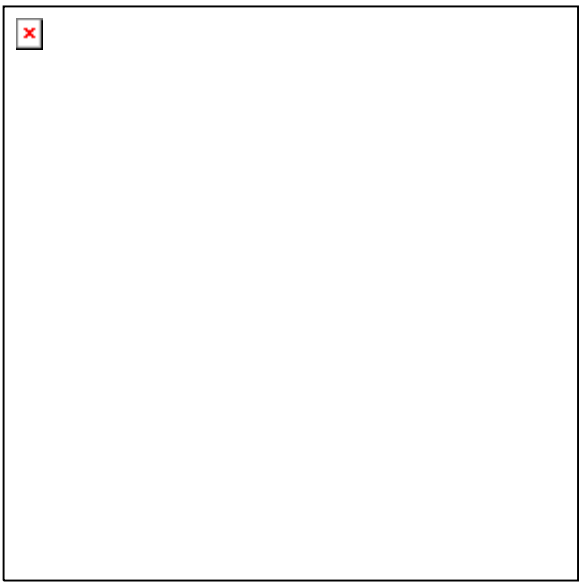
3D Reverb is a unique and powerful algorithmic reverb effect. It produces the sound of a wide range of acoustic spaces, from small rooms to large concert halls. It works equally well in stereo and 3D multi-channel applications with up to 10 audio channels.

When used on a surround AUX bus or surround master, the input signals are routed into the virtual room according to their 3D panning, which supports the spatial impression of surround sound mixes.

The calculation of early reflections is based on a geometric spatial model that is automatically adjusted to the audio channel configuration to obtain a reflection pattern that exactly matches the positioning of the incoming audio signals. This allows the existing 3D panning in a project to be used directly to create a reverberation that matches the placement of the source signal to produce a convincing 3D sound. Late reflections are based on a 16th order feedback delay network (FDN16).

Decay times range from <100 ms to infinity and are independently adjustable for high and low frequencies. For further processing of the frequency spectrum of the room signal there is an attenuation filter. The early reflections can be diffused, the late reflections can be made more lively with modulation. On the output side, there is an equalizer and a Spatial Balance control for adjusting the spatial placement of the reverb signal.

Overview



1. **Peak meter:** The peak meters for the input and output signal are located at the top left and right.
2. **Visualization:** There are three different views for visualizing the generated reverb signal: Impulse response, spectrum or waveform display.
3. **Room:** This is where you'll find the controls for editing the early reflections, consisting of an equalizer for attenuation of the reflections and controls for room size and decay time.
4. **Color:** Here you will find the controls for the sound processing of the late reflections as well as a crossfader for adjusting the mixing ratio between early and late reflections.
5. **Output:** This is where you will find the controls for processing the output signal: an equalizer, the Spatial Balance control for positioning in 3D space, and the controls for the output level and the mix between dry and reverberated signal.
6. **Presets/settings:** In the area Presets and settings presets can be loaded, there you can also reach further settings and the bypass function.

Peak Meter



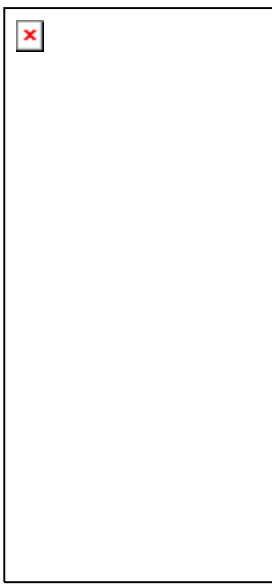
At the input and at the output of the effect there is a peak meter section where you can check the input and output level of the channels.

The number of peak meters displayed dynamically adjusts to the number of channels used by the plug-in.

If clipping (level above 0dB) has occurred in a channel, the red clipping indicator lights up. The indicator can be reset with a single click.

Visualizations

At the top center is a visualization area where the input and output signal can be displayed as a spectrum or waveform as well as the impulse response of the Hall effect.



Impulse Response The graph shows the reflections generated by the reverb algorithm including all processing over frequency and time.

If a change has been made to the parameters, it will take a short moment for the display to update.

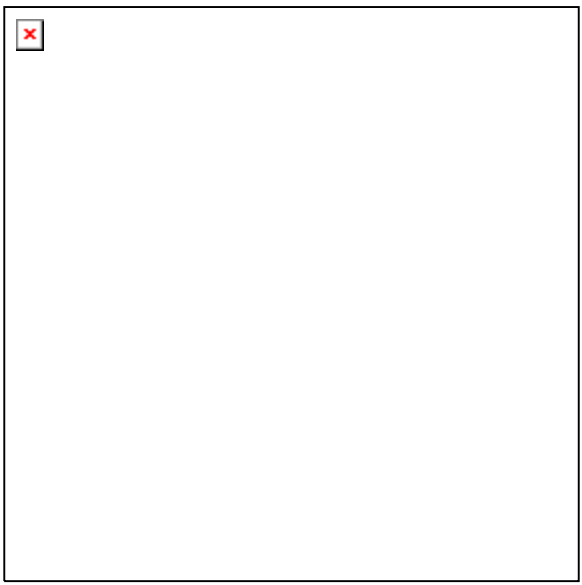


Spectrum: With Spectrum the frequency distribution of the input signal (gray) and the output signal (blue) is displayed in real time.



Waveform: Waveform represents the envelope of the waveform of the input signal (gray) and the output signal (blue).

Room



Size: Sets the room size and thus the timing of the early reflections. The larger the room, the more clearly separated the early reflections are perceived.

1. In combination with the mixing ratio between early and late reflections, see Color, you also use it to determine how far away the sound source appears.

2. **Decay:** Determines the decay time, i.e. how often the signal is reflected between the "walls" of the virtual room and how much sound is absorbed in the process.

3. **LF Decay:** Here you can set the factor (25%-400%) by which the decay time for the low frequencies should be shorter or longer.

4. **LF X-Over:** Crossover frequency for the LF decay time. Determines at which crossover frequency the LF Decay is applied.

5. **Equalizer:** With the equalizer you can influence the frequency behavior of the room reflections. Drag the two dotted lines to adjust the attenuation for the low and high frequencies. The blue dot controls a peak filter: the horizontal position determines the frequency, the vertical position the boost or cut by that frequency. Use the mouse wheel to adjust the resonance of the filter.

Color



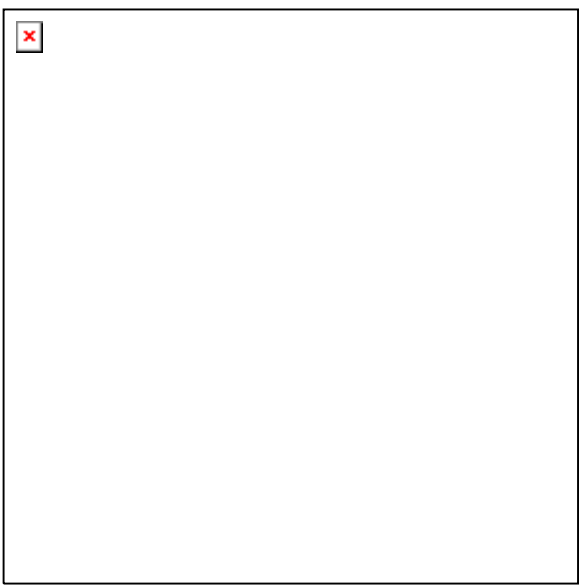
1. **Early/Late Reflections:** This control adjusts the mix ratio between early and late reflections.

2. **Density:** There are two different models for calculating late reflections, **Density Normal** and **High**. Normal is more suitable for small rooms, High for large rooms.

3. **Diffusion:** This control determines the amount of diffusion applied to the early reflections. The diffusion determines the density or scattering of the individual reflections. At low values, the individual reflections can be perceived separately, at higher values they tend to merge.

4. **Movement/Speed:** The delay times of the later reflections can be modulated for a more natural sound. **Movement** sets the strength of the modulation and **Speed** sets the modulation frequency.

Output

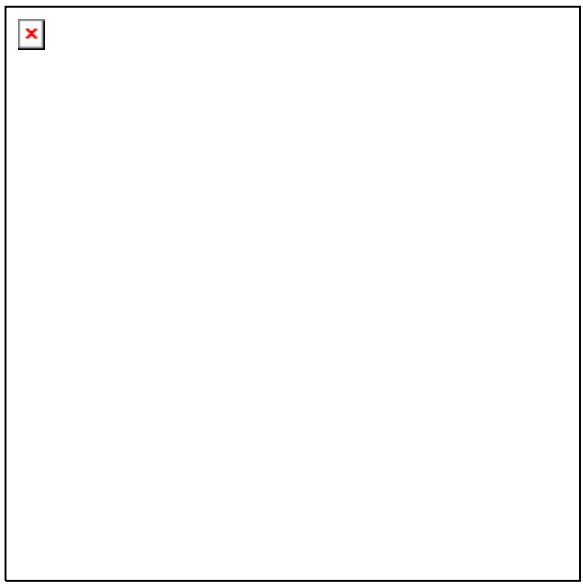


1. **Output EQ:** For the final sound shaping of the reverb signal there is an equalizer consisting of a low cut (high pass filter) for the complete removal of the low frequencies and a low and a high shelving filter for boosting and cutting the low and high frequencies. Drag the dotted line to adjust the low cut frequency. The two blue dots control the frequency and boost/cut of the low and high shelf filters.

2. **Spatial Balance:** Use these controls to adjust the 3D position of the reverb signal.

3. **Wet Level:** The output level of the reverb signal can be adjusted independently of the dry/wet mix.

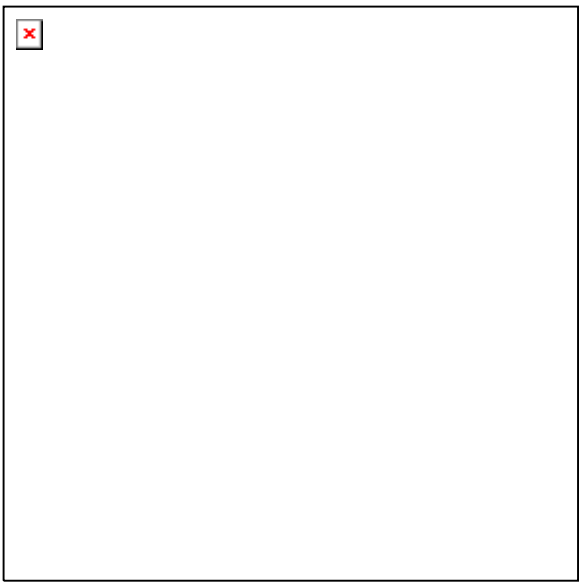
   Mix: specifies the mix ratio of the original sound and the processed signal.

4. 

   With the lock symbol , the slider can be fixed, so it won't change when loading a preset. This makes it easier to compare different presets in the context of a mix.

Presets and Settings

At the top of the plug-in window, you can select presets, adjust view settings and temporarily disable the effect.



1. **Previous/next preset**
2. **Preset menu:** In the menu you can select the presets from different categories.
3. **Settings:** Here you can disable the tooltips. You can also switch the labels of the controls: At **Value** the set value is always displayed, at **Dynamic** the value is only displayed when you move the mouse over it and otherwise its function. **Show tutorial** opens the tutorial, which explains the function of the effects.
4. **Bypass:** Use this button to temporarily disable the effect.

Vandal

Virtual guitar & bass amplifier

Sequoia is a complete simulation suite for guitarists and bassists. This plug-in is capable of simulating the entire signal chain, from input to stomp boxes, amplifiers, microphoned loudspeakers and post-processing studio effects, all in top quality.

We focused on several aspects while designing this program, which make Sequoia a very special plug-in in terms of its amp modeling.

Special Features of Vandal

- High-fidelity analog modeling featuring accurate models of tube amplification stages, genuine real-world positive characteristics like distortion, dynamics, and audio complexity.
- Not the application of impulse responses during speaker simulation (conventional), but rather genuine physical modeling that calculates components like speaker, housing, recording space, and microphone individually based on physical situations, offering enormous freedom of sound design and realtime control.
- Easy-to-learn interface with clear workflows
- The focus is on the important components of a guitar or bass setup.
- Not a copy of a famous amplifier, but rather true "Custom Amplification Design" that features a sound of its own and versatility in terms of audio variety.
- Completely latency-free Sequoia generates no signal delay by doing away with impulse responses. Only your audio interface and driver determine the input and output latency during live playing.
- Full MIDI control of all parameters in each component.
- 4-times oversampling (below 44/48 kHz) for components that create distortion. This avoids typical digital artifacts (aliasing)
- Comparably low CPU load In a live setting your computer needs to be stable and have "room to stretch". In the studio, you may require multiple instances of the same plug-ins in one song. For example, for one specific metal riff you may require 10 guitars instead of two for that real "punch in the gut"; simply let Sequoia rise to the occasion in 10 separate instances...

VANDAL Quick Start

VANDAL Overview



We consciously designed the VANDAL interface to feature a realistic look. As bassists and guitarists ourselves, it seemed important to us to feel right "at home" here. The only unusual detail is that the floor pedals (and the gray carpet) are viewed from above, while the other effects are presented from the front. We admit, there was no way around this...

Also, no one in real life would place an insanely heavy amp on a 19-inch rack. But in this case, it seemed logical because the signal path flows just like how you are read these lines – from left to right, and from the top down.

The area at the top of the plug-in is called the "console". This is the so-called "switchboard" of VANDAL. The second half of this document (view page 1081) will explore these areas in more detail.

Input level: VANDAL and the real world

Independent of the preferred software, amp modeling has to feature impeccable integration with audio hardware and ASIO drivers. Please ensure stable and latency-free operation and the largest possible signal-to-noise ratio of the converter or the complete analog section when selecting appropriate equipment. If in doubt, ask your audio equipment dealer for advice and consider your entire computer system. On-board sound systems are not recommended due to their rudimentary or unsuitable driver sets and the often below-average signal quality.

Before making any "serious" endeavors in VANDAL, please make sure that your audio interface is supplied with a sufficiently controllable signal level from your instrument. The input signal at the audio interface should be "saturated", but it should not be overmodulated. Compared to the possibly desired distortions created within the plug-in, an AD converter distortion will sound abrupt and horribly digital, and the the worst part is that you will never be able to remove it.

If very loud velocities sometimes reach the upper modulation limits and the average level stays at -12 or even -20 dB, everything is normal. The less noise your audio hardware has in the process, the better. We recommend working with devices that feature a resolution of at least 20 or 24 bits. A simple 16-bit sound card will stop being fun to use at the very latest when high-gain sounds come into play.

If possible, use an interface with a high-resistance instrument input (often labeled "HiZ"). Line or microphone inputs are not very well-suited for this; due to the relatively high impedance values of (passive) guitars or basses, this sort of input has a dampening effect on the instrument. The sound will seem dull and "dead" as a result...

If the interface does not feature an instrument input, insert a separate guitar pre-amp, a DI box, or even a simple pedal (not switched to "true bypass).

Instruments with active pickups or electronic components are less affected by this adjustment problem.

Please take this opportunity to also check the latency or the buffer size used by your audio system. A good ASIO driver should let you play with a latency of only a few milliseconds between the stroke and the calculated sound output from the software. Values under 5 ms are considered good. You may have to experiment to find the lowest limit at which your system functions stably. If you notice ugly crackling or even long drop-outs, then try a higher latency.

Input level reloaded: VANDAL's input level



After you have made sure that the external levels are clean, you should take a closer look at the input level; this controller is located at the top left of the console. Just like with a genuine guitar or bass setup, it's important to ensure the highest possible input level in order to be able to work optimally. This is even more important for distorted sounds and natural high-gain playing styles. You should make use of metering for this as well (the LED indicator next to the input control). If needed, activate the noise gate and adjust it so that it lightly suppresses the input signal during pauses in playing. VANDAL does not cut the input as hard as classic gates do, but instead regulates them finely via the signal energy, beginning with the highs, at the point where noise is most audible.

VANDAL preset and scene selection

Would you like to know about everything that VANDAL can do as quickly as possible? Just start strumming away and listen or flip through some of the included presets. These are available via the list in the upper edge of the console.



A preset includes all settings for the main elements of VANDAL, i.e. stomps, amp settings, cabinet simulation, studio effects, and a series of control parameters.

All settings for the main elements (such as the amp) are subdivided into "scenes" within a preset. This "scene memory" is located to the right of the preset list. You may create up to four variations of a preset using scenes. Numerous included presets make regular use of the scene concept. We will examine this working method (view page 1082) more concretely later on.

Presets may be switched by clicking on the list or using the arrow symbols. You can also use the left and right cursor keys.

Scenes may be switched by clicking on the adjoining number (1 to 4) or on the name field of a scene. Alternatively, you can use the up/down arrow keys.

Of course, presets and scenes in VANDAL may also be controlled via MIDI (view page 1085).

Presets are located in a subfolder within the VANDAL folder on the hard drive. These may be changed however you like, sounds may be exchanged with peers using VANDAL, and your preset library may be expanded without limits.

Components

Stomp Boxes (pedals)

The real world has produced a series of effect devices popular with guitarists and bassists in the "stomp box" format. We've also included a rich palette of these devices.



VANDAL includes four "stomp slots" for use with effects equipped from the list. The signal flow within this chain runs from left to right. Stomps can be moved around freely, by dragging an unused area on a pedal's surface.

When a stomp is assigned to a slot that's already occupied, the two change their places. Dragging while holding the [CTRL] key creates a copy of the pedal at the new location.

Each stomp box may be detached from the signal path using the "foot switch" or the blue button next to the selection list. An inactive device does not utilize the CPU.

Some stomps may be familiar to you; we reproduced some real prototypes with regard to functionality or sound, or simply got some inspiration from others. Some, in contrast, were created from the ground up and offer original sound ideas.

We're constantly developing the selection of existing stomps that are available. With each plug-in update, it is our intention to constantly supply you with fresh "food for thought" in addition to technical improvements. Look for updates to VANDAL at http://www.vandalamps.com.

The following stomps were available at the time this manual was printed (each in its own category):

Overdrive/Distortion



La Crema: Inspired by the world's most famous distortion pedal: the Ibanez Tube Screamer. La Crema offers soft musical overdrive sounds for solo parts, or simply serves to provide the amp with a prepped, contoured signal. The available gain is comparatively moderate; this is a typical pedal for classic rock and blues.



Halvar: The name may sound a bit clumsy and coarse. In reality, however, this blue Scandinavian is a real softie and is a homage to Yngwie J. Malmsteen and his preference for DOD distortion pedals. Halvar tidies up nicely in the bass range, and it's outstanding for shredding and styles of play where crystal-clear response without any bassy noise is important. The gain factor is comparable to La Crema, but Halvar distorts "harder" and has less accentuated mids.



Hellfire: Welcome to guitar hell! This pedal will roast everything, deliver gain until the cows come home, basses from the pit, cutting highs, and it easily creates the typical "scooped" sound of popular metal bands.



Fuzz: I can't get no Satisfaction, Foxy Lady. That basically says it all...

The Fuzz effect is famous for its singing-saw sound. Where overdrive and distortion effect the signal in a comparatively mild way to let it "breathe" within limits, Fuzz attempts to rebuild the sound from the ground up according to its own principles.



Bass Distortion: This pedal is of great service for bass players in search of juicy & meaty tones. For massive, chunky overdrive sounds, it is important to reduce low-frequency content from a distortion device. With classic fuzz tones, it's quite the contrary.

This distortion device delivers both, and many facets in between. It is equipped with a dual filter network, located before & after the distortion process. Simply tweak your favourite sound texture to taste.

Modulation effects



Chorus: The application and sound remind us of our personal favorites: those good old Boss pedals. Our Chorus offers modulation that's soft as butter but still crunchy and always assertive.



PhaseShifter/Tremolo: A phaser alone is already quite spacy, and a tremolo can sound a bit psychotic, but both together... Whether soft or choppy modulation (->Wave), slightly out of tune, or with sharp resonance effects... (-> Shift Q), it can sound as if it's up in the clouds or totally sick!



Flanger: Although technically related to the Chorus function, the Flanger uses shorter delay times and internal signal feedback to produce slightly edgier modulation effects. This Flanger pedal certainly sounds soft and wide in the same way that analog flanger pedals sound.



Bass Chorus: The essential sound of this pedal is very similar to the "normal" Chorus. Frequency response, delay times, and modulation curves are certainly optimized for use with bass. The Bass Chorus produces enormous width, but it never becomes muddy, no matter how deep it gets.

Delay / Reverb



Digital Delay: A simple delay pedal without too many frills that features maximum sound transparency. Simply set the time (via the delay controller), the feedback, and the damp level, mix it in, and that's it. As required, the delay time may be snapped in to the host software's currently active tempo via the sync switch.



TimeTunnel: Compared to the DigitalDelay, this pedal is a real vintage item. The (virtual) switching corresponds to that of the built-in "bucket brigade delay" in other stomps. Properties such as loss of highs in case of long delays or the "compander" switch for noise suppression of "breathing" sounds created by high feedback settings are clearly audible. The BBD memory's virtual sampling rate may be modulated via the "Mod" controller similar to the Chorus effect, which enables very lively delay textures.



TwängBäng Spring Reverb: This is truly an authentic recreation of a spring-reverb system, incorporating all those lovely boingy, springy & slinky noises admired for decades. TwängBäng lets you select from a spring system using two seperate springs, or from one consisting of three springs. Both have similar delay times as of certain well-known US spring devices, but each one does have a distinct colour and reverb pattern on its own.

The 2-spring system sounds quite "vintage" while the 3-spring mode is a bit more balanced and open. Using the "tension" knob, the "springiness" and therefore the delay and attack structure can be altered. "Tone" & "damping" decide over the overall color and reverb time. Using a fair amount of „drive" makes the simulated transducers sweat a bit, creating a thick "glue" to the reverb sound.



SpaceCake Echo Reverb: This pedal is a perhaps unusual amalgam of a muti-tap delay and a reverb systems known from the late 1970s. Using proper BBD (bucket-brigade delay) emulation techniques borrowed from "TimeTunnel", the signal feeds two seperate delay lines here. Each delay has 4 taps on the output, which have user-adjustable weighting. This alone already lets you dial quite unusual reflection textures.

In "reverb" mode, things can go more wild. Now, the feedback signal of a delay line is derived from the sum of all 4 taps, and the two feedback paths are cross-coupled. Depending on the actual delay-time ratio of the two delay lines, very complex echoic and reverberant tails can be created. Just keep in mind that by summing the taps in "reverb" mode, the system is prone to instability. Luckily, you have the "mod" parameter at hand, slowly varying the delay times and keeping feedback under control. And it greatly enhances that "spaced-out" feeling.

Volume/Dynamics



Volume Pedal: No explanation needed. Step on it, give it some gas. You are, after all, driving with tube equipment! If the amp is set correctly, you can control distortion levels precisely just by varying the volume. Common swelling sounds are also no problem when using MIDI controls.



Compressor: The heart of this pedal is the modeling of a simple floor-board effect's typical control circuit, which is based on an FET (field effect transistor). FET compressors are famous for their speed, high degree of sustain, and above all for their character at comparably low switching complexity. Of course, the guitar signal may also be compressed subtly or only the attacks may be enhanced specifically.



Bass Compressor: The basic circuitry of this Stomp Box is comparable to the "normal" (guitar) compressor. Under the hood, adjusted specifications control at decisive positions to provide bassists with consistently solid sounds. The Bass Compressor works frequency selectively, i.e. very deep frequencies trigger the process less intensely than mids or highs. This enables the Drive controller to be used to produce a good foundation and thrust.



Twin-C: This compressor pedal divides the instrument's frequency range into two ranges (low & high) and sends these to separate compressor stages. This separation helps to avoid pumping artifacts or other typical problems caused by broadband compressors. The stomp selects control times automatically. Basically, very short attack times are applied to the upper band, while longer times are applied to the lower band. Twin-C enables slap bass sounds with extreme punch to be created easily, which offers a very sturdy foundation and a sound that "goes straight at you".



Tube Compressor: This stomp box uses virtual tube circuitry. Although in this case, it doesn't neccessarily mean the process is about distortion. It's more the fact that the gain-reduction process is a function of tube operation: like the "variable-mu" concept used in popular concepts from the 1950s onward, a tube can be externally controlled to change its amplification factor.

Of course, such a circuit can be driven hard by purpose and produce typical tube-like saturation textures.

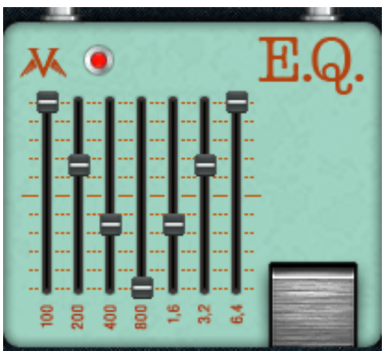
In order to keep things well-balanced and controllable, a (symmetric) push-pull circuit is used in this design.

Using low to moderate "drive" settings, this pedal is a very smooth & gentle dynamics compressor.

Filter/EQ



CheWahWah: This little red rebel will prick up your ears. It can scream out your solos and fills loudly and clearly while keeping its cool and staying smooth. This is one of the few digital wah pedals that doesn't sound digital. On the normal setting, the pedal acts like a regular wah pedal; in auto mode, the filter circuit reacts directly to the incoming signal's amplitude. In this case, the pedal varies the sensitivity of the auto wah effect.



EQ: This pedal enables the guitar signal to be set precisely in seven frequency bands. This is not only suitable for corrective adjustments, it's also great for supplying any subsequent overdrive or distortion stages with perfectly prepared sound. It also performs well in solo passages, e.g. in case the mid range needs to be highlighted.
The heart of this EQ pedal is a circuit based on an analog model which features parallel filter stages. In this case, the typical disadvantages of serial filters such as phase problems and sound degradation are avoided. This EQ features the desired properties of real circuits, including added harmonics to the input or individual filter stages.



TrebleBooster: This effects pedal is ideal for that "British" sound or any kind of crunchy style. It effortlessly provides crystal clear highs to the amp or to other connected stages. Rhythm guitars come to life, while leads sparkle out of the speakers.
The underlying circuit remains unique in spite of a certain likeness to vintage treble boosters. The adjustable corner frequency ("freq" controller) is the key player: depending on the setting, dramatic changes to the frequency and phase responses may also be heard, which makes the booster extremely versatile.

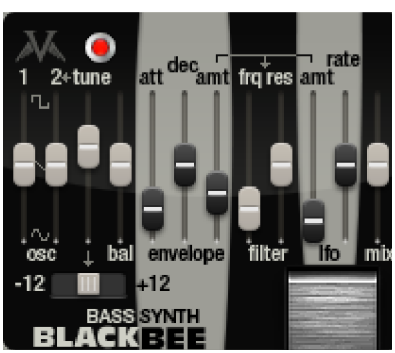


FunkFrog: The ideal companion for sixties & funky styles, but also great for a lot more. In principle, this pedal is an auto wah; however, the frequency band is not only controlled by the input signal via an envelope follower ("env" controller). The "mix" controller allows an LFO (Low Frequency Oscillator) to be mixed in as the controlling signal. This creates a kind of rhythmically croaking auto wah, or maybe an oscillating, whirring filter thingy..

Others



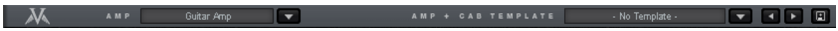
Octaver: This effect uses classical frequency division switching to produce to sub-harmonics from the (mono) end signal. The controller "oct1" defines the portion of the first sub-octave (-12 semitones), and "oct2" defines the portion of the second one (-24 semitones). The direct portion of the signal can be mixed into this.
Both filter controllers enable the sharpness to be removed from the waveform that is produced, e.g. in order to thicken it up just for the bass range. As usual with octavers, clean tracking requires single tones; chords should not be played. In general, the selection of the neck pickup is also important, since this is where the basic sound is transferred most clearly.



BlackBee: This bass synthesizer can be used to manipulate the bass signal in a variety of ways. It offers the corresponding envelope and filter parameter plus two oscillators and an LFO. The "tune" controller can be used to adjust the pitch of the signal.

Amplifier

Vandal essentially offers two different amplifiers; a guitar amp and a bass amp. These can be selected from the list in the "Amplifier panel" below the stomp view.



As we mentioned, in Vandal we found our own way and avoided modeling specific amplifier brands and models. The amplifiers are set up variably so you to get a number of different sound characteristics out of your Sequoia amp. Internally, these circuit designs work exactly the same as the real versions. At a few points that are decisive for the final sound we've gone with our instincts and let our ears tell us what sounds right.

However, if you need a bit of help to get started with your sound you can go to "AMP + CAB TEMPLATE" and use one of the many presets.

Amp/Cabinet Template

The extremely variable concept for amps and cabinets within VANDAL is great for individualists, but it would take a bit of time to imitate a band mate's 40-year old Marshall with speakers of the same vintage. If you want to use a combination like this often, then it's nice to have a library for it, which is exactly what the templates are for.

Our team of experienced musicians has compiled a multitude of settings for the amp and speakers. These templates may be changed at any time, or you may create your own if a particular combination of amp and box is especially appealing.

Guitar Amp



Circuit Philosophy



The VANDAL guitar amp offers three different pre-amp modes and two switchable end stage models. This basic configuration may be changed via the wrench symbol on the right side of the amp.

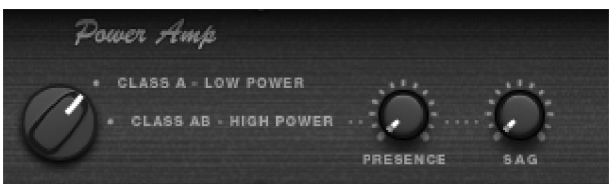
The pre-amp has three modes of operation:



- Classic (think Fender, early Marshalls, or the first Mesa/Boogie amps)
- British (oriented on Marshall Super Lead / "Plexi")
- Modern High Gain (design similar to "souped-up" amps like Rectifier, Soldano or Peavey 6505)

More about pre-amp modes and their properties resulting from channel selection can be found in the chapter "Pre-amp Channels" (view page 1071).

The last stage has two modes:



- **Class A, low power:** A simple, low-performance, but musically pleasant distortion. In "A" mode, only a half-wave of the signal is amplified. To do this, the signal is brought to approximately half of the amplification range of the end tube(s) through bias voltage, because tubes always amplify only a half-wave of an alternating circuit. A class-A circuit may be put together using only a few components and sounds very "warm" because of its constantly asymmetrical characteristic curve shape (some even harmonics appear). This amplifier, however has the disadvantage of low efficiency or low output performance and a comparatively high liability of erosion. The latter has been left out of the modeling, of course...
- **Class A/B, high power:** In this case, each half-wave has its own tube, which (almost) doubles the efficiency. Class A/B router amps are more complicated to implement (and calibrate). Compared to Class A, the sound characteristics include descriptions such as "sovereign" and "powerful", but somewhat "colder", because they result in almost exclusively odd harmonics. Using positive and negative feedback via the output transformer, an additional "sound design" is often added here. For this reason, VANDAL's A/B power amp features more juice in its lows and more bite in its highs.
- **Presence:** On many amplifiers, this pot is located next to the EQ section, although a presence boost actually takes place inside the power amp circuit. Most push-pull amps use negative feedback (from the output transformer back to the power amp's input) to linearize the amplification process.

Lowpass-filtering this feedback signal and mixing it (anti-phased) with the input results in boosting the mid and treble region. Using presence, the sound takes on a livelier and more up-front character.

- **SAG control:** Many older tube amps use rectifier tubes to transform AC current to DC (instead of conventional semiconductor diodes used today). However, a tube is a high-resistance component, and can't produce an even current flow during steep load changes. This "sagging" feature of a cranked-up tube amp is the acoustical result of these short-lived interruptions. Moderate sagging is initially perceived in the attack; it sounds somewhat compressed, but in a "lively" way. If the effect is even stronger, it changes the entire signal. Besides the dynamics, the harmonics spectrum also changes, because operating point of the tubes is shifted.
- Sagging occurs in this form only in push-pull end phases; in principle, Class A amps always draw constant (maximum) current from the mains.
- Intensive sagging will result in creating less "presence", as the overall signal gain in the power amplifier decreases, thereby generating only small voltage excursions to feed back on the power amp's input.

Preamp Channels



The guitar amp is set up with three channels in all preamp configurations (Classic, British, Modern High Gain).

- **Clean:** This channel should create no distortions. If the incoming channel is "hot enough", then it will be distorted in a typical tube fashion. The model will use only half of a double triode; in principle, a simple amplification takes place at the "working level".
- **Crunch:** Amplification stage number, circuit and, last but not least, the resulting sound all vary:
  - Two amplification stages are used in the "Classic" preamp setting. This model uses a long-established design that is similar to the classic Fender amps or the first Marshalls. The "classic" preamp delivers saturated lows that are only slightly dampened by the second stage. As a result, this circuit delivers the typical "brown" sounds of older vintage amps.
  - The "British" switch is inspired by dual input stages, such as Marshall "Plexi" variants (JTM, "Super Lead", etc.). The signal is sent to half of each triode circuit, and these favor different frequency ranges ("warm", "bright", etc). The "British" circuit in the VANDAL amp uses a set mixing ratio for both tube portions. In contrast to the "Classic" version, this preamp setting sounds more "alive" and reacts much more sensitively to dynamics, pick-up selection, and gain settings/guitar volume.
  - "Modern High Gain" delivers the typical fat sound that is rich in highs, similar to American amps like the Mesa/Boogie & Co. in 3 cascading amplification stages. Before the second and the third stages, the proportion of lows and

highs is regulated, so that low tunings and quick passages are amplified without any "mud" and the signal always remains clear and assertive.

- **Lead:** In all preamp modes, the lead channel consists of an additionally cascaded tube circuit. In the "Classic" and "British" preamp setting, the lead channel may be viewed as more of a modification of "tuning" of vintage aspects. The "Modern High Gain" variant really gets going here. In this setting, VANDAL playfully takes on even drop-C or -D tunings.

Tip: Even if you don't want to use up all your gain in the crunch & lead channels, you can still try to go "a level higher" and take the (pre) gain level down a little for a slightly distorted sound. This increases the sound's complexity, and it will also seem more lively.

Pre & Post Gain



You can set up the desired amplification factor with the pre and post gain controllers. The pre-gain controller corresponds to the normal "gain" controller on most amps.

Using the post-gain controller, the currently active channel's overall volume may be adjusted to another channel, or a certain channel may be raised (e.g. for a solo).

Don't worry about switching settings around in the middle of things; the amplifier will remember the gain settings when channels are changed.

Voicing



There are a number of more or less open secrets with regard to the different signature sounds of guitar amps. This one might sound "bluesy", but doesn't sound so great with more gain, and the other is contoured for grindcore metal and fat sounds, but maybe it's a little anemic otherwise... How does this work?

Most amps in the real world use more or less the same or similar circuitry designs. The important thing is the number of amplification stages involved. Each new stage not only increases the level of audio complexity, it also has a large influence on how the way the signal is treated before and after that stage. Filtering is the key:

In addition to preset filtering circuits between the amplification stages, we gave VANDAL a little something we call "curve EQ". Assuming that you are using an EQ pedal before the amp, you can change the distorted sound drastically, because instead on the overall sound, you can favor certain frequencies. Curve EQ does something similar: it's located (in some cases in multiple instances) at strategic

points between individual amplifier stages, and it filters the signal before it is distorted by the next stage. This is known as "voicing". For fun, turn a curve in both directions and move around a bit with the frequency controller in the spectrum. This will give the amp a completely different character. By combining channels, pre-amp modes and voicing, you can create a completely individual amp sound and even imitate other amps.

Equalization



The actual sound control (the "tone stack") functions rather conventionally: VANDAL offers low, mid & high settings.

Everything functions like the passive sound regulation network in genuine amps so that the controllers influence each other to produce numerous variations. Depending on the preamp mode, the activation frequencies are slightly different.

Reverb



Genuine studio reverb devices are great, and VANDAL has two of them included in both multi-effect units. However, styles such as surf, twang, sixties, and others require a spring reverb. We've drawn on famous reverb spirals for our modelling.

Everything sounds natural with complete authenticity. This means that it has an old-school style rattle and typical rippling on the attack.

Master



The master control may be used to determine more than just the total amp volume. Since this is located before the end tubes in real amps, you can also use it to control the end stage distortion level.

This is very easily perceived in the class A mode; it doesn't just get louder, it also becomes more compressed and distorted.

If you are used to amps without a master volume, simply leave this at the maximum setting and control gain via a pre-stage and the final volume and the cabinet simulation or at the console output.

Bass Amplifier



During development of the VANDAL bass amp, it was our main goal to create a recipe consisting of the best amp designs in a single amp featuring an all-tube design that would be simple and easy to use.

The core features of the VANDAL bass amp are:

- Multi-stage tube preamp (with extra distortion stage)
- Contour control for extra-fat bass and glittering highs
- Controllable optical-electrical compressor
- 4-band sound control
- Class A/B tube end stage

Gain Control



The input level of the first amplification stage is set in the bass amp via the gain controller. Low to medium values leave the signal relatively neutral, and high values softly increase saturation.

Contour



The contour circuit is a filter stage that works similarly to the "Loudness" function. This control lets you give the bass sound an underlying character without additional EQ measures.

Turned all the way left, no effect will be present. However, the more the control is turned to the right, the more the mids will be thinned out and lows and highs raised. It's sort of like an "instant slap".

Comp (Opto Compressor)



After the contour circuit, the signal passes through the compressor stage (Comp). This is a simple but extremely musical "optical-electrical" design: the bass triggers a light source (e.g. an LED or a luminescent film) that is coupled with a photo resistor.

The louder you play, the more light falls on the photo resistor, which in turn dampens the signal. This may already be familiar to you from the most famous studio compressor for bassists, i.e. Urei LA2A, which functions according to the same principle.

There aren't that many settings to change in such a simple compressor. The control times are mostly determined by the sluggishness of the photo resistors. The comp control only sets the signal strength, which is routed to the light source.

Drive



After any possible compression, "Drive" provides the option to take the bass sound to the next level. The drive circuit is an additional amplification stage, which is proportionally mixed to the main signal. Signal distortion takes place depending on the frequency:

In spite of a level of distortion, the basses remain relatively clean and contoured. This guarantees complete "traction" of the signal when the controller is set all the way to the right.

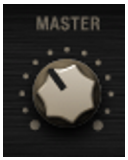
Equalization



The following equalization stage offers 4 frequency ranges, in which case the two mid bands are variable.

Some bass amps allow their EQ circuits to have a drastic effect on the signal. We decided on a softer filtering with an edge slope of 6 dB/octave to avoid fully counterfeiting the essential character of the instrument in spite of the four bands.

Master



The final master volume controller sets the volume at the final stage. Similar to the guitar amp, the end tubes are also engaged in this case as much as remains sensible.

Through the interaction of gain, drive & master, countless saturations and distortion textures may be modeled as needed.

Cabinet Simulation

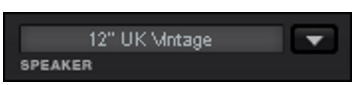


As was mentioned at the start, we went our own way in simulating the speakers and cabinets. In place of impulse responses from microphoned cabinets, we've calculated the individual components of this sort system:

- Loud speakers
- Housing
- Recording space
- Microphone

True to our "custom amplification" philosophy, this process provides total freedom and a certain measure of personal dynamics, which is the final deciding factor at the end concerning whether the system feels "lively". VANDAL also takes subtle effects into account, such as the reaction of the speaker's housing, the effect of this on the end stage that is applied, and so on.

Speaker selection



In the speaker selection list there are a wide variety of speakers in different sizes and with different sound characteristics. These are sorted according to guitar and bass types:

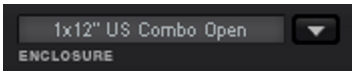
Guitar

- 10“ UK Vintage, 12“ UK Vintage, 12“ UK Modern, 12“ UK Blue, 10“ US Alnico, 12“ US Ceramic,15“ US Alnico, 15“ US Fat

Bass

- 10“ Custom Rock 10“ Custom Alu 15“ Custom Rock 15“ Custom Alu

Housing Selection



Various housing types may be combined with the available speakers in the housing list. This selection list is also organized according to guitar and bass types.

Guitar

- 1x12“ US Combo Open 1x12“ US Combo Closed 2x12“ UK 4x12“ UK 4x10“ US Tweed

Bass

- 4x10“ Ported (Bassreflex) 1x15“ Ported (Bassreflex)

The housing has a significant effect on the resulting sound with a selected speaker. In general, open housing types are somewhat weaker on the lows, but go lower and sound somewhat more neutral than closed types. Larger housings also sound "bigger" and maybe "fatter". However, this is not always desired; for a guitar solo or for certain styles like blues, a small combo housing may be much more assertive or simply more "charming".

It's up to your creativity whether you would like model the cabinets on realistic role models (e.g. 12" vintage speakers in a 4 x 12 housing) or place a 10-inch design in a 15-inch box. Refer to the "Advanced settings" a little further on for all other variations to help you create your own "personalised" speaker.

Microphones



Cabinet simulation offers two separate microphones for the recording of the virtual speaker in a modeled recording space (MIC 1 & MIC 2). Similar to a real-life situation, you may position a virtual microphone stand in a space to achieve interesting mixing ratios and stereo effects.

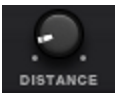
For each of the two available microphones, you may select from a list which includes several popular studio types featuring characteristic sound models:

- Condenser, Dynamic 1, Dynamic 2

For each selected microphone, a range of sound design parameters is available:



Axis: The "on axis" (controller to the left) setting corresponds positioning of the microphone directly in the middle of the speaker cone. In this position, the sound will have many highs, but may sound piercing. For this reason, the microphone will often be positioned a bit "off axis" in order to achieve a softer sound.



Distance: This control allows you to remove the microphone from the speaker and position it further away in the recording studio. This way you can make the sound more alive and loosen it up acoustically. Using two microphones with different panoramas (see below) enables you to create extremely realistic stereo images (note that the axis parameter has a decreasing effect in relation to the increasing distance).

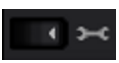


Pan: This parameter distributes the microphone signal in the stereo sum at the output of the cabinet simulation.



Level: This adjusts the microphone volume.

Advanced Settings



Clicking the wrench symbol accesses additional cabinet simulation parameters:



Speaker Distortion: As discussed at the start of this manual, we have also modeled the non-linear aspects of a speaker. These dramatically change the sound, depending on how hard it is working. Under heavy loads, not only more distortion is created, but the dynamics and the frequency spectrum also change. This control allows you to set the effect of the non-linear behavior within certain limits. Please note that the acoustic effect of this parameter also depends on the amp's end stage.



Cabinet Balance: We have already explained that the housing is an important part of the overall sound. This parameter allows you to set just how much effect this has. Turned all the way to the left, the speaker exists almost "free in space", turned to the right, you are picking up sound directly "in the box".



Enclosure Damping: This parameter sets the level of housing resonance. The normal value is in the middle of the control range. When turned all the way left, the housing sounds "emptier" (increased resonance). Turned all the way right, the speaker sounds as if it were covered in insulating material, and typical resonance features are reduced.



Room Size: This adjusts the size of the recording space being modeled. The more the controller is turned to the right, the bigger the recording studio's dimensions become.



Room Damping: Similar to enclosure damping, the materials involved and the room's capacity to absorb may also be changed. When turned to the right, the room is dampened to the maximum.



MIC 1 / 2 Delay: Turning either knob (or both) delays the corresponding mic signal to up to 30 milliseconds. This can be handy in creating spatial effects (thereby taking advantage of the "Haas effect", allowing localisation through time delay). Interesting new tonal textures are also possible by using just very small amounts of delay on either mic.



Phase Flip: These switches regulate the phase length ("in-phase"/"out-of-phase") of both microphones. This is helpful in case the signal from VANDAL needs to be mixed with others or in order to create interesting sounds by changing the positions of the virtual microphones.

Rack effects (FX1/FX2)



There are effects that don't always work well when placed before the amp, e.g. reverb or delay, especially when they are distorted. Normally, these effects are better placed at the end of the signal chain.

For final processing and enrichment or whatever else you need, we offer two separate studio-quality effects units that behave just like real 19" rack devices.

Many of these algorithms create a stereo signal. As required, make sure that the channel strip is being calculated as stereo in the audio sequencer track.



Effects units may be selectively operated one behind the other (serial) or parallel. Switching may be changed via the mode switch.

The following algorithms & effects are available:

- **Mono Delay (msec & tempo sync):** Selectable as a simple delay with freely adjustable delay time or synched to the sequencer tempo with a musical raster. In case of higher feedback values, a reduction of the damping frequency is required to provide natural echo sound.
- **Stereo Delay (msec & tempo sync):** Features two models, just like mono delay. Repetitions may take place on separate channels (feedback controller to the right: dual delay) or in ping-pong mode (controller to the left), in which case the signal alternates between the sides.
- **Chorus:** Produces a typical "floating/shimmering sound" via modulated detuning of a signal to "thicken" the sound or spread it across the stereo field. Detuning is achieved via a short delay, the length of which may be varied via the modulation. This produces the so-called "Doppler" effect and broadens the signal.
- **Flanger:** Similar in terms of the algorithm that is applied to chorus, but different in that the delay time is significantly lower and delay works via repetitions (feedback). The flanger sounds more cutting and up-front than chorus.
- **Phaser:** A modulation effect just like chorus & flanger, but in this case no detuning takes place. Filter components periodically alter the signal's "phase response" (principle of the "phase shifter"). When mixed into the original, characteristic notches are produced in the frequency spectrum response (comb filter effects).
- **Room Reverb/Hall Reverb:** Reverb offers realistic simulation of realistic reverberation. Room creates the impression of a small to mid-sized recording room, while reverb produces the impression of a concert hall. A particular is that both effects algorithms provide a modulation parameter, which may remove possible resonance at low dosages and can produce a soft chorus effect at higher values.
- **Vintage Plate Reverb:** The algorithms used in this type of reverb are quite similar to the ones that popular hardware effects units back in the 80s used to emulate that certain dense space of reverb plates. The resulting sound is correspondingly "wide", spacial and responds very directly, with immediate dispersion and no single echoes. Modulation of the reverb tail is also possible here, to minimize the typical ringing of the underlying metal-plate model.
- **LoFi:** Depending on its setting, this algorithm adds grit to the sound or a certain measure of signal destruction. Turn down the internal sample rate as much as you like to steal a few bits from the sound's resolution. Definitely an unconventional method...
- **Vintage Compressor:** Ideal for thickening up the signal a little. The algorithm emulates an older popular circuit design that is similar to studio legends like the Urei 1176 or simple compressor pedals. A so-called "FET building block" simply,

effectively, and quite musically controls the volume via the input level, the set compression level, the compression ratio, and the response responses (attack and release).

- **3-Band EQ:** This sound controller works like a conventional mixer with a controller for the bass, the highs, and two controllers for the variable mids. This adds the final polish to your sound.

Console



In the quick start (view page 1056) chapter, we mentioned that the console is the central switchboard for all of VANDAL's software. This interface is divided into various sections:

- Controllers for input and output levels, noise gate & limiter
- Management of all underlying components as a preset file
- Management of 5 remote controls for steering any parameters
- Chromatic tuner
- Advanced settings (MIDI, global)

Preset Menu



Presets are files within the VANDAL folder that consist of four scenes (but not all four must me used).

The default presets are organized in (sub)folders on the hard drive and may be located via the navigation structure in the presets menu.

An altered, but not yet saved preset is marked with an asterisk (*) in front of its name.

To save a preset, click on the diskette symbol and enter a descriptive name in the dialog that opens. Navigate to a subfolder or create a new one as required.

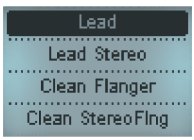
The number of possible presets available within VANDAL is unlimited. The same applies to variations to settings saved as new files. For the sake of preserving a clear overview, we recommend saving your own preset files in specific subfolders. The depth of the folder tree is unlimited.

Remote Parameters



The five controllers provide an elegant option for adjusting any desired parameter in currently activated components. These may be controlled via MIDI and via VST automation in the sequencer program. The remote control is also the interface between VANDAL and the outside world. The controls may be assigned via the advanced settings (view page 1083).

Scene memory



Imagine you have to put together a set list for a gig. A good idea might be to create a list in Vandal in the form of presets.

It might be an idea to create a new folder and enter the location and date of the gig as a title. Then save the individual presets for each song in it.

As a guitarist, you will probably want more than a single setting for each of your songs. If more gain or volume is needed in a particular solo section, it wouldn't be very productive to create an entire preset for this.

The solution Vandal offers is a range of four variations for each preset, these are so-called "scenes". These may be switched using keys 1 - 4 beside the preset list or by clicking the names. All presets which are made from now on will only affect to the currently active scene. Scenes can be changed around at any time without discarding settings.

Settings may be transferred easily to other scenes simply by pressing the copy button for the source scene, and then switching to the target scene and pressing paste.

Copy & paste also works for presets, i. e. the clip board is not cleared when you change the presets, this enables settings to be saved easily into a new preset.

Scenes may be switched externally via MIDI (view page 1085).

Chromatic Tuner



The chromatic tuner in the console has an automatic circuit which analyzes the recently played sound and shows the note and deviation from the idea (in cents) on the display. Simultaneously, the point moves up (too high) or down (too low) along the scale or stops in the center when the tuning is exact.

The arrows indicate the direction of the tuning (up or down).

The mute button switches the output signal from VANDAL to mute for as long as the chromatic tuner is activated (tuning fork symbol).

Advanced Settings

Clicking on the wrench symbol below the tuner opens a separate display.



On its right hand side, there is a sub menu for acessing in-depth adjustments of the software:

- remote ctrl (Control): Remote parameters for external control, either from the sequencer or via MIDI, can be assigned here.
- patch list: The underlying sub page serves for mapping MIDI or VST programs to actual presets. Because the preset library is a dynamic collection of arbitrary files on the hard drive, but the MIDI program scheme is tied to 128 possible programs, a proper mapping regime is needed. A "patch list" is used for that exact matter.
- preferences: Here, global settings can be made, affecting all instances of Vandal, regardless of the current preset.
- about: Shows the current installed plugin version and provides clickable links for the product web site and additional downloads.

Remote Control



You can use a standard MIDI controller in conjunction with Vandal to externally control any of the program's parameters. For example, preset changes present no problem either using program change commands, or using the wah-wah pedal via MIDI control change. A MIDI configuration may be set for each preset and each scene saved in Vandal.

A maximum of ten different control commands may be assigned at the same time (assignable in two rows of five). The source for controlling may be external MIDI, but the five remote controllers below the preset display are also available. An individual remote-controller/MIDI controller can control up to four parameters.

For each target parameter, a selection may be entered for the type and level of the control. For example, "Add to param" indicates that the value of the external controller or remote controller should be added to the parameter that is being controlled.

Let's assume that the controller, which is sending MIDI CC 7 (Volume), should also control the guitar amp's gain in the lead channel from zero up to the current control setting. Simultaneously, the master volume should be reduced to compensate for the volume. Proceed as follows:

- In the Remote 1 view click on MIDI learn in the MIDI controller range or select CC 7 volume from the list By doing this you have specified that you want to control the remote memory with your pedal.
- In the first target list, select "Guitar amp -> PreGain lead" as the target.
- In the list next to this, select "Multiply" as the control method.
- Move the fader right to the 100% mark. This indicates to the control that the depressed pedal should match the current controller maximum. If for example the gain fader is at 12 o'clock (i.e vertically) then a half-pushed pedal would give you a gain of 25%.
- In the second list, select "Guitar amp -> Master volume".
- Select Subtract as the method this time.
- Try setting the fader to 10-15%. Of course, the level of compensation depends on the concrete sound; if any at all... This is, after all just an example.
- Enter a sensible name for "Remote 1" by double clicking it, e .g. "Pedal gain".

The MIDI/Remote configuration may be left again by clicking the wrench symbol. The main display will now feature "Pedal gain" in the first remote storage section and the assigned MIDI controller (volume).

Patch List



Because the preset library is a dynamic thing and can hold huge numbers of files (sheer unlimited), they can't simply be accessed by MIDI program change messages from outside. MIDI knows only about 128 programs, and it doesn't have a notion about the plugin's file organisation, for instance sub folders, or even multiples of those. For that reason, a "connection" between these two worlds is needed. That's where the patch list comes in.

This list is shown as pages of 16 programs / program slots. Each represents a MIDI/VST program.
Using the "edit" menu, the current page to edit is brought up. Alternatively, an external MIDI or VST program can be sent; the underlying page is automatically picked and the selected program is highlighted in the display. Presets can be simply assigned to programs by choosing an existing file from the corresponding menu.

A patch list can be saved using the disk icon. Just like the presets regime, multiple patch lists can be created and you can set up a library of different lists, each serving a particular purpose.
This can be handy for musicians gigging a lot, and changing their set very often. Using seperate patch lists makes programming an external controller much easier, as it only needs to be fed with the programs you really need for a show.

A selected / saved list can be stored as your preference, by hitting the "as default" icon. This is a global setting, affecting all Vandal instances. You'll find this setting again on the "preferences" page, labelled "startup patch list".

Preferences





On this page, numerous in-depth settings to the software can be accessed. These act as "global" modifiers, which means they affect all (following) instances of Vandal, and they work regardless of the current chosen preset.

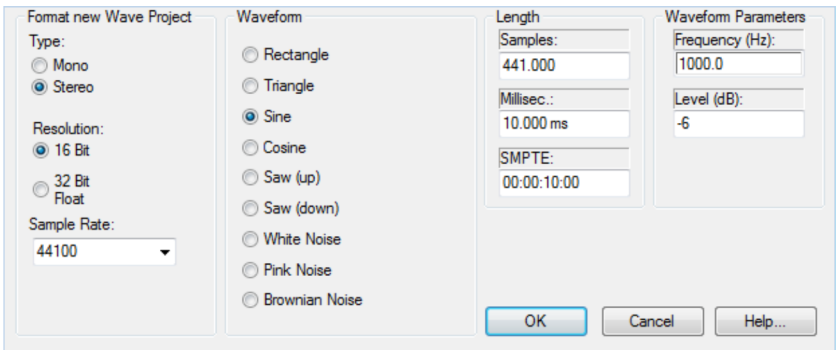
- startup preset: Here, the default preset is specified, which will be loaded at plugin instanciation. Keep in mind that your sequencer program can freely override this preference, by sending a MIDI/VST program change command as last in the boot-up sequence, which might trigger a preset from a "patch list". So, this setting should probably be considered a fallback, in case no default patch is found.
- startup patch list: As mentioned earlier, this is the default list of preset files that get assigned to MIDI/VST program numbers.
- direct switch scene ← midi: The 4 scenes of a presets can be switched independently via MIDI. You can specify here, which scene is accessed by which command. Possible data types are either MIDI note number, or program change messages.
- scene skip -/+: for external controllers having up/down switching capability, proper control commands for skipping scenes can be chosen here.
- skip mode: You can specify, whether skipping should stop at either end, or whether further commands should result in a round-robin fashion (e.g. from scene 4 to scene 1).
- preset skip & skip mode: Just like with scenes.
- default preset scene: This menu specifies whether the first scene is activated upon preset load, or whether the preset should recall the one that was last selected by the time it got saved.
- input & gate level: Typically, the knob settings for the console's gate and input level are read directly from the current preset. Choosing "use current for all", you can specify global values as the reference. The values stored are taken directly from the current knob readouts.
- window resize: Normally, the two little buttons on the console's left side (below the "V" symbol) switch the stomp and the amp/cabinet/fx section on or off. This manual mode can be changed to automatic. Then, presets containing stomps and an amp will automatically maximize the window, while the view will collapse as soon as either the stomp or amp component slot isn't occupied. The window can also be forced to stay always maximised (so that the above mentioned console buttons have no function). The maximised mode might be helpful with some host programs that show dynamic window resize issues.
- mouse mode: Choose your favourite method here for handling knob movements. Knobs can either be turned in a linear fashion (up/down), or as a circular movement.
- tuner reference frequency: In case your music asks for a tuning other than standard 440Hz, the tuner can be recalibrated here.
- carpet / guitar amp overlay image: You can "skin" Vandal's stomp background (the "carpet" as well as the guitar amp's front cover (cloth) to your personal liking, by specifying own images.
  A "custom image" has to be in PNG format and reside inside the "_custom" folder (found inside the Vandal installation branch).
- image opacity: "Custom" images with transparency/alpha information are supported. Likewise, an image can be softly blended in or replace the background completely.

Cleaning & Restoration Suite

The Cleaning & Restoration Suite consists of DeClicker/DeCrackler, DeClipper, DeHisser (view page 935), DeNoiser, Brilliance Enhancer (view page 895) and Spectral Cleaning (offline) (view page 948).

Waveform generator

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.

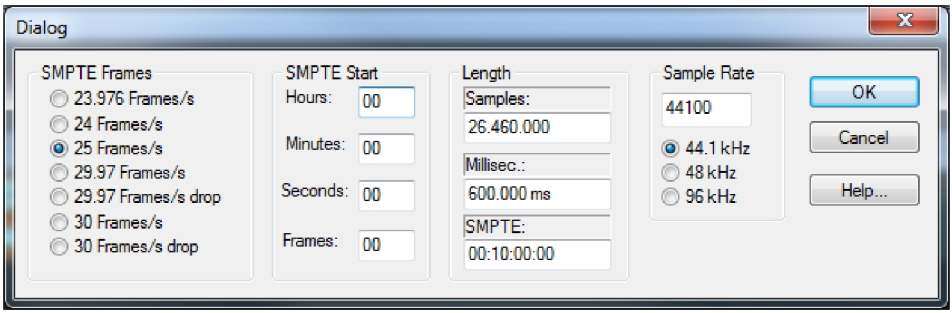
The following **sample rates** are available: 22050, 32000, 44100, 48000, 88200, 96000, 176400, 192000, and 384000.

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.

SMPTE Generator



If you have no external MTC to SMPTE converter to synchronize to analog equipment, this dialog allows you to create audio files containing an SMPTE time code. To do this, place the generated SMPTE signal onto an empty track and route its output to the SMPTE sync input of the analog recording device.

You can choose from the following specifications as **SMPTE frames** per second: 23.976, 24, 25, 29.97, 29.97 drop frame, 30, 30 drop frame.

The **SMPTE start** value can be entered in hours, minutes, seconds, and frames.

Specify the **length** in samples, milliseconds, or SMPTE code.

You can choose from the following **sample rates**: 44.1kHz, 48kHz, and 96 kHz.

Process Only Left/Right Stereo Channel

Choose whether you wish to edit the left or right channel here. Of course, this is only useful if the signal you wish to edit is a stereo signal. Clicking again deselects everything.

Process Effects Offline

All effects opened using this menu are calculated destructively, provided the option "**Apply effects offline**" is active. The option to work with a copy is available in order to preserve the original audio material. The "**Create copy**" option is already selected in the corresponding dialog.

CD/DVD Menu

This menu contains special functions for the CD/DVD mastering process like setting CD tracks, sub-indices and dialogs for creating CDs or Audio DVDs.

Sequoia allows you to directly burn CDs from any project virtual project or any stereo HD wave project. A sample rate of 44.1 kHz is required. 24 bit objects are converted into 16 bit objects in the CD burning/CD trackbouncing process.

"CD arrangement" mode is especially useful if existing wave files or wave projects are going to be used to burn a CD. Wave projects are arranged behind one another on one track in the VIP when loaded. The distance is set according to the CD pause time ("CD/DVD menu -> Set pause time (view page 1116)").

Tip: If you want a finished virtual project to be burned as a track onto CD, use the command "CD/DVD -> Create CD -> Generate completely new file" (view page 1092) beforehand to convert this multitrack project into a single wave project. Don't put any CD's into the drive and end the burning process after bouncing. The newly created Wave file can then be placed into the new CD VIP as a track object.

Import audio CD track(s)

This function allows the import of audio data from CD/DVD drives. The data is imported digitally which eliminates any loss in sound quality. The CD tracks are imported into the project as wave files.

To import audio tracks you should proceed as follows:

1. Open the "**CD/DVD**" menu or go to "File -> Import -> Import audio CD track(s)" (view page 620).

2. If you have more than one drive installed, click the button "CD 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 "CD Drive Options" along with read speed and copy mode displays.

If the drive letter of the CD/DVD burner has changed in the meantime, you will have to reset the drive list in order to guarantee correct allocation.

3. Select the desired track(s) using the keyboard combination "Shift" or "Alt" and the arrow keys.

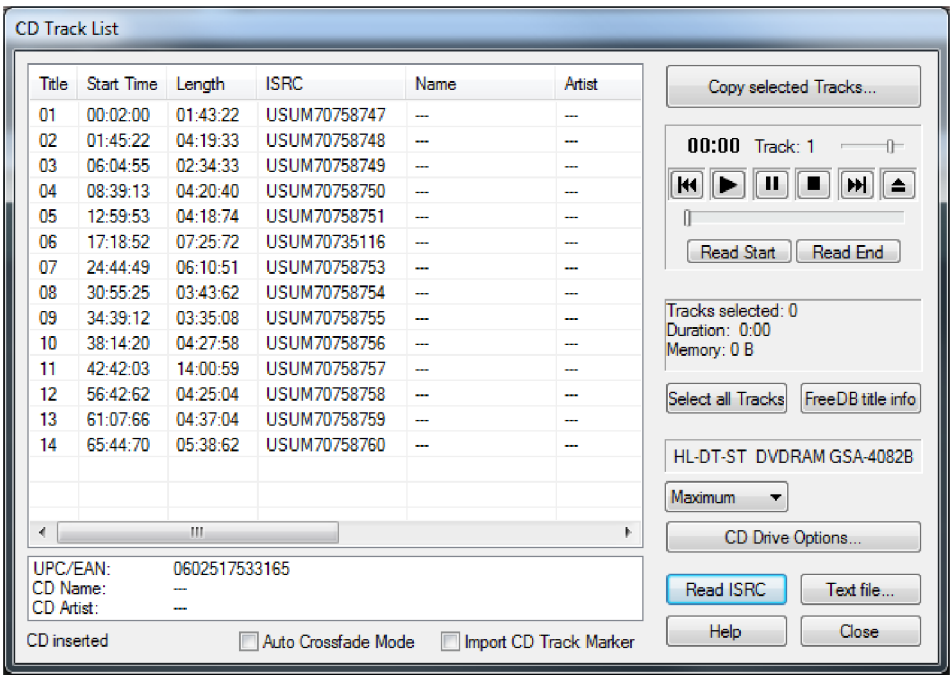
4. Click on "Copy selected CD tracks..."

5. Select a file name for the created WAV file or HD Wave Project and click on OK.

6. The audio material is now copied onto your hard drive from the CD. A progress bar indicates the status.

7. Close the track and drive lists. New objects will appear in your VIP containing the audio material from the CD.

Track List Dialog



Copy Selected CD Tracks: This button begins the audio copying process. All selected tracks are copied into a WAV file or into an HD wave project. A new object is created in the active virtual project for every track.

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.

CD Drive Options: Configure the system drive list here.

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.

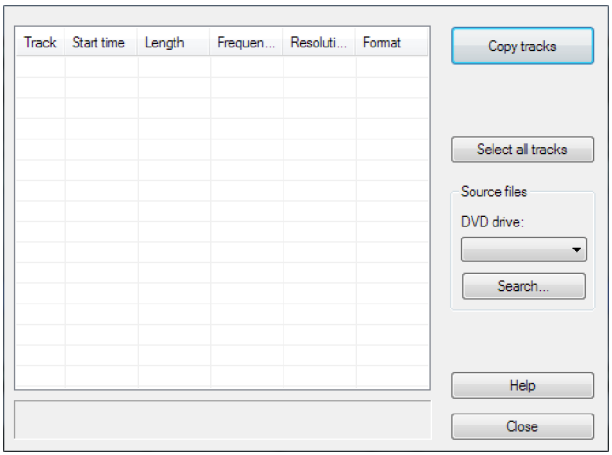
Text file: Exports all of the information listed as a text file.

Auto Crossfade Mode: Switches on Auto Crossfade Mode when importing audio tracks. Crossfades will be automatically set between imported tracks.

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

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

Import Audio DVD...



This function allows the import of audio data from DVD drives. The DVD tracks are created as wave files and integrated into the virtual project during importing.

To import audio DVD tracks into a virtual project, proceed as follows:

1. Open the dialog via "**CD/DVD**" or via "File -> Load/import -> Import audio DVD".

2. If more than one drive is installed, click the "DVD drive" list to open the drive list dialog. Select the desired DVD drive.

3. Select the desired track(s) using the keyboard combination "Shift" or "Alt" and the cursor keys.

4. Click "Copy tracks"

5. Enter a file name for the resulting wave file and click "OK".

6. The audio material will be copied onto the hard drive from the DVD. A progress bar indicates the status.

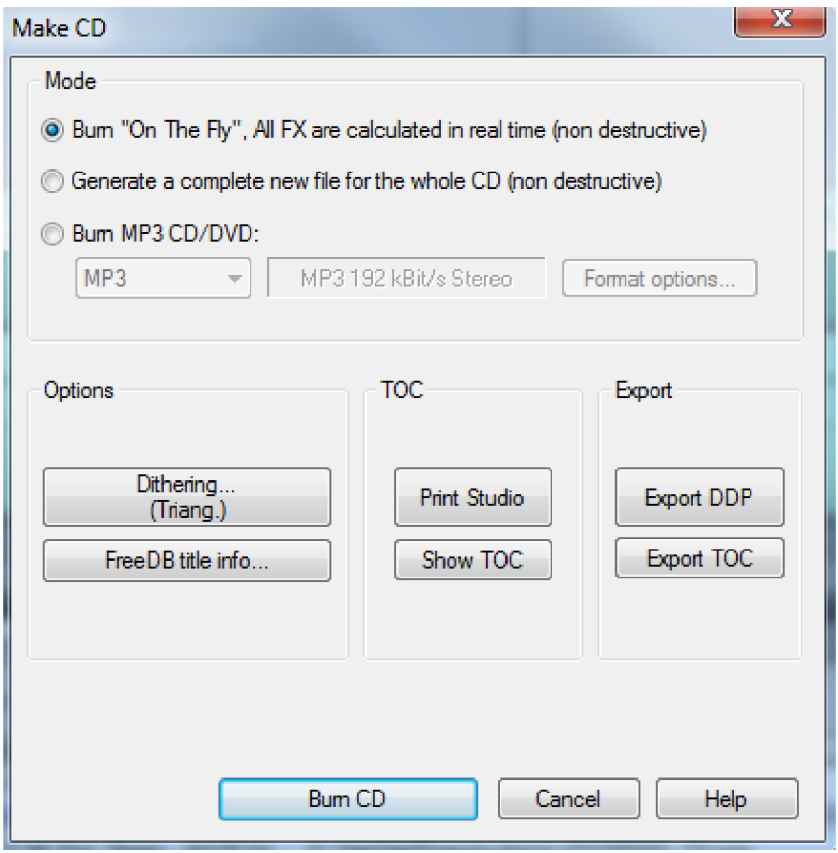
7. Close the track list. New objects will appear in your virtual project containing the selected audio material from the DVD.

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

Make CD

This dialog starts the CD writing process. Sequoia contains high-quality and constantly updated CD burn routines which are licensed from "Point Software & Systems".

Sequoia creates a so-called TOC file (Table of Contents) before the CD burning process starts, and this features the name of the current project and the file extension TCX. This file is saved via the same path as the project itself.



"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. Mainly it deals with:

- Object effects, volume and panorama settings
- Fades/crossfades
- Mixing tracks
- Mixer track effects

- Effects from the mixer master section
- Plug-ins used in the mixer
- 32-bit float to 16-bit conversion and dithering

Generate a completely new file for the whole CD: Use this function to create the CD if your 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 MXCDR tool. This provides the opportunity to load more files or begin burning a data CD.

Detailed information about **dithering options** is available in "Options -> Program preferences -> Dithering options (view page 703)".

Get FreeDB title info...: This function causes Sequoia to search the freedb.org database for information that corresponds to the track and displays it if found.

Print Studio: This button starts the external "Print Studio" program. Use this tool to easily print the content file of the current CD. You can choose between a text style format to print the production documentation and a formatted printout for the CD jewel case.

Show TOC (view page 1114): This button opens a text window with the current table of contents. You can also open this data with current CD contents in an external editor.

DDP export

The additional tool DDP Export will be launched.

DDP Export is available only in Samplitude ProX Suite und Sequoia.

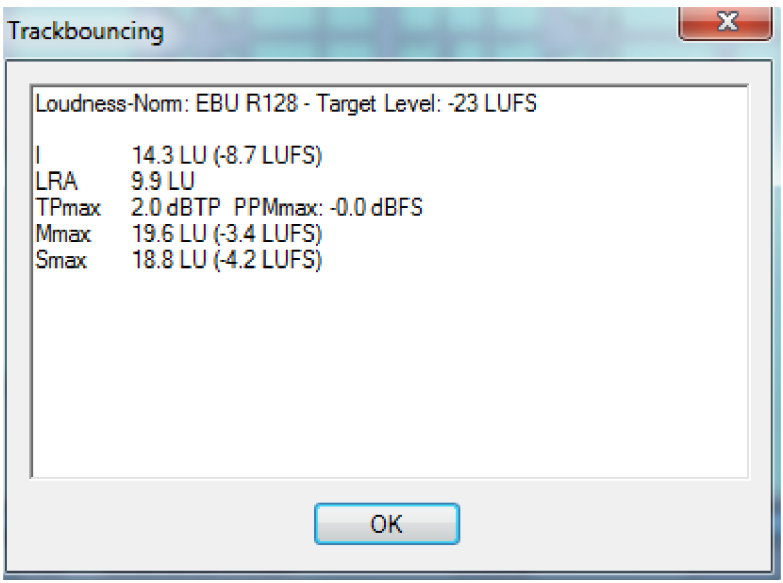
This additional program enables the export of a CD project that is compatible with DDP-CD mastering standards. It can be opened by pressing the "Export DDP" button in the "Create CD" dialog.

During exporting, the following files are created: **DDPID (DDP ID stream)**, **DDPMS (DDP map stream)**, **Image DAT**, and **PQDESC (PQ description), CHECKSUM.TXT (CRC32), and CHECKSUM.MD5**. Zip these before you send them to a CD manufacturer **via FTP**. If the manufacturer can unzip the files properly, you can be certain that the files are consistent.

The DDP ID stream (DDPID), the DDP map stream (DDPMS), the audio data (Image DAT), the PQ description (PQDESC), and the check sums CRC32 and MD5 can also be **burned to DVD** for delivery to the manufacturer with the note "Replica: CD-DA".

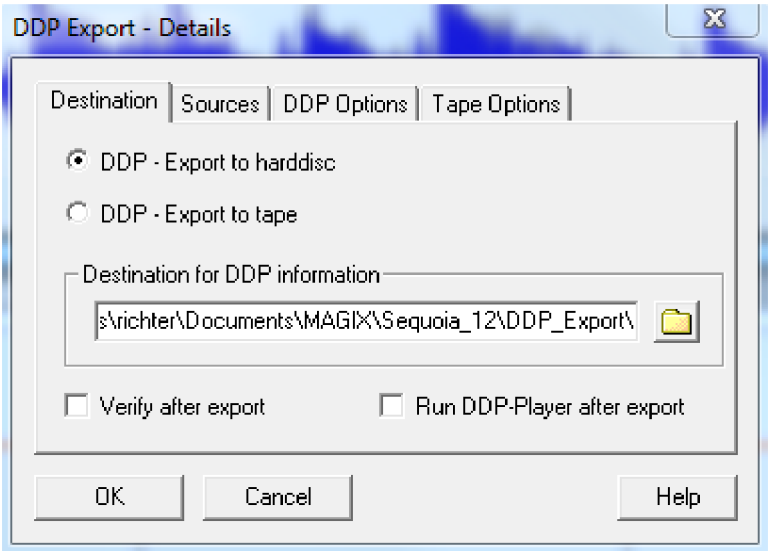
Note: Sequoia creates an additional WAV file during DDP export. This does not need to be transferred to the manufacturer, since the audio data is already included in the Image DAT file.

Simply enter a name for the project's CD audio files. Sequoia will now conduct trackbouncing of the virtual project and show the volume at the end.



At the end, the tool for DDP export will appear.

The project can be exported either to hard disk or onto tape (for example, Exabytes EXB-8505, EXB-8500). A project exported to the hard disk can be transferred to a common medium like a DVD or sent per FTP to the manufacturer.



Destination: Enter a target folder for **exporting to hard disk**. Enter the path to the tape drive in "**DDP - Export to tape**". Use the function "**Verify after export**" to start importing the CD images for verification.

The "**Run DDP player after export**" option opens the "MAGIX Sequoia DDP Player by Sonoris" directly after export.

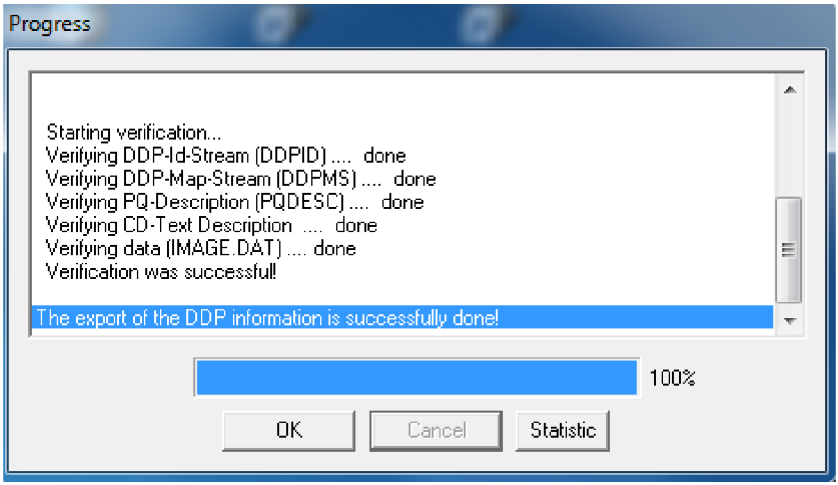
Sources: The **CD info file (*.tcd)** associated with the project is created automatically by Sequoia. If the DDP export was not initiated from within Sequoia, but rather another program outside of the Sequoia program folder, select the corresponding files here (CD info *.tcd + audio file/data file). "**CD info details**" shows the contents of the CD info file.

DDP options: The formats of the **1.01 and 2.00 standard for DDP export** can be used and a **Master ID entered**. This ID is used by the manufacturer for testing purposes.

Tape options: Enter a "**Volume ID**" and an "**Owner ID**" for archiving.

Press "**OK**" to export the CD project. Sequoia calculates and compares the check sums CRC32 and MD5 and combines them with the DDPID, DDPMS, Image DAT, and PQDESC files in the **DDP_Export** folder within the target path.

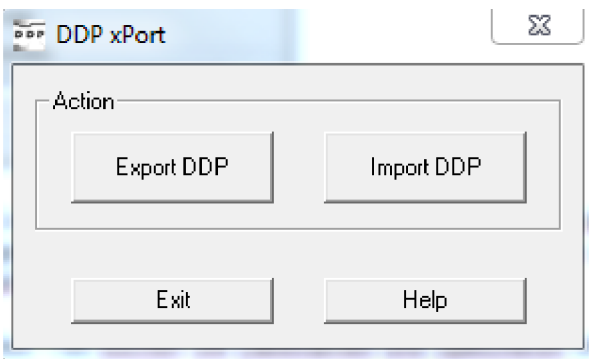
After this process is complete, you can view some statistics regarding the export process.



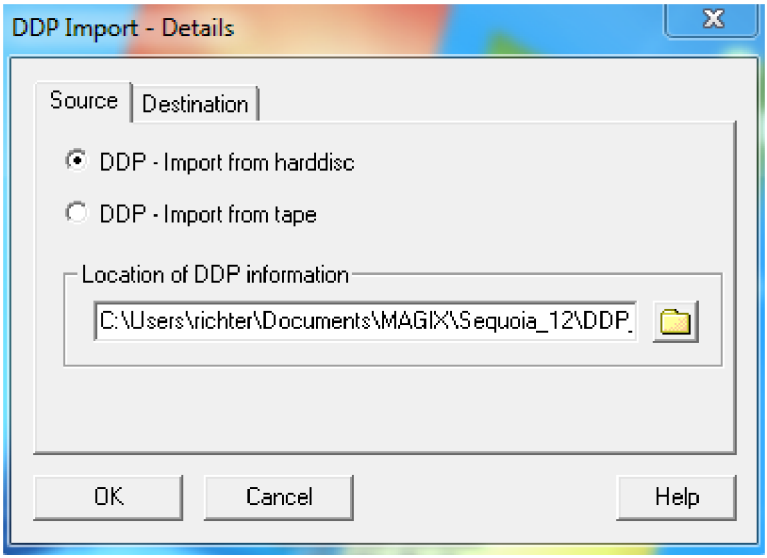
Import DDP files

To import a DDP project into Sequoia, proceed as follows:

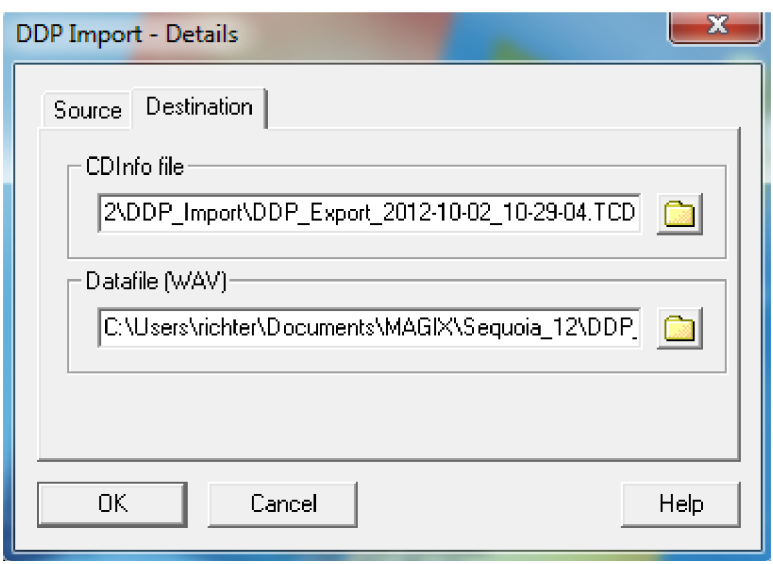
Click on the "DDP Export" button in the "Create CD" dialog or start the DDP export program from the Sequoia folder and select "Import DDP".



In the "**Source**" you can select whether you want to import from the hard disk or a tape deck. If you picked "**DDP - Import from hard disk**", indicate the folder with DDP data under "**Location of DDP information**".



The "**Destination**" tab allows you to select the file name and save location for the CD info file (*.tcd). The associated audio file will be automatically named accordingly.



Start the importing process by pressing "**OK**". The **CRC32/MD5** values are checked automatically if the CHECKSUM.TXT or CHECKSUM.MD5 are available. The result will be an audio file with the CD's entire audio data and a corresponding CD info file containing all track and other relevant CD information. All files used for this are located in the folder "**DDP_Import**" according to the target path.

A further option for importing DDP files is available through "**File > Open project > DDP import file**". A new arrangement opens including the complete CD project.

Sonoris DDP

Introduction

The Sonoris DDP Player is a standalone application that imports DDP 1.0x and 2.00 images for playback. You can playback tracks, audition gaps, see all PQ codes, ISRC, MCN and CD-Text data and automatically check the project for RedBook compatibility.

Features

- Import DDP 1.0x and 2.00 image files (loadback)
- Enhanced CD support
- Playback of tracks
- MD5-Check
- RedBook compliance checking
- Time display selectable between disc time or track time

Sonoris DDP – Basics

Audio format

A CD holds 16 bit, 44.1KHz stereo audio. This audio is stored in a whole number of frames, where every frame consists of 2352 bytes of audio. At a sample rate of 44.1KHz this means that the frame rate is 75 frames / second.

Tracks and indexes

A CD can hold 99 tracks. Normally each song gets a track number, but it is also possible to have more tracks in a song. This can be useful if you want to give sections in a song a separate track number, as is common on classical recordings.

Another, less common way to divide a track in sections is to use indexes. Some players show them but it is not widely supported. Every track can hold 99 indexes. Index 0 is a special index, it defines the pre-gap or pause before the audio starts. Most players show index 0 by a negative countdown to zero. However you don't need to use the Index 0. If you don't use it then you end up with a gapless CD (if burned in SAO or DAO mode). However, the very first index on a CD does have to be an index 0 entry of 150 frames (2 seconds) length minimum as is defined by the RedBook standard. It is also mandatory to have an index 1 for every track.

An index 0 entry normally defines a pause between two tracks, but it is possible to fill it with audio. While this is not strictly RedBook compatible, it is supported by many players.

ISRC, MCN

It is common to add a media catalog number to a disc. This MCN number the standard UPC/EAN number and should conform to the specifications of the UCC and EAN. EAN numbers are 13 digits long and UPC numbers 12. UPC numbers should therefore have an extra 0 at the beginning. The last digit is a checksum of the preceding digits.

ISRC numbers are issued per track and have the following format:

CCXXXYYNNNNN, where CC is the 2-character country code, XXX is the alphanumeric registrant code, YY are the last two digits of the year and NNNNN is a unique 5-digit number.

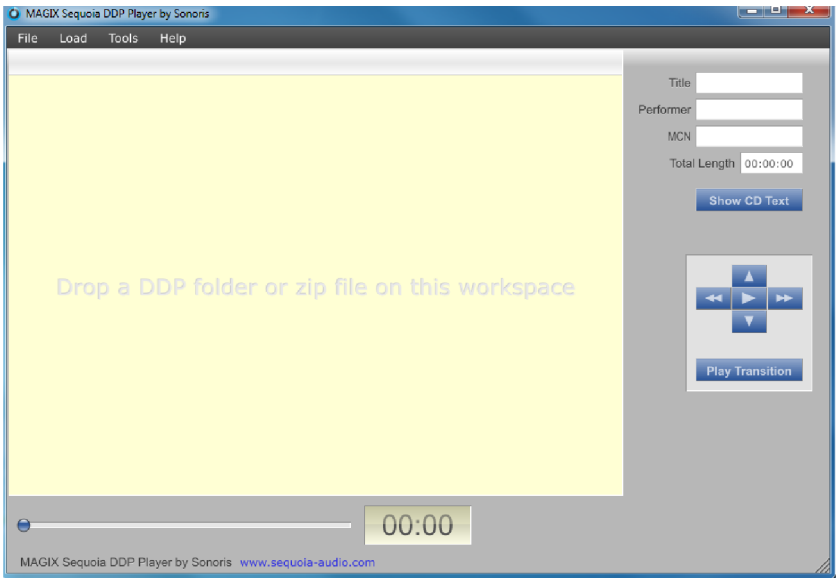
Important: There should be only one ISRC code on a track and that it is only valid on the first index entry for each track.

CD-Extra

Besides audio data a CD can also hold computer data like images, videos, song lyrics or even program files. A CD with additional data is called an Enhanced CD. It is possible to add data in the space before the first track or as a separate data session on the CD. The DDP Player supports the extra session as this is the most common and compatible format. When an Enhanced CD is played back in a normal CD player it will play just like any other audio CD. But when this CD is opened on a Windows PC or Mac it will show the data session as an extra drive.

Sonoris DDP – Main Screen

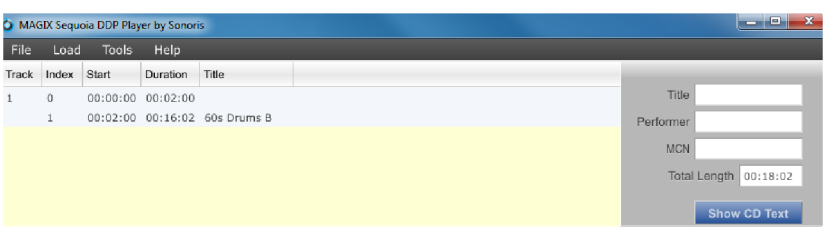
When started the "MAGIX Sequoia DDP Player by Sonoris" program interface will appear. On the left there is a large window with the project workspace. On the right there are some text boxes and a few buttons.



Workspace

The workspace consists of a table with the following columns:

- Track: track number of the entry, ranging from 0 to 99.
- Index: sub index of the entry, ranging from 0 to 99 within a track.
- Start: absolute position of the index entry.
- Duration: length of the index entry in frames.
- File Name: file that holds the audio of the index entry. This can be a wav, aif(c) or DDP image file
- Title, Performer, Songwriter, Composer, Arranger and Message: Alphanumeric information. This information is used for CD Text only .
- ISRC: ISRC code of the track.



You can adjust the columns by dragging the headers or the separators between headers. You can also hide/display columns by right-clicking on a header and selecting or deselecting the desired column.

View Disc Data

The CD Text information for the disc can be viewed by clicking on the "Show CD Text" button.

Playback tracks

To play back a track, select an index in the workspace and click the "Play" button or press the Space key. Clicking the button or pressing the Space key again pauses the playback. Playback will stop when the user presses the pause button or at the end of the project.

It is possible to navigate during playback by clicking on the left and right arrow buttons. Holding them will fast forward or rewind. The playback slider displays the relative track position and can be adjusted to go to any position quickly.

Playing a transition between two tracks

You can audition the transition between two tracks by pressing the "Play Transition" button. Clicking the button or pressing the Space key again pauses the playback. You will hear the end of the last index of the current track played into the first index of the next track and if a pre-gap exists it will be played in between. The pre-roll time can be set in the Settings menu.

Sonoris DDP – File Menu

Exit

Exits the program.

Sonoris DDP – Import

DDP Image

Loadback (import) of a DDP 1.0x or 2.00 fileset, regardless of the image data is stored in one file or more. CD Text data is imported if it exists. After the import, the project window shows the current DDP image as a regular project. You are prompted for a folder holding the DDP image.

This function enables you to import a DDP images made by any application to verify PQ points, CD text, ISRC, MCN and other data.

Sonoris DDP – Tools

Check Project

The current project is tested against a number of RedBook rules. The following criteria are used:

- MCN code must have the proper format
- First pre-gap must have a duration greater than 2 seconds
- Maximum length is 80 minutes
- Maximum number of tracks is 99
- It is also mandatory to have an index 1 for every track.
- The maximum number of index points for a track is 99
- Every track must have a duration greater than 4 seconds
- CD Text data on an index point other than 1 is not allowed
- ISRC codes must have the proper format

This test is automatically started before all export functions and PQ Sheet printing. If any of these criteria is not met an error is shown. Non critical errors can be ignored.

MD5-Check

Prompts to open a checksum file (with extension md5). It tests whether the checksums of the files found in this file matches with the actual checksums of these files. This function allows you to check the integrity of a DDP fileset after downloading or reading them from disc.

Settings

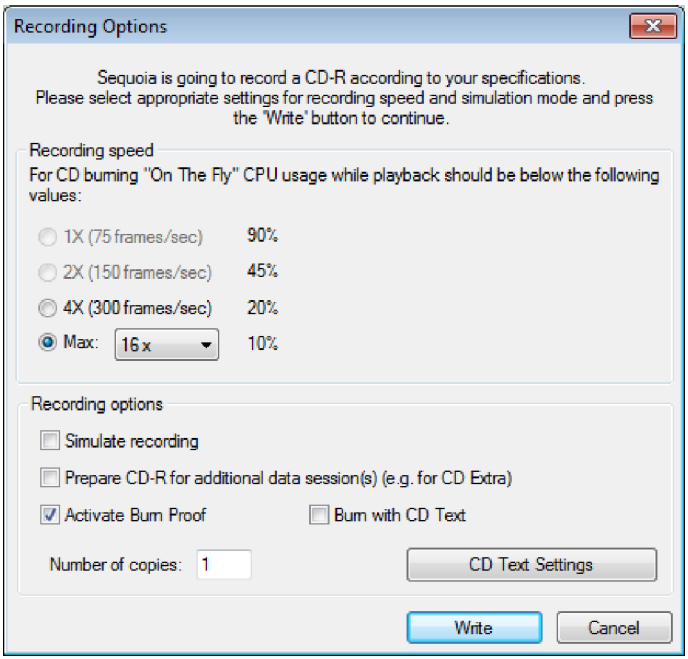
In this dialog you can fine tune the program settings. These settings are not saved with a project but will be saved as program settings.

- Pre-roll time: this setting is used during the playback of a transition between two tracks and defines how much time at the end of the first track will be played before the transition and how much time of the second track will be played into. Enter a value between 5 and 20 seconds.
- Time display: selects whether the main time display shows the track time or the disc time.

TOC Export

Clicking on the „TOC Export“ button saves the information from the CD content folder in a *.toc file (view page 1114).

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 (view page 1116) editor.

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

Create Audio DVD

With Sequoia you can burn DVD audio discs with any DVD burner. You can burn DVDs in the formats +R/-R/+RW/-RW.

Note: DVD audio discs can only be played back on players with DVD audio capabilities. This is indicated by a visible DVD Audio logo somewhere on the device.

For multichannel playback your DVD audio player requires multiple analog outputs.

Sequoia burns so-called "black discs", i.e. DVD audio without graphic menus. You can create video DVDs with menus, videos and slideshows using the video editing and DVD authoring program "MAGIX Movie Edit Pro".

On a DVD audio disc you can record audio in 16-bit and 24-bit with sample rates of 44.1 and 48 kHz or even doubled and quadrupled sample rates. Up to 6 audio channels are supported in 5.1 Surround, stereo, or 4.0.

The only limitation is that the maximum guaranteed data rate of around 10 Mbit/s for hardware players cannot be exceeded. For example, for 5.1 Surround 24-bit sound, a sample rate of only up to 48 kHz is possible, since at 96 kHz, the 10 Mbit threshold would be exceeded.

The following playback times are approximations for the available space on a single layer DVD-R (44.1 kHz sample rate):

Stereo 16-bit	approx. 7 h
Stereo 24-bit	approx. 4.5 h
5.1 Surround and 24-bit	approx. 1.5 h

Create DVD-Audio...

First, set CD track markers in your virtual project. The same project may be used to burn an audio CD and an audio DVD immediately in sequence.

When creating an audio DVD, the "trackbouncing" dialog will open. Select either "**Create DVD audio**" or "**DVD video with stereo PCM**".

Note: Note that the creating a "**DVD video with stereo PCM**" does not provide authoring or image information. For extensive authoring options, we recommend "MAGIX Movie Edit Pro".

Format:

You can choose between **16 and 24-bit audio**. If "**Create DVD audio**" has been selected, then the sample rate may also be defined.

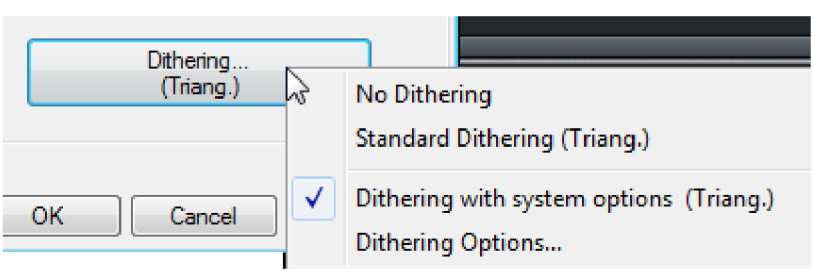
An external program will open if "**Path**" is pressed. The path to this program will be preset, but it may also be changed.



Options:

Calculate maximum amplitude: Display the maximum volume in dB after bouncing in order to correctly set equipment for additional editing or to correct the master level. Once bouncing has been completed, the corresponding information window will be displayed.

Dithering: Every trackbouncing process may feature its own dithering process, independent of the global settings. This enables dithering to be bypassed or to apply the standard dithering (dithering featuring triangular noise).



Use this dialog to apply dithering according to the system options or to access the dithering options in the system settings. The button value in brackets (e.g. **Triang.** or **POW-r 1**) indicates the currently set dithering algorithm.

Detailed information about this is available in "Options -> Program preferences -> Dithering settings (view page 703)".

DVD info:

Learn details about **project length**, **maximum DVD length**, and select either **PAL DVD** or **NTSC DVD**.

Source:

Selected range only: In this case, trackbouncing is only applied to the length of the selected range in the arranger. This function does not work track-selectively, i.e. all tracks of the selected range, except for muted tracks, are used in bouncing.

From the project start to the last object: In this case, the bounce process includes all objects from the project start to the end of the last object, plus the time it takes for the sound to dissipate.

Complete project: If this option is selected, the entire virtual project is bounced.

Surround trackbouncing:

Save the individual tracks in the Surround project as mono tracks or stereo tracks.

Note: Double the storage space of a DVD is required on the hard drive, since the project is bounced first and the DVD image is then created from the complete audio data that is burned.

The burn dialog allows burner and speed selection. The red "Start" button creates a DVD image first, and the writing process starts after this is completed.

Indices (Track markers)

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.

Use the function "Set Track Indices on Object Edges (view page 1109)" to create a track marker automatically at each object in the virtual project.

To manage the markers or rename them, use the "CD track/index manager (view page 1110)".

Shortcut: Ctrl + Alt + I

Set CD Sub Index

Choose this option to place a sub-index marker at the current playback marker position. The numbers of the subsequent sub-index markers are customized automatically.

Sub-index markers are not required for CD creation, but they allow CD players to cue to specific points 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 playback marker 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 command marks the end of the CD. Setting an end marker corresponds with two main applications:

1. It may be the case with some projects that the last object reverberates after its actual ending. To prevent Sequoia from cutting this section of audio during burning, you can place the CD end marker at an appropriate distance after the last object.

2. If you only want to burn a project partially to a CD, place the first track marker at the desired position in the arrangement and mark the end of the burn process with the CD end marker. 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.

Set Track Indices on Silence...

If you are working with a longer audio file that contains multiple tracks, this function can be used to set track markers between the titles automatically.

Min time: Period that the signal must be below the set level before a marker is set.

Threshold db: A marker is set at the positions in the arrangement where the signal sinks below the threshold volume for the set period of time.

Start number: Marker number after which automatic indexing starts.

Prefix: Enter additional characters which will precede the marker numbers set by this function. This makes it possible to easily differentiate them from markers that may already exist.

Add time (ms): Use this value to specify how many milliseconds will be set in front of the markers in the arrangement.

Delete all markers with prefix: Deletes all markers in the project featuring the set prefixes.

Delete all markers: Deletes all project markers.

Set Track Indices on Object Edges

Choose this option to place a track marker at the beginning of every object in the VIP.

Set Track Indices on Object Edges - Options

Set also Pause Indices on Object Ends

This option automatically generates pause markers at the object ends. This option allows you to set an offset for the generated track markers.

Time Offset for Indices on Object Edges

This dialog provides options for adding an offset before the object edges when placing indices.

No indices on Object Crossfades

This option prevents track indices from being placed between objects that are connected by a crossfade.

Remove Index

This function deletes a previously set track or sub-index marker. First, click the marker and access this function.

Shortcut: Del

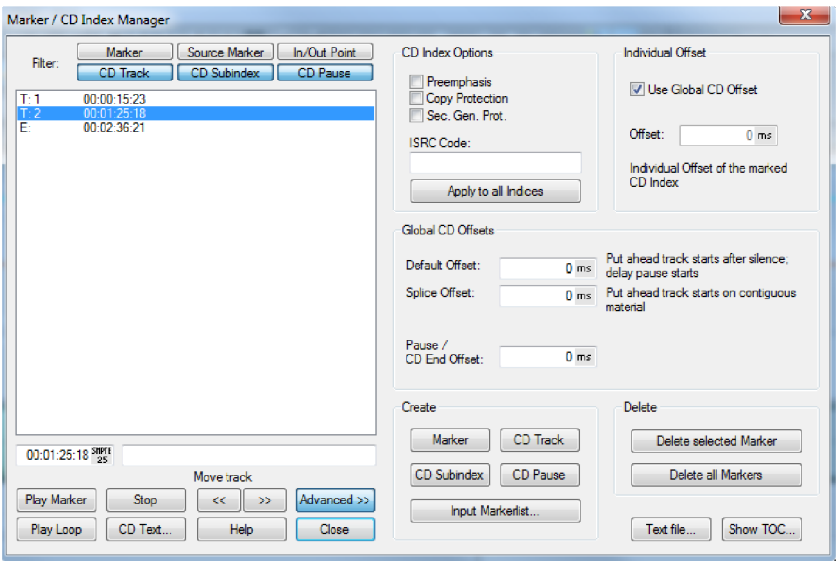
Remove All Indices

Choose this option to delete all previously set track and sub-index markers.

Shortcut: Ctrl + Alt + Shift + I

CD Index Manager

Keyboard shortcut: Ctrl + Shift + Alt + D



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. "**Play marker**" starts playback from the position of the selected marker. "**Play loop**" plays a loop around the marker. "**Stop**" stops playback. "**CD text**" opens a separate dialog field for entering CD text information. Use the **double arrow buttons** to jump to the previous or next marker.

The "**Filter**" buttons at the top of the dialog enable you to display only those markers that are relevant to your needs.

CD Index Manager - Advanced

If this button is activated, additional CD mastering options will be displayed.

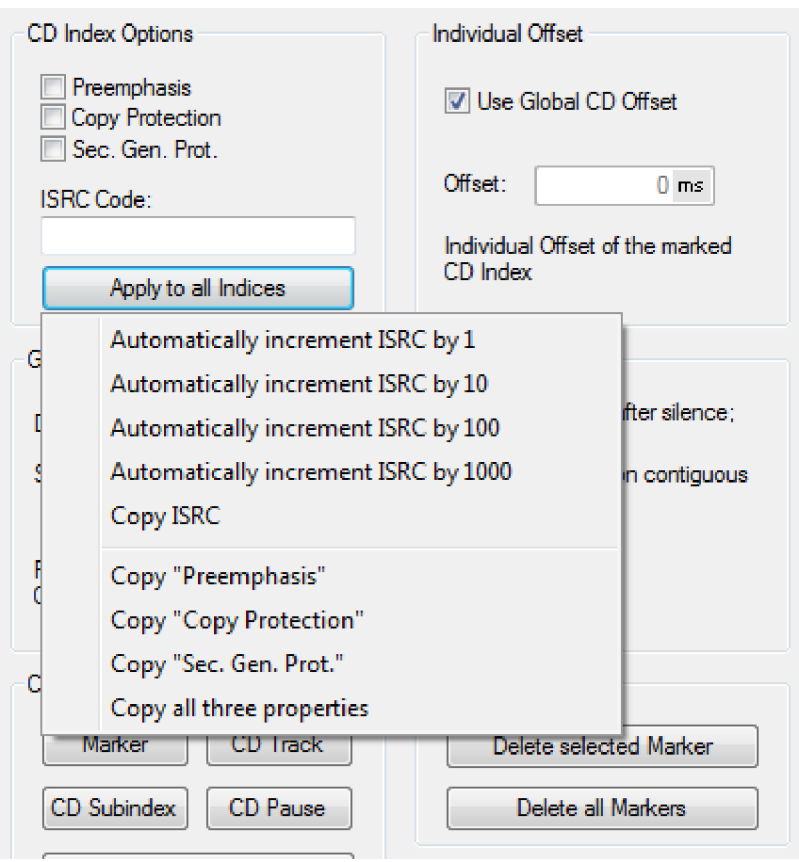
CD index settings

Several options are available for the individual CD tracks, including **copy protection (SCMS)**, **pre-emphasis**, **second generation protection**, etc.

Pre-emphasis means that high frequencies are accentuated before A/D conversion when creating a CD's. 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.

Furthermore you can enter the **ISRC code** (view page 1090) and automatically copy or increase it to 1, 10, 100, or 1000 for the individual pieces of music.



Pre-emphasis, **copy protection (SCMS)**, and **second generation protection** may be transferred and copied together for all indexes.

CD Offsets

Generally speaking, these CD offsets are used to move markers while burning a CD relative to the position, at which they are visible/audible in the project.

As a result of technical limitations, playing a track on the CD player may cause delays in relation to latency-free reaction of a hard disk recording system. A CD player requires a certain amount of time between the initiation of playback and the moment at which it is capable of producing sound. To make sure that a track is correctly reproduced from the start, the Red Book defines some minimum time spans.

For this reason, the track, pause, and end markers of a CD should be set somewhat earlier than in the corresponding virtual project so that the CD player can play the track exactly from the position specified in Sequoia.

Individual Offset

By deselecting "Use global CD offset", you can apply an individual offset value for each selected CD track or pause marker.

Note: Track markers are moved to the left when an offset is applied, and end and pause markers move to the right.

Global CD Offsets

These functions apply the same offset values to all CD tracks. Start markers are written earlier, and pause and CD end markers are set later.

Track offset: Move the track start point forward for quiet track changes. A common value here is about 330 ms.

Splice offset: For CDs with continuous tracks (e.g. typical classical music), use a lower offset value, the so-called splice offset. A common value here is about 180 ms.

Pause/CD end offset: Pauses and the CD end marker may only be moved back.

Note: These offsets don't change the timing of the audio data transferred to the CD burner, they just change the relative position of the track markers relative to their position in the project.

Create CD Marker

Use the respective buttons to create **markers**, **CD tracks**, **CD sub indices**, or **CD pause markers** at certain time positions. Write the time position into the time fields in the marker list input.

Marker list input

Detailed information about marker list input is available in "Play -> Marker list entry (view page 843)".

Delete CD Marker

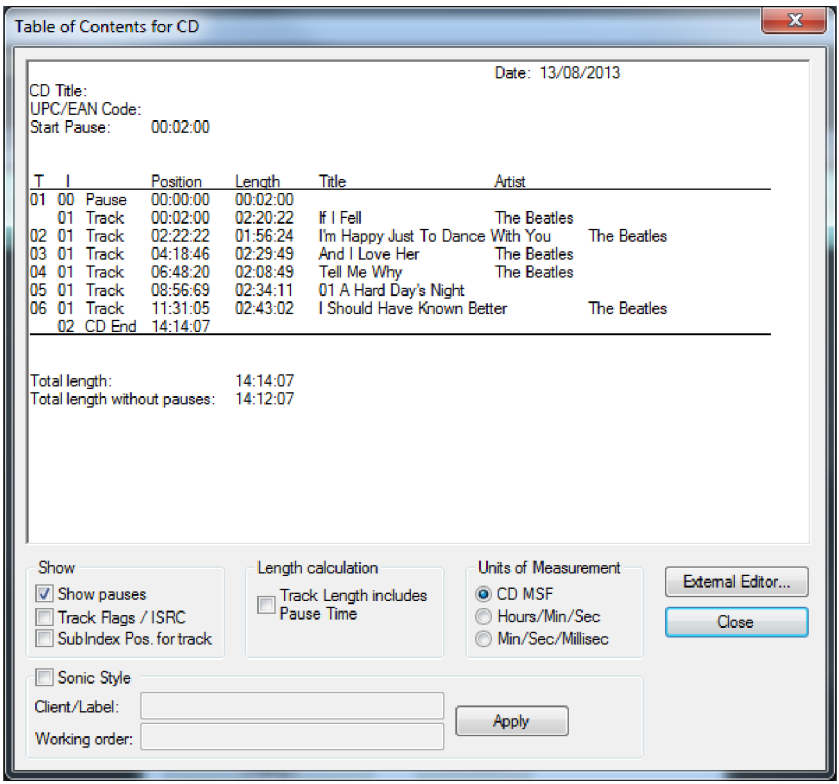
Use the respective buttons to create **markers**, **CD tracks**, **CD sub indexes**, or **CD pause markers** at certain time positions. You can also delet all of the markers at once.

Text file...

This function creates a text file (*.txt) in which all CD markers are listed.

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 (view page 1090)
- 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.

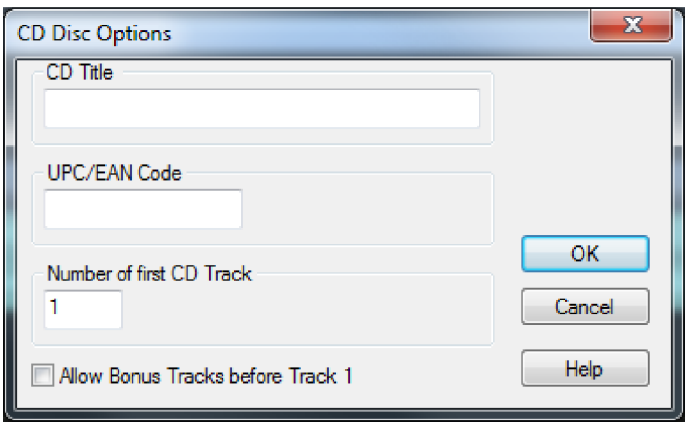
The following measurement units are available: "CD MSF", "Hours/Min/Sec" and "Min/Sec/Millisec".

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.

CD disc options

This dialog opens the settings for the current CD.



CD title: The title which will be written to the CD with the CD text.

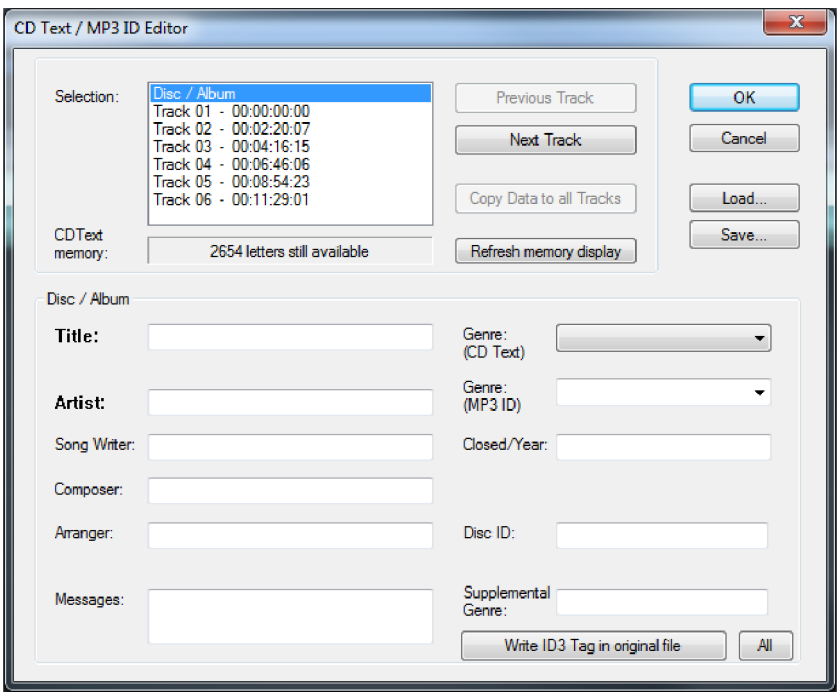
UPC/EAN Code: The EAN code (European Article Code) is a 13 digit number which is used in retail. The 12 digit UPC (Universal Product Code) is exclusively used in Canada and America and can, if necessary, be extended with a final '0'.

Number of first CD track: Under some circumstances (such as "track at once" writing), the title number of the first CD track may be set. This is not important in "disc at once" mode, since the CD will always start with title no. 1.

Allow bonus tracks before track 1: If this option is active, a so-called "hidden track" may be burned on the CD. The first track marker will now be applied to the second audio object. The "hidden track" may only be accessed on standalone CD players via the "STEP BACKWARD" button.

CD-Text/MP3-ID-Editor

Here you can enter text information for the CD you are going to burn. The CD track names are adopted from the CD track marker's labeling in the VIP. All information is also contained in the project, but in this dialog window you also have the possibility to save all entries in a separate file. (*.cdt)



Note: The number of maximum available characters for the entire CD is 2824.

If you can used MP3s in a VIP that have ID3 tags then using of the "Write ID3 tags to original file" button you can ensure that data made in the CD text/MP3 ID editor is written back to the corresponding MP3 file without separate export. This will be applied to all corresponding MP3s by pressing the "All" button.

Set Pause Time

Use this function to set the preset pause duration between objects. Audio files that you load into your virtual project will be ordered sequentially on one track. The value specified here will be set as standard pause.

Set Start Pause Time

Use this function to set the length of the pause in front of the first track. A regular value for the start pause would be about 2 seconds.

CD Arrange Mode

If this is activated, Sequoia arranges recently added objects to insert a Red Book Standard-compatible pause between the objects.

- Open a new project.
- Activate "CD arrange" mode.
- Load wave files, audio tracks, or make a recording using a microphone.

You will see gaps between the individual objects in the project. These are the inserted pauses. The duration of inserted pauses may be specified in the "Set pause duration" dialog.

Show CDR Drive Information...

Choose this option to display information about the burner.

This includes the manufacturer, drive name, hardware revision, cache size, and the features supported by the drive mechanism:

- Supports disc-at-once
- Supports indices
- Supports second generation copies
- Supports catalog numbers
- Supports ISRC
- Supports CD text
- Supports CD RW

The "disc at once" feature is extremely important for Red Book-compatible audio CD production. CDs burned in this way are accepted as masters by CD manufacturers.

Show CDR Disc Information...

This dialog displays information about the blank CD inserted into the drive, including type, track count, session count, total disk space, remaining space, and status.

The most important value is the maximum length under "Total memory" in minutes and audio frames, which cannot be exceeded during production (e.g. 359995 audio frames or 79 minutes and 59 seconds and 70 MSF).

View Menu

This menu contains tools for changing the screen view of Sequoia.

Mixer

Use this menu item to open the mixer.

Detailed information about the mixer is available in the chapter "Mixer (view page 363)".

Shortcut: M

Monitoring

This menu item opens the Monitoring section (view page 395)..

Shortcut: Shift + M

Transport Console



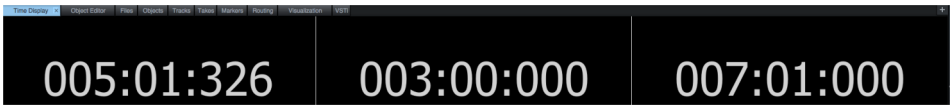
Shortcut: Ctrl + Shift + T

Detailed information about the transport control is provided in "Screen elements -> Program interface -> Overview -> Transport console (view page 65)".

Time display

Shortcut: Ctrl + Shift + Z

This function shows a zoomable time display. This makes it possible to maintain a good overview of the project even at longer distances. The time display is opened in the docker with three display fields, one for position, one for range length and one for the range end.



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..."). In the next section you will find a list of all the possible display fields.

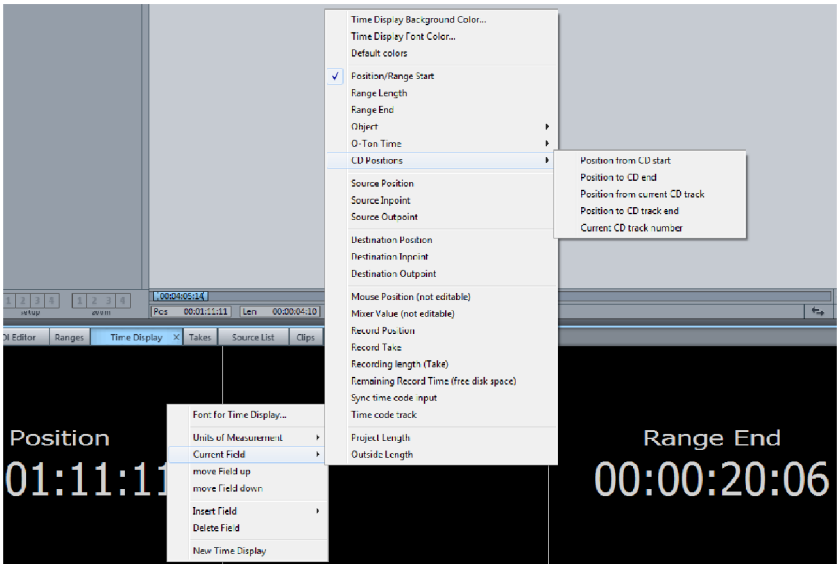
Additional Settings Options in the Context Menu

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 colors 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.



Time display - Current field

Position/Start of range: Displays the current position/current playback marker or the beginning of the range. When moving the objects you can see the starting position of the object here. entereing a + or - before the number, the playback marker or range start is moved relatively back or forward by the entered value.

Range length/object length: Specify the range length/object length here. When entering a new range length, the range start time is kept (the range end changes). If a negative amount is entered, the range end will be kept.

Range end/object end: Set the range end/object end here.

Cue times: Here you can set the current playback time as well as the current remainder playback time of your cue recording (view page 828).

CD position: A display of the following current positions:

- Position starting at the CD start
- Position until CD end
- Position from the current CD track
- Position until CD track end
- Current CD title number

For the **source destination cut** you can display and edit the following positions:

- Source position
- Source in-point
- Source out-point
- Destination position
- Destination in-point
- Destination out-point

Mouse position: This value displays the current mouse position and cannot be edited.

Mixer value: This value displays the mixer fader that was just changed, as well as the volume or pan fader in the track header (cannot be edited).

Recording position: This value displays the current recording position.

Recoding take: this value displays the current recording take.

Take length: This value displays the length of the current take.

Remaining recording time: This value indicates how long recording may continue until the hard disk is full.

Sync timecode input: This value shows the timecode that is applied via synchronization.

Timecode track: This value shows the timecode position of defined timecode track (view page 776).

Total length: This value shoes the total length pf the project.

External length: This value shows the external length, i.e whole legnth of the project - actual range length.

Visualization

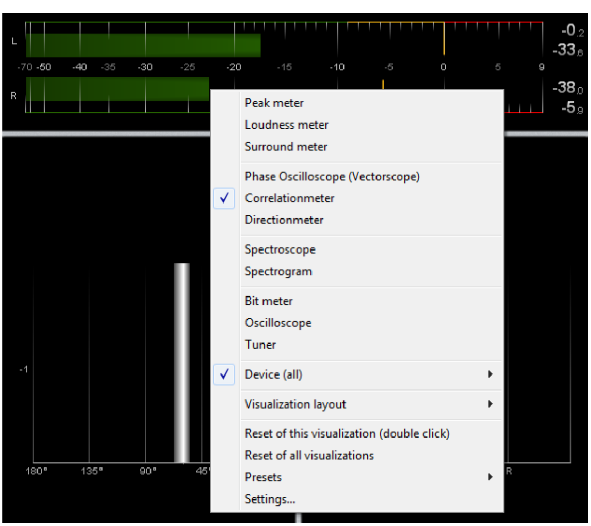
Shortcut: Ctrl + Alt + Shift + V

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 109) 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), **Loudness meter (Sequoia/Samplitude Pro X4 Suite only)**, **Surround meter, Phase oscilloscope (vector scope)**, **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. For example, if you mix more than four submix busses that are each routed to a different device, you can display each bus individually.

Visualization layout: Here you can load the preset **Layouts** and save your customized **Layouts**. To customize a layout, first load a layout in the format of your choice (1x1, 2x2, etc.). Then, right-click the individual sections of the layout to adjust the visualization for each.

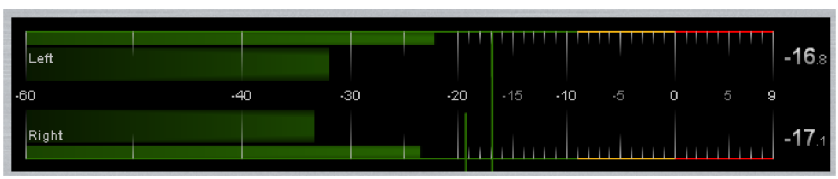
Reset this visualization (double-click): This resets the display of the current visualization.

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. Depending on the display instrument (which can also be changed under "Instrument" in the General tab), this dialog will show different additional tabs with settings options (see below).

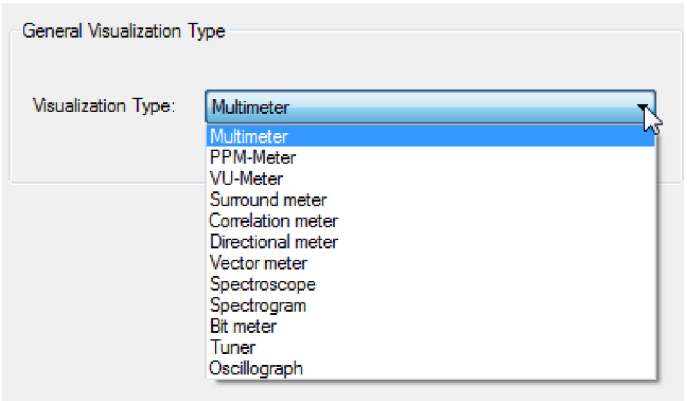
Peak Meter



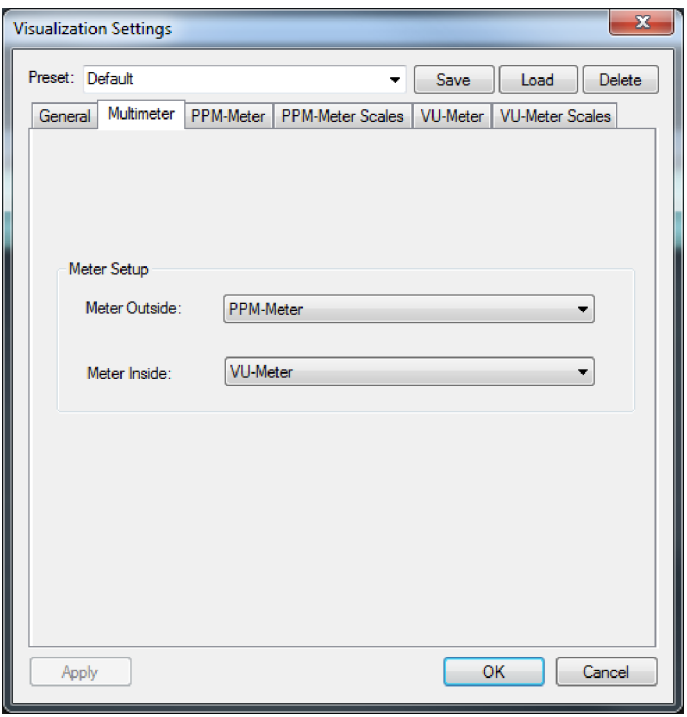
The (multi-) peak meter (Instrument multimeter) displays the volume during playback in dB. The thinner, outer bars show the **Program Peak Meter (PPM Meter)** and the thicker, inner bars display the **VU meter**, which you might be familiar with from analog equipment.

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.

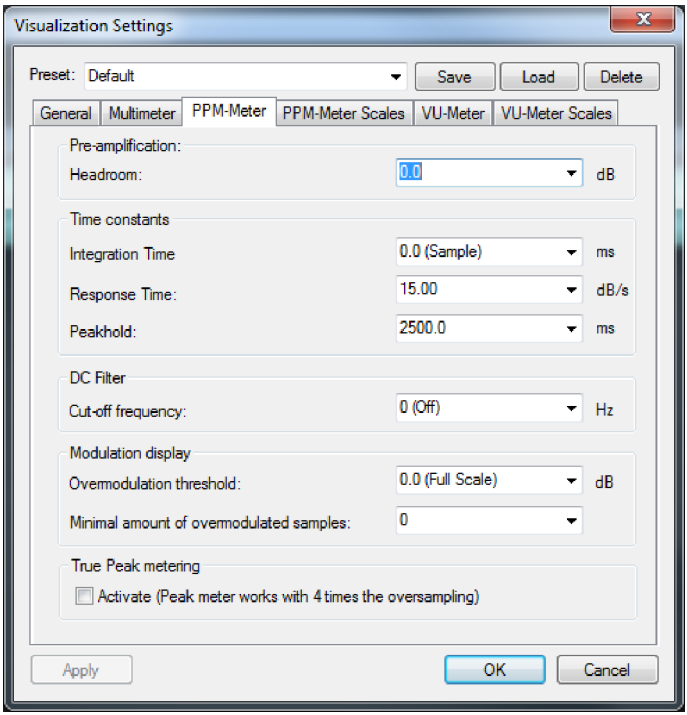
If you only want to see either the PPM display or VU display, select the corresponding display instrument.



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 from a maximum value. 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. With "manual" peakhold, the level peaks remain until you double-click the visualization.

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.

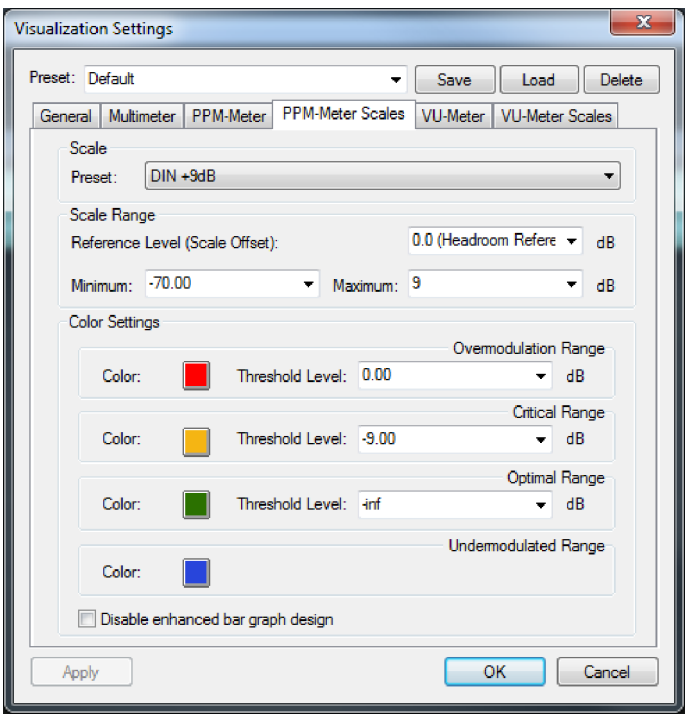
Clipping threshold (dB): This values indicates from which dB value clipping is displayed, i .e. when the display hits the red.

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.

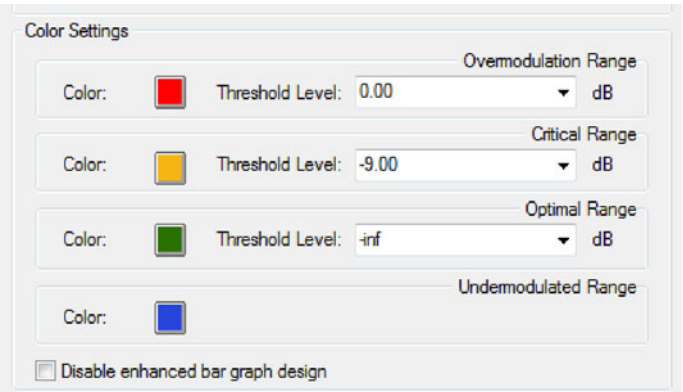


Scale: Select from a range or different scale display settings. In the "Presets" you will find a selection of peak meters with different scales and display characteristics based on the standards in various European countries.

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.

K-Metering

In the visualization area of the peak meter there are a number of presets for the K-metering system. These presets can be opened by right-clicking on the peak meter display. Here you'll find the presets „K-12 Broadcast“, „K-14 Music“ and „K-20 Cinema“. The selected scale also appears in the settings dialog of the context menu under "PPM Meter Scale > Preset".

K-metering enables uniform reference volumes for different media especially when mastering under normalized listening conditions. The peak hold display continues to show signal peaks and can be used to avoid clipping.

The "K-system" refers to the metering system developed by Bob Katz, and it has become the standard for monitoring audio signals during mastering. K-system metering enables uniform calibration and monitoring. You can use it to easily exchange audio material between different studios and have matching monitoring results. The K-metering system focuses more on musical dynamics and less on loudness. Setting the level to 0 dB sets the reference volume which no longer matches the maximum level, as often used to be the case.

Depending on the application and audio material, three different meter scales can be used:

K-20 = 0 dB reference (83 dB SPL) at -20 dB FS

K-20 is recommended when using audio with large dynamics like classical music or film sound.

K-14 = 0 dB reference (83 dB SPL) at -14 dB FS

K-14 is recommended for rock and pop productions or for Surround sound.

K-12 = 0 dB reference (83 dB SPL) at -12 dB FS

K-12 is recommended for radio and television networks.

The corresponding scale for setting up the monitoring volume is calibrated with pink noise. If you set pink noise to 0dB, you get a level of 83dB SPL, a volume reference that originated in the film world.

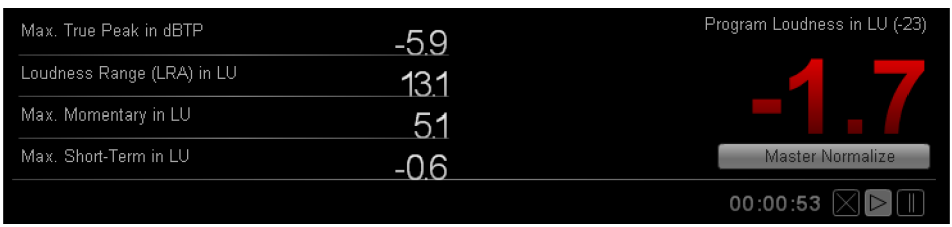
Loudness Meter

Loudness refers to the subjective volume sensitivity when listening to audio. This sense varies among listeners, depending on several listening conditions such as level, music style, age and even psychological state.

EBU Recommendation R128

The "EBU Recommendation R 128" deals with a method for measuring loudness which is being used more and more widely. The goal of this recommendation is to standardize all audio productions and level metering. This offers a replicable method for loudness metering. R 128 is based on the ITU-R BS.1770 standard which was established by the International Telecommunications Union.

The measurement is based predominantly on a consistent, average loudness impression. This impression is created by the integrated loudness, averaged over the entire duration of the piece. This value is then used for normalization. According to EBU R 128, the target value for normalization is "-23 LUFS".



By changing to loudness modulation and normalization, a uniform loudness level can be achieved, which can be retained for the whole production process. The end result is an even loudness impression for the listener and audio material that contains fewer artifacts caused by excessive dynamic compression.

The hope is that loudness normalization can stop the "loudness wars" and at the same time improve the audio material quality.

The EBU Recommendation R 128 focuses on three parameters that are essential for the properties of the audio signal:

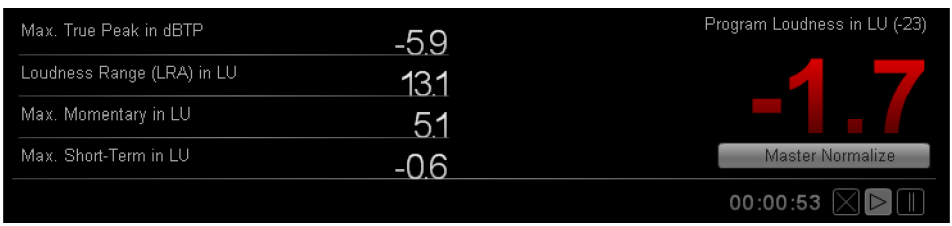
- **Program Loudness:** Integrated long-term loudness for the duration of the audio material - measured in LUFS (Loudness Units, referenced on Full Scale) or in LU (referenced to a target value, e. g. 23 LUFS according to EBU 128). This parameter indicates the average program loudness. It calculates the mean value over the length of the piece. The term LU is the level difference to the target value. A measured calue of -20LUFS for example corresponds to a difference of 3 LU from a target value of -23LUFS.
- **Loudness Range (LRA):** Variation in measuring the loudness of the audio material - measured in LU (Loudness Units). The value indicates the difference between the loudest and softest measured value (except silence)

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- **Loudness Range (LRA):** Variation in measuring the loudness of the audio material - measured in LU (Loudness Units). The value indicates the difference between the loudest and softest measured value (except silence)
- **Maximum true peak:** The maximum value of the audio signal during continual measurement on the timeline. When converting digital to analog signals level peaks could happen that were not displayed before. Thats why the displayed peak level of a True Peak Meter is typically higher than that of a usual digital

(QPPM - Quasi Peak Program Meter). The R128 standard alows a maximum True Peak value of -1dBTP.

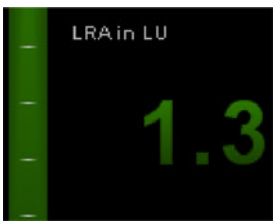
The combination of these parameters allow more dynamic mixes without affecting the loudness.

Loudness meter - Parameters

INT in LUFS/LU: The program's loudness value for "Integrated loudness" in measured loudness units (LU).



LRA in LU: Loudness range, measured in LU as a relative value.

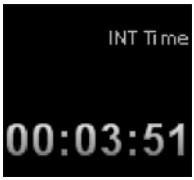


True Peak in dBTP: Peak level measured in Oversampling. When converting digital signals into analog signals, level peaks may occur which were previously not displayed. Therefore the peak level of a true peak meter is typically shown above that of conventional digital meter (Quasi Peak Program Meter - QPPM).



Using Oversampling-True Peak Metering according to ITU-R BS.1770 true peaks can be ascertained (dBTP - referenced on Digital Full Scale). Here the accuracy of the measurement depends on the Oversampling frequency. Oversampling Peak Meters create a good estimate for an audio signal's True Peak. The R 128 standard permits a maximum True Peal Value of -1dBTP.

INT Time: Time value for the entire duration of the current loudness measurement up to now - i.e of the program's and loudness range.



With the three buttons below you can interrupt measurement (Pause button), continue (Play button) or reset (Reset button). By clicking the "Reset" button all display values are reset.



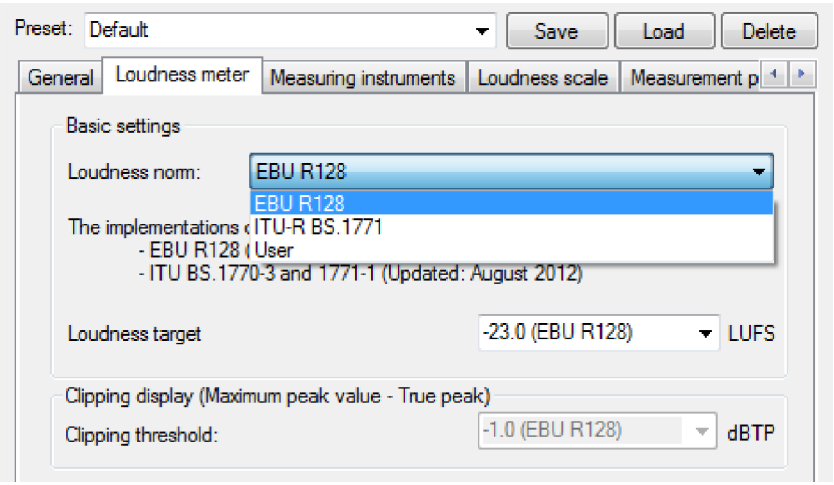
Gate: This display lights up as soon as certain threshold values are exceeded during loudness measuring. While the Gate is lit up loudness measuring is interrupted so that recordings with long silent passages or quiet background noises won't be estimated too low in long term integrated loudness.



Loudness Meter - Settings

You can access the loudness meter settings dialog by right-clicking on the graphic display ("Settings..."). In the **„General"** tab go to "Instrument" > "Loudness Meter" to work with the entire instrument section. The "Multimeter", "VU Meter" and "PPM Meter", in contrast, produce only a simple bar display similar to that of a peakmeter. A range of presets is available in the preset list. Confirm your selection with "Apply".

In the „**Loudness Meter**" tab you can set the standard that should be used under "Loudness Norm". If you select "User" as loudness norm you can set the parameters yourself. If you select "EBU 128" or "ITU-R BS. 1771" the parameters will be set according to the corresponding standard. Windows which are no longer available for editing will be displayed grayed out.

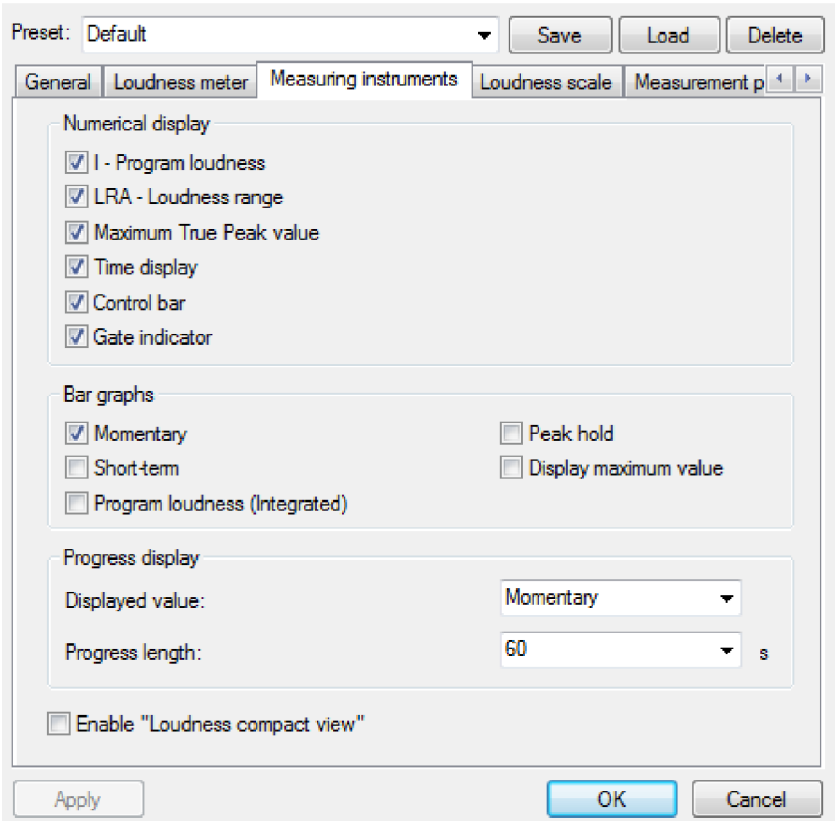


Here you'll also find the important parameter "Loudness Target Value". The target value "-23 LUFS" is specified for EBU R128. This level represents the default value for content production in broadcasting.

Clipping Threshold: Set the clipping level, i.e. from what point clipping should be indicated. This value is standardized for EBU R128 at -1 dB. This results in a measured True Peak-peak level of -0.8 dBTP being displayed as clipping for example.

Automatic reset at playback start: With this option you can reset the measurement each time you begin playback. This option represents the "Reset" button in the graphic display.

In the „**Measurement Instruments**“ tab you can activate and deactivate various display instruments such as the "Numeric Display", "Bar Graphs" and "Progress Display".



For „Numeric Display“ select between „I - Program Loudness“, „LRA - Loudness Range“, „True Peak Level“, „Time Display“, „Control Bar“ and „Gate Indicator“.

Under "Bar Graphs" you can select which time constants in the bar graph display should be displayed for the measure mode: "Momentary", "Short term", "Program Loudness (Integrated)", "Peak hold" or "Display Maximum Values".

Under "Displayed Value" you can select which time constants in the program diagram should be displayed for the measure mode: "Off", "Momentary" "Short term", "Integrated" or "LRA" or "M + S".

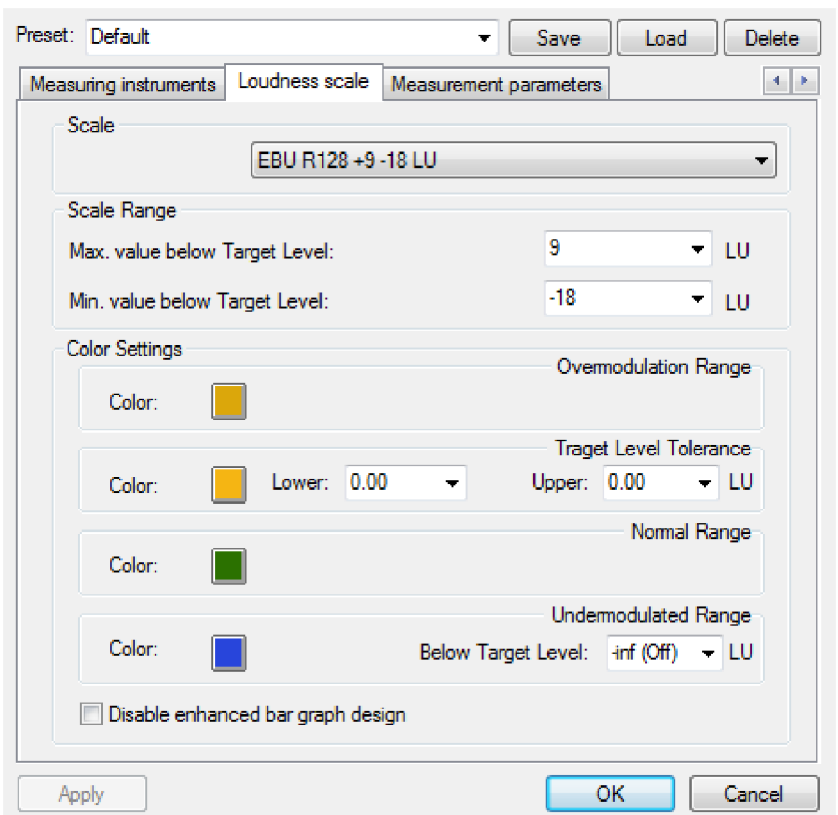
Here you can also set various event lengths between 10 seconds and an hour.

If you prefer a compact display of loudness, activate the corresponding option.



Now you have a better overview of the values for "Max. True Peak in dBTP", "Loudness Range in LU", "Max. Momentary in LU" and "Max. Short-Term in LU".

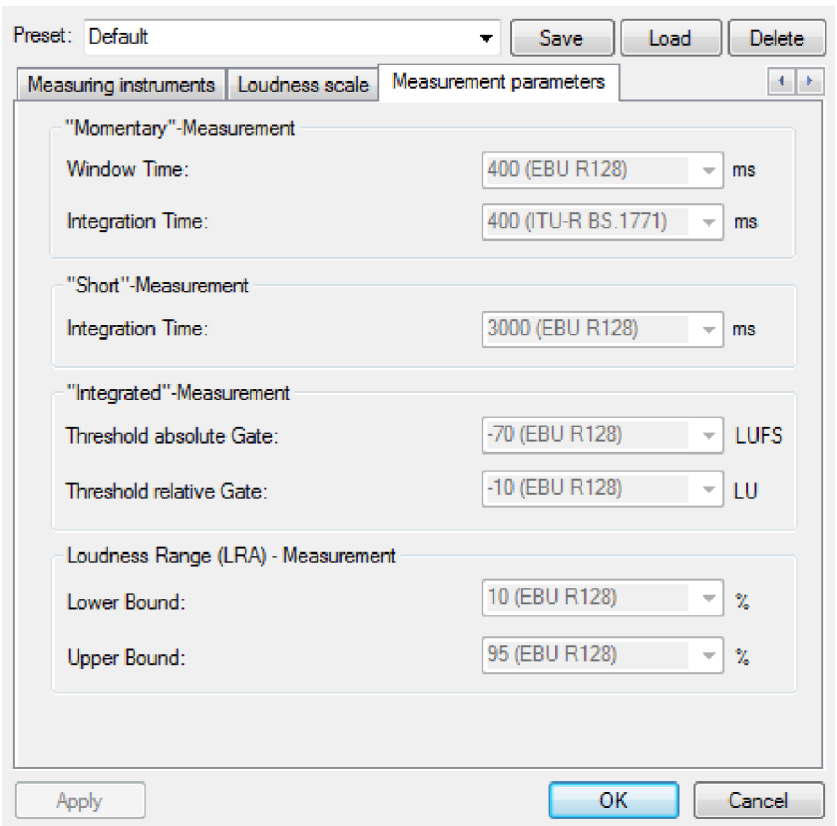
In the "**Loudness Scale**" tab you can switch between different standardized scale norms. The range of values can be defined freely. The entry "EBU R128 +9" means that the scaling display is sufficient until +9 LU is reached.



In the scale area you can determine by how many LUs the maximum or minimal value is allowed to exceed or fall short of the target value.

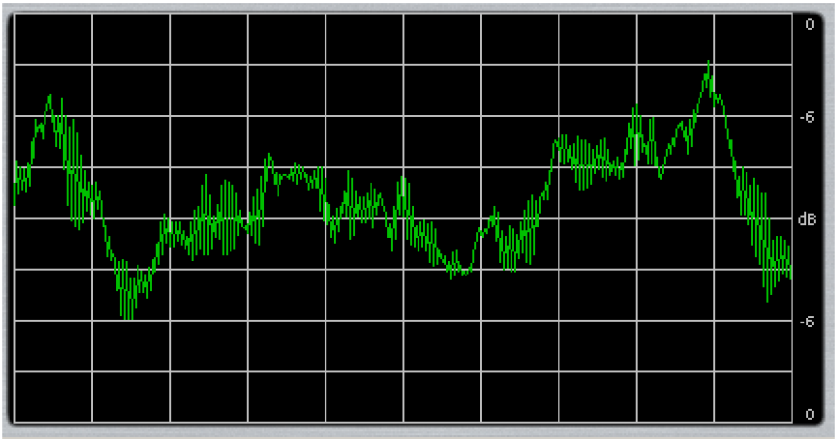
Under "Color Settings", you can define color coding as well as thresholds for the clipping section, tolerance section around the target value, the normal value and the weak signal section.

In the "**Measurement Parameters**" tab you can set the time parameter for "Momentary" and "Short" measuring, the gate threshold for "integrated" measuring as well as the upper and lower limits for the "Loudness Range (LRA)" measuring. The specified default values for EBU R128 are marked in a noticeable way.



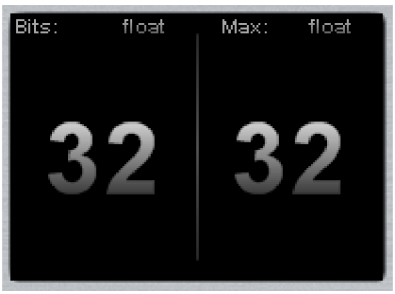
In this section when changing the norms you can adjust the affected values and save them as new presets. Absolute and relative values are displayed in relation to the selected scale. L or LK denote relative scales, LUFS and LKFS denote absolute scales.

Oscilloscope:



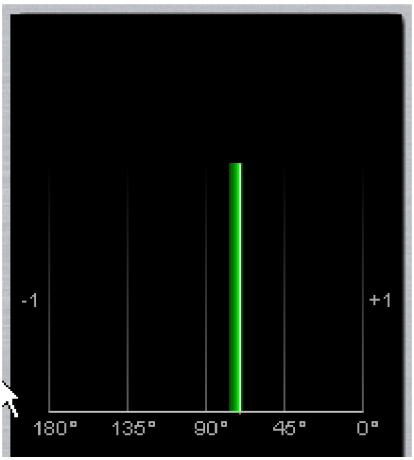
The oscilloscope displays the signal amplitude over the time curve.

Bit Meter



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

Correlation meter

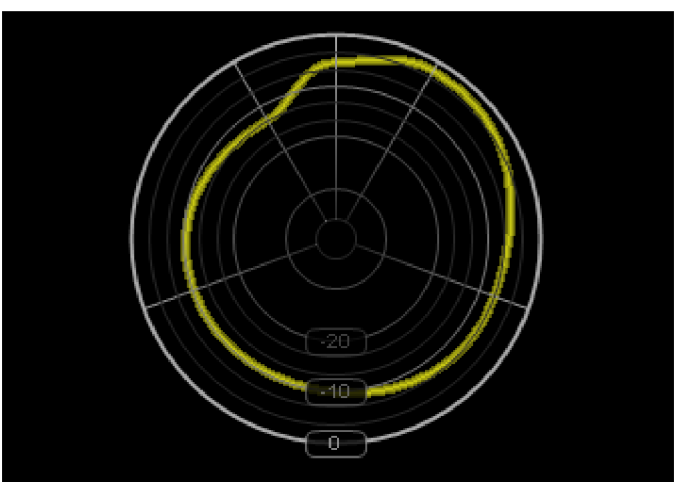


With the correlation meter you can read phase offsets between the two stereo channels, and examine how much has been deleted. If the signal display is in the left, red area between 90° and 180° the signal will no longer be reproduced properly over a mono receiver.

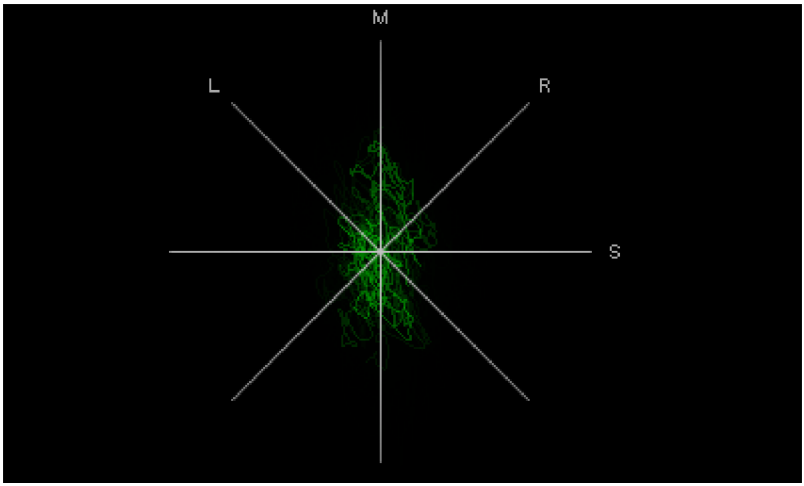
In the settings dialog you can select the colors for Mono and Stereo and mono compatible ranges, threshold values for the ranges can be set too.

Surround Meter

The surround meter displays the signal using the surround settings that are set as global in the „Surround Setup“ (view page 477) dialog. It shows each speaker's level in a polygon on a sound representative area.



L/R Oscilloscope (phase correlation):

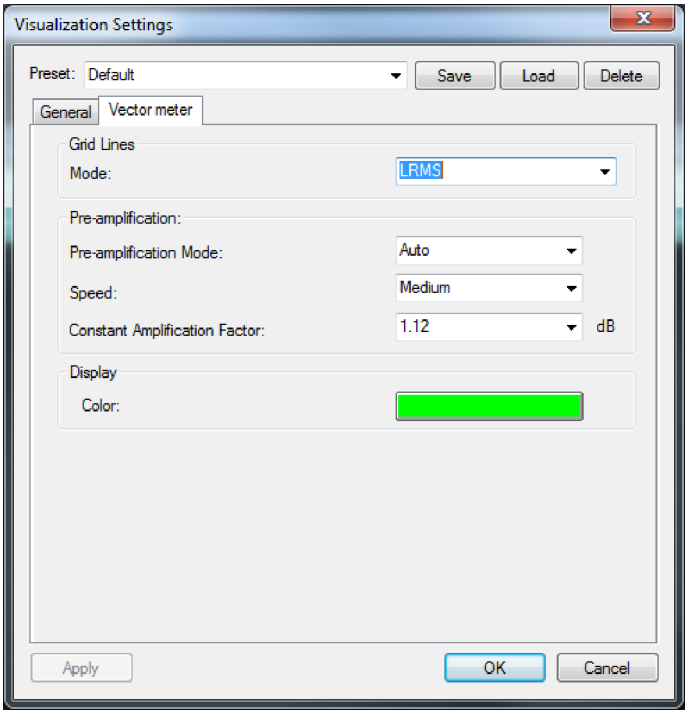


Phase correlation provides information on the distribution of the stereo image in your recording. A mono recording would be displayed as a vertical bar in this view. A song produced in stereo is by contrast shown as "diffuse ball," because multiple instruments were assigned to the mix in different panoramic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

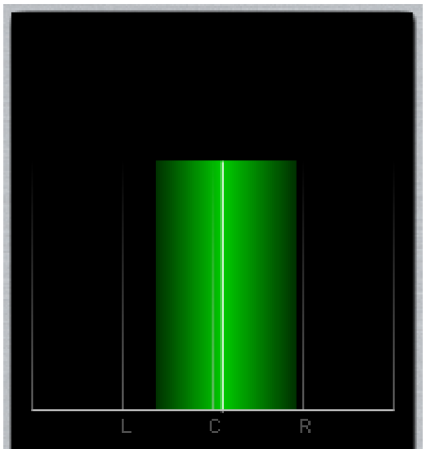


The image is correspondingly increased in pre-amplification mode, to ensure readability. You can also switch off this mode and set a constant amplification factor instead.

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

Finally you can set the display color.

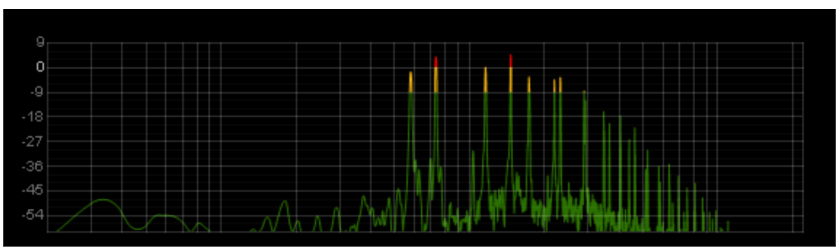
Direction meter



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.

Spectroscope settings

Line/bar representation: Select the representation type you want to use. **Line (smoothed)** is set by default. Other options include **Line** (connects the points with straight lines instead of interpolated curves) and Bar, where you select the number of bars.

Stereo Options: Left+Right is set as the default (i.e. mono sum). It is also possible to set up two spectroscopes, one each representing the left and right signals. The side signal can also be represented by selecting Left-Right.

FFT parameters

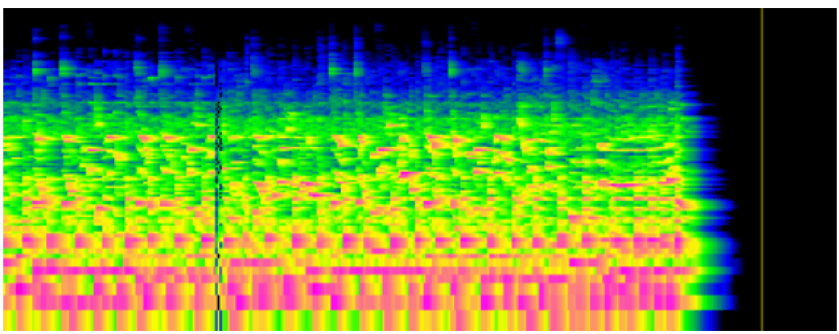
Here you can adjust the parameters **FFT Size**, **Overlap** and **Window** for determining the spectral values.

Time constants

The time constants correspond to those of the peak meter (PPM). (However, the peakhold value has no meaning for the spectrogram).

The **Spectroscope Scales** settings correspond to those of the peak meter (view page 1123).

Spectrogram

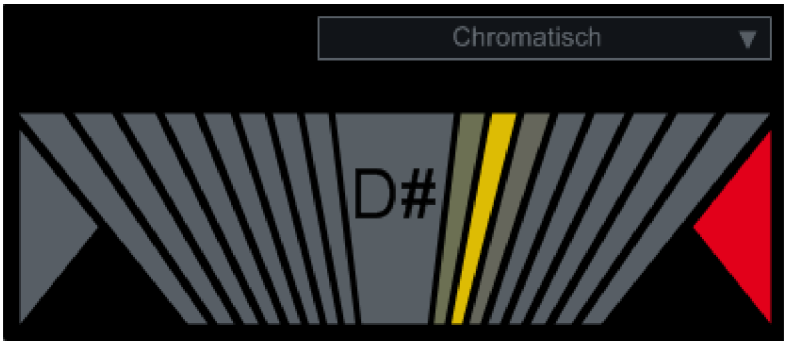


The spectrogram displays the signal as frequency levels along the timeline. The volume of frequencies is visualized via a color code or via its brightness.

The spectrogram is useful for detecting sudden disturbances in a recording. Audible distortions louder than the music are usually limited to a certain frequency spectrum. These are highlighted with colors in the spectrogram.

Sequoia enables audio disturbances to be removed with the help of Spectral Cleaning (view page 948).

Tuner



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

This is how you can tune your guitar:

1. Select your guitar in

Step 2 Open recording options by right-clicking on the record button in the transport console.

3. Activate the "Monitor" button and use the monitoring mode on "Track FX Monitoring" or "Mixer FX Monitoring" to be able to control tuning acoustically.

4. Activate the "Visualizer" button and select the "Tuner" visualization in the graphic display.

In the middle you will see the note name of the string being played. The calibration marks show deviation of played to 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.

Video Window

This function opens the video window, e.g. to apply a media link.

Track Editor

This menu item opens the track editor on the left edge of the arrangement window. This enables access to all important track parameters for the selected track.

Detailed information about the track editor is provided in "Screen elements -> Track editor (view page 67)"

Shortcut: Ctrl + Alt + Shift + E

Keyboard

Use the keyboard (view page 213) to access an on-screen keyboard to play MAGIX synth sounds and record them directly.

Manager/Docker

You can open the Manager/Docker here. In addition to the Manager windows (view page 178) now Visualization, Time display, Transport console, Sound pool, Object Editor and MIDI Editor are grouped as a tab window by default in a "Docker".

Manager

The manager features the following sub-windows:

- ClipStore (Ctrl + Shift + Alt + C)
- File browser (Ctrl + Shift + B)
- Object manager (Ctrl + Shift + O)
- Track manager (Ctrl + Shift + S)
- Marker manager (Ctrl + Shift + Alt + M)
- Range manager (Ctrl + Shift + Alt + B)
- Take manager (Ctrl + Shift + Alt + T)
- Source list manager (Ctrl + Shift + Alt + S)
- VST instruments manager (Ctrl + Shift + I)
- Routing Manager (Ctrl + Shift + R)
- Soundpool manager

Detailed information about the individual manager sub-windows is provided in the "Manager (view page 178)" chapter.

Control bars

Main Toolbar

Use this command to hide or show the toolbar. A check behind the menu item indicates that the toolbar is visible.

Main Toolbar 2

Use this command to hide or show the second toolbar. A check behind the menu item indicates that the second toolbar is visible.

Statusbar

Use this command to hide or show the status bar. A check behind the menu item indicates that the status bar is visible.

Autoscroll

Switches Autoscroll on/off

See "Playback -> Playback Options (view page 830)".

Soft Autoscroll

Toggles between page and soft autoscrolling.

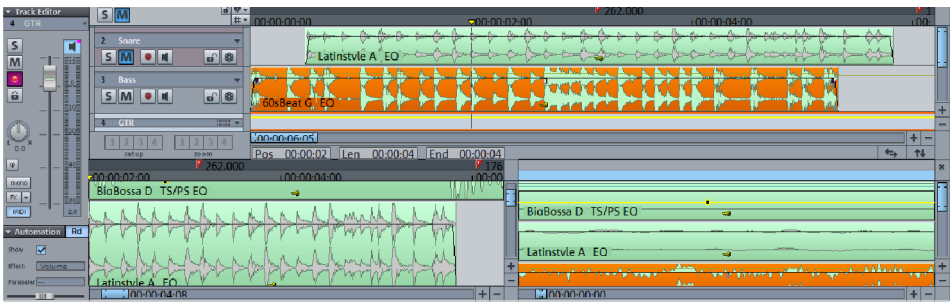
See "Playback -> Playback Options (view page 830)".

Rebuild Graphic Data

The graphical display of a project is recalculated in this case. This is usually unnecessary, but does help with errors or inaccuracies in the display after offline editing.

Sections

These Menu items allow you to take advantage of Sequoia's ability to simultaneously display **one two or three** different "views" of the same project.



If you select the option "**2 (sections)**", two view windows, each of which can be worked in independently, open **one under the other**. This way, for example, you can view the complete sample in one window, while magnifying a specific area in the other.

The "**3 (sections)**" mode can be especially helpful when searching for loop points in the wave window. In the **top large window the entire sample** can be viewed, in the **bottom left window area the beginning** can be viewed and **in the bottom right window area the end of the loop** is displayed. This view can also bee seen by using the shortcut "B". The corner values of the three windows can be independently adjusted.

You can determine the areas through window borders once and for all by selecting a start of a segment by clicking the clip bar at the corresponding position, and then selecting the end of the segment in a different clip with the Shift key held down.

When using the zoom function on a specific clip, it is necessary to first select it by using scroll bars to the right or to the bottom.

Shortcuts:

Show range:	Shift + B
Split range (shows 3 sections)	B

Activate destination

This command activates the previous/upper section.

Shortcut: Page up

Activate source

This command activates the next/lower section.

Shortcut: Page down

Hide Submix/AUX Busses

Use this command to remove the buses from view in the arranger window. This may improve the display in case you are working with numerous tricks.

Attention: This function keeps bus tracks hidden, even if a check is set in the track manager next to the "Arrangement" column.

Grid view

Grid

This command activates the coordinate grid for the project window.

Grid Lines

This provides a choice between various line layouts for the grid display. The display of the grid helps makes the positions in the arranger window more legible.

Units of Measurement

Specify the units of measurement for the grid view here.

The following units are available: samples, milliseconds, hours/mins/secs, SMPTE, SMPTE/milliseconds, CD-MSF, feet and frames 16 mm (40 fpf) and feet and frames 35 mm (16 fpf), noise reduction (meter).

The units of measurement selection will affect the start and length of the current range in the project window, the grid width and the details of the position of the playback marker (position line).

Snap active

This functions turns the grid on or off.

Snap/Grid

Here you can set the grid type. You can choose from:

- Objects
- Range
- Beats/Bars
- Bars (relative)
- Grid/Frames
- Frames (relative)

Detailed information on snap and grid settings can be found under "File > Project Properties > Snap and Grid Setup (view page 656)".

Snap and Grid Setup

Detailed information on snap and grid settings can be found under "File -> Project Properties -> Snap and Grid Setup (view page 656)".

Show 2nd Grid Line

This command displays a second grid.

Exchange Grids

This command exchanges the upper and lower grids.

VIP Display Mode

- Definition...
- Mode 1
- Mode 2
- Toggle between mode

Detailed information on display options can be found in "File > Program Properties > Display Options (view page 694)".

- Spectral Display
- Waveform display

Representation of the waveform as Spectrum or WaveColor Detailed information on display options can be found in "File > Program Properties > Display Options > Waveform Color".

Overview Mode

Detailed information about Overview Mode can be found in the chapter „Working in the Project Window“ > „Zoom“ > „Zooming with Overview Mode“ (view page 114).

Zoom to selected object

When this command is called, the project section jumps to the selected object. If you execute the function again, the original section is displayed again.

Horizontal

This menu contain horizontal functions for clip control, which are also partially represented in the bottom position bar as buttons.

Shortcuts:

Half section left	Ctrl + Alt + Left arrow
Half section right	Ctrl + Alt + Right arrow
Cut to playback marker	Ctrl + Alt + ,
Clip to the start of field	Ctrl + Alt + B
Clip to end of field	Ctrl + Alt + N
Zoom in	Ctrl + Left arrow, Up arrow
Zoom out	Ctrl + Right arrow, Down arrow
Show all	Ctrl + Alt + Up arrow
Zoom to range	Ctrl + Alt + Down arrow

If you can't see the playback marker in the current VIP's image section, then the command "Section to play cursorplayback markerposition" moves the visible section to the current playback marker position. In a stopped state the picture section jumps to the previous playback position.

Vertical

This menu contains vertical functions for clip control, which are also partially represented in the bottom position bar as buttons.

Shortcuts:

Half section up	Shift + Page down
Half section down	Shift + Page down
Zoom into waveform	Ctrl + Page up
Zoom out of waveform	Ctrl + Page down
Zoom Out Volume Automation	Ctrl + Shift + Page up
Zoom Out Volume Automation	Ctrl + Shift + Page down

Tile

This function arranges all open windows one one top of the other.

Shortcut: Enter

Untile

This function arranges widows in the same way as they were prior to "tile vertically".

Shortcut: Shift + Enter

Window Menu

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

Close all windows

This closes the currently opened project. A confirmation will ask if you want to save the project.

Shortcut: Ctrl + H

Show/Hide Effect Window

With this function you can minimize an open effect dialog window or maximize a hidden effect dialog window.

Set focus on project window

This commands activates the project window.

Shortcut: Ctrl + P

Activate docking window

This command activates the docking window.

Shortcut: Ctrl + B

Close current docker tab

This command closes the currently opened docker tab.

Keyboard shortcut: Ctrl + F4

Next docker tab

This command moves you from one docker tab to the next and activates it.

Minimize docker

This command minimizes the docker.

More...

Select Workspace/Mixer snapshot/Mixer View snapshot

With these commands it is possible to load Workspaces, mixer snapshots or mixer view snapshots. Under these menu items you will find submenus with the available workspaces and snapshots, from which you can select the desired entry with the arrow keys and confirm with the Enter key. If you also assign a keyboard shortcut to the menu items, you can switch workspaces and snapshots completely by keyboard control.

Store Position and Zoom Level

Save the current position and zoom level here. These may be retrieved via the menu item "Get position and zoom level" in the "View" menu and on the number pad of your keyboard.

The first three numbers on the keypad may be used to save presets. These configure the setup buttons 1, 2, and 3 in the arranger window.

Shortcut: Ctrl + Num 1

Ctrl + Num 2

Ctrl + Num 3

Get Position And Zoom Level

Load the saved position and zoom level here. This may be retrieved via the number pad on the keyboard.

The first three numbers on the number pad can be used to save presets.

Shortcut: Num 1

Num 2

Num 3

Store Zoom Level

Save the current position and zoom level here. These may be accessed via the menu item "Get position and zoom level" in the "View" menu and via the number pad.

The numbers 4, 5, 6 on the keypad may be used to save presets. These configure the zoom buttons 1, 2, and 3 in the arranger window.

Shortcut: Ctrl + Num 4

Ctrl + Num 5

Ctrl + Num 6

Get Zoom Level

Load the zoom level here. This may be retrieved via the the number pad on the keyboard.

The numbers 4, 5, and 6 on the keypad may be used to save presets.

Shortcut: Num 4

Num 5

Num 6

System Information

Use this function to open a new window with the system information.

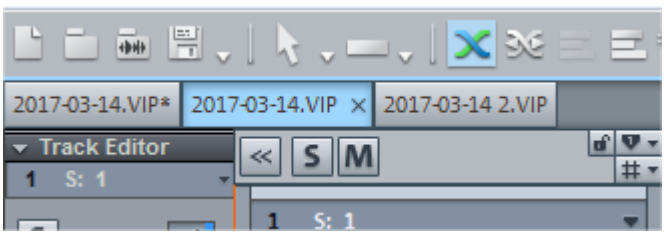
Display the date/time, number of open files, the free memory on all connected hard drives, the configuration of Windows resources, and the amount of memory used by Sequoia.

FTP Download...

This menu item opens an FTP download window. Enter your FTP access data and click on "Connect..." to establish an FTP connection immediately. Files may be downloaded, but not uploaded.

Activating a Project

Sequoia shows a list of the currently opened projects in the last section of the "Window" menu. The active project is indicated by a check mark. Select a project from this list to activate it in the project window. All of the open projects appear as project tabs in the arranger.



When you click on one of these project tabs, Sequoia will open the desired project.

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

Help Index...

Choose this option to display a list of available help contents.

Context Help...

This command turns the mouse cursor into an arrow with a question mark. Click any menu item or button in one of the toolbars to receive more information.

Shortcut: Shift + F1

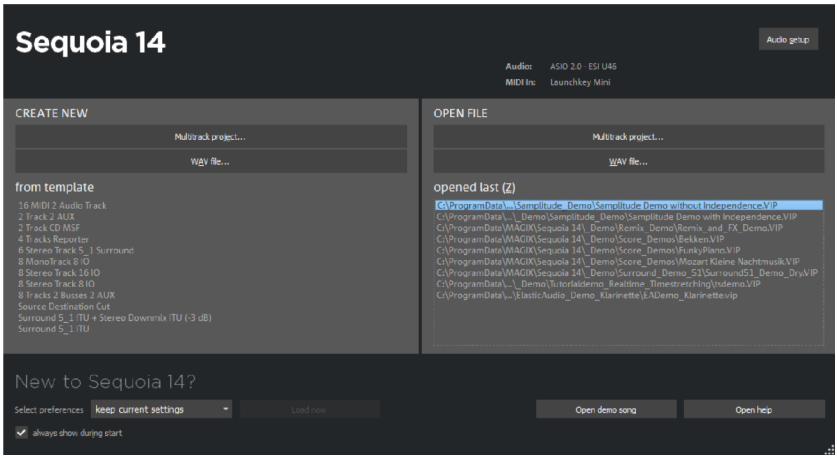
Manual

Use this command to open the Sequoia manual (PDF document).

About Sequoia

This displays copyright information and the version number of Sequoia. At the very bottom, you can see the serial number required for support purposes.

Start selection



The Start Selection opens every time you start Sequoia. You can also open it via the "Help" menu at any time. It offers several options to start working with Sequoia.

At the top the chosen **Audio and MIDI driver** is displayed, with the "**Audio Setup**" you get into the according settings dialog to change it if necessary.

On the left side are options for creating new wave projects or virtual projects. For virtual projects you have direct access to the list of project templates from the New Virtual Project (view page 610) dialog.

At the right side you can load saved projects from a file explorer window or chose from the list recently edited projects.

At the bottom of the dialog there are useful options for getting started with Sequoia with quick access to the demo project and the help file.

Load a complete set of program settings with the "**Select Preferences**" list box. More information about program settings management can be found in Options management (view page 683).

Attention: To help Sequoia beginners switching to Sequoia we changed the program behavior at some places, for instance the mouse wheel scrolls in VIPs vertically (tracks) by default instead of horizontally (time position). To make Sequoia behave exactly like in the versions before, chose "Sequoia 13 Shortcuts/Mousewheel behavior" from the list box. For more information, read the section Mouse Wheel at the end of Editing Keyboard Shortcuts and Menus (view page 688)

If you don't like the Start Wizard opening at each program start, just uncheck the checkbox "Always show during start".

Product Registration

Detailed information about activation is provided in the chapter "Installation > Activating Sequoia".

Download more instruments

Selecting this menu item will take you to the MAGIX website where you can browse through the latest selection of Vita Instruments and download the ones you want.

Update online

This command lets you update Sequoia 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. Sequoia restarts in the updated language.

Glossary

A

Audio markers

Markers in wave projects are saved in the audio file (*.wav) as audio markers and are available in this form in other applications as well. Audio markers are coupled directly with the audio material and visible along the upper edge of an audio object. The purpose of audio markers is to mark positions within the audio material so that the selections remain independent of their placement within the virtual project.

Audio markers may also be made visible in the display options ("Shift + Tab") in the "Objects" area by marking "Audio markers" with a check.

The markers displayed in a virtual project's object are identical to the markers in the associated wave project. If you set new project markers in a wave project (this happens automatically while recording a take), then all audio markers in all associated objects of the virtual project will be visible at the same position in the audio material.

Note: All time information for the audio marker relates to time positions in the audio material, and not to positions in the virtual project.

Automation

Automation can be used to change the envelope of certain parameters in Sequoia such as panorama, volume, surround and AUX parameters, MIDI controllers, and VST parameters. Changes can be displayed with the help of an automation curve.

There is a separate mouse mode available for editing automation curves.

AUX bus

An AUX bus is a track featuring all the abilities of a normal track, i.e. it can include track effects, and panorama and level can be automated. It has one input and one output. The most important difference is that the audio signal is diverted to the bus of any track with a lower track number than the AUX bus. This is done via the AUX send faders in the mixer.

AUX buses are typically used for effects like reverb or echo, which may be equally required for various tracks, but in different amounts. Objects may also be routed to AUX buses.

C

Clip for audio files

The clip is a buffer or a clipboard where samples from wave projects can be copied into or from which you can paste samples into wave projects. The content of the clip can also be mixed with the data of an audio file. The clip always assumes the properties of the project where the data originated (bitmap resolution, sample rate, and mono, LR or stereo mode). The clip is presented on the screen by a window with Clip in the title.

The clip window is normally hidden; however, you can make it visible with the command "View > Window> Audio file window > Iconize all Wave projects". Otherwise, the clip is a project like any other, i. e. you can play it, edit it, and save it with another name.

Clips in ClipStore

In the Sequoia ClipStore, a clip is a reference to one or more objects. This reference contains the exact path name as well as the start and end point. It also includes all object-related settings.

Context menu

Nearly all parts of a VIP and the mixer include a context menu that is opened via the right mouse button. It is called a context menu because different menus appear that are always adapted to the current context, i.e. depending on the VIP or mixer area that is clicked.

Crossfade

Crossfading two objects on one track within the Virtual Project (VIP). A standard crossfade can be created automatically when cutting to avoid clicking ("Auto crossfade" mode). Crossfades can also be edited in Sequoia in detail with the crossfade editor.

D

Destructive editing

In this mode, wave projects are edited directly on the hard drive. This means that changes to the audio material which occur acoustically and visually in the project window are also included in the audio file. The file does not have to be saved after editing, since editing of the audio data can be performed directly on the hard drive. Of course, undo steps are also available, but these are only available for as long as the file is open. After that the changes are permanent!

Note: Destructive editing can be used in virtual projects if offline effects are applied to selected objects or when working with wave editing in "Destructive" mode.

F

Fade

An audio transition. These may be fade-ins (volume increase) or fade-outs (volume decrease). You can also fade between objects in the VIP (crossfade).

H

Handles

Handles are the five little rectangles that appear on the outline of a selected object.

The **length handles** are on the bottom left and right corners of the object. These may be used to adjust the length of the object.

Fade handles: Located on the top right and top left. These can be used to softly fade an object in or out.

Volume handles: These are located in the center. Use these handles to adjust the track volume.

Hybrid Engine and economy tracks

Very small audio buffers ("Low latency" mode) increase the CPU strain of the playback engine. For optimal use of the PC resources, we recommend using the mixer tracks in "Low latency" mode only. This includes:

- Monitor tracks
- Tracks with VSTis

Tracks that play hard disk content (MIDI or audio files) may be taken out of "Low latency" mode in the Hybrid Engine by assigning the "Economy" property to them. These tracks will then use the play engine's buffer (VIP buffer size).

L

Latency

Audio latency and response latency are important aspects of working with audio applications.

Audio latency is the delay of audio data when being processed through an audio function unit (effect plug-ins, DAW audio engines, digital mixers, hardware effect

devices, DA/DA converters, etc.). The entire audio latency is made up of the sum of the latency of the sound card (ASIO latency) and the latency of the effects used in monitoring tracks and the master. When monitoring, these latencies are disruptive even at 5 ms.

The response latency is the delay between the use of the sound-creating or sound-changing control element and the audible change or generation of the sound. The response latency is the sum of the output latency from the sound card, the latency from the applied effects, the latency of the play engine and, in some cases, from the MIDI input latency as well as the latency of the graphical user interface of the operating system.

When using MIDI controlled synthesizers (VSTis) the response latency just as critical as the audio latency for monitoring (disruptive after 5 ms).

When playing files with the play engine, response latencies created by effects and the play engine itself are much less problematic (disruptive after approx. 250 ms).

The internal effects show the latency which causes the effect in the lower control element list of the effects. The latency of the sound card in conjunction with the driver being used is displayed in the System Options / Audio Setup dialog (Keyboard shortcut "Y"). The entire latency of all effects is shown in the status bar once playback begins.

Latency comparison

When using effects which cause latency, you must make sure that there is no time delay with parallel tracks, objects, or AUX channels. Depending on the location where effects which cause latency are integrated, various latency comparison methods may be applied. When using latency-causing effects in live input tracks (monitoring), all other tracks must be compensated.

When using effects which cause latency in tracks that are playing content on the hard disk, latency may be balanced using a read offset. The other tracks do not have to be delayed in this case.

When using internal effects in the object and in the master under MME, a real-time latency balance method (intelligent audio stream preview management) is applied.

The advantages of this process:

- When linking the latency-filled effects or increasing latencies by modifying the latency-relevant parameters, passages containing silence will not be created.
- When unlinking the latency-filled effects or the reduction of latencies by modifying the latency-relevant parameters, errors/skipping will not occur.
- Synchronization always remains the same (real-time latency balance – playback not necessary)

- Time changing parameters (timestretching or resampling) may also be used without causing any synchronization problems.

A delay of other audio streams is not necessary.

Lock

This button sets a lock on individual objects or all objects of a track.

If the lock button is active, then individual objects or all objects on a track may be locked in the track header.

Other parameters of locked objects depend on the "Lock definitions (view page 708)" option in the "Object" menu.

M

Markers

Markers act as reminders for position points. They are visible as named orange bars along a special line at the top edge of a project. Markers may be placed during playback as well as during recording.

MIDI object

MIDI objects in Sequoia are edited like wave objects.

Object-oriented editing for audio allows a VIP object to be used for playback instructions of audio files. MIDI objects are structured according to this concept: they refer to a recorded or imported MIDI file and may be copied, split, and trimmed. They have fade handles as well as a middle volume handle.

MIDI objects may also be edited with real-time effects in the object editor. Sequoia displays a different type of object editor specially adapted to MIDI objects for this purpose.

Monitoring

The signal to be recorded is played back through the output device. Different modes are available depending on the driver model used and the monitoring mode. In a strict sense monitoring has nothing to do with the playback of the recorded signal. It really only deals with the signal at the input.

O

Object editor

The object editor enables you to edit the properties of any audio object exactly. For example, the length of the object may be entered numerically or the fade characteristics may be determined here.

Real-time effects, EQ, and dynamics editing may be applied to the object without having to access track effects in the mixer. The selected processing does not depend on any following changes to the object position.

Object-oriented

Object-oriented work describes a method that makes it possible to apply various changes to the audio material connected with the object virtually, i.e. without affecting the actual audio material (data on the hard drive). Sequoia features an independent audio engine behind every object, which enables virtual editing of all DSP functions on the object level.

Objects

In virtual projects, the audio data is represented by rectangles (referred to as "objects") on multiple tracks. An object is a representation of the sample or a segment marked within the sample.

In addition, each object has certain attributes which may be modified using Sequoia's object editor. The rectangle features several handles which may be used to modify the object's start time, length, and volume attributes.

Objects that overlap in one VIP track may be faded into one another. For exact crossfade settings, try the crossfade editor.

The lock symbol positioned on the lower edge of each object may be used to protect a single object against accidental modification. The attributes to be protected can be selected in "Lock definitions".

P

Play cursor

The play cursor or the position line is the vertical line that moves during playback to display the current position in the audio material.

The starting position of the play cursor is moved by clicking in the project. The mouse must be in "Universal" or "Range" mode. If a range is opened, it will be cleared by this action.

R

Range

A range can be stretched or selected in projects using the mouse. As soon as a range is selected, pressing the spacebar will play it back. A range is defined by its start and end (horizontal) as well as by its upper and lower edges (vertical).

The range specifies sections where certain operations are executed (e.g. cutting, inserting, normalizing, fade in/out, move up, calculate effects, and delete).

The range is also used to define loops that are included during project playback.

Routing

Internal rerouting or assignment of signals.

S

Scroll bars

The scroll bars are displayed on the lower and right edges of the arranger window. These bars may be used to navigate the project or to enlarge or reduce the window section.

Scrub

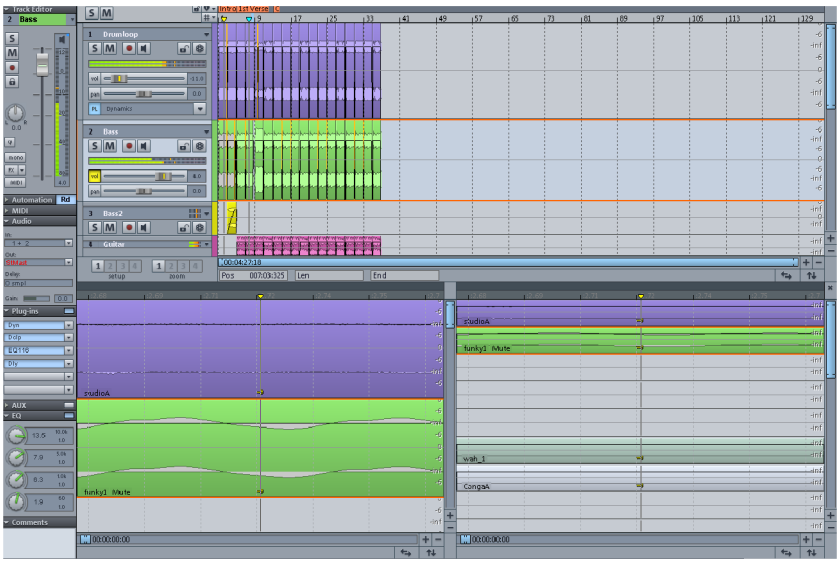
Scrubbing refers to "searching" through the audio material at different speeds to locate a certain position "acoustically".

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

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

Section

"Section" refers to the visible part of a project in the project window. Which section of the project this is depends on the position of the section and the zoom.

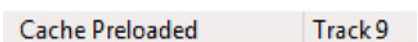


There are many commands for moving (scrolling) the visible clip and adjusting its size (zooming). These commands may be opened via the "View" menu, via the position bar, and via the shortcut keys.

Up to three different sections can be displayed in a project at any one time. This way, the entire project can be displayed in one section, while a smaller part where you are working is shown in a different section.

Status Display

The status bar is located at the bottom part of the Sequoia window. For longer actions or calculations a bar is displayed whose width shows the current state of operation.



Action instructions and explanations also appear.

The status bar can also be hidden by going to "View > Toolbars > Status Display" and removing the checkmark.

Submix bus

A submix bus is a track featuring all the abilities of a normal track, i.e. it can include track effects, and panorama and level can be automated. It has one input and one output. The important difference is that you can route the output signal of every track with a lower track number than the submix bus to this bus (instead of routing to the master or an output device).

AUX buses are typically used for effects such as reverb or echo, which may be equally required for various tracks, but in different amounts. In Sequoia, objects may be routed to AUX buses as well.

Surround bus

A Surround bus is required if Surround effects are to be applied. Tracks that include Surround effects therefore have to be routed to a Surround bus where the appropriate parameters may be set.

All tracks routed to a Surround bus have the Surround Editor instead of the normal panorama fader for setting the output level of the track. As soon as a Surround bus is available, the output signal of each individual object may also be routed to this bus and positioned in the Surround panorama independent of the track panorama settings.

From a technical viewpoint, the Surround master is also a Surround bus in a Surround project, since Surround channels may also be assigned to the audio devices and it cannot be routed further to a Surround bus.

System settings

Settings that are not project-specific are described as system settings. These include settings that apply to the entire project.

T

Tempo markers

A tempo marker indicates a tempo change at a certain project position.

Time input field

All fields that represent a time position can act as an input field by double clicking with the mouse. The time units may be changed by clicking on the right section of the field.

Title bar

The title bar is located at the top of the window. It contains the name of the application and the project.



To reposition the window, simply click the title bar and move it as desired. You can also move around dialog boxes by moving the title bar.

Tooltips

Tooltips provide additional information about the function of a certain control element. Hold the mouse pointer over an element for a few seconds, and more information will be displayed. Text featuring the name or a short description of the element will appear. Keyboard shortcuts and operating instructions are often also included.

Track

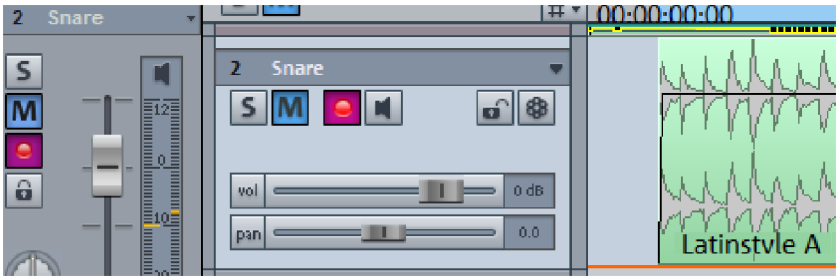
The audio track. A track can contain audio and MIDI objects. Automation curves are also represented in the track.

Track Header

The track header is the front section of a VIP track. It contains various functions including mixer elements, buttons for loading plug-ins, and other track parameters.



Track header with activated mute



You can show or hide the track header using the "Tab" key. The display options can be opened with "Shift + Tab". The "Buttons/Slider" option in the "VIP" section ensures that the track header is displayed as the default standard.

V

VIP - Virtual Projects

Virtual projects (VIP) are the top level of audio editing in Sequoia. In virtual projects, objects can arranged to create complex arrangements. All editing steps carried out in the arranger window are applied non-destructively. This allows you to select the proper cut position, cut length, volume level, effect setting, etc. without losing the original audio material or altering it.

The actual audio data is represented by objects on the various tracks. An object is a representation of the sample or a segment marked within the range. In other words, the object references the audio file.

Visualization

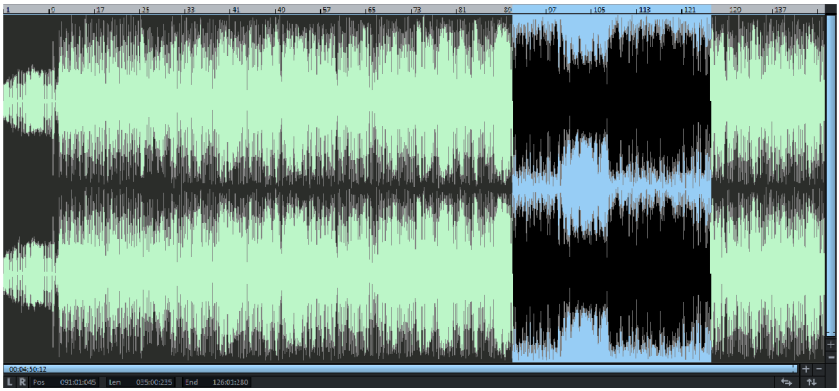
The visualization feature displays the audio material visually in different ways.

W

Wave editing

Non-destructive wave editing lets you take advantage of combining direct and virtual editing of audio material (wave projects), just like with VIP projects.

Wave projects



A wave project contains audio data. Objects in the VIP refer to this audio data.

If you use Sequoia as recording/mastering DAW you usually don't have to edit wave projects as all editing options can be performed quicker and safer in the virtual project. For using Sequoia as a wave editor, refer to "Working in the project window -> Sequoia as a wave editor (view page 62)".

The windows of the wave projects used by VIP objects are usually hidden. These may be accessed via "Edit wave projects" in the "Object" menu or by pressing "Shift" and double clicking an object used by wave project.

All loaded wave projects may be made visible as icons via the command "View > Window > Iconise All Wave Projects". To hide them again, go to "View > Window > Hide All Wave Projects".

The name of the wave project, sample length, and the resulting memory requirement may be entered in the title bar of the wave window. To activate a wave project you have to click into the corresponding window. Sequoia can manage an infinite number of wave projects on the screen.

Work area

Work areas are for adapting the program to certain work styles, i.e. they may be changed, renamed, and created. The work area is opened via the "Workspace bar". Work areas are available as wave editing or virtual projects.

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