/// MAGIX

S E Q U O I A

PRODUCTION | BROADCAST | MASTERING

MANUAL

Version: 17.250305.0913

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WHAT'S NEW IN SEQUOIA 17?

The most important innovations in Sequoia 17 at a glance:

Version 17.3

- **Dolby Atmos / ADM** (↗532):
  - **ADM and Dolby Atmos metadata editing & import**: Using the new ADM Editor, metadata can be created from scratch, modified, or adapted from existing projects. Sequoia now allows importing ADM and Dolby Atmos metadata from ADM/BWF files.
  - **Monitoring and renderer output options**: Monitoring playback is available through the ADM or Dolby Atmos renderer. When using the Dolby Atmos renderer, stereo or binaural output can be accessed separately from the monitoring section. Additionally, a loudness display provides real-time level information, and a trim control allows scaling all renderer inputs to achieve the desired loudness.
  - **ADM file export and Dolby Atmos loudness management**: Projects can be exported as BWF files with ADM metadata for use with encoding tools or distribution services. Users can choose to export either the entire project or a selected range. Additionally, when exporting for Dolby Atmos, a separate loudness report can be generated as a text file for precise level management.
  - **Additional resources and Dolby Atmos template**: For more details, press F1 in the ADM Editor. Alternatively, try the new Dolby Atmos template (Dolby Atmos 7.1.4.VIP) to get started with Dolby Atmos.
- **Surround improvements**:
  - **Surround presets**: The surround presets have been restructured and renamed for better organization, including new and updated Dolby Surround setups and additional ITU surround presets like Sound System D (5.1.4). Additionally, a quick access feature now enables faster selection of frequently used presets.
  - **Surround Aux sends**: AUX send panning can now be displayed in the track panning dialog. Enable this via "Show AUX Sends" in the '+' button menu. By default, surround AUX sends follow the main surround panner in linked mode, which can now be disabled for independent panning. Stereo AUX sends can toggle between linked mode and "Follow track pan" behavior.

■ **New program option – invert docking behavior**: A new setting allows you to invert the docking behavior of windows while dragging. By default, docking is enabled and holding Ctrl disables it. When the option is switched off, docking is disabled by default and can only be enabled by holding Ctrl.

■ **Multichannel support for Algorithmix Renovator** (Enterprise version / maintenance contract only): If Sequoia detects a Renovator installation, you can access the command directly by selecting one or more objects, opening the context menu, and navigating to Effects (Offline) > Restoration > Spectral Editor Renovator. The selected tracks will be sent to Renovator and the tool will open. You can now play back your edits directly in Renovator while the project continues to run simultaneously, allowing you to compare your adjustments in context.

Version 17.2

■ **Markers and range markers**:

- **Marker manager**: The marker manager can also show audio markers and warp markers of a selected object. You can have multiple marker managers open that can have different settings (window positions, columns etc.) which are individually saved.

- **Range markers and range marker track**: Saved ranges are shown as range markers in a separate track similar to the marker track. Range markers are shown in the marker manager like project markers, hence the range manager was removed. There is a unified keyboard shortcut for saving markers and range markers with Shift + number key. With marker lines, markers and range markers can be extended across all tracks.

■ **Improved metering**: There's a new option to switch the track peak meter to show the hardware input level automatically when the track is armed for recording. Also, any track input is available as an input source for a visualization window.

■ **Export Improvements**:

- **Improved single track export**: When exporting the project into individual tracks, there's a new option to include the content of AUX or Submix busses. This means that if tracks are routed to submix buses that contain their own effects or if tracks send to AUX buses, the influence of these buses is also included in the exported files of the individual tracks.

- **Export selected range markers**: A new time selection option "Selected range markers" was added. This allows you to export each selected stored range in a separate file.

- **Recently loaded or saved files**: Now up to 16 entries of the last used files are available in a dedicated File menu sub-menu and in the start dialog. You can access these quickly with the keyboard shortcut Ctrl + Backspace.

- **FX I/O Matrix** (Enterprise version / maintenance contract only): The FX I/O Matrix now also allows existing sidechain inputs to be routed to plug-in inputs.

Version 17.1

- **FX I/O Matrix** (Enterprise version / maintenance contract only): Opened from the "Plug-in" menu of a VST Plugin dialog to route track audio channels into plug-in input channels and plug-in output channels to track audio channels.

- **Additional waveform scaling option** for scaling with object fades and volume curves only, without track volume curves.

- **More GUI improvements**:
  - Use Ctrl key while dragging a window to avoid docking.
  - The horizontal mixer size is now stored globally, i.e., the width of the mixer window is now maintained when tracks are added or the project is switched.
  - Lock groups are now displayed as colored spot on lock buttons.
  - New menu command to select the active track only (default shortcut Shift+Alt+Enter).
  - New commands in the automation menu to show only Vol/Pan/AUX/EQ/Plug-in automation lanes.

- **Comping with take lanes improved**: Audio and MIDI takes can be switched together in take lanes. Obsolete takes that occur when working in the take lanes can be removed in the dialog "Project status".

- **More ARA plug-ins supported**: RipX DAW can be used, compatibility with the Synchro Arts VocAlign and Synchro Arts Revoice plug-ins has been improved.

Version 17

- **Revised design**: A sleek, modern flat design ensures a modern visual language and fatigue-free working.

- **New export dialog**: The export dialog (↗661) combines the functions in the menu **File** > **Export...** for file export to the various audio formats and the track bouncing dialog in a new, uniform dialog, with the option of exporting to several file formats simultaneously and generating file names from project properties (track names, object names, date etc...).

■ **Revised comping**: Comping, i.e. combining sections from different recording takes, can now be carried out directly in the project instead of in a separate Take Composer window. For this purpose, there are the new take lanes (↗138) below the tracks.

■ **Auto-conforming** (↗628): Transfer cuts from a video recording to a multitrack project using the EDL exchange format.

■ **AudioWarp** (↗366): Correct timings quickly and efficiently directly on the objects. To do this, warp markers can be set in audio objects, allowing the audio material to be grabbed and moved in order to correct irregularities in a recording or to change the rhythm of the audio material.

■ **Integration of external hardware**: External hardware effects can be conveniently integrated into the Sequoia mixer as a latency-compensated ExternalFX plug-in (↗225).

■ **MIDI plug-in support**: It is now possible to use MIDI plug-ins (↗380) such as sequencers, arpeggiators or chord utilities together with virtual instruments on one track.

■ **Marker tracks**: For a better organization and overview, markers are now displayed in a separate track. The height of the track can be freely adjusted so that long marker names are no longer cut off by following markers. The markers (↗112) can now also be colored, the Marker track head contains buttons for creating, coloring and locking markers.

■ **Revised dockable mixer**: The Mixer (↗478) can now be docked anywhere in the program window. The controls adapt dynamically to the available space. It is also possible to increase the height of the mixer window beyond the respective space requirements of the individual sections in the mixer channels, allowing the mixer channel faders to be extended.

■ **Redesigned Track Editor**: The Track Editor (↗59) has been reorganized for better clarity, the important controls for the AUX send, track effects and automation are now together in one section.

■ **Dockable plug-in browser**: The plug-in browser (↗216) is available in a simplified variant as a dockable window that can always remain open. It allows effects to be inserted into tracks and objects simply by dragging and dropping.

■ **Help** and **manual** have been completely revised and updated.

■ **Selectable input of the visualization (↗1)**: The input signal of the visualization can now be selected directly on the visualization window from a menu.

■ **PreSonus FaderPort support**: Sequoia now natively supports PreSonus FaderPort V2 controllers and higher (FaderPortV2/FaderPortV2 8/FaderPortV2 16). For more information, see the document `PreSonus FaderPort Support in Samplitude DE.pdf` in the program folder.

■ **Softtube Console 1 support**

- Many **improvements for working in the project window**:
  - New, meaningful mouse pointers for the editing areas in the track
  - Numerically adjustable volume directly at the object
  - optimized display of notes in a MIDI object
  - improved default settings for many display options
  - Color button in the object editor sets the object background color.
- VST3 plug-in compatibility and performance improvements.
- **ARA2 plug-ins on the track**: When loading ARA plug-ins into a track, the plug-in is automatically inserted into every object on the track. This also works with objects added to the track later, for example if an object from another track is moved to the track.
- **Overload-proof float recording**: If your recording device supports true 32bit float recording, clipping can no longer occur in **32bit (float)** recordings, as levels above 0dB are also saved correctly in float format.

DOCUMENTATION AND HELP

There is a variety of information available to assist you in working with Sequoia.

- Help features
- Online support area
- Online user forum
- PDF manual

Help

The menu item **Help** (keyboard shortcut: **F1**) provides detailed explanations of a specific program function at many points in the program. Via menu **Help** > **Help Index...** you open the start page of the help in your standard browser. There you can jump to specific topics in the help using the table of contents on the left or search for information in the help using the search box in the upper right.

In addition, there's a search box (↗52) in the upper toolbar provides you with for finding menu items and help topics. You can also perform the search via menu **Help** > **Search...** or keyboard shortcut **Ctrl + F**.

> 💡 *Using the search field in the program, you search only the keywords and headings of the help. If you do not find what you are looking for there, use the search box in the help itself to search the full text of the help.*

Forum

Also visit the Pro Audio user forum. Here, as a registered user, you enjoy the benefits and support of a professional forum community and can contribute yourself.

PDF Documentation

All information from the help can also be found in a PDF document in the program folder of Sequoia. It can be opened via menu **Help** > **Open Manual...**. In the program folder there is also a document with the most important keyboard shortcuts to be printed.

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

Other Functions in the Help Menu

About Sequoia**...**: Opens the About Box with copyright notices. You will also find the exact version number of Sequoia and your serial number. You will need both if you need to contact customer service.

Start selection: see below

Product registration: If you have installed a demo version, you can register the full version via this menu item.

Deactivate program license (P3)...: The Sequoia license allows use on a maximum of 2 computers. You can use this command to deactivate the license on this computer to be able to use it on another computer.

Download more instruments...: Selecting this menu item will take you to the MAGIX website, where you can browse through the entire latest range of Vita Solo Instruments and download your desired additional instruments.

MAGIX Auto-Update: Use this to check whether product updates are available for Sequoia. If so, you can install them right away.

Language: Select the language of the program interface and help. Sequoia restarts in the updated language.

> ℹ️ *Help is available in English and German only.*

Start Selection
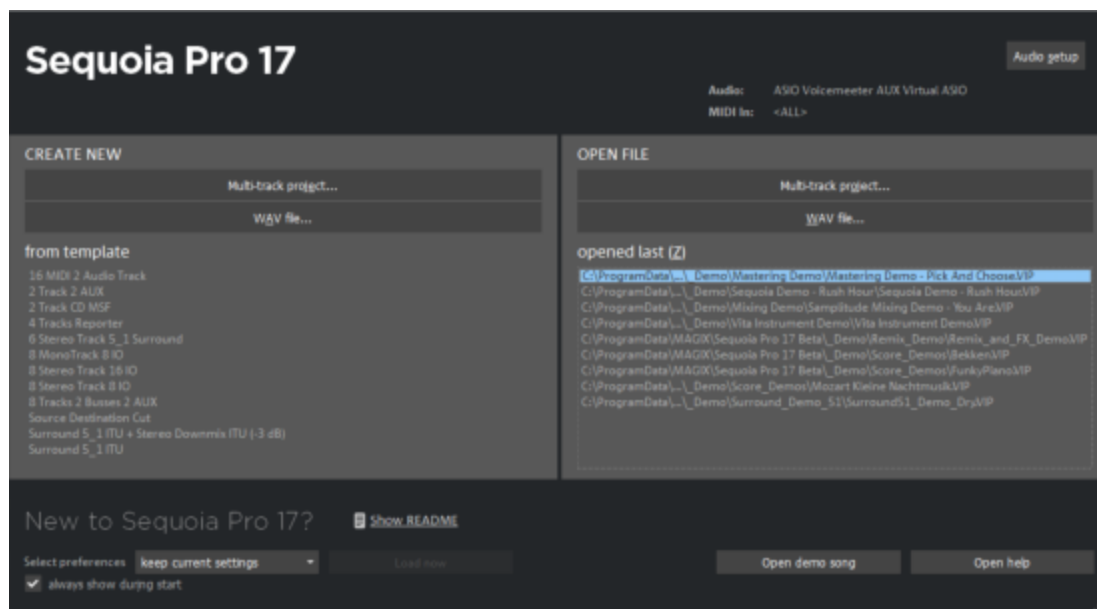
The start selection opens every time you start Sequoia, but can also be opened from the menu **Help**. It offers several quick options for working with Sequoia.

For checking purposes, at the top there is a display for the selected **driver**, the **Audio setup** button takes you to the corresponding settings dialog to change the audio driver, if necessary.

On the left side are options to create a new wave or multitrack project. For the latter, you can directly access the project templates from the New Virtual Project (↗641) dialog.

On the right you can open existing projects, choosing from the list of recently edited projects.

Below are useful options for getting started working with Sequoia with access to the demo song and help.

In the drop-down menu **Select preferences** you can load a complete set of program settings. For more information on managing saved program settings, see Loading and saving settings (↗730).

> ℹ️ **Attention**: *To make it easier for program beginners to switch to Sequoia, the program behavior has been changed in some places compared to older versions of Sequoia, e.g. the mouse wheel in VIP no longer scrolls horizontally, but vertically. To make the keyboard and mouse behave exactly as in previous versions, select the entry Sequoia 13 Shortcuts/Scroll wheel behavior from the list. For more information, see Keyboard Shortcuts and Edit Menu (↗709) in the Mouse Wheel section.*

If you do not want the startup selection dialog to be displayed at every startup, uncheck the **Always show at startup** option at the very bottom.

Contact

Support

Registered users get technical support. For more information, visit our online support area.

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

Supported operating systems

- Microsoft Windows 10 or 11, 64-bit

Minimum system requirements

- **Processor**: 2 GHz and higher

- **RAM**: 8 GB

- **Screen resolution**: min. 1280 x 1024 pixels, 2nd monitor recommended.

- **Hard disk**: Memory requirement 2 GB with minimal installation, 90 GB with Independence Sampler Content. SSD recommended.

- **Internet connection**: Required for registration, activation, downloading additional program content and individual program functions.

- **Audio playback**: Sound card (ASIO-capable sound card recommended).

- **Optional**: CD/DVD recorder, MIDI interface

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. If you have an installation DVD: Insert it into the DVD drive.
   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".
   In the installation screen, you can learn about the contents of the installation DVD, visit our website, or install additional programs such as CodeMeter RuntimeTo manage the license and check its validity by Sequoia, the software Code Meter Runtime with the included CodeMeter Control Center is also required. It is automatically installed together with Sequoia. (↗34).

2. If you have purchased a download version, run the downloaded installer.

3. You can select the language in which you want to install Sequoia.

   > ℹ️ *The program documentation is only available in German and English.*

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

5. Follow the instructions and click "Next" each time. If you choose the "User defined" installation style, you can also specify the destination 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.

> ⚠️ *For Sequoia to start, a valid license must be present on the computer or on a connected dongle. For more information on this, read the section* License Management *(↗34).*

Sequoia License activation

To start Sequoia you need a valid license. This license can either be stored locally on the computer or on a dongle for mobile use on different computers. Depending on the scope of the license, other deployment models are also possible, such as deploying the license on a license server in the local network or lending licenses over a specified period of time.

After purchase, the licenses are made available in a web depot. Activation and transfer to your computer or into a dongle is done via the Program **License Central**. Other administrative operations, such as moving or renewing licenses, can be performed on your Web Depot website.

To manage the license and check its validity by Sequoia, the software **Code Meter Runtime** with the included **CodeMeter Control Center** is also required. It is automatically installed together with Sequoia.

If necessary, start the **CodeMeter Control Center** program via **Windows Start** > **CodeMeter** > **CodeMeter Control Center**. You can access the interface of the program via the tray icon 🌀 .

> ⚠ *Note: You need version **6.90** of CodeMeter Runtime. **License activation does not work with older versions!** This version is included in the Sequoia installation program.*

Activate license

When Sequoia is started, the program **License Central** opens.
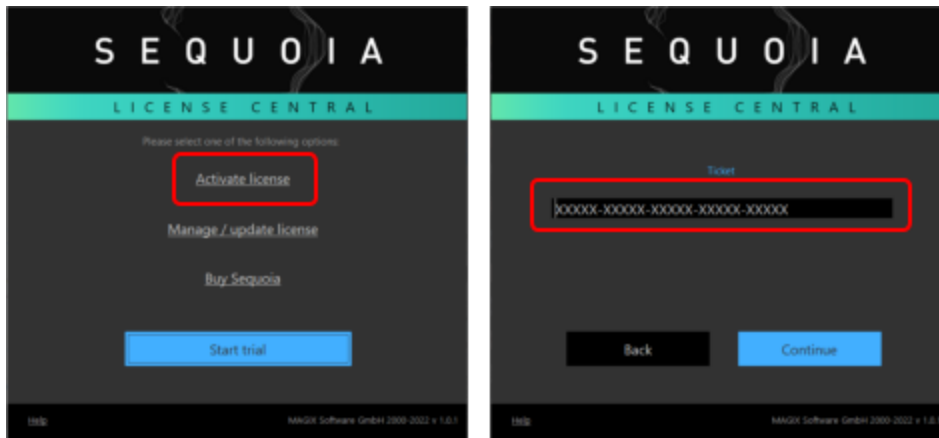
Click **Activate License** . After purchasing their license, you had received an email with a ticket number via the email address provided during registration. Now enter this ticket number here and click on **Continue**.

> ⓘ *If your ticket is a link to your web depot's website, in the form https://depot.license-central.magix.com/get.php?id=XXXXX-XXXXX-XXXXX-XXXXX-XXXXX, the string at the end is your individual ticket.*

You can now transfer this license either to a local file on the computer or to a CodeMeter dongle.

Dongles of the 1st and 2nd generation - not suitable   *3rd generation dongle - suitable*

Also 3rd generation dongles that shipped with a given product code along with an older version of Sequoia will not work with the new licensing system.

Both options are referred to by the CodeMeter licensing system as **CmContainer**. You can think of the local container file as a kind of virtual dongle.

Select the container type. The license manager automatically detects connected CmContainers (dongle or file) via the CodeMeter Runtime program. With **Activate now** Sequoia transfers the license there.

ℹ️ *To transfer the license from the Web Depot to a local CmContainer, the computer on which you want to use Sequoia must be connected to the Internet at least for this transfer. If this is not possible, you can transfer the license to a CodeMeter dongle on another computer, which you can then use on the (offline) computer with Sequoia. Or you can use the **file-based license transfer**.*

If the license transfer was successful, the license is displayed in the CodeMeter Control Center.
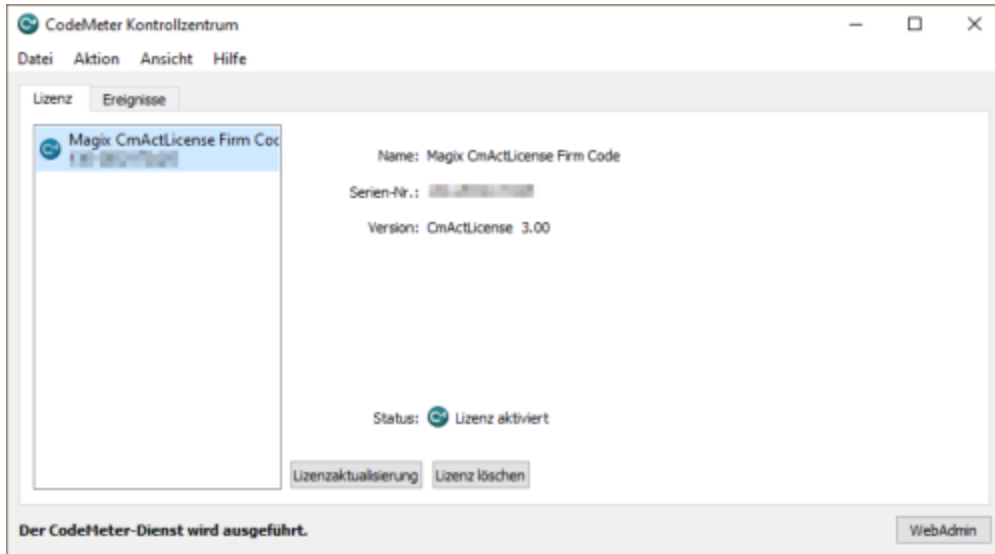
Renew subscription

If you purchased a license as a subscription, the license must be renewed at the end of each billing period. To do this, open the program **License Central** and select **Manage/Update License** and then **Auto Update**.

License Central checks online if the subscription is still valid and updates the local license to the next expiration date.

> ℹ️ *You will be warned in time a few days before the subscription expires when you start* Sequoia.

Moving and restoring licenses

Via **Manage/Update License** and **Go to web depot** you get to the overview of your licenses in your web depot.

A license cannot be copied, but only transferred between different CmContainers. It can only exist exactly once, either in the Web Depot or in exactly one CmContainer.

Use **Move Licenses** to transfer a license from a CmContainer back to the Web Depot. It can then be moved again into another CmContainer on another computer or into a dongle.

If a computer or dongle is broken or lost and a CmContainer is no longer usable, the license is not lost. You can transfer it again into a CmContainer with **Restore License**. In this case, it must be ensured that the lost container really no longer exists. It will be marked as invalid and cannot and must not be used again.

> ℹ️ *For example, if a lost dongle is found again after the license has been restored in another CmContainer and you try to use this dongle with the same license, it will be permanently locked. In this case contact the support, you will have to send in the dongle for reprogramming.*
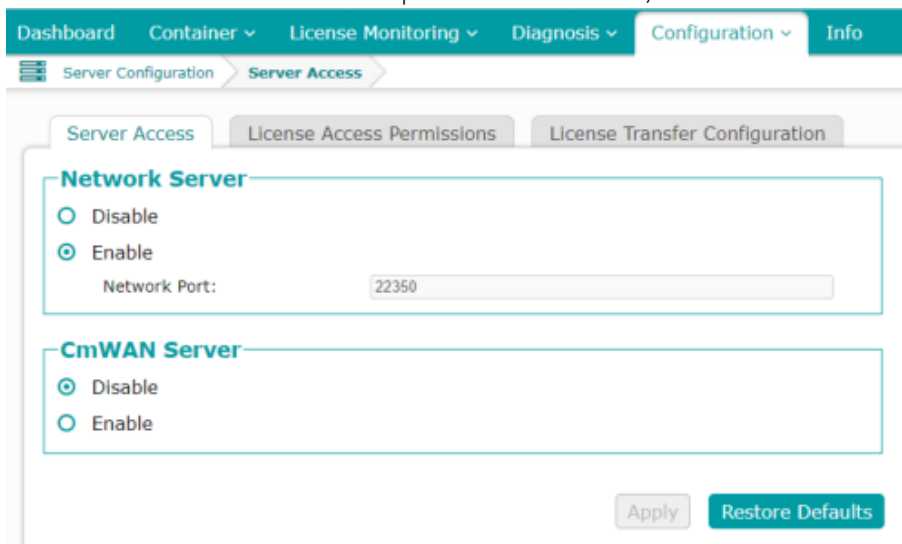
Network licenses

If your purchased license includes the name suffix *net license*, you can make the licenses stored in a CmContainer (local file or dongle) available to other computers in the same local network. Then a CmContainer with the necessary licenses has to be connected on only one computer. These are so-called "floating" licenses. This means

that you can run any number of installations on the network, but only as many of them can be used simultaneously as there are licenses in the container.

Running CmContainer as CodeMeter Server

1. Install a current CodeMeter Runtime on a network server or on a PC (Windows, Linux, Mac OS X) that is available in the network. If there is a Sequoia installation on the computer in question, the CodeMeter Runtime is already installed.

2. Transfer Transfer the licenses to a CmContainer on this computer.

3. Start the CodeMeter Control Center with **Windows Start** > **CodeMeter** > **CodeMeter Control Center**. Open the program via the tray icon ⟳ .

4. Click **WebAdmin** in the lower right corner. Your Internet browser opens with the web interface of the CodeMeter Control Center.

5. In the WebAdmin menu, go to **Settings** > **Server** > **Server Access** and click near Network Server on **Enable**. At **Network Port** the network port used for client/server communication is displayed.

> ⚠ *Network communication via this port must not be blocked by the firewall. If the local firewall detects network communication through this port, the firewall may display a warning message. In this case, click on "Allow access" so that a corresponding rule is created in the firewall.*

6. Click the **Apply** button. The server is now ready.

> ℹ **Note**: *In the WebAdmin under* **Container** *all existing CmContainers are listed. Click on a container to view your available license count.*

Use of the program on the client computers

1. Install **Sequoia** on the client machines.
2. Check if the current CodeMeter Runtime is installed and running on the client machines.
3. Launch **Sequoia** as usual. The server with the network licenses is automatically found in the network and **Sequoia** starts as long as the number of available licenses is not exhausted.
4. If the license server was not found, check the settings of your firewall. If you have any problems, deactivate your firewall as a test. If there are several license servers installed in the network, you can change the license server used at the very bottom of the WebAdmin.

Borrow licenses

If you run a license server with a CmContainer that contains a multi-user license, the clients must be able to access this server over the network to use these licenses. If clients are not on the network, you cannot access the licenses. For this case there are licenses with a special property: it is possible to borrow these licenses to the client for a certain period of time. During this period, the client computer therefore no longer needs to access the license server because it can use the locally stored license.

If you have such licenses, you can use the program **Sequoia** **License Manager** for this purpose.
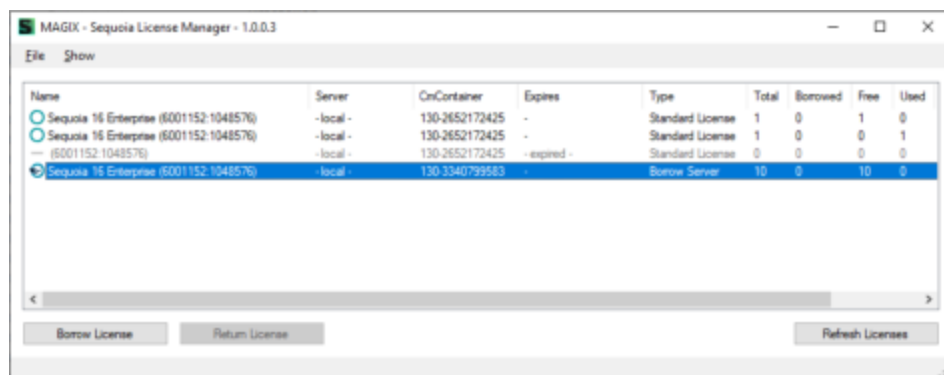
1. Start the **Sequoia** **License Manager** program on the client without permanent network access. He must have access to both the CmContainer with the multi-user license (on a dongle or a license server) and the container to which the license is to be borrowed, typically a local CmContainer on this machine.
2. Select the container with the license from the list. In the column **Type** the container must be marked with **Borrow Server**.
3. Select **Borrow License** and enter the security password.

4. Select the target container and enter the duration of the rental period. Click **OK** to transfer the license to the target container.

5. The computer can now use **Sequoia** for the duration of the rental period. After this period the license is no longer usable. It is automatically returned as soon as the computer has access again to the container from which the license was lent.

> ℹ *For more information on acquiring appropriate licenses, License Manager and security password, please contact* **Sequoia** *Sales.*

Notes

Internet Access

Communication with the CodeMeter stick or the local license container is done 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 must not be blocked by a firewall.

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

Restarting the CodeMeter Runtime Routines

Should access to the CodeMeter stick be blocked on startup or while the program is running, restarting the CodeMeter Runtime may help. To do this, start the **CodeMeter Control Center** and execute the **Restart CodeMeter service** command in the **Action** menu. This restart of the CodeMeter service can be made even while the program is running.

System Boot Problems on Certain Systems

When 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.

> ℹ **Note**: *If you experience recurring problems with your dongle please contact our* Support

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 on a Separate Computer with an Internet Connection

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

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 e-mail address in the start dialog.

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

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

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).

==

Serial number:

Email address:

ComputerID:

==

4. Now go to www.magix-audio.com via your Internet computer and log into the support area. If you do not yet have a user account, you can create an account.

5. After logging in select „Register product".

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

7. Then select your program version and click "Complete registration".
8. Click on "My products" in the start dialog of the Service Center.
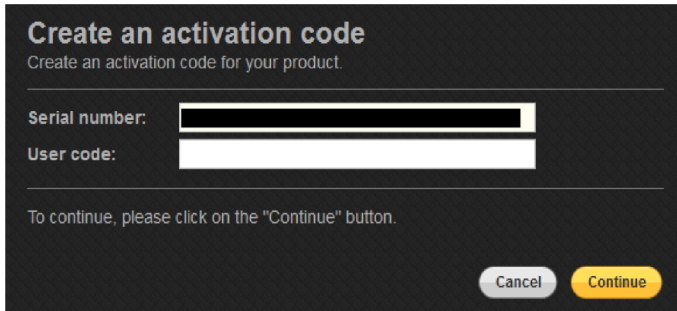
This will take you to an overview of your registered products.

9. Now click on the option „Create new activation" behind the recently registered product and enter your usercode (ComputerID) from the form.
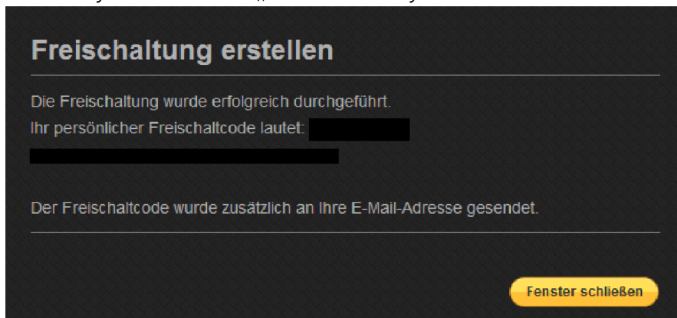
10. When you click on „Continue" you'll see an activation which is also sent by email.
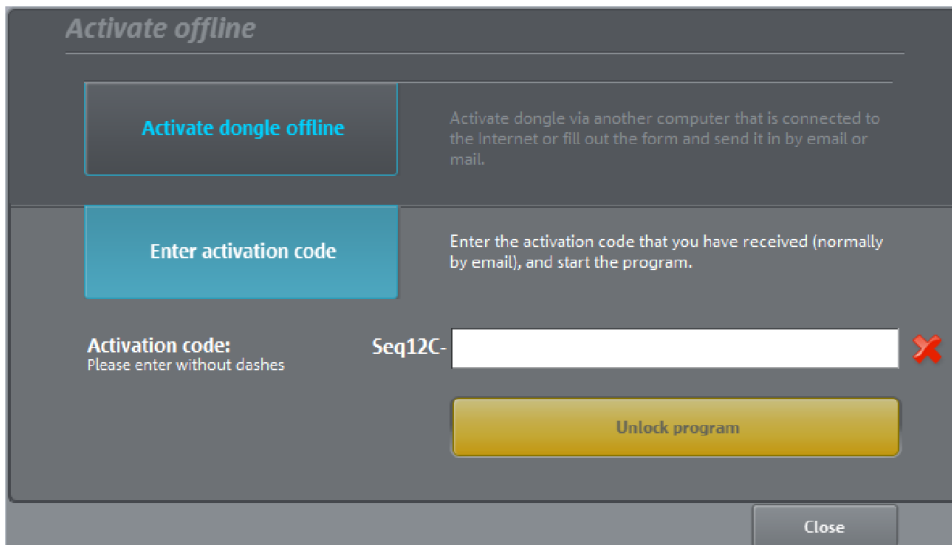
11. Return to the activation code dialog on your studio computer and enter the activation code in the corresponding field. Confirm the activation process by clicking "Unlock program".
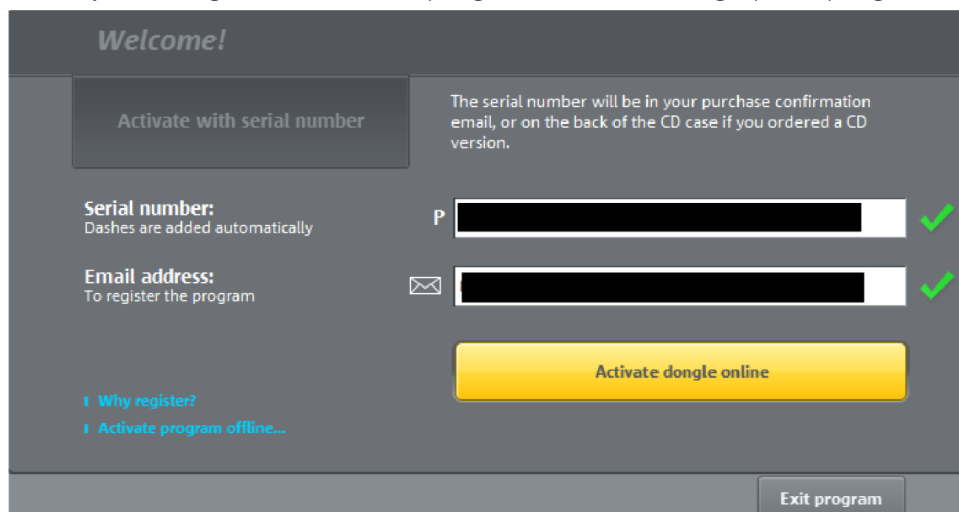
This completes the registration of **Sequoia**. You now have unlimited use of the program.

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 e-mail address in the start dialog.

3. Select "Activate program offline", click the "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).

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).

===

Serial number: █████████████████████████

Email address: ██████████████

ComputerID: ████████████████████████

===

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

Activation without direct internet access - by email/ post/fax
Activation by email/post/fax is an alternative to activation without a direct internet connection using the Service Center.
Please fill in the form below completely. When you're finished, add as an attachment and email to upgrade@magix.net

Or print it out and send to:
- Fax: + 49 (0)5741 / 310768
- Post : MAGIX AG, Borsigstraße 24, D-32312 Lübbecke, GERMANY

MAGIX will send you the activation code as soon as possible.

OVERVIEW OF THE PROGRAM INTERFACE

❶ **Title bar**: The title bar at the top of the program window contains the program name as well as the name, sample rate and length of the current project.

❷ **Menu bar**: The menus are located directly below the title bar. Each menu item can be assigned a keyboard shortcut (↗709).

❸ **Toolbars**: The two toolbars (↗50) contain buttons that can be used to execute commands or toggle functions.

  💡 *By right-clicking on an icon you can* customize the toolbars (↗54) *by removing buttons or adding other buttons.*

❹ **Grid bar, range marker track and marker track**: Marker track and range marker track (↗112) are located above the first track of the arranger. Markers can be positioned there. Above it is the grid bar that displays the project time in the selected unit of measurement. Ranges can also be selected here.

❺ **Arranger:** This is where the audio and MIDI objects are arranged on the tracks of the project. Around it there are scroll bars and other controls to move (↗104) the visible section of the project in the arranger.

6   **Track head**: The track head contains all the controls for a track, such as mute and solo functions or recording activation. At higher vertical zoom levels, some elements are hidden. More information about the individual controls of the track head can be found under Track head (↗63).
Arranger, track heads, grid and marker bar together form a **Project window**. Several project windows can be open at the same time.

7   **Track Editor**: The Track Editor window is opened by default on the far left of the track heads. It contains controls for all properties of the selected track. More information about the individual controls can be found under Track Editor (↗59).

8   Preset **Docker** (↗57) for various windows: managers, visualization, time display, keyboard, object editor and MIDI editor. Each window in Sequoia can be detached from a Docker by dragging its title bar and freely positioned or docked at a different position. The **+** button (far right) can be used to open additional windows.

> 💡 *The project window itself is also a docking area where you can place any windows. New project windows are opened by default in this docking area (project dock).*

9   **Transport console** (↗67).

10   **Status display**: The status display at the bottom of the program window shows information about CPU usage, latency, buffers as well as the progress of longer calculations. Information about commands selected in the menu is also displayed.

> 💡 *For a little more space on the screen, the status display can be hidden via menu **View** > **Toolbars**!*

Menus

In the menus at the top of the program window there are commands to control Sequoia. The vast majority of program functions can be operated more quickly using the controls and windows directly on the program interface of Sequoia. Nevertheless, all program functions are also located in the menus, because the menu entries serve as a basis for the definition of keyboard shortcuts (↗709).

> ℹ️ *The new, revised help does not explain all menu items again in detail in a menu reference, but refers to the corresponding menu items in the description of the program functions. If you are looking for a specific menu item, use the search field (↗52) in the upper toolbar. Help for a specific menu item or topic can also be found directly in the help file using the full text search.*

Therefore, here is only a brief overview of the contents of the menus:

- **File**: In this menu you will find all commands for importing and exporting files, loading and saving projects and configuring **Sequoia**.

- **Edit**: Here you can find all the commands for range selection and editing functions related to ranges, as well as other assistants for editing projects.

- **Track**: Here are all the commands related to the tracks and the mixer.

- **Object**: All commands for cutting, moving and editing objects are located in this menu.

- **Playback**: All functions related to playback and recording, to positioning the play cursor and to setting and navigating with markers can be found here.

- **Automation**: This menu contains all commands for automation.

- **Effects**: Use the menu items in this menu to apply the corresponding effects to selected objects.

  > **ⓘ** *Note: The effects are calculated* destructively *(⬈227) if you activate the option* **Apply effects offline** *at the very bottom of the menu.*

- **CD/DVD**: All functions related to burning CDs and DVDs and collecting and editing metadata can be found in this menu.

- **View**: This menu contains all the view options and commands to show or hide the individual managers and windows and to control the project display.

- **Help**: Here are all commands for program help, unlocking and registration, program update and additional content.

For a better overview, not used menu items can be hidden. If the help mentions menu items that you cannot find in the program, they are probably hidden. Hiding/displaying menu items is also done in the keyboard shortcuts/menu (⬈709) dialog in the program preferences.

Toolbars

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

By right-clicking a button, you can display the buttons as large icons, toggle the skin (⬈714), and customize (⬈54) or reset the toolbars.
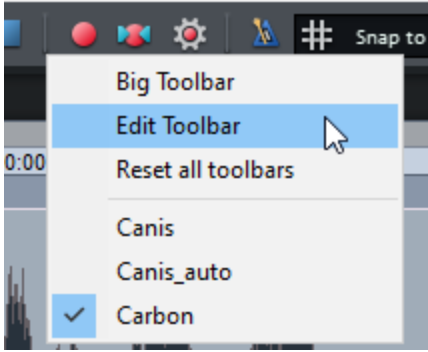
Upper Toolbar

The following overview shows the default toolbar in the workspace (↗58) **Default**.
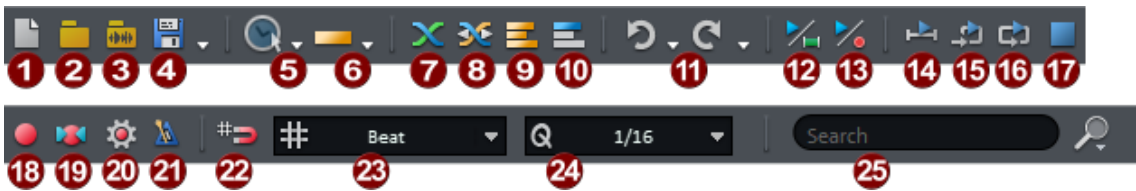
1	New Virtual Project
2	Load project
3	Load audio file
4	Saving and Exporting
5	Mouse mode (↗70)
6	Object mode (↗155)
7	Auto Crossfade On/Off
8	Crossfade Editor (↗170)
9	Group
10	Ungroup
11	Undo/Redo

12	Continuous playback while editing (↗88)
13	Playback and editing independent of recordingPlayback & Editing independent of Recording (↗95)
14	Play
15	Play into the loop
16	Play as loop
17	Stop
18	Record
19	Punch RecordingPunch Recording (↗102)
20	Recording optionsRecording Options (↗95)
21	Metronome (↗367) on/off
22	Snap (↗79) on/off
23	Snap menu, see Snap (↗79)
24	Quantization menu
25	Search menu commands and help topics (↗52) and enter macro commands (↗632)

Search Menu Commands and Help Topics

Sequoia provides an input field for finding menu commands and help topics.
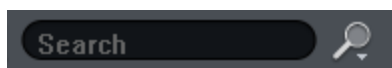
Search 🔍

Click in the search field or use the keyboard shortcut **Ctrl + F** and enter a search term. Up to five menu commands containing the searched term and the corresponding path in the menu are displayed as well as corresponding help topics.
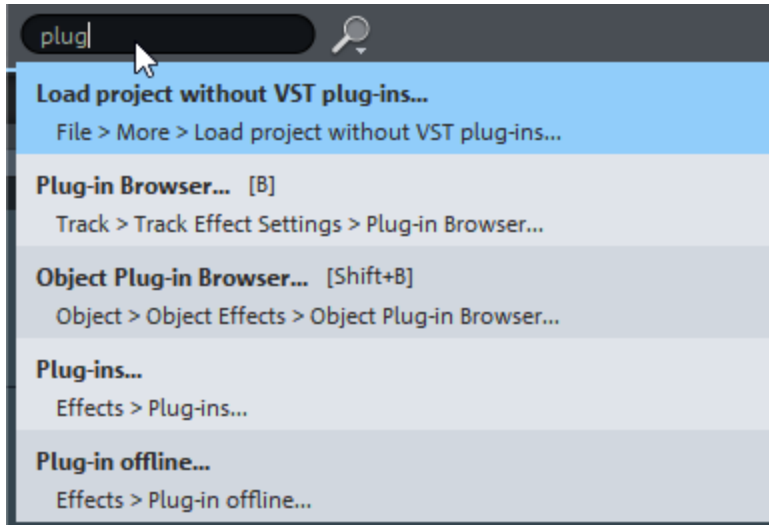
You can execute the commands in the search results directly, if you select a help topic, the help with this topic will be opened.

💡 *You can also use the search field to search for track and object names or to execute specific commands! For more information, see* Macro functionality *(⭷632).*

Lower Toolbar
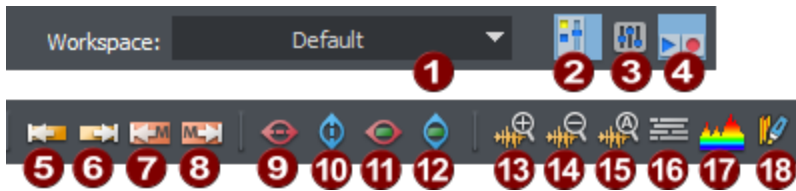
❶	Workspace (⭷58)
❷	Show Track Editor
❸	Open Mixer
❹	Display transport console
❺	Play cursor to previous object edge
❻	Play cursor to next object edge
❼	Play cursor to previous marker

8	Play cursor to next marker
9	Show all
10	Show all vertical
11	Zoom to range
12	Zoom to vertical range
13	Zoom into waveform
14	Zoom out of waveform
15	Display waveform normalized (⤢717)
16	Overview mode
17	Spectral display
18	WaveColor waveform display

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 Toolbars**.

This opens the **Edit Toolbars** window which contains all of the available icons.
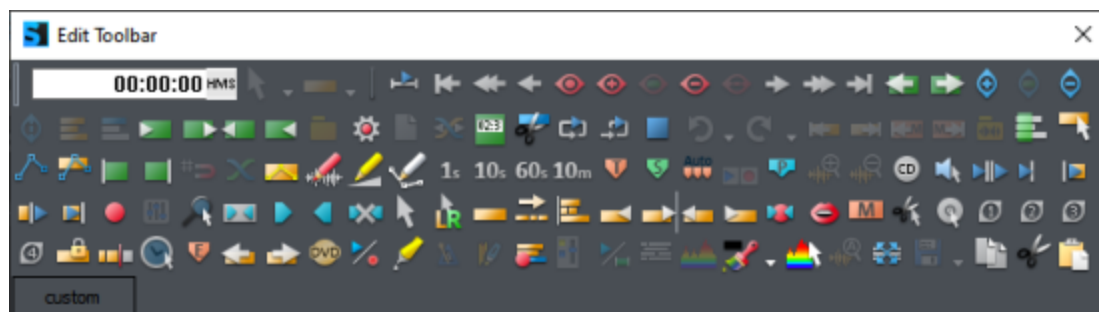
Icons that are already on the toolbars are grayed out. The others you can drag with the mouse to one of the toolbars to add them to the bar.
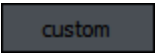
To remove an icon from a toolbar simply drag it off. To change the arrangement of the icons, drag them to the desired position on the bar.

Creating Your Own Buttons on Toolbars

You can create a button for each menu command from Sequoia.

1. Click on the **Custom** button. custom
    The dialog **Customize toolbar icons** opens.

2. In the dialog, click on **Add new** below **Icon**.
    Choose an icon to be displayed on the button.
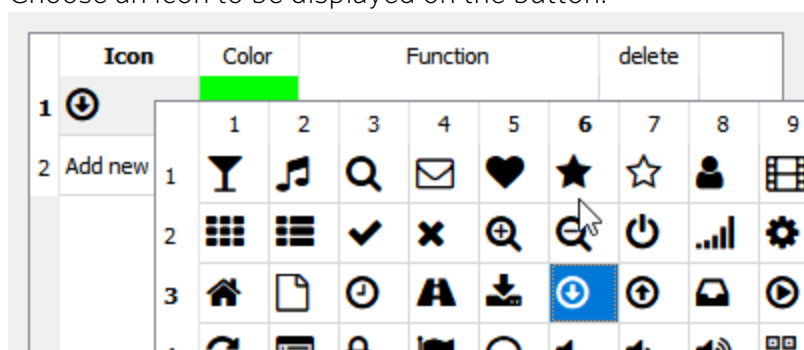
3. Under **Color** you set the color of the symbol. Click in the column **function** to open the complete main menu of Sequoia. Select the menu command to be executed with the button.
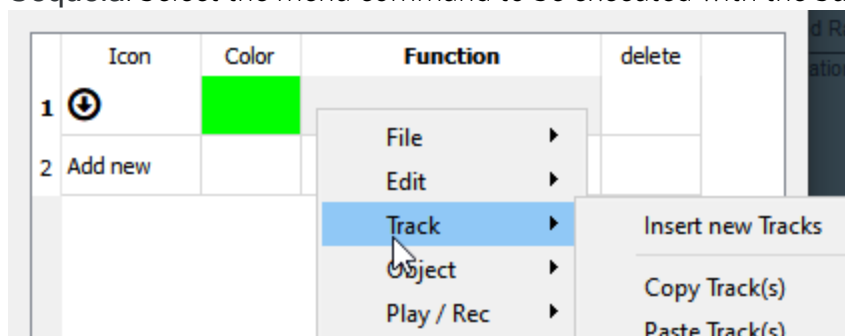
4. For another button, click on **Add new** in the next line. To delete a button, click the field in the column **Delete** in the corresponding row.

5. After closing the dialog, the new buttons are available in the Button Palette and can be added to the toolbars as described above.

Docking

Using docking, you arrange the windows of **Sequoia** on the program interface.

Each window (projects, manager, visualization, object editor, MIDI editor...) can exist on the surface as undocked ("floating") or docked. Undocked windows can be placed on top of other windows, resized as desired and feature a title bar. Docked windows share a screen area with other windows.

Docking Windows

You can dock undocked windows by grabbing them in the title bar and moving them to the designated areas on or in other windows. While moving, arrow icons will appear on the surface of the corresponding target window to show the possible docking positions.
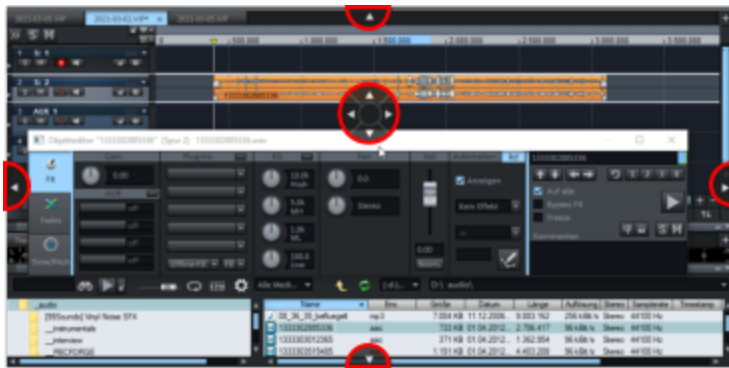
The docking areas have the following meaning:

- Dragging on one of the outer arrows will dock the window outside of the existing windows, thus creating a whole new docking area.
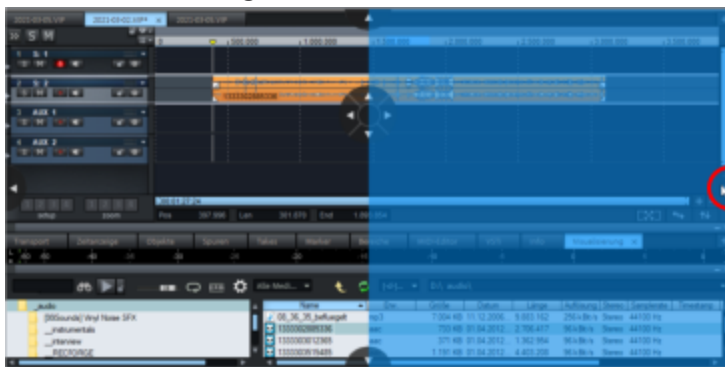
- Dragging on one of the inner arrows will dock the window in an existing docking area, the two windows will then share the screen space previously occupied by the window alone.
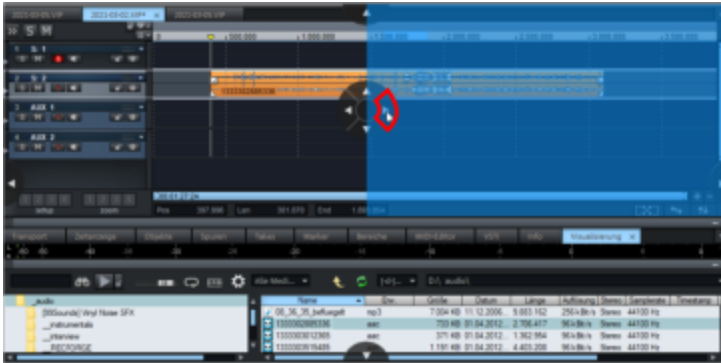
■ When dragging on the circle in the center, the windows are placed on top of each other. They now occupy the screen area together and are grouped in a **Docker**.
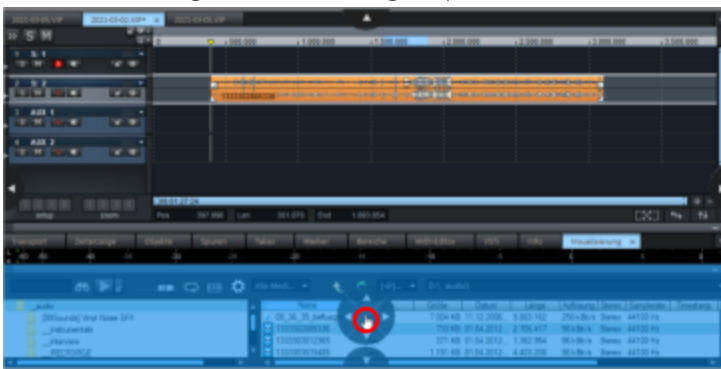
■ You can also place and arrange new windows inside a docker itself. To do this, simply move the respective window using the mouse onto one of the arrow symbols in the docker.
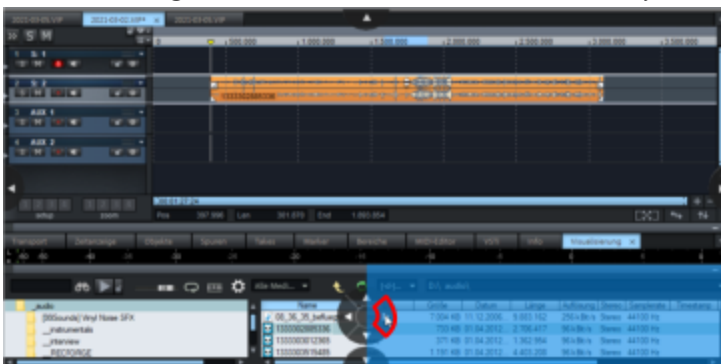
💡 *To prevent unwanted docking of windows at the docking positions, also hold down the **Ctrl** key when dragging the windows with the mouse.*

Docker

A Docker is a group of windows that occupy the same screen area.
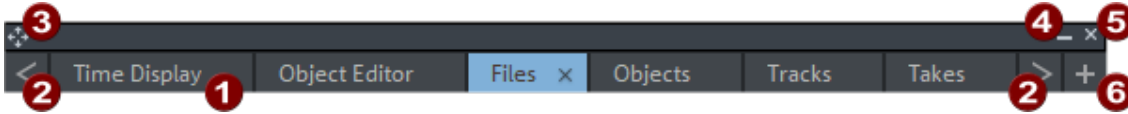
1 **Tabs**: You can switch between windows by clicking on the corresponding tab.
To rearrange the Docker tabs simply drag & drop them to a different position in the tab bar. If you want to detach a window from a Docker, drag it out of the Docker by its tab.

2 If more windows are open in the docker than tabs can be displayed, you can scroll through the tabs using the arrow buttons.

3 **Docking handles**: To move a Docker as a whole, drag the Docking handle ❖ at the very front of the Docker's title bar.

4 A 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. By dragging the header, the height of a Docker can be adjusted.

5 Close the Docker with the close button.

> **ⓘ** *The Manager/Docker can be shown and hidden in the* ***View menu***.

6 The **+** button opens the Docker menu, which allows you to open new windows in the same Docker.
It contains a list of all windows that can be integrated into the Docker. Windows that are already integrated are marked with (+). Entries for windows that can only exist once in the program are grayed out in the menu in this case.

Two Dockers are permanently available in Sequoia:

- All managers and many other dialog windows are opened by default in the **Manager/Docker**

- New project windows are first combined as new tabs in a **project docker**.

> **ⓘ** *In the project docker, the* ***+*** *button opens the* ***New Virtual Project*** *dialog.*

- Use the keyboard shortcuts **Ctrl + B** to switch to the Manager/Docker, **Ctrl + P** to switch to the Project Docker.

- With **Ctrl + Tab** you can switch between the individual tabs within a docker by keystroke.

- Press **Ctrl + F4** to close the active tab.

Workspaces

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

To do this, you can customize the Toolbars (↗54) (right-click on the toolbar) so that they contain only the icons 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 workspaces are already predefined, the "Default" Workspace is the default. It contains the toolbars in their default configuration and can be used as a starting point for your own customizations.
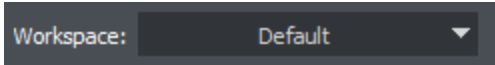
To customize a Workspace, right click on the drop-down menu and select **Edit Workspace**. In further menus you can select which toolbars are displayed. There are also entries for customizing the toolbars (↗54). Use **Save Workspace** to save your adjustments, preferably under a new name.

> 💡 *In the menu* **View** > **More** > **Workspace** *you will find all workspaces for selection at* **Select workspace**. *If you set a keyboard shortcut for it, you can switch the workspaces in this way completely keyboard controlled.*

Track Editor

🔲 The track editor shows the settings options for the selected track. It can be shown and hidden with this icon above the track header.

The Track Editor gives you quick access to all the important parameters of the active track. By default, it is located at the left edge of the project window, but like any other Sequoia window, it can be docked as desired (↗56).

In the Track Editor, settings of a single track are clearly displayed in individually expandable sections. This allows you to edit all relevant settings of the selected track in the Project window even without an open mixer or with low track heights. The controls correspond to those in a mixer channel, so you can also think of the Track Editor as a single mixer channel for the active track.

> ℹ️ *You can also open the Track Editor via the corresponding button of the* bottom toolbar (↗53) *or via the menu* **View** > **Track Editor** *(keyboard shortcut: Ctrl + Alt + Shift + E).*

> ℹ️ *When docking in landscape mode, a different layout is used, the controls are identical.*

Main

The section **Main** contains the most important settings for any tracks.

1 **Track number display and track name**: Right-clicking on the track name opens the "Track Options" dialog.

2 **Record symbol**: Activates the track for recording.

3 **S**: mutes all other tracks. For detailed information, see Global Solo Modes (↗505).

4 **M**: Mutes this track. (Mute)

5 **Mono**: sets the track from the input to the pan control to mono processing, i.e. all track effects before the pan control calculate in mono. However, the Submix and AUX return buses always remain stereo.
In the default routing, only the post-VST plug-ins and the post-AUX sends are located after the pan control and are thus calculated in stereo. The routing position of the pan knob can be set in the **FX Routing** dialog. Again, right-click to access the Stereo panning dialog (↗492).

6 **Lock**: Prevents the objects in the track from being accidentally moved or deleted.

7 **Panorama**: Controls the positioning in the mix. Right-click to access the Stereo Panning dialog (↗492), which allows you to make further settings for the stereo image, stereo width and phase.

8 **Phase reverse**: Rotates the phase of the signal by 180 degrees. Right-clicking also takes you to the Stereo Panning dialog/stereo editor (↗492).

9 **MIDI**: sets the track to MIDI recording.

10 **The speaker icon** turns on the Monitoring (↗695), i.e. the playback of incoming signals when the track recording button is activated. "MIDI thru" will be set active here for MIDI tracks.

11 **Volume fader**

12 **Level meter**: The two LED chains show the output signal of the track, and the input signal when monitoring is activated.

13 **Volume input field**

⑭ **FX**: opens the Effect Routing dialog (↗220). Right-clicking on this button gives you access to the menu for the track effect chains (↗220).

> 💡 *If you select the **Mixer** option in the menu, the mixer will be opened at the track's position.*

⑮ **AUX**: Here you can set the AUX sends (↗495) or right-click to switch to output assignment (↗487) or sidechain sends display (↗378).

⑯ **Plug-ins**: Here you can reach the track insert effects. Clicking on the empty insert slot opens the plug-in browser (↗216), 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 plug-in's interface. In the menu at the insert slot ▾ you reach further functions, e.g. you can open the plug-in browser again to exchange or remove the plug-in. The **Plug-ins** button at the top turns all effects in the respective effect chain on and off. A visual indicator (*) for plug-ins shows that they were previously active and will be reactivated the next time the **Plug-ins** button is clicked. The order of the plug-ins can be changed by dragging and dropping a filled slot to another position.

⑰ Use the button **Track Auto** /**Lanes + Obj. Auto** in the **Track Editor (↗59)** to switch the display of the automation curves in the track:

- ■ In **Track Auto** mode, track automations are displayed in the track over the objects, object automations are hidden.

- ■ In the **Lanes + Obj. Auto** mode, the object automations are displayed in the track over the objects, the track automations are displayed in lanes (↗340).

⑱ **Automation mode**: Left-click to quickly switch the automation mode (↗342) between the last selected mode and Read.
Right-click to access all automation modes.

⑲ An automatable parameter can be controlled directly in the track editor with the **automation controller**. Click on the parameter field above the slider to display the automation menu (↗338) and select the effect and parameter.

MIDI

The section **MIDI** contains the specific settings for MIDI tracks.

① **In :** MIDI input menu

② **Out**: MIDI output menu
In addition to the hardware MIDI outputs, the menu also gives you quick access to all MAGIX software instruments and lets you send the track's MIDI output to a VST effect.

3 **Channel In**: MIDI input channel

4 **Channel Out**: MIDI output channel

5 **Program:** With Program you can select the presets of the software instrument selected at **Out**. If a MIDI hardware output is set, the default GM program names are displayed. The program change commands are sent at each playback start.

6 **Bank MSB/LSB**: Additionally device specific bank change messages can be sent to control an external MIDI instrument.

7 **Map**: Here you can select a Drum Map (↗438) for the assignment of the MIDI notes to device-specific sounds.

8 **Transpose**: Here you can transpose the MIDI output. This function works in real time, i.e. the MIDI data in the MIDI objects is not changed.

> **ⓘ** *If a drum map is selected, the transpose function will not be available because the pitch is fixed by the drum map.*

9 **Velocity Dyn**: This button activates the Velocity Dynamics (↗463) as a real-time track effect.

10 **Input Q**: Input quantization. Incoming data is then already quantized during recording according to the MIDI quantization settings (↗449). You can restore the original positions with the menu command **Object** > **Quantize** > **Reset Quantization**.

11 **MPE**: Sets MIDI recording to MPE mode (↗467)

12 **Note Expression Map**: Here you can select a VST3 Note Expression Map (↗469).

Audio/EQ/Comments
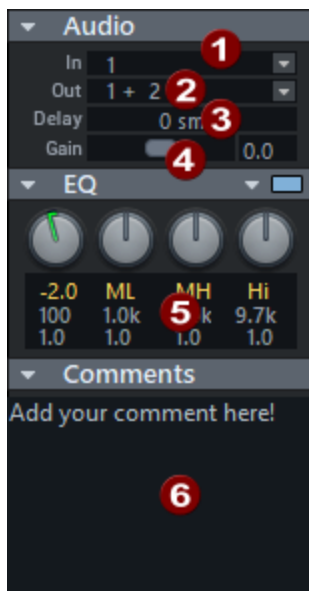
1 **In**: This sets the input of the track, which can be the audio input devices, outputs of other tracks or the outputs of software instruments.

2 The **Out** slot determines the audio output device. This can be, for example, a hardware output, the master or a bus.

3 **Delay**: This sets a delay time for the track.

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

5 **EQ**: The parametric EQ for the track. Right-click to access the advanced settings dialog with six frequency bands.

6 **Comments**: The Comments area corresponds to the track comment in the Infomanager (↗337).

Track Head
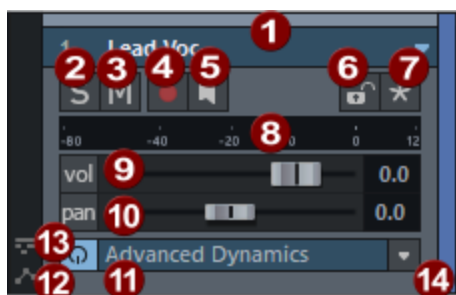
The track head is displayed with fewer options depending on the vertical zoom level. In order to see all options, you may need to zoom (↗104) into the display.

> 💡 *Tip: For more space on screen, if you do not need the track head, you can use the **tab** key to switch to the alternative project display mode, where the track head is hidden.*

❶ 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. By clicking on the arrow you can select further track-related functions from a context menu.

❷ The **S** button mutes all tracks except the selected track (Solo).

❸ The **M** button mutes the track.

> ℹ️ *The S and M buttons above the track header of the first track turn all solo and mute states on and off together.*

❹ Record: This button activates the track for recording (↗91).

❺ The speaker symbol activates the monitoring (↗692) (listening to the input signal).

❻ The **lock button** prevents the objects in the track from being accidentally moved or deleted.

❼ Revolver tracks: With the help of Revolver tracks (↗137) you can create different variants of arrangements of the objects in a track.

❽ Peak Meter: Both LED rows display the input and output signal of the track.

❾ Volume Fader: This controls the track volume. The **vol** button toggles the volume automation curve (↗338). Using this, level changes in the track can be controlled via an automation curve and the channel fader movement can be recorded.

❿ Panorama Slider: This controls the position in the mix. The button **pan** in front of the slider toggles the panorama automation.

⓫ Plugin: You can load track effects into the effect slots (↗215) or assign VST instruments to the track.

⑫ Automation Lanes: Use the icon to show and hide the Automation Lanes (↗340) of the track.

⑬ Take Lanes: Use the icon to show and hide the Take Lanes (↗340) of the track.

⑭ Color selection: If you click on the right edge of a track head, a selection menu will appear for specifying the color (↗78) of the track and its objects.

Time Display

Keyboard shortcut: **Ctrl + Shift + Z**

There is a separate time display window so that you can easily read the current position and other information even from a greater distance from the screen. 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.
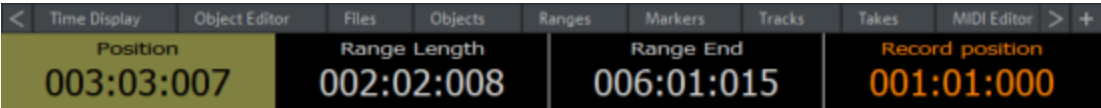
Using the context menu of a field (right mouse button) you can add more fields (**Insert field...**), remove existing fields (**Remove field**) or select the time position to be displayed in the field (**Current field...**).

Font...:	This is the font type for the display
Units of Measurement:	Unit of measurement for the display: Choose from 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). If you have multiple time display windows open, by default they all use the same unit of measurement. Activate the **Independent time format** option in the corresponding window to use a different unit of measurement there. By double-clicking on the numerical value, you can edit most values directly in the time display.

Current field...:	■ **Background color**: Lets you change the background color for the time display.
	■ **Standard color**: Here you can reset the default color for the time display.
	■ **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. By entering **+** and **-** in front of the number, you change the position relative by the entered value.
	■ **Range length**: Specify the range length here. When entering a new range length, the range start is retained. If you enter the new range length as a negative number, the range start will be moved instead, the range end will remain.
	■ **Range end**: Display of the range end.
	■ **Object**: Display start position, end position, length or file offset of the last clicked object and the summed lengths or the number of selected objects.
	■ **Cue time**: Here you can set the current play time as well as the current remaining play time of your cues (↗87).
	■ 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

- **CD position**: The current position of the playback marker is displayed in relation to existing CD track markers:
  - Position from CD start
  - Position until CD end
  - Position from the current CD track
  - Position until CD track end
  - Current CD track number
- **Mouse position**: This value represents the current mouse position and cannot be edited.
- **Mixer value**: Displays the value of a currently edited control in the mixer or the volume and pan controls in the track header, and cannot be edited.
- **Recording position**: This value displays the current recording position.
- **Recording take**: Name of the currently recorded take.
- **Recording length (take)**: This value represents the length of the currently recorded take.
- **Recording length (project)**: This value represents the length of the current recording in the project.
- **Remaining recording time**: This value indicates how long recording can continue until the hard disk is full.
- **Sync timecode input**: This value displays the timecode coming in via the SMPTE audio input.
- **Timecode track**: This value shows the timecode position of a Timecode track (↗602).
- **Total length**: This value shoes the total length pf the project.
- **External length**: This value shows the external length. This is the total length of the project minus the selected area.
- **Marker**: you can display the name of the current marker, lyric marker or CD track

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 if too many fields in the window can no longer be displayed appropriately.

Transport Console

The transport console contains the buttons for controlling playback, recording and positioning of the play cursor. (Show and hide with keyboard shortcut: **Ctrl + Shift + T**)

1 **Ranges**: Use the buttons **1** and **2** to save ranges. Click on the arrow symbol to restore the last range selections.

2 **Marker**: Opens the Marker manager (↗325).

3 **Marker 1-12**: A current playback position can be saved by clicking on one of the 12 marker buttons. Clicking this button again sets the play cursor to this position.

4 **Punch In/Out**: Sets the in/out point for a punch recording (↗102). With additionally pressed **Alt** key you can set additional Punch In/Out markers.

5 **Recording modes**: Here you can set up the 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.

- **Playback & editing independent of recording:** 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 independently of this and is then used, for example, as a "read after write check". Set the play cursor to 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, only the range between the punch in marker and punch out marker is recorded.

6 **MIDI record modes** (↗99): Select from the following MIDI record modes: normal, overdub or replace.

7 **Tempo/Signature**: Here you can change the tempo and the time signature of the arrangement. The objects can be adapted (↗354) to the new speed.

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 (↗89) options.

⑨ Arrows: The arrows below start the slowed down playback forward and backward.

⑩ Grid: You can activate the snap (↗79) here and have the grid displayed (↗81) here.

⑪ Click: Here you switch on the metronome. A right-click opens the Metronome options window (↗367) to configure the pre-count and click volume.

⑫ Loop: Switch to Loop modePlayback Range/Loop (↗84) here.

⑬ Sync: This button opens the Synchronization (↗599) options dialog.

⑭ Mon: Activates monitoring. All tracks with activated recording show the incoming signal on the peak meter. With a right click on the "Mon" button you can select the different monitoring modes. Detailed information about monitoring can be found in Monitoring Setup (↗692).

⑮ Punch: Activates the punch recording (↗102) mode.

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

PROJECT WINDOW

In the Project window the audio and MIDI data are arranged. The basic principle is the same as in almost any DAW:

- Events (in **Sequoia**: objects) are arranged on the horizontal time axis, which runs from left to right. During recording and playback, the objects are played back one after the other according to their position on the timeline.

- Vertically the project is divided into tracks, objects arranged on different tracks on top of each other are played simultaneously. The volume levels of the tracks can be adjusted separately and each track can be assigned different effects. Each track corresponds to a channel in the Mixer.

1　**Project dock**: Switch between different loaded projects here. See Docking (↗56) for more information on docking windows

2　Controls above the track heads:

　　　S M The **M** and **S** buttons above the track header of the first track can be used to switch all solo and mute states on and off together.

　　　🔒 **Lock markers**: All markers on the marker track are locked against moving.

　　　▽▾ **Marker menu**: Opens a menu containing all commands for all marker types.

　　　#▾ **Grid menu**: Opens a menu that contains all commands related to the grid.

3　**Marker and range marker track**: In these two tracks, markers and range markersMarkers (↗112) are displayed, which can be used to save and restore certain positions and ranges in the project.

4　**Track Header**

5　**Selected track**: Click on the track number or in the track name to select a track.

⑥ Playback range

⑦ Play cursor: The play cursor marks the current playback position.

⑧ Object: Objects hold audio or MIDI data, more about this in chapter Objects (↗145)

⑨ Memory locations for position and zoom level, see Saving and restoring position and zoom depth (↗110)

⑩ Time display fields: The time display fields show important time positions in the project. Their functions correspond to those of the fields in the Time display window (↗64)

> 💡 *Using the keyboard shortcuts **Alt + numeric keypad 1 ... 5** (**Edit menu** > **More...** > **Edit time display**) these fields can be edited using keyboard shortcuts.*

⑪ Zoom control/zoom menus

Mouse Modes

Different mouse modes can be selected for working in the project window of **Sequoia**. Depending on the mode, the function of mouse-clicks in the project changes. For most tasks the default Universal Mode is sufficient but there are other mouse modes specifically for editing objects, ranges and automation curves.

The menu ▾ next to the icon with the current mouse mode can be used to select the different mouse modes.
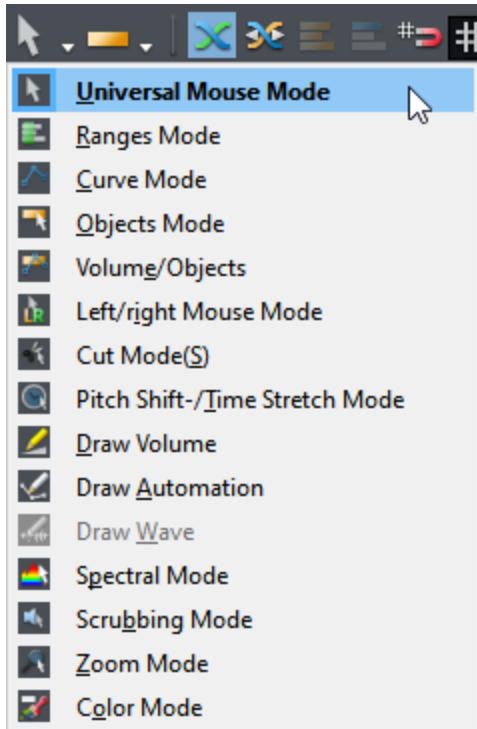
The various mouse modes can also be chosen through menu **Edit** > **Mouse Mode**. To open this selection menu anywhere in the project window, you can:

■ Click *and hold* the right mouse button and then click the left mouse button or

■ Press **Shift** + click right mouse button or

■ Press **Shift** + Context menu key (next to the right **Ctrl** key).

Universal Mode

This is the preset mouse mode for Sequoia. Depending on the click position within a track, left-clicking performs different functions. By clicking in the upper half you can mark areas and set the playback marker. In the lower half, objects can be selected and moved. Right-clicking opens a context menu.

Upper half of the track - ranges

■ **Click**: Set Play Cursor

■ **Shift + click**: Select a range between play cursor and click position

■ **Shift + click outside of an existing range**: Extend range to click position

■ **Click and drag within existing range**: Move range borders

- **Click and drag without existing range or outside of existing range**: Select a range

- **Shift + click + drag within existing range**: Move range

> ℹ️ *If you want to keep an existing range, move the play cursor by clicking above the track on the grid, not by clicking in the track.*

Lower half of track - objects

- **Click on object**: Select individual objects

- **Ctrl + click on object**: Add single objects to selection

- **Shift + click on object**: Add all objects between last selected object and clicked object to selection

- **Click between objects and drag to the right**: Span a selection frame ("object lasso"), all objects touched by it are selected

- **Shift + click between objects and drag to the right**: Span a selection frame, all objects touched by it are added to the existing selection.

- **Click between objects and drag to the left**: Span a selection frame, all automation points touched by it are selected

- **Double click on object**: Open the Object Editor (↗183).

- **Double-click between objects**: Select all objects of the track.

- **Clicking on an object + dragging**: Move an object, object selection or group of objects to the desired position.

- **Shift + click on an object + drag**: Move objects with lock of horizontal (time) position, to move to another track.

- **Ctrl + click on an object + drag**: Duplicate an object, object selection or group of objects and drag to the desired position.

  > ℹ️ ***Ctrl*** *and* ***Shift*** *can be combined.*

Object handles

Various object parameters can be set on the five object handles, which become visible on selected objects. When moving the mouse pointer over a handle, the mouse pointer changes shape to indicate the handle's function.

- **Click on middle handle + drag up/down)**: Adjust the object volume. The exact value in dB is displayed in the tooltips. This also works with multiple selected objects and object groups.

- **Click Side handles (top) + drag right/left**: Set the duration of object fade-in and fade-out. The duration can be adjusted more finely by additionally holding down **Shift** key. With additionally held **Alt** + key the fades of a multiple selection can be edited together.

- **Click Side handles (top) + drag up/down**: Adjust the curve shape of the fade-in/fade-out.

- **Click on lower handles + drag to the right/left**: Set the start and end position of the object. The position can be adjusted more finely by additionally holding down the **Shift** key.

> ⚠️ *For objects that are crossfaded with other objects (for example, the auto-crossfade that occurs when you split an object), you can also move the crossfade itself with the mouse. The handle for the start position of the rear object is located a little to the right of the crossfade.*

Automation curves

> 💡 *For detailed information on automation curves, see the section* Automation *(↗338).*

- **Double-click on automation curve**: Create and delete (double-click again) curve points. Selected points can also be deleted by pressing the **Del** key.

- **Click on curve point**: Selection of individual curve points.

- **Ctrl + Click on curve point**: Add individual curve points to the selection.

- **Shift + click on curve point**: Add all curve points between last selected curve point and clicked curve point to selection.

- **Clicking between objects and dragging to the left**: Spans a selection frame ("curve lasso") for curve points, all curve points touched by it are selected.

- **Shift + click between objects and drag to the left:** Spans a selection frame, all curve points touched by it are added to the existing selection.

- **Click on Curve Point + Drag**: Move the selected curve points.

- **Shift + click on curve point + drag**: Move the selected curve points only in the vertical direction, that is, the time positions are preserved.
  Holding the **Alt key** in addition to this will move the curve points only horizontally. (So you'll be changing their positions and not their values).

- **Click on curve section + drag**: Move the curve section. The curve points at the two ends of the section are selected and moved together.
- **Click on curve segment in range selection + drag**: If there is a range selection, you can move the entire curve vertically with all existing points within the range limits by dragging on a curve segment. This creates two new curve points on the range edges.
- **Right-click on a curve point**: Numerical editing of the value and definition of the curve shape For more information, see Editing automation curves (↗349)

Range Mode

The mouse functions of the Universal mode for the upper half of the track apply to the entire track in this mode. Thus, in this mode, only ranges and the playback marker can be manipulated. The Range mode therefore is a "safe mode" because objects or curve points cannot be moved accidentally.

Object Mode

The mouse functions of the Universal mode for the lower half of the track apply to the entire track in this mode. In this mode, you can only edit objects and their start and end position, fade-in and fade-out phase, and object volume. The range selection is retained.

Curve Mode

This mode is exclusively for editing automation curves (↗338).

- **Click on the curve**: Creates a new point; a **double-click** or the **Delete** key deletes it.
- **Click next to the curve + drag**: Selection frame for selecting multiple automation points.
- **Click on point**: Selects point.
- **Ctrl + click on point**: Adds point to selection.
- **Shift + click on a point**: Selects all points between last selected point and clicked point.
- **Click on selected point + drag**: Move all selected points.
- **Ctrl + click + drag a selected point**: Move only the clicked point if multiple points are selected.
- **Right-click on a curve point**: Numerical editing of the value and definition of the curve shape For more information, see Editing automation curves (↗349)
- **Right-click**: Context menu

Object/Curve Mode

⬜ This mode is a combination of Object mode and Curve mode. In this mode, you can move objects and edit curves.

> ℹ️ *To move the play cursor or select ranges in this three modes (Object mode, Curve mode, Object/Curve mode), click or click and drag on the grid bar above the first track. To select ranges in the track, for example to delete or copy ranges, double-click the selected range in the grid bar.*

Left/Right Mode

⬜ This mode combines range and object editing, i.e. the functions of the Universal mouse mode on the upper and lower track halves in such a way that you use the left mouse button to access the functions for ranges, and the right mouse button to access those for objects.

In return, there is no context menu in the project window in this mode.

Cut Mode

⬜ In this mode you can quickly split objects. Click on the object with the left mouse button to split it at the corresponding position. When the snap is active, it also affects the mouse pointer according to the snap settings.

> 💡 *In* Program Settings > Keyboard/Menu/Mouse > Special Keys *(↗709) you can set a key to temporarily activate the cut mode.*

Pitchshift/Timestretch Mode

⬜ In this mode, you can change pitch and playback speed of objects, edit Tempo Markers (↗354) and Warp Markers (↗366).

- **Click on middle object handle + drag**: The pitch of the object can be changed in the range +/- 6 semitones. The pitchshifting algorithm used can be selected in the object editor.

- **Click on lower right object handle + drag**: Change the length of the object by timestretching. The same time section (original length of the section in the audio file) can be lengthened or shortened by time-stretching, so the object will play faster or slower. The time-stretching algorithm used can be selected in the object editor.

- **Clicking into an object (upper half)** sets a **Warp Marker** into the object.

- **Click + drag a Warp Marker**: Move the Warp marker (upper half) or move the anchor point of the Warp Marker (lower half).

- **Shift + click on grid bar**: Create tempo markers. By dragging vertically, you can set the tempo right away.

- **Shift + click on tempo maker + drag vertically**: Set BPM of tempo marker.
- **Alt + click on grid bar**: Create grid position markers.
- **Right-click**: Context menu

Draw Volume Mode/Draw Automation Mode

In these two modes you can draw automation curves (↗338) with the left mouse button.

- **Click**: Create a curve point on an existing curve.
- **Click + drag**: Freehand drawing function for automation curves.
- **Shift + Click + Drag**: Create a straight line starting from the previous curve point.
- **Right-click on a curve point**: Numerical editing of the value and definition of the curve shape For more information, see Editing automation curves (↗349)
- **Right-click**: Context menu

For the most part, the modes have the same function: They work identically when drawing in Automation lanes. But when the automation curves are displayed in the track (↗340), the Draw automation mode edits the active curve and, if **Show all curves (selectable)** is active, the clicked curve. However, in Draw volume mode, the volume curve is always edited, regardless of which curve is displayed.

Draw Wave Mode

If you are destructively editing audio files (↗569), you can draw directly in the waveform of a file in this mouse mode. The display of the waveform changes to a suitable zoom level. This is particularly suitable for manually removing very short interference pulses.

Spectral Mode

With the Spectral mode you can remove noise from an object directly in the project window.

To be able to work particularly effectively with the spectral mode, set the option Spectral display (↗721) for the project display.

The audio material is represented in the spectral display by a spectrogram that shows the frequency components over time. The volume of frequencies is visualized via a color code or via its brightness.

Audible distortion that is louder than the desired signal are usually limited to a certain frequency spectrum. They are highlighted with colors in the spectrogram. A continuous sound is displayed by a pattern consisting of horizontal lines, which correspond to the sound components (overtones) of the sound. An impulse-like noise becomes recognizable as a vertical spike.

Select the noise by clicking in the spectrum and dragging a rectangle around it. With the handles you can still adjust the selection rectangle afterwards.
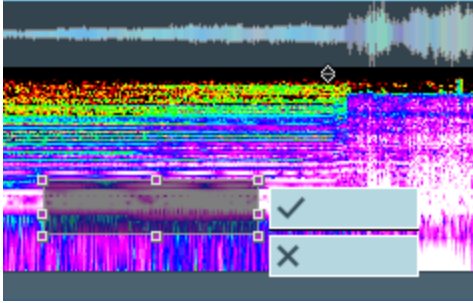
To avoid an audible gap, extracted components of the original frequency spectrum are interpolated from the audio material surrounding the noise. Confirm the correction by clicking the button with the checkmark.

> 💡 *You can also edit the left and right channels of the signal separately by activating the option **Stereo in 2 waveforms** in the **Project display options** (keyboard shortcut: **Shift + Tab**) > **Object** and then dragging the selection rectangle accordingly only over the upper or lower part of the spectrogram.*

Scrubbing Mode (Pre-Listening)

 This mouse mode enables you to monitor and control the playback speed (↗90). In this case, the project can be played back in a forward or reverse scrubbing direction.

- **Left click**: Pre-listen with play speed control.
- **Right-click**: Context menu

Zoom Mode

 Use the right mouse button to zoom (↗104) out of the project, and the left mouse button to zoom in.

- **Left click**: Zoom in
- **Right click**: Zoom out
- **Click + drag** : Zoom in selected area (horizontal)
- **Shift + click + drag**: Zoom in selected area (vertical)

  > ℹ️ *There's an option in **Program Settings** > **Keyboard/Menu/Mouse** > **Mouse** by which you can zoom horizontally and vertically via selection frame even without the **Shift** key.*

Color Mode

Color mode enables you to color objects or entire tracks. After selecting the color mode, open the mouse modes menu again, in the lower part of the menu you can now select a color. Further down in the menu there are other color palettes to choose from. Click on a color palette to replace the colors in the menu with those of the selected color palette.

- **Click on object:** The object and the waveform are colored.

- **Click between objects + drag**: Dragging a selection rectangle and coloring all touched objects.

- **Shift + click on object/ right click on object**: Colorize the waveform.

- **Ctrl + click on object**: Colorize the object background.

  > These modifiers also work with the selection rectangle.

- **Click on track header**: The track and all objects contained in it are colored. The track color is also used as the background color in the track name fields in the Track Editor and Mixer, and on the right edge of the track header. There you also have the option to open the color palette for each track and set a new color for the track. If multiple tracks (↗132) were selected beforehand, the color selection will apply to all of these.

If you do not like the default color selection, you can change individual colors. At the very bottom of the menu, choose **Select your own color...** and open the **Colors** dialog. There you can change any of the 16 available colors. With **Create new color palette...** you can save the colors together in a new palette, which can then also be selected in the menu.

Temporarily Switching to Other Mouse Modes

By pressing and holding certain keys, you can temporarily switch to another mouse mode. This way you can combine several modes without much clicking.

. (Dot): Temporary change to **Object mode**. As long as you keep the button pressed, you can select objects and edit them with the handles. This button is useful in the Draw Automation mode, in the Range mode and in the Cut mode, in which you cannot otherwise access the objects that quickly.

- (hyphen): Temporary change to **curve mode**. As long as you keep the key pressed, you can edit automation curves in curve mode. For example, you can work in Universal mode and use the slightly different (↗348) editing options of Curve mode at the touch of a button.

-ᦰ- *You can also specify a second key for switching to Draw automation mode and then be even more flexible.*

Insert/Number Pad 0: This key is assigned twice, so it works independently of the status of the NumLock key. It allows you to temporarily activate scrub mode and thus move around in the arrangement by clicking.

Z key: Temporary change to **Zoom mode**. Then you can zoom in by left-clicking and zoom out by right-clicking.

Not defined by default, but available are also a toggle switch for scrub mode (i.e. scrub mode remains active until the key is pressed again) as well as keys for the **Draw automation mode** and the **Cut mode**.

Snap

The snap makes it easier to work with the mouse when moving objects and selecting ranges by snapping the moved markers or objects to specific positions, so you don't have to position the mouse exactly and still get exact positions. Beat divisions, markers, object edges, frames or the set quantization value can serve as snap positions.

Snapping means that if a marker, an range border or an object is moved close enough to a possible snap-in position, it automatically jumps to exactly this position, it "snaps" into this position.

ℹ️ *"Close enough" refers to the screen display, i.e. the distance of the object from the desired position in screen pixels. If you have zoomed in to just a few bars, but have the snap set rather coarse (e.g. 1/2 notes), it is possible that objects can be moved to positions between snap positions.*

Using the **Snap and Quantize menu** on the top toolbar, you can define the most important snap settings and turn the snap on and off.

ℹ️ *The complete **snap and grid settings** can be reached via the dialog **Project options - General** (↗643) (keyboard shortcut **I**).*

To turn the snap on and off, click the button with the magnet or select Menu **View** > **Grid** > **Snap Active** (keyboard shortcut: **Ctrl + '**).

In the **snap menu** next to it you can select the type of the snap:

- **Snap to objects**: With snap to objects the range selection, the play cursor and other markers snap to the edges of all objects in the project when they are moved. In addition, either object start or object end snap to all markers and to the edges of the other objects when moving objects. Whether the start or end position of the object snaps depends on whether the object is clicked in the front or rear half to move it. The shape of the mouse pointer indicates which edge is being snapped to:

  - ⬌ snaps to the left edge of the object
  - ⬌ snaps to the right edge of the object

- **Snap to beats/bars**: This option activates a snap grid that uses beat subdivisions as a basis. The beat snap ensures that when selecting ranges and moving objects and object edges, the play cursor and other markers, the time positions "snap" to defined values determined by the snap width. So, for example, if you have set a beat snap with a one bar snap width, you can click on the grid bar only approximately near the beginning of a bar to place the play cursor exactly at the beginning of the bar.
  The beat snap also determines the step size when moving play cursor and range boundaries with the keyboard (**arrow keys** and **shift** + **arrow keys**).

  > ⚠ *If you have set a very coarse snap width (e.g. 2 bars), but zoomed in strongly, it may seem as if objects cannot be moved at all, because possible snap positions are very far away from the starting position on the screen or outside the visible section.*

- **Snap to beats/bars (relative)**: This option also activates a snap grid that uses beat subdivisions as a basis. Relative means that for example the position of an object can only be changed stepwise in the set snap width, as with the normal beat snap, but the distance to the respective snap position remains the same. It therefore does not snaps on the beats, but in each case at the same distance before or behind them.

- **Snap to frames**: This sets the snap width to the length of one SMPTE frame, according to the project frame rate set at **Project Settings** > **Synchronization**. This setting is important for video dubbing, the time positions then correspond exactly to the frames in video files.

- **Frame raster (relative)**: As with the relative beat snap, the frame snap here preserves the relative distance to the respective snap point.

- With **Snap to range** the grid is similar to the beat grid, except that the snap width is defined by selecting a range before activating the grid. The length of this range then corresponds to one bar, and the snap width can be divided according to the **Q** value as with the beat snap.

In the lower part of the menu there is a selection of settings for the snap width for the beat snap:

- **Snap to quantization**: Select the snap width in the **Quantization menu** (see below) and activate this setting to use this value also for the beat snap.

- **Every 4 bars, 2 bars, Full bar**: To keep the menu manageable, all smaller values for the snap width are missing here. Set the values in the quantization menu and select **Snap to quantization**. Only the very large values for the snap width (which are also not useful as quantization values) can be set directly in the menu. If you absolutely want to set a value other than the quantization value for the snap width, you can do so in the Project Options under **Snap/Grid**. To do this, go to the menu **File** > **Project properties** > **Project options...**.

- **Beat**: The denominator of the time signature is used as the snap value, i.e. ¼ in 4/4 time and ⅛ in 3/8 time. The snap thus follows the time signature and takes time signature changes into account.

In the **Quantization menu** you set the quantization value **Q**, which is used for the audio- (↗612) and MIDI- (↗449)quantization and also serves as a setting for the snap width in the snap setting **Snap to quantization**. By default, the Q values of audio and MIDI quantization (↗698) are linked to each other. Right-clicking on the field opens the MIDI quantization settings.

Grid

At the top of the project window there is the Grid bar, where the project time is displayed in different units of measurement.

The grid are vertical guide lines that show this time information as an orientation guide over the tracks.
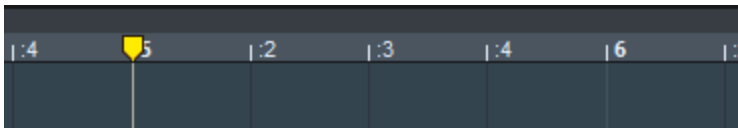
All the grid-related menu commands described below are located in the menu **View** > **Grid**. This submenu can also be accessed by right-clicking on the Grid bar:

- To show or hide the grid, select **Show grid** or press **]**.

- Various line types can be selected at **Grid lines**.

- In the lower section of the Grid bar menu or under **Grid** > **Units of Measurement** you can choose between different **units of measurement**: Samples, Hours:Minutes:Seconds, SMPTE (Hours:Minutes:Seconds:Frames), SMPTE/Milliseconds (Hours:Minutes:Seconds:Milliseconds), Beats/Bars and others. Depending on the unit of measurement, the grid spacing dynamically adapts to the zoom level. For example, with the unit of measurement samples, a grid subdivision is drawn every 10000 samples at a low zoom level, and every 1000 at a high zoom level. With the unit of measurement bars, there are main subdivisions (bars) and secondary subdivisions (beats) on the grid, if they can be meaningfully displayed at the selected zoom level.

- **Show 2nd grid line**: Above the grid line a second grid line can be displayed, whose unit of measurement can be set independently from the first grid line so that you can display two units of measurement combined, e.g. bars/beats and time position.
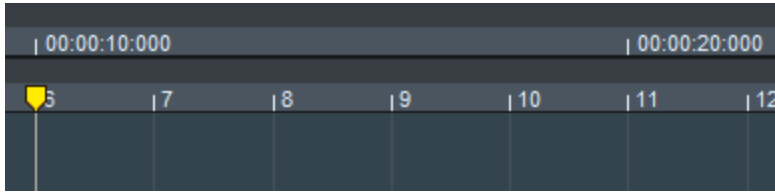
- **Exchange grids** allows you to swap the units of measurement of the two grid bars and thus display the grid based on the setting of the other grid bar.

Displaying Project Windows

Configure Project Display

The visual appearance of the project window and the objects it contains can be configured very flexibly. You can switch between two project display modes using the **Tab** key. The default setting switches between the normal view and a simplified view without track heads and with objects without waveform representation.

You can access the display options via the menu **View** > **VIP display mode** > **View Options...** or with the keyboard shortcut **Shift** + **Tab**.

> ⓘ *For detailed information on project display options, see* **System Options** >**Project Display**.

Arrange Project Windows

By default, each new project window is opened docked in the Project docker. Click on the respective project tab to activate the desired project.

> 💡 *To work with dockers and the associated keyboard shortcuts, read the section* Docking (↗56)

To display all open project windows side by side, press the **Enter** key (menu **View** > **Tile**).

To restore the previous view press **Shift** + **Enter** (Menu **View** > **Untile**).

If windows become unreachable because in a previous session you were working with multiple monitors, but now only one monitor is connected, you can reach these windows again with the command Menu **View** > **Windows** > **All windows to main screen**.

> ℹ️ *If additionally the **Shift** button is pressed when the command is selected, the saved window positions of windows that are not currently open are also adjusted so that they are opened on the main screen.*

To close a project, click **x** on the respective project tab or choose Menu **File** > **Close Project**.

Recalculate Graphic

In rare cases, inaccuracies in the display of objects may occur after offline editing. Use menu **View** > **Rebuild graphic data** to update the graphical waveform display for the entire project.

Project Playback

> ℹ️ *All the commands for playing the project can also be found in the menu **Playback** to be able to assign keyboard shortcuts to them or change the shortcuts if necessary.*

Simple Play and Stop

By simply clicking on the grid bar or the marker tracks above the first track, you set the play cursor to the desired position in the project. You can also click anywhere in the project in range mouse mode or in the upper half of a track in universal mouse mode to move the play cursor.

> ℹ️ *When clicking on the grid bar or in the project, an active* snap *(↗79) is respected; when clicking in the marker tracks, the play cursor can be placed at any position.*
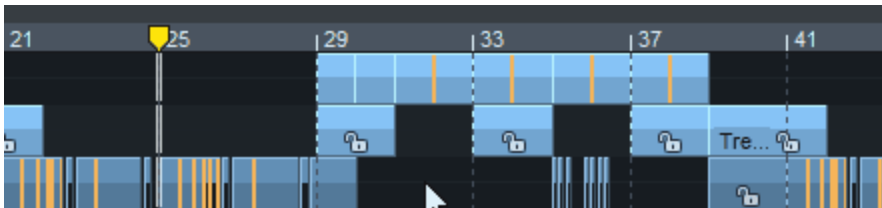
The **Space bar** or the Play button on the Transport control (↗67) starts playback at the current play cursor position, the play cursor moves through the arrangement showing the currently played position.

With **Ctrl + Space** only the **selected object(s)** will be played. To do so, the play cursor is moved to the beginning of the first selected object.

Pressing the **space bar** again or the Stop button on the Transport control stops playback. The play cursor jumps back to the start position.

To stop at the current position, press the **,** key (above the space bar or on the numeric keypad) or the **Pause** key, or click the Play button in the Transport control again.

ⓘ *This default behavior can be changed in the **Program Settings** > **Playback** (↗702) by selecting **Stop at position** there.*

With the command **Restart play** the play cursor jumps back to the starting position during playback and plays again from the beginning.

Playback Range/Loop

By clicking and dragging on the grid bar, you define a playback range. The play cursor is then located at the beginning of the playback range.
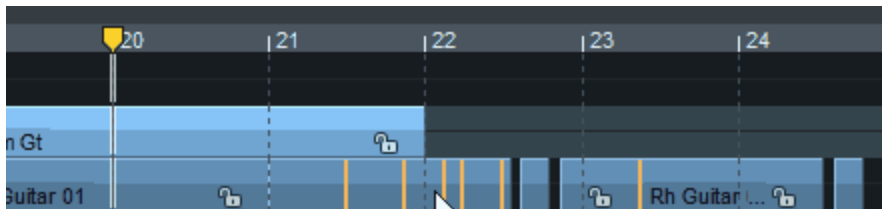
Without Loop, playback stops at the end of the playback range.

Moving the play cursor deactivates the playback range, but it can be restored by double-clicking in the inactive range.

💡 *With **Shift + Backspace** you can restore the last 5 ranges, more about this in the* Ranges (↗118) *section.*

The button **Loop** on the transport control (also in the menu **Play/Rec** > **Playback mode** and as keyboard shortcut **Shift + L**) activates the **Loop mode**.

The playback range is colored blue. When the play cursor reaches the end of a playback range while loop mode is active, playback repeats at the beginning of the range and continues until playback is stopped.

> ℹ️ *If no range is defined and **Loop** is active, playback will resume at the start of the project when the end of the project is reached.*
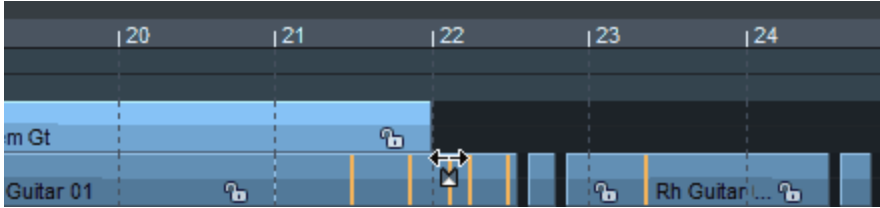
*When **Loop** is active, the playback range is preserved when the play cursor is moved.*

You can set a playback range as a loop and place the play cursor at a position before the loop range and "play" into the loop.

If Loop mode is not active, you can achieve this behavior by **Play In Range/Loop** (keyboard shortcut: **Shift + P**): With this command you start the playback and activate the loop mode simultaneously.

If (the other way around) you want to clear the loop range even though the loop mode is active, move the play cursor by **double-clicking** on the Grid Bar.

> ℹ️ *If you start playback behind the loop range, **Loop** will be disabled during playback to indicate that playback will never occur in the loop range.*

Playback ranges can also be re-selected during playback, with the play cursor immediately jumping to the new range start. However, the range boundaries can be moved with the mouse without affecting the playback.

> 💡 *Playback ranges are also always created when selecting ranges in the project window. For more explanations on editing, saving, restoring, etc. of ranges, see the* Ranges *(↗118) section.*

Other Playback Functions in the Play/Rec Menu

Play with Pre-Load

(Keyboard shortcut: **Shift + Space bar**)

All playback buffers are preloaded before playback starts. This makes no difference during normal playback, but when Sequoia is controlled by MIDI sync, this ensures that playback starts without delay after the arrival of a corresponding MC/MTC command.

Play Cut

Assuming that a current range selection is to be the basis of an edit, you can use these commands to play back the project at specific positions around a playback range.

Playback is performed with a defined lead pre-roll and post-roll time: when playing back to a point, it starts before the point by the amount of the pre-roll time and ends there; when playing back from a point, it starts at the point and runs for the duration of the post-roll time. Pre-set pre-roll time and post-roll-up time are 2 seconds. You can change the time at **Program Settings** > **Playback** (↗702).

> ⓘ The commands are also used in the Crossfade Editor (↗177) to play parts of a crossfade, for which a separate pre/post-roll time can be specified in the Crossfade Editor preferences (↗181).

Command	Range selection	Keyboard shortcut
Play to Cut Start (In Point)	Play until the beginning of the range	**F5**
Play from Cut Start (In Point)	Play from the beginning of the range	**F6**
Play to Cut end (Out Point)	Play to the end of the range	**F7**
Play from Cut End (Out Point)	Play from the end of the range	**F8**
Play over cut	Play until the beginning of the range and after the end of the range. This allows you to simulate cuts: Playback occurs as if the selected range had been deleted.	**F4**
Play into cut	Playback of the range with pre-roll and post-roll time.	

> ⓘ The menu commands **Fade-out/Fade-in/Play upper audio/Play lower audio** are specifically intended for playback in the Crossfade Editor (↗177) and are only available there.

Other Playback Modes in the Play/Rec Menu

Cue Mode

The Cue Mode is a special live playback mode that can be used in broadcasting and theater for starting feeds such as sound bytes or sound effects. Objects (cues) contained in the first track (cue track) can be played back in sequence with it. Simultaneously, you can record to other tracks.
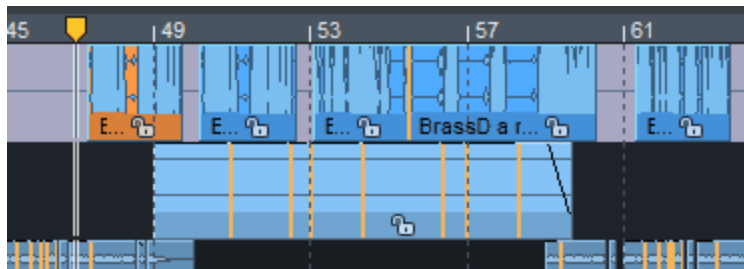
The cue track is highlighted with a special background color.

The definition of the individual cues can be done via objects or track indices:

- **Objects**: When playback is started by pressing the spacebar, a playback range is selected above the next object after the play cursor and played back. After stopping playback, with the spacebar or automatically at the end of the object, a range above the next object is selected, which is then played at the next playback start, and so on.

  > ℹ️ *Directly bordering or overlapping objects are treated as one object. In multi-track projects, only the objects on the first track are considered cues.*

- **Track Marker**: CD track indices in the project are considered cues here. To do this, select Program Settings > Playback (↗702) at **Cue Mode** > **At CD Track Indices**. The playback ranges are always set between a CD track index and the next one.

> ℹ️ **Note**: *The Cue mode only works when **Loop** is not active.*
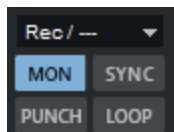
 In the Cue mode, the Recording Mode also changes to **Playback & editing independent of recording** (↗95). You can recognize this by the display in the transport console.

In this recording mode, the playback of the project is controlled independently of the recording. For example, you can record a narration on one track and select the Cue track as the recording source for another track and then play the cues at the desired points one after the other.

In the **Program preferences** > **Playback** (↗702) you can use **Solo switch for first track** to specify that the cue track is automatically soloed in cue mode.

Via the **Pre-roll for Cue Mode** you can enter a delay with which the Cues are played after the playback start. The delay can be recognized in the project by the fact that the start of the playback range above the objects is shifted forward by this amount of time, so playback starts accordingly before the object start.

Change Play Direction

Activate this option in the menu if you want to play the project backwards. If you change the option while playback is in progress, it will play forward again with the next play back.

Continuous Playback While Editing

The default behavior of Sequoia is that when you select a range, the play cursor is moved to the beginning of that range. If you want to perform editing operations that require range selection during playback, this behavior is disruptive because it interrupts playback and resumes it elsewhere in the project. The **Continuous playback while editing** option prevents the play cursor from being moved to the beginning of the range selection while playback is in progress. This allows you to edit your project even while it is playing.

⚠️ *However, the play cursor is moved when you click on the grid bar or select an range by clicking + dragging.*

Playback at Different Speeds

With **Play with speed 1 - 4** : you can start playback at four different speeds: 1/4x, 1/2x, 1x and 2x original speed. If you select one of the four menu options while holding down the Shift key, you can individually adjust the speed for the selected option.

Move Play Cursor

In the menu **Play/Rec** > **Move play cursor** there are further commands with keyboard shortcuts to control the play cursor with the keyboard:

- **To Beginning/To End**: Use the **Home** and **End** keys to move the play cursor to the beginning or end of the project. There are other menu commands to move the play cursor to the beginning or end of the selected range or to the beginning of the visible section.

- **Left/Right movement in Page/Scroll mode**: With these commands you move the play cursor with the cursor keys (right arrow /left arrow). Movement is possible in Page mode or Scroll mode.
  In **Page Mode** (keyboard shortcut **Right Arrow**/**Left Arrow**) the play cursor is moved in the visible section until it reaches the edge of the section, then the section is moved by its entire length so that the play cursor is

displayed at the opposite edge of the next or previous section. So the play cursor moves through the project page by page.

In **Scroll mode** (keyboard shortcut **Alt** + **Right arrow**/**Alt** + **Left arrow**), the play cursor is moved to the center of the section and remains there while the project is moved under it.

> ℹ **Note:** *When a range is selected, the **Left Arrow**/ **Right Arrow** keys control the start of the range.*

- **Object edge left/right**: These commands move the play cursor to the previous/next object edge of the selected track.

  Keyboard shortcut: **Ctrl** + **Q** / **Ctrl** + **W**

- **Marker left/right**: These commands move the play cursor to the previous/next marker in the project.

  > ℹ **Note:** *These commands also include audio markers in the object if they are displayed in the project. (Menu **File** > **Program settings** > **Project display...**). Also included are other markers such as CD track indices or tempo markers.*

  Keyboard shortcut: **Alt** + **Q** /**Alt** + **W**

  F2 / **F3**

  These commands are also available on the lower toolbar.

- **Section to play cursor / Last stop position**: If the option **Stop at current position** at menu **File** > **Program settings** > **Playback parameters** is not active, the play cursor jumps back to the starting position after stop. With this command you can get the play cursor back to the last stop position.

  > 💡 *To keep the play cursor at the same stop position, stop the playback with the **,** key.*

  Keyboard shortcut: **Ctrl** + **Alt** + **,**

- **Recall last position(s)**: This command allows you to set the play cursor to the last five stop positions.
  Keyboard shortcut: **Backspace**

- **to maximum peak for selected objects**: Sets the position of the play cursor to the position in the project where the highest peak level occurs of all selected objects.

Varipitch / Varispeed

The Varispeed function allows you to set the playback speed of the project in the range of -/+ 200%. Negative values mean that the project will be played backwards. So you can use Sequoia like a tape machine with adjustable

tape speed.

To use Varispeed, open the **Playback, Scrub and Varispeed Settings** dialog via menu **File** > **Program Preferences** > **Varispeed/Scrub Settings...** or via right-click on the Scrub Control knob of the transport console.
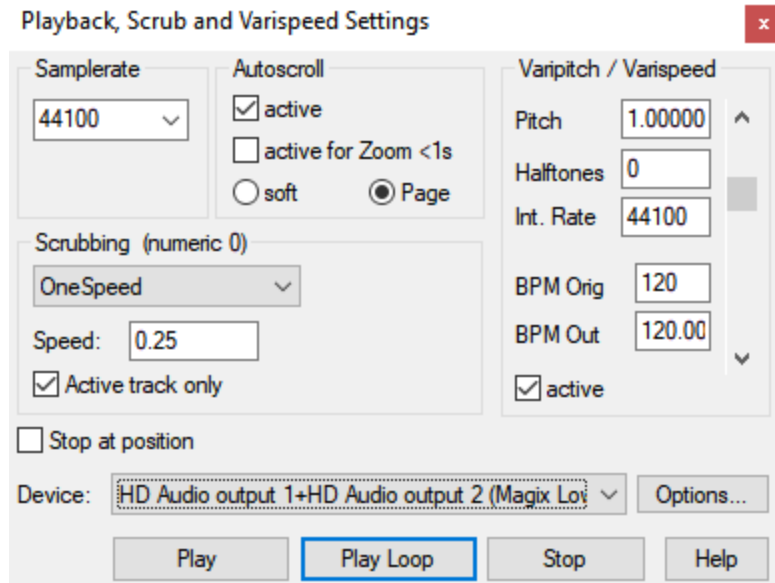
> ⓘ *In addition to the Varispeed function, this dialog also provides access to important other* playback settings *(↗702)*
> *such as scrub control, autoscroll or the standard output device.*

Activate Varipitch with the checkbox **Active**. Now you can set the playback speed with the fader on the right. For an exact setting you can enter the desired values in one of the fields next to it, the other fields will be adjusted accordingly: The speed can be defined as a factor **Pitch**, but also as a pitch change in **semitones** or as a tempo specification, whereby the pitch factor is determined from the specification of the original tempo **BPM Orig** and the target tempo **BPM Out**. **Internal Rate** is the resulting sample rate used during playback, based on the original sample rate and the pitch factor.

> 🔆 *This function can also be used during recording!*

Examples of use:

- Play passages of a recording more slowly to transcribe the notes of a complicated solo.
- An old studio trick: record a solo along with a slower speed playback in a lower key. If you then play the project at the original speed, you will get a solo in correct pitch and "inhuman" speed!

Scrubbing

Scrubbing is the process of "going through" the audio material at different speeds to find a specific location by ear. Here, Sequoia behaves like the "edit" mode of a tape machine: The motors are switched off, but the tape is still

present at the tape head. By turning the tape reels by hand, the tape is slowly moved along the sound head to precisely approach a specific location.

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

In Sequoia scrubbing can be done by dragging the mouse in a special **Scrubbing mouse mode**.

Activate the Scrubbing mode with the **Ins/ 0** key on the numeric keypad or the key combination **Alt + Shift + down arrow** or by selecting the corresponding mouse mode (↗70) on the upper toolbar 🔊 .

If you click and drag into the project starting from the play cursor, you will hear the underlying audio material.

To scrub with the keyboard, use the keyboard shortcuts **Alt + Shift + Left Arrow** and **Alt + Shift + Right Arrow**.

> ℹ️ *You can also use the scrub control of the Transport console for scrubbing. The more you turn the knob to the right or left, the faster the playback will be in the respective direction. (However, this is actually only useful in combination with a hardware controller).*

Scrubbing Modes

Different scrubbing modes can be set in the **System options** > **Playback** (keyboard shortcut **P**):

- In the default Scrub mode **OneSpeed** playback is at original speed. If you additionally press the **Shift** key, playback is at half speed. With the **Ctrl** key, the scrubbing speed set in the **Speed** field (default: 1/4) is used.

- **Two speed**: Two speeds are provided for scrubbing. Depending on the distance of the play cursor to the mouse position, the object is either played fast or slowly, whereby a speed of 0.25 is preset for slow scrubbing. Fast scrubbing is set to 1.0, i.e. original speed. Change the slow playback value in the **Speed** field.

- **Shuttle**: The distance between the play cursor and the mouse position is also used for tempo control, but the tempo control is continuous. So the play cursor follows the mouse movement quickly at first, and then the closer it approaches the mouse cursor position, the slower it moves.

- **Absolute**: In this mode, the absolute position of the mouse in the window is used to control the tempo.

   Mouse pointer on the left border   = double speed backwards

   Mouse pointer in the middle   = no playback

   Mouse pointer on the right border = double speed forwards

The **Scrubbing for active track only** option restricts scrubbing playback to the selected track.

Recording

For a quick start, Sequoia is already set up " from factory " so that a recording can be made immediately:

① Arm the tracks to be recorded to for recording by activating the red record button.

② By default, monitoring is active, you should now already be able to hear the input signal through the track and the track peak meter should show the level of the input signal.

> ℹ *Note: You control the input level with the mixer application of the audio hardware or directly at the audio hardware. The track volume control is used to set the playback or monitoring volume and not the input level.*

③ If you want to play at a certain tempo by metronome, set the tempo in the transport console and activate the button **CLICK**.

④ Start recording with the keyboard shortcut **R** or click on the Record button of the transport console.

⑤ The recording will start and you can immediately see the recorded audio on the track. Clicking Record or Stop again ends the recording. In the subsequent query, you can decide whether to keep or discard the recording.

The play cursor is returned to its starting position. You can now use the recording or record another take at the same place.
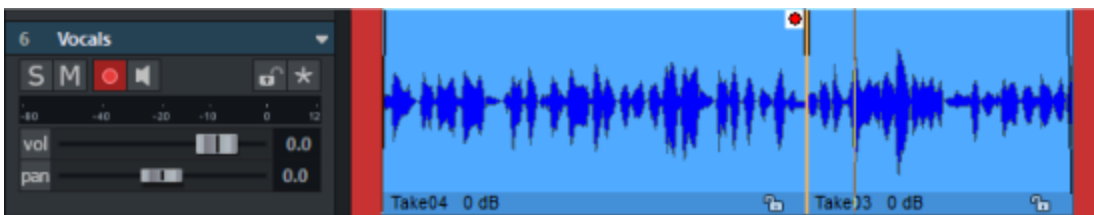
The existing object is overwritten by the recording, but the previously recorded material is still available in the project. After the recordings, there is an object in the track that references all the recording takes. You can manage the different take lanes via the Take Lanes (↗138).
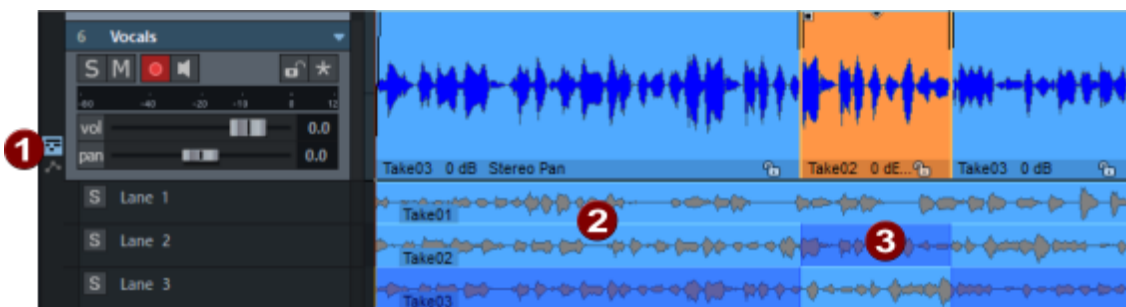
① Select the object on the lane and open the take lanes with the icon on the lane header. All recording takes are displayed under the object.

② Click a take to use that take for the object.

③ Click and drag over a take to use a section from this take to combine the different recording takes.

For multitrack recordings, the takes on all tracks will be exchanged accordingly if you leave the recorded objects grouped after recording as preset.

You can now use the object handles (↗147) or the Crossfade editor (↗170) to precisely set the transitions between the takes and then edit the project further using the editing functions for ranges (↗123) or objects (↗158).

To tailor the recording function to your needs, you can customize the following:

- **Input routing**: There are controls in the mixer, track header and track editor for assigning recording devices to tracks. In the dialogs Track Settings (↗141), Mixer Settings (↗646) and in the Routing Manager (↗332) there are methods to assign several tracks to several recording devices at once.

- For **monitoring**, i.e. the playing of the input signal during recording, there are a number of different monitoring modes for the audio engine (↗692), which can be used to configure monitoring for the playback of tracks with effects.

- **Mono/Stereo**: Recordings can be made in mono or stereo files. This means you can record stereo signals into either two mono tracks or one stereo track. You can use the routing functions in the Mixer Settings dialog (↗647) to adjust several tracks or the entire project accordingly and make the necessary pan settings.

- **Files**: You set the default folder for projects in Program Preferences > Program > General (↗706). The recorded audio files are saved next to the project file in the project folder in the form `project name_ track number. wav`. As long as you do your production entirely within Sequoia, you can leave these names on the default defaults. If you want to use the recorded raw material elsewhere or pass it on, you have the option of naming these files according to specific naming schemes (↗143) for better identification.

- In the **Record Options dialog** (↗95) you can specify a different file format for the files and set other options for recording. For even more settings for recording, see Program Preferences > System Options > Recording (↗699).

Controlling the Recording Levels

To check the recording level, use the **peak meter** of the track. It is very important that the recorded signal does not exceed the maximum level of 0dB at any time. Such digital "clippings" are audible as very unpleasant noise in the recording.

> ⓘ *An exception to this is if your recording device supports real* 32bit float recording. *Then no more clipping can occur when recording in 32bit (float), as levels above 0dB are then also stored correctly.*
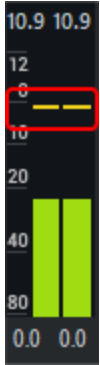
 The maximum level can be seen as a marker in the peak meter in the track editor and mixer. To reset the maximum level, left click on the peak meter.

What the peak meters display depends on whether monitoring is active:

If monitoring is active in a track, the peak meters show the level of the input signal; if not, they show the level of the audio signal being played back on the track. The measurement takes place at the end of the track effect chain, so the level is calculated including the influence of track volume and effects.

ℹ️ *With the preset* Monitoring switching behavior *(⤴695)* **Tape machine monitoring***, monitoring is automatically active for all tracks that are armed for recording when playback is stopped. With manual monitoring, monitoring can be switched on and off independently of the playback status.*

The level display after the influence of track volume and any effects ensures that the level displays are consistent for recording and playback.

When recording, however, it is sometimes helpful to check the level of the signal directly at the input of the audio hardware, as not only the recording itself but also the input converters can be overloaded if the input signal is too high.

To do this, click on the peak meter on the right and select **Peak meter for audio input** from the menu.
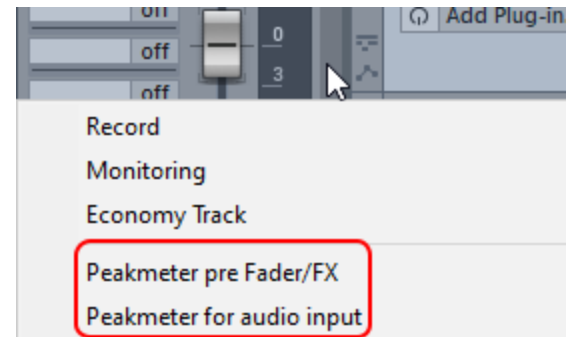
This option is activated automatically as soon as you arm a track for recording. If you do not want this behavior, deactivate the option **Peak meter on audio input when recording** at the bottom of the monitoring switching behavior in the **Program preferences** > **Audio Setup**.

With the option **Peakmeter Pre-Fader/FX** the measurement takes place after a possible level compensation by the track **Gain** controller.

Recording Modes

The recording can be done in two different transport modes.

You can change the mode in this menu in the Transport Console, in the **Recording Options dialog**, or with the options in the **Playback menu** > **Recording Mode/Punch In**.

Standard (Playback during Recording)

In this mode, you record additional tracks to existing audio tracks that are being played back. To do this, arm the tracks that are to be additionally recorded. In these tracks, monitoring is activated by default and you hear the input signal you want to record for these tracks. The peak meters of the tracks also react to the input signal.

Recording starts at the playback marker position as soon as you click the **Recording button**.

If a range is selected behind the play cursor on the grid bar and loop mode is also active, recording will not start until the play cursor reaches the start of the range. Before that, Sequoia signals by a flashing record button that a recording will be made. In the selected range, recording is done in loop mode.

Playback & Editing independent of Recording

In this mode, recording takes place completely independently of playback.

As soon as you start recording, the recording starts at the play cursor position. When monitoring is active, the input signal of the recorded tracks is played back. The captured objects are displayed in the project.

The playback marker does not move with the recording. Instead, you can play back the project at any point, playing back the audio material present on the tracks, including the material you just recorded, rather than the input signal (monitoring). Editing operations with areas or objects are also possible independent of the recording. You can use it to edit a project while it is still being recorded.

> **i** *The playback option* Continuous playback while editing *(↗88) is helpful for this.*

The recording continues regardless of this until you press the record button again.

Recording Options

With **right-click on the recording button** of the transport console or with menu **Playback** > **Recording options...** (keyboard shortcut:**Shift + R**) you open the dialog **Recording options**.

All recording settings can be changed here. The dialog can be closed and also reopened during a recording. The dialog also allows you to start, pause and stop recording and set markers during recording.For this task there is optional (↗699) also a reduced version of the dialog that contains only the controls for this.
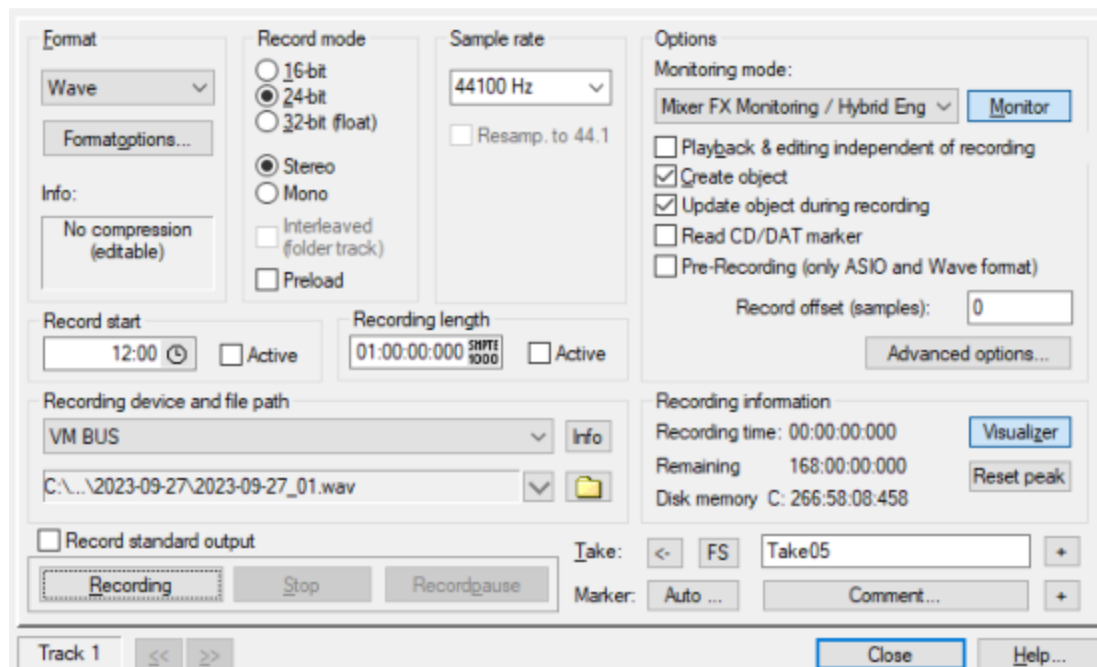
Files and Inputs

On the right side of the dialog you specify the format and paths of the files to be recorded to and the recording devices used.

- **Format**: Here you can set the respective recording file format (Wave, MP3, MPG, 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 can be edited directly in Sequoia and which data compression is used.

- **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. See more at Interleaved recording (↗101).

  > ℹ️ *If your recording device supports true 32bit float recording, clipping can no longer occur when recording in **32bit (float)**, as levels above 0dB are also saved correctly in float format.*

  If **Preload** is active, when you start recording with the record button, all playback buffers are loaded first and a confirmation dialog is displayed. After confirming with **OK** the recording is started without delay.

- **Sample rate**: This option allows you to select the sample rate of the recording. By default, the project sample rate is used. However, you can also record at a different sample rate, in which case a object resampling (↗660) will be applied to the recorded objects after recording.
  Resamp. to 44.1: If you have set the recording sample rate to a value other than 44100 Hz and activate this option, Sequoia will resample to 44.1 kHz during recording.

ⓘ *This is an deprecated option. Modern audio hardware allows you to set all common sample rates via its driver settings and we recommend working with a uniform sample rate for driver, project and recording.*

- **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. With **Recording length** you determine the duration of the recording. This gives you the option to perform scheduled recordings even in your absence. If no length has been selected, recording continues until the hard disk is full and then stops automatically.

- **Recording device**: Display and selection of the recording device for the track. With **Info** you open the driver settings of the recording device.

- **File path**: Specify the path to the files where the data should be recorded. The folder button opens a query for selecting the path and file name. In the menu below the down arrow there are naming schemes to derive the file name from various track properties (track name, track number...). For more information, see Dialog Track Options (↗141).

  ⓘ *Use the **<</>>** buttons at the very bottom to switch between tracks. However, stereo/mono recording, recording device, and file name are the only options in this dialog, which may differ from track to track. So you can't record wave file on one track and MP3 files on another.*

- **Record standard output**: Enable this option to record the standard output of the operating system, for example the playback of DVD sound, the browser or games. 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 accidentally remaining active, it must be reactivated for each such recording operation.

  ⓘ *Note: Some audio devices integrated in the mainboard (onboard sound) also offer the entire audio output as a recording device, often under the name "Stereomix" 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 inside the audio device, so a double conversion - from digital to analog and back - takes place.*

  ⓘ *Tip: Since everything that is played back in the operating system is actually recorded, you should disable system sounds, website notifications, instant messengers and the like.*

Options

In the **Options** section of the dialog you will find the most important options for recording. Further options can be found in the Program Preferences > System Options > Recording (↗699), which you can also open directly via this

dialog with the button **Advanced Options...**.

- **Monitoring mode**: For detailed information on monitoring, see Program Preferences > System options > Monitoring setup (↗692).

- **Monitor**: With this button you activate the monitoring. All tracks with activated recording show the incoming signal on the peak meter.

- **Playback and editing independent of recording**: This option switches from standard recording mode to Playback and editing independent of recording (↗95) mode.

- **Create object**: All recordings create objects on the corresponding tracks. You could disable this default behavior here.

- **Update object during recording**: During the recording, the graphical representation of the objects updates. The option is available only for recording to wave files. You could disable this default behavior here.

- **CD/DAT markers read**: DAT devices and some professional CD players output marker information via the SPDIF output (CD track or DAT marker). If this option is active, these markers are transferred to the project, provided that the audio device supports this.

- **Pre-recording (ASIO and Wave format only)**: Use this function to add audio material that you have played before starting the recording to the beginning of the current recording.

  > ℹ️ *For more information on pre-recording, see Program Preferences> System Options > Recording (↗699).*

- **Recording offset (samples)**: If recordings exhibit a constant, undesired shift in relation to the existing audio material in the project, set an offset here which can be used for positioning all recordings. The option **+ASIO latency** can be used to deactivate the compensation of ASIO input and output latency, which is always included by default, for certain recording and monitoring situations, for example when recording the master to a track and additionally recording the input.

Recording Information: In this section you can find information about the last recording you made, such as **Recording Time** (length of recording), **Remaining Time** (remaining recording time), and **Disk Space** (remaining space on the hard disk).

Visualizer: Opens or closes the visualization window.

Reset Pk.: Reset the visualization's peak hold display.

Recording control

You can also start and stop recording in the Recording Options dialog. The button **Recording** (keyboard shortcut: **A**) starts the recording. **Stop** (keyboard shortcut: **S**) ends the recording.

You can pause the current recording with the button **Recording pause**. Click again on **Pause recording** to resume recording at the same position. Unlike stopping and restarting the recording, this does not create a new take. When stopping the recording from the paused state, the play cursor remains at the current position, the next recording then starts at this position.

Take: Each recording take is given a name and a sequential number. In the name field you can change the preset name that will be used for the next recording. To rename the currently recorded take while it is still recording, enter the name in the take name field and click the button **<-**. The consecutive number can be increased by one with the **+** button.

If you press the button **"FS (False Start)** during the recording, the acquired object will be split at the current position - but the recording will continue. The name of the separated part of the take is completed with "FS".

ᶭ *For more information on takes, see* Take Manager *(↗328) and* Take Lanes *(↗138).*

Auto: During recording, you can use this button to set markers at the current recording position, e.g. to mark an incorrect passage for later correction. 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.

MIDI Recording

In Sequoia there is no difference between audio and MIDI tracks. Each track can contain both audio and MIDI objects. This allows you to manage software instruments entirely within a single track: The track contains MIDI objects that control a VST instrument whose output is played back through the same track.

However, you can only record from one particular device on a track at a time, so it is not possible to record audio and MIDI to the same track at the same time. By default, audio is recorded on a track

To record MIDI on a track, click the **MIDI** button MIDI in the Track Editor, or select **MIDI Record** in the input assignment menu (right-click the record button in the track or the **In** area in the mixer channel). When loading a VST instrument (↗371) into the track, MIDI recording is automatically turned on.

In Sequoia MIDI data is recorded in a similar way as audio data. You create an object for each recording, which overlaps existing objects.As with audio recording, you can record multiple takes that you can combine using the Take Lanes (↗138).

ⓘ **Input Q**: *In the **MIDI** section of the Track Editor next to the Velocity Dynamics button you will also find the button **Input Q**. If it is active, MIDI notes are already quantized during recording according to the current setting. You can restore the original positions with the menu command **Object** > **Quantize** > **Reset Quantization**.*

The peak meter works as an indicator for incoming MIDI signals.
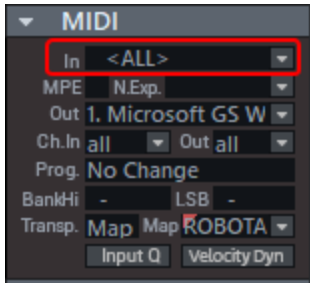
You can select the MIDI input device in the MIDI section of the Track Editor, by default all available MIDI input ports are selected.

> ℹ️ *For more information on configuring the MIDI ports, see **Program preferences** > **System options** > **MIDI** (↗697)*

MIDI recordings are started just like audio recordings by clicking on the record button in the transport console.

MIDI Recording Modes

MIDI data can be recorded in three different MIDI recording modes. They differ in how newly recorded MIDI data is inserted into the VIP if there are already MIDI objects at the recording position.
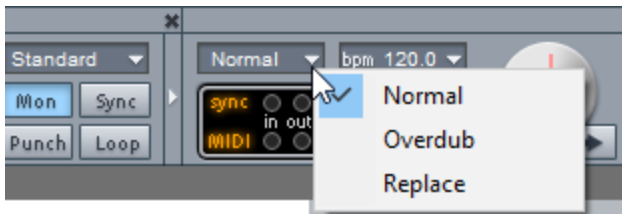
In this menu in the transport console or in the menu **Play/Rec** > **MIDI recording mode** you can select the MIDI recording mode.

- ■ **Normal**: In this recording mode, a new MIDI object is created on top of the existing object for each recording operation, just like with audio recording. The existing object remains unchanged and you get multiple takes that you can select in the Take Manager (↗328) and combine in the Take Composer.

- ■ **Overdub**: In this mode, the newly recorded MIDI data is additionally written to the existing MIDI object.

- ■ **Replace**: In this mode the MIDI data of existing objects will be replaced by the newly recorded ones.

Retrospective MIDI recording

The **Retrospective MIDI recording** allows you to integrate notes played on the keyboard into the project afterwards even without an activated recording, e.g. if you improvise an additional solo to an existing project with a VST instrument.

Activate the option **Retrospective recording** in the program settings (keyboard shortcut: **Y**) > **MIDI** at **Recording** and set the time period for which MIDI inputs should be stored retrospectively with the menu **Buffer**.
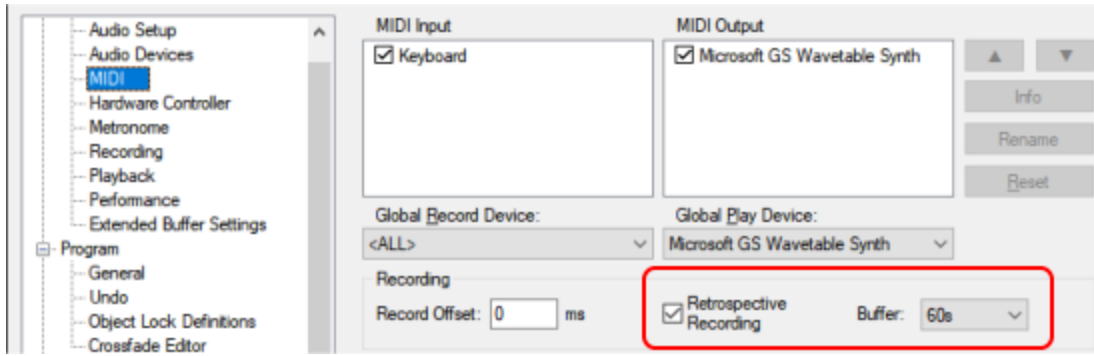
Then, if at some point you played along to a project on your MIDI keyboard and want to save that performance, choose **Playback** > **Retrospective MIDI recording**.

Select whether you want to insert the MIDI data at the position in the project where you played it or at the current position of the play cursor. Accordingly, Sequoia inserts a MIDI object the size of the buffer into the track that contains the played notes.

> ℹ️ *Tip: Recording MIDI notes using the "Retrospective MIDI recording" feature also works when stopped, i.e. without simultaneous playback of the project.*

Interleaved Recording

You can conduct a multi-channel recording as interleaved recording. This means that an arbitrary number of audio channels are recorded into one wave file.

Here's how you do it:

1. 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 as you like.
2. Select all tracks by clicking the track head of the first track and, with the **Shift** key held down, the track head of the last track.
3. Now choose **Track** > **Insert New Track** > **New Folder Track** from the menu.
4. Now right-click on the record button in the track head of the folder track and choose **Interleaved (Foldertrack)** from the context menu.
5. Now activate recording in the folder track with a left click on the record button on the track head.
6. Use the record button on the transport console to start recording as usual.
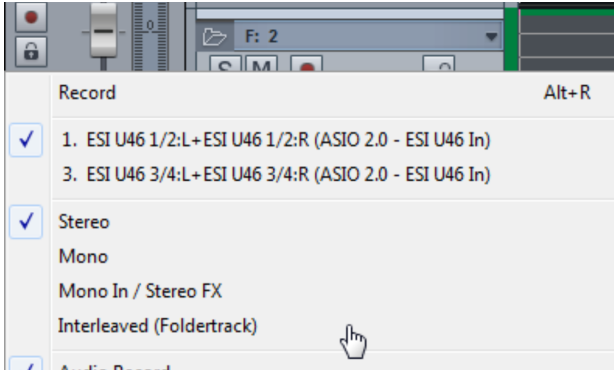
Notes:

- When loading an Interleaved Multi-Channel-Wave a Foldertrack will also be created by default, optionally a surround master will be created. You can deactivate this behavior in the Options for loading audio files" (↗658).

- 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.

Loop Recording

If you want to record in a loop, first select the range in the project window that you want to record.
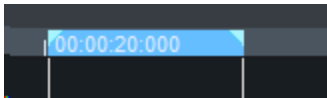
Then activate the **Loop** button in the Transport Console.

You can place the play cursor in front of the selected range, if Loop is activated the range selection will be kept. Start recording, playback starts from the position of the play cursor. The actual recording starts when the play cursor reaches the loop range.

The range will loop until you stop recording again. A new take is created for each pass.

The Take Manager (↗328) and Take Lanes (↗138) can be used to organize and edit the recording takes.

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.

"Punch-in/Punch-out" is a recording process that can be started and ended while playback is running.

If you have not set punch markers before, they will be set automatically when recording if a range was selected before recording started. If no range is selected, you can also set the In marker during recording to define the recording start position and the Out marker for the recording end position.

> **i** *If you set the option* **Prepare all tracks for Track Punch record** *in the* Program Preferences > System Options > Recording *(↗699), you can subsequently add audio tracks to a recording that were not yet armed when the recording was started, or remove individual audio tracks from the recording. To do this, click on the record button of the desired track.*

If the recording is started when punch markers have been set, the recording button will flash until the In marker has been reached and again when the Out 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. However, if you click the **stop** button, playback will also stop.

Punch Recording with Markers

If the range in which a punch recording is to be made is known precisely in advance, you can use markers to perform the punch process.

1. Select the range to be recorded and click the **In** (sets the Punch In marker) and **Out** (sets the Punch Out marker) buttons in the Transport console (↗67).

2. Now you can set the play cursor in position.

3. If **Punch** is activated, start the playback 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 play cursor is not yet in the punch range, the "Record" button will blink. During punch recording, it lights up red continuously.

By right-clicking on the **In/Out** buttons on the transport control, you reach a menu where you can delete the markers: **Delete Marker** deletes the marker of the respective right-clicked button, **Delete Punch Markers** deletes all regular punch markers.

With **Additional Punch Start Marker** and **Additional Punch End Marker**, any number of additional punch in/out markers can be set. These correspond to a simple marker named **Punch In+** and **Punch Out+**. Whenever the play cursor reaches the project section before these markers during a punch recording in a project with such additional markers, the regular punch in/out markers are moved to the positions of the additional punch markers.

This means that you can mark all the places in a project over which a punch recording is to be made with such additional markers and then complete all punch recordings in one recording pass.

Delete additional punch markers deletes all additional punch markers.

> 💡 *All punch marker commands can be found in the menu **Playback** > **Marker** and can thus also be provided with keyboard shortcuts.*

Recording Track Outputs

It is also possible to record the output of a track, a bus or the master on a track. To select the output of a track as the source for audio recording, right-click the Record button in the Track Head, Track Editor, or Mixer, or click the **Input** button in the **Audio** section of the Track Editor, or at the very top of a mixer track, to open the Track Input menu.
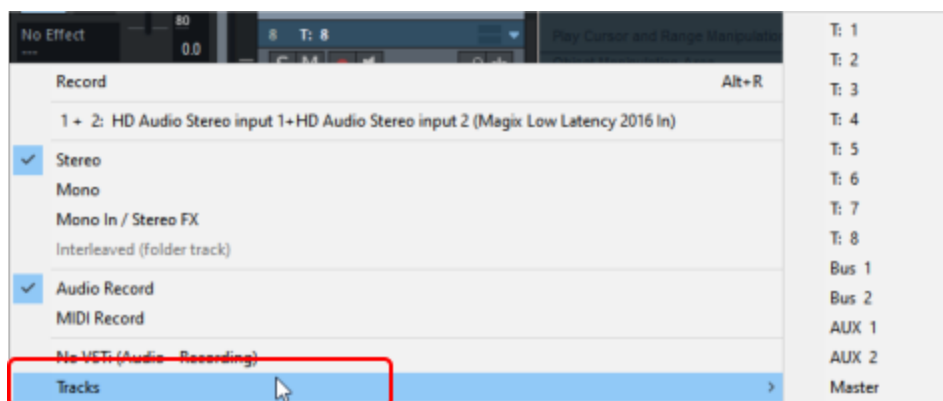
At the very bottom under **Tracks** you can select one of the tracks or masters of the project as the recording source.

For example, you can quickly output an intermediate status of the project without having to think about the export settings by recording the output of the master on a new track.

Section, Scrolling and Zooming

A Sequoia project contains a lot of graphical information: Objects with their waveform representation, automation curves, markers and much more. An overview of the entire project is needed just as much as an exact representation of the fade-in of a single object.
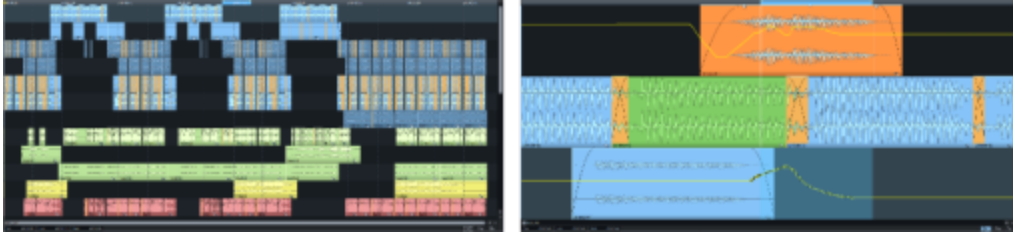
The part of the project in the project window that is visible at the same time is called the **section**, and the size of the display of this section is called the **zoom level**. Changing the zoom level is called **zooming**.

Moving the visible section is called **scrolling**.

Zooming

There are a number of ways to zoom in Sequoia using the keyboard and mouse.

Zooming with the mouse wheel

To zoom with the mouse wheel, hold down one of the following modifier keys and turn the mouse wheel. You can zoom in by rolling the mouse wheel upwards. You can zoom out by rolling the mouse wheel downwards.

Description	Modifier
Vertical Zoom	**Ctrl + Mouse wheel**
Horizontal Zoom	**Ctrl + Shift + Mouse wheel**
Vertical + Horizontal Zoom	**Ctrl + Alt + Mouse wheel**
Zoom waveform display	**Alt + Mouse wheel**

*The modifier keys used can be customized in the **Program Settings** under **Keyboard/Menu/Mouse** > **Mouse Wheel**. With the **Reset** button, you can switch there between the default mapping and an alternative mapping where the mouse wheel scrolls horizontally without a modifier key.*

Zooming with the keyboard

- Use the key combinations **Ctrl + right arrow** and **Ctrl + left arrow** to zoom horizontally in the project, i.e. in project time. Alternatively, you can use the **up arrow**/**down arrow** keys.

- Use the key combinations **Ctrl + up arrow** and **Ctrl + down arrow** to zoom in and out of the waveform display.

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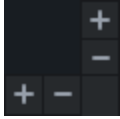
In the lower right-hand corner of the project window, the **+** and **–** buttons may be used for more precise adjustment of the horizontal and vertical zoom levels.

Right click on the zoom buttons or left click on these buttons opens the menus View > Vertical and Horizontal as context menu with all commands for zooming and scrolling.

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

1	Entire project (horizontal)
2	Entire project (vertical)
3	Vertical zoom to range
4	Horizontal zoom to range
5	Zoom out of waveform display
6	Zoom in on waveform display
7	Display waveform normalized (↗717)

You can add (↗54) additional zoom buttons via the **Edit Toolbar** dialog from the context menu of the toolbar.

Zooming with the scrollbars

The scroll bars can be used for zooming:

- ■ **Click and drag** the **ends** of the scroll bars at the bottom and right edges of the project window to change the size of the displayed section or the height of the displayed tracks.

- ■ **Click and drag** the **bars** to move the horizontal or vertical section.

- ■ **Double click** on the **bar** sets the zoom level to the maximum value (display of the entire project or all tracks).

The length of the section, i.e. the duration of the project section displayed in the project window, is shown at or on the scroll bar.
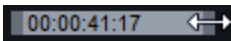
Zoom mouse mode

There is also a special mouse mode 🔍 for zooming Use the right mouse button to zoom out of the project, and the left mouse button to zoom in. Also, in this mouse mode, you can draw a selection rectangle over the project window to zoom in on a specific time period.

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

Zoom to selected object

Click this button at the bottom right of the arranger or choose Menu **View** > **Zoom to selected object** to adjust the section across all selected objects.

If you click this button again, Sequoia will display the original section again.

ℹ️ *The previous section is always restored when you deactivate the function, regardless of whether the object selection or section was changed while it was active. Attention: It will be restored even if you have switched to another open project in the meantime. This can lead to unexpected results.*

Overview mode

The Overview mode shows the entire project in reduced size below the arranger tracks.

Open the Overview mode via the button in the lower toolbar or via the corresponding menu command in the menu **View**.

In the overview, click and drag to select the section to be displayed in the arranger. By simply clicking in the overview, you position the section at the desired location in the project. Conversely, the selected section in the overview also follows the visible section in the project.

ⓘ *The overview mode only determines the horizontal zoom level, not the height and number of displayed tracks.*

Zooming by Vertical Mouse Dragging

Zoom your project by **clicking and dragging vertically**. At the same time you can change the play cursor position or the range borders by **dragging horizontally**.

This allows you to position the play cursor precisely "in one go".

1. Roughly set the desired position by clicking in the marker bar.
2. Click in the **Marker bar** above the top track and drag down to zoom in, drag up to zoom out.
3. At the same time, correct the play cursor position by dragging horizontally. Dragging up zooms back out.

ⓘ *Tip: If you press the **Ctrl** key before releasing the mouse button, the program automatically returns to the old zoom level.*

And this is how you select an exact range with this method:

1. Using the method described above, position the play cursor where you want the range to begin.
2. Now click below the play cursor in the **grid bar** and drag the mouse to the right to roughly define the end of the range.
3. Click the end of the range on the grid bar and drag down again to zoom in and adjust the end of the range precisely. Press the **Ctrl** + key and release the mouse.

ⓘ *If you do not want to change the zoom level when dragging the mouse, disable this feature in Program Preferences (Y key) > **Keyboard/Menu/Mouse** > **Mouse** with the option **No zoom when dragging mouse vertically in timeline**.*

Scrolling

Scrolling is moving the section of the project that is visible in the project window.

Autoscroll

By default, the visible section follows the play cursor in **Autoscroll** mode during playback. This means that the section "jumps" to the next time section shortly before the play cursor moves off the right side of the screen (Page Mode). In **Soft Autoscroll** mode, the section moves steadily under the play cursor, which is fixed in the center of the project window.

You can switch between these scrolling modes in the **View** menu. Use the **Scroll** key to turn the autoscroll mode on and off in general, use **Shift + Scroll** to turn the soft autoscroll on. Without autoscroll, the play cursor just moves out

of the visible section. To move the visible section of the project back to the play cursor, use the command **Section to play cursor/Last stop position** (keyboard shortcut **Ctrl + Alt + ,**). In stopped state, the section jumps together with the play cursor to the position where the playback was stopped.

Scrolling with the keyboard

The play cursor can also be moved using the arrow keys on your computer keyboard. If you additionally hold down the **Alt** key, you can move through the project in soft autoscroll mode.

The **Home** and **End** keys move the play cursor and the visible section of the project either to the very beginning or the very end.

To move the visible section without moving the play cursor, use the commands in the menu **View** > **Horizontal** and **Vertical**. Many of these scrolling commands can be executed by keyboard shortcut.

Horizontal:	Half section left	Ctrl + Alt + Arrow left
	Half section right	Ctrl + Alt + Arrow right
	Section to play cursor	Ctrl + Alt + ,
	Section to range start	Ctrl + Alt + B
	Section to range end	Ctrl + Alt + N
	Zoom in	Ctrl + Arrow left, Arrow up
	Zoom out	Ctrl + Arrow right, Arrow down
	Show all	Ctrl + Alt + Arrow up
	Zoom to range	Ctrl + Alt + Arrow down
Vertical positions:	Half section up	Shift + Arrow up
	Half section down	Shift + Arrow down

> **ⓘ** *Tip: Many of these commands are also accessible from the* bottom toolbar *(⚹53). Other scroll buttons can be* added *(⚹54) in the* **Edit Toolbar** *dialog.*

Scrolling with the Mouse

In the grid and marker bar: Click and drag on the grid and marker bar with the **Alt** key held down to move the visible horizontal section.
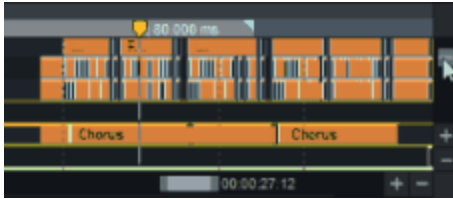
Using the scroll bars: By moving the scroll bars, the contents of the window can be moved.

With the mouse wheel: Turn the mouse wheel to move the visible vertical section (i.e. the displayed tracks). Hold down the **Shift** key to move the horizontal section.

> 💡 *The modifier keys used can be customized in the **Program Settings** under **Keyboard/Menu/Mouse** > **Mouse Wheel**. With the **Reset** button, you can switch there between the default mapping and an alternative mapping where the mouse wheel scrolls horizontally without a modifier key.*

With the middle mouse button: Click and drag anywhere in the project with the middle mouse button to move the visible section.

Storing and Recalling Position and Zoom Level

There are four memory locations to save a specific section (zoom level and position) in the project and restore it later. The zoom level can be saved in four additional memory locations.

The buttons for the memory locations are located below the track header of the lowest track.

Left-click a button 1-4 at **Setup** to save the current zoom level and the position of the section in the project. The number on the button is displayed brighter if it is occupied. Left-click on an occupied button to restore the section. The selected memory location is highlighted in color. To delete a memory location, click on the button with the **Shift** key held down.

Use the buttons at **Zoom** to save and load only the zoom level in the same way.

Corresponding menu commands for this can be found in the menu **View** > **More**. There are keyboard shortcuts for the first three memory locations in each case:

Command	Keyboard shortcut
Store Position and Zoom Level	Ctrl + NumPad 1, 2, 3
Get position and zoom level	NumPad 1, 2, 3
Store Zoom Level	Ctrl + NumPad 4, 5, 6
Get zoom level	NumPad 4, 5, 6

Multiple Sections

You can display up to three **sections** of the same project via menu **View > Sections**.
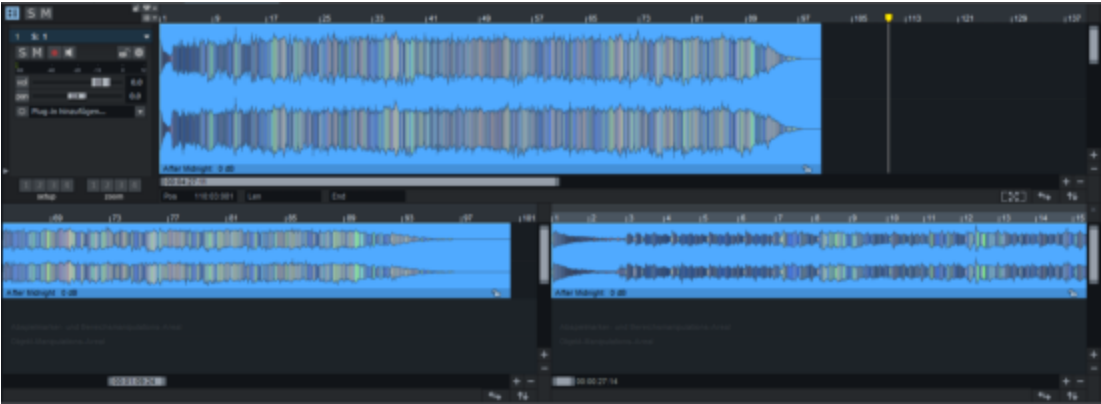
If you select option **2** for two sections, a second view will open under the project window, whose section can be set independently of the first view. This way, for example, you can view the complete project in one window, while magnifying a specific area in the other.

The **3** (sections) mode can be especially helpful when searching for loop points in a wave window. The upper, large window can display the entire file, while the lower left pane shows the beginning and the lower right pane the end of the loop range.

ⓘ *To stop displaying multiple sections, select Menu **View** > **Sections** > **1***

Split range

If you select **Sections** > **Split range**, the project is displayed in three sections. The lower two sections show the start of the range and the end of the range at the maximum zoom level. In this way, the range limits can be set precisely. **Split range for Video** does the same thing, but sets the zoom level for the range edges to 1 frame for working with video footage.

Use the **Page up** / **Page down** keys to switch between the sections.

Markers

You can use **markers** to indicate places of special importance in your project, for example, in a song arrangement, the beginning of each song part, such as intro, verse, or chorus. The time positions marked in this way can be accessed quickly using a menu entry, button or keyboard shortcut. Selected ranges can also be provided with a **range marker**. In addition to the position, it also has a length and can be used to quickly restore a play range or to export individual time segments of the project to separate files. A project in Sequoia can contain any number of markers.
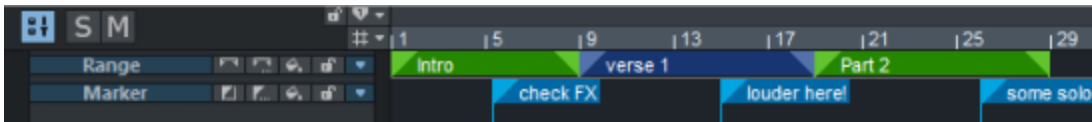
ⓘ *In addition to these markers, which are displayed on special tracks, there are other markers for marking time positions in projects or objects: Audio markers, markers for CD recording functions and metadata, tempo markers, bar markers, grid position markers. These are shown above the grid bar and discussed elsewhere:*

- *Markers for **CD functions** (↗579) are used to mark CD track positions when a project is to be burned to CD, but also to save metadata for export to compressed formats such as MP3.*

- ***Tempo markers**, **Beat markers** and **Grid position markers** are markers for controlling tempo changes (↗354) in the project.*

- ***Audio markers** (↗116) are markers in wave projects that can optionally be displayed in the associated objects.*

- ***Warp Marker** (↗366) are set in objects and used for time correction of audio events in audio material.*

- ***Lyrics markers** (↗472) enable you to enter song text, comments or production directions.*

Markers are displayed in a special **Marker track** above the top track in the project; for range markers there is also an **Range marker track**. Operation is identical in both tracks; the following explanations on markers and the marker track also apply to the range markers in the range marker track.
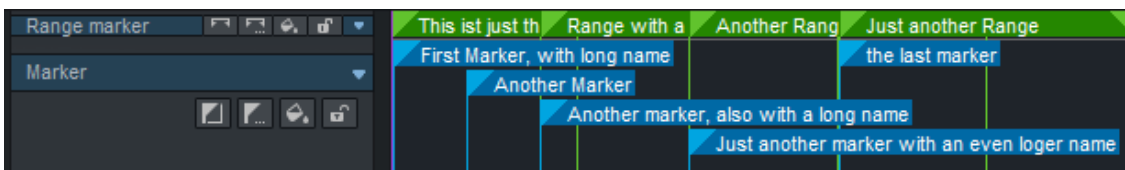
The height of the marker tracks can be adjusted by clicking and dragging the lower edge of the track. Depending on the height of the track, the marker names are displayed in such a way that they are optimally legible. Right-clicking on the marker track head opens the Marker manager (↗325).

Starting from the marker, marker lines are drawn over the entire height of the marker track. The option **Show marker lines/range marker lines across all tracks** in the context menu of a marker can be used to display the marker lines across the entire project.

> ℹ️ *The marker track/range marker track can be hidden separately using* **Marker track** *and* **Range marker track** *in the context menu of a marker or in the* **View** *menu. The markers are then not displayed in the project, but can only be accessed via the marker manager.*

Markers can be set with the buttons in the track head of the marker track, by menu command or keystroke and then jumped to directly.

Setting Markers

Markers can be set both in stop state and during playback or recording. They are placed at the current position of the play cursor. All marker commands can be found in the **Play/Rec** > **Marker** menu. You can also access the menu by right-clicking on the marker track or using the menu at the marker track head.

Marker command	Keyboard shortcut
Set markers with number 1...0: For fast saving of markers use the number keys with held **Shift** key. Depending on whether a range is selected, an range marker or a simple marker is set.	**Shift** + **1...0**
Set marker with next higher number: This command or a click on this button in the track header of the marker track ▨ sets a new marker with the next higher number.	**Shift** + **'**
Set marker with any name: With this menu command or this button ▨ in the track head of the marker track, the marker can be named arbitrarily when it is set.	**?**
Set new markers at recording position: This option (active by default) determines whether markers set during recording are set at the current recording position or, if not active, at the playback marker position instead. It is only relevant in the recording mode **Playback & editing independent of recording** (↗95), because in standard recording the play cursor and recording position are the same.	**Alt** + **?**
Get marker: Clicking on a marker moves the play cursor to the corresponding marker position. The marker positions 1...0 can be jumped to directly from the keyboard using the number keys. Clicking on an range marker selects the corresponding playback range.	**1...0**

> 💡 *Tip*: With **Alt** + **F-Keys** *you can also save and restore ranges with the keyboard, for more information see* Saving and restoring ranges *Saving and Restoring Ranges (↗127)*

Coloring, Renaming or Deleting Markers

Markers can be assigned colors for more clarity. Select a marker by clicking on it, by clicking with held **Ctrl** key you can add markers to the selection. Then click the Marker Color button 🎨 or choose **Marker Color...** from the Marker menu. Select a color from the menu.

> ℹ️ *The menu corresponds to the color menu used to colorize tracks and objects, for more information on colors, see* Color mode *(↗78)*

To **rename** a marker, double-click a marker and change the name in the dialog.

To delete a marker, select the marker by clicking on it, multiple markers can be selected by clicking and holding **Ctrl** key. Then press the **Del** key to delete. In the marker menu there are also commands to delete all markers in the selected range(**Delete markers in range**) or to delete all markers (**Delete all markers**).

Moving Markers

To move a marker, click and drag it to the desired position. You can also select multiple markers by clicking and holding **Ctrl** key and then move them together. To lock the markers against accidental moving, click the **Lock markers** button 🔓 in the marker track header.

If you move a marker or several selected markers while holding down the **Ctrl** key, all subsequent markers are also moved by the same amount. This also applies to the end of range markers, i.e. if you change their length.

The option in the menu **Object Mode** > **Link Markers to Objects** lets you move markers along with objects in the top track of the project when you move them. Moving objects in object mode **Link all tracks to the right** then moves the markers independently of the track of the moved object.

Via the submenu **Link markers** of the marker menu 🔻 you can specify in detail how existing and newly set markers behave when moving objects: In the upper section **Current marker** you define whether the marker at the playback position is moved along with objects of the first track (**First track**), of the selected track (**Current track**), all tracks (**All tracks**) or not at all (**No tracks**) when moving them. The lower section **New Marker** defines which behavior is applied to newly set markers. For example, you can first set markers to be moved with the objects and set fixed markers at a later time.

> 💡 *For a clearer representation of which markers are moved along with objects on which tracks when you move them, use the **RefTrk** column in the* marker manager *(↗325). For each marker, it specifies the track (or all tracks or no track) whose objects move this marker.*

Other Actions with Markers

To select a range between two markers, first click on the first marker and, holding down the **Shift** key, click on the second marker or double-click in a free area in the marker track between the two markers.

Marker on range borders: Use this function to place a start maker **S** and an end marker **E** at the borders of a selected 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 Menu **Play/Rec** > **Move Play Cursor** > **Marker left/right**.
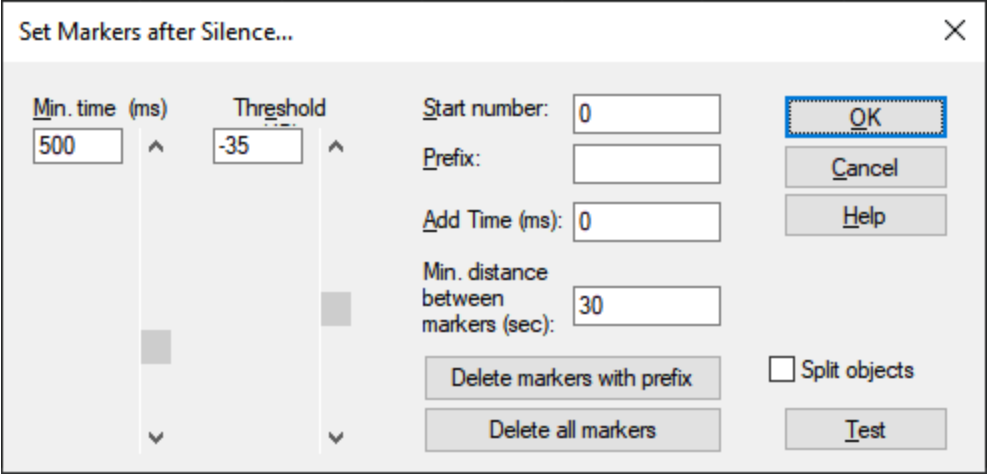
To quickly switch between range markers, use the keyboard shortcuts **Shift + Alt + W**∕**Shift + Alt + Q** or select **Play/Rec** > **Move play cursor** > **Select previous/next range marker**).

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, right click the track head of the marker tracks or select from the **Marker menu** or **View** > **Manager** menu **Marker Manager...** or use the keyboard shortcut **Ctrl + Shift + Alt + M**.

Markers after Silence

"Silence" is defined as a place in the audio material where the level falls below a minimum level, the **Threshold (dB)** for a certain **Minimum time (ms)**.

For naming the generated markers you can enter a **Start number** from which the markers are incremented, additionally you can enter additional characters/letters with **Prefix** which are prefixed to the marker number. This makes it easier to distinguish them from existing markers.

Add time (ms): Here you can enter the time span by which the markers will be shifted backwards (to the left).

Delete markers with prefix: Deletes all markers that have the given prefix. This allows you, for example, to delete all markers that were created by a previous application of **Markers after Silence...**.

Delete all markers: Deletes all markers in the project

Test: With the button **Test** you can generate the markers temporarily at first to check if the correct markers are generated with the set threshold and time constants. If too many or too few markers are displayed, change the values and click Test again until you are satisfied with the result. Then click **OK** to close the dialog and finally create the markers.

Split objects: If you select this option, the object will be additionally split at the generated markers.

Audio Marker

In Wave Projects (wave files) markers can be stored directly in the audio file (*.wav). They are thus also available in other projects or applications. When wave files containing markers are loaded into a virtual project, these markers can be displayed at the top of the object.

> 🛈 *Markers in wave files set in other audio applications (e.g.* **MAGIX SOUND FORGE Pro**) *are also recognized.*

The display of audio markers is disabled by default. To display audio markers, activate the option **Audio marker** in the **Objects**section of the View Options (↗716) dialog in the **Program settings**.
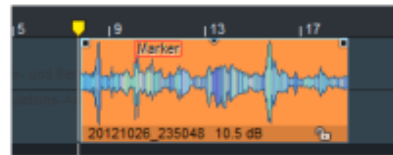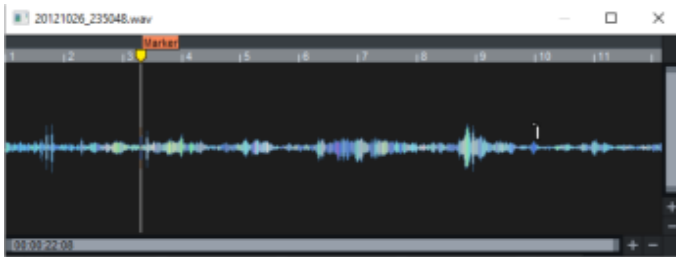
If you right-click an object and choose **Wave Editing...**, *the audio file will be opened for destructive editing as a wave project.*

Markers that you set in the wave project appear in the virtual project as audio markers in the object.

If the audio marker display is turned on, you can right-click the top of an audio object in the virtual project to open a menu that lists all the audio markers. Select an audio maker to move the play cursor to its position. The option **List all audio markers of the track** in the menu determines whether the menu contains only the audio markers of the clicked object or of all objects of the track.

Set New Audio Marker

To set a new audio marker directly from the virtual project, choose the command **Set new audio marker**. Use this command to set an audio marker in the selected audio object at the play cursor position. If there is already a marker at this position in the virtual project, its name is taken also for the audio marker.

VIP Marker to Audio Marker/Audio Marker to VIP Marker

Select an object and choose **Play/Rec** > **Marker** > **Copy audio marker to VIP marker** to copy the audio markers of the audio file to the virtual project according to the object position within its object boundaries.

Copy VIP markers to audio marker vice versa, copies all markers of the project that are temporally between the object boundaries to the audio file of the object.

Audio Marker Manager

The Audio Marker Manager is a special variant of the **CD Index Manager** and is used to conveniently edit all audio markers of an audio file. For information on operation, refer to CD Index Manager (↗581).
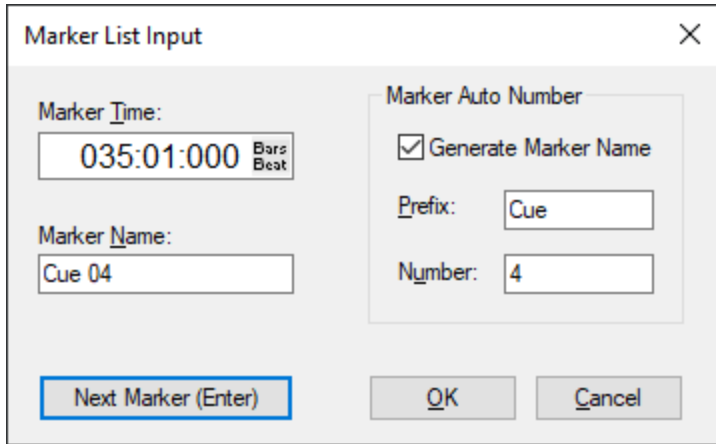
> **ℹ** **Note**: *The Audio Marker timings in the Manager refer to time positions in the audio file, not to positions in the virtual project.*

Marker List Input

With the marker list input, a larger number of markers (e.g. a handwritten marker list of a recording session) can be entered quickly and effectively into the project. Open the marker list input with menu **Play/Rec** > **Marker** > **Input marker list...** or keyboard shortcut **Ctrl + Shift + L**

And this is how you enter markers:

1. Enter the time position of the marker into the **Marker time** editing field. In the field on the right of the time entry you can set the unit of measurement.
   For the time "001:02:010" enter only "00102010", in the edit field the numbers move automatically to the left, colons don't have to be typed.

2. If necessary, press the **tab** key to change the default marker name.

3. Use the **Enter key** to enter the marker. The marker is created, then the time input field is selected again and you can enter the next marker.

4. When you have entered all markers, click **OK** and exit the marker list entry.

The default marker name results from the settings in the input fields **Prefix** and **Number**.

- The **Prefix field** specifies a fixed part of the marker name, while the **Number field** contains the sequence number, which is incremented by one with each new marker.

- If **Generate marker names** is deactivated, the entered marker name is no longer changed and thus remains as a name for the creation of further markers. However, the marker number continues to count up with each marker applied.

Ranges

You can select parts of the arrangement as **Range** to perform cutting operations: The objects and automations within a range can be cut or copied and pasted to another location. The range selection can extend over any number of tracks, and the selection range is not limited to objects and does not have to coincide with the object edges. During the cutting operations, the objects are split at the range boundaries.

Selecting and Editing Ranges with the Mouse

To set a range, click and drag the mouse on the tracks over the objects. In the Universal mouse mode , you must click in the upper half of the track to do this, otherwise you will move the objects. You can also use the Range mouse mode   and can thus use the whole track for range selection.

You can see from this shape of the mouse pointer that an range selection is possible:
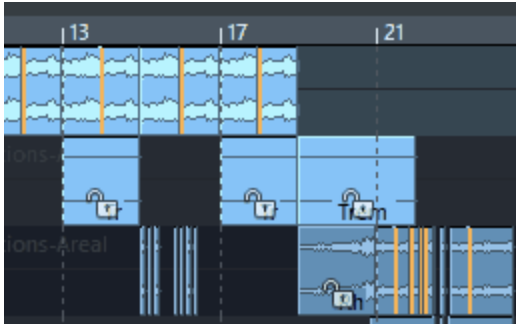
Ranges can also extend over several tracks

> **ⓘ** *When selecting a range, a corresponding* playback range *(↗84) is always specified as well. Moving the play cursor therefore always cancels the range selection. However, it can be restored with* **Shift + Backspace** *(see below).*

You can edit an already selected range. To do this, click inside the range selection and drag in the appropriate direction:

- Drag the playback range borders on the grid bar to change only the time selection. **Shift + click and drag** in the range on the grid bar to move the whole range.

- First drag up or down to add more tracks to the selection or remove tracks from the selection. The time selection (horizontal) is retained.
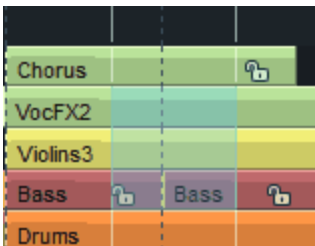
- First drag to the right or left to edit time and track selection.
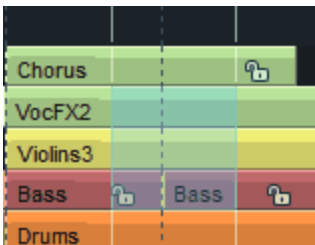
■ A **double-click in the playback range** on the grid bar switches between three different variants of the range selection: Range over current track, Range over all tracks and Range not over tracks (playback range only).
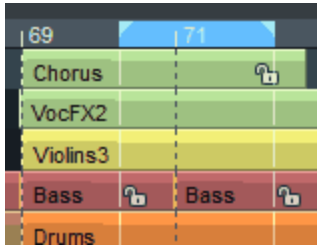
> ℹ️ *If a range was selected over several but not all tracks, the track selection is lost in the process.*

> 💡 *For the mouse actions to expand/reduce the selection over the tracks and the double-click actions, there are also menu items in the menu* **Edit** > **Range** > **Edit Range** *that can be provided with keyboard shortcuts:*
> **Extend/reduce Range top/bottom** *and* **Range not over/over current/over all tracks** *and* **Remove range**.

Editing Ranges with the Keyboard

In the menu **Edit** > **Range** > **Edit Range** are the commands to control the range selection with the keyboard.

Command	Description	Keyboard shortcut
Range all	Set the range over the entire duration of the project. Repeatedly applying the keyboard shortcut toggles between Range over Selected Track, Range over All Tracks, and Playback Range Only.	**A**
Move Range start left/right	The start of the range is shifted one grid unit (↗79) to the left or right.	**Arrow left/right** **Alt** + **Num ÷/×**
Move Range end left/right	The end of the range is shifted one grid unit to the left or right.	**Shift** + **Arrow right/left** **Alt** + **Num -/+**
Flip range left / right	The current range is flipped by its length to the left or right. With **Flip Range left**, its new end corresponds to its former beginning, with **Flip Range right**, its new beginning corresponds to its former end.	**Ctrl** + **Shift** + **Arrow left/right**

Range start to left marker	The start of the range is moved to the nearest marker to the left.	**Shift** + **F2**
Range end to right marker	The end of the range is moved to the nearest marker to the right. ℹ️ **Note:** *These commands also include audio markers in the object if they are displayed in the project. (Menu **File** > **Program settings** > **Project display...**). Also taken into account are other markers such as CD track indices or tempo markers.*	**Shift** + **F3**
Range over all Selected Objects	The range is set over all selected objects. It includes all tracks between the top and bottom selected object over the time duration of the leftmost and rightmost selected objects. 💡 *So with this command you can quickly define a range by clicking the first object in the desired time period, holding down the Ctrl key and clicking the last object, and then pressing **Shift** + -.*	**Shift** + **-**

Unselecting and Reactivating Ranges

If you move the play cursor to another position, by mouse click into the project or a command from the menu **Play/Rec** > **Move Play Cursor**, the range selection is removed.

ℹ️ *A double-click on the Grid bar also cancels the* playback range *(↗83).*

To restore a previous range selection, enter the keyboard shortcut **Shift + Backspace** or choose Menu **Edit** > **Range** > **Recall last range**. By repeatedly executing the command, you can restore the last five ranges. You can also execute this function by clicking this button in the Transport Console.
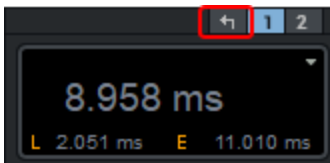
Finding Zero Crossings

"Zero crossing" is what we call a sample value that lies between a positive and a negative sample value (ideally 0 or almost 0). If samples are not cut at zero crossings, audible value jumps can occur when these samples are cut directly one after the other.

ℹ️ *When* Auto Crossfade mode*Auto Crossfade Mode (↗169) is activated, zero crossings do not matter because objects are then faded in and out at their boundaries.*

If you do not want to fade in the samples, for example to best preserve the transients of percussive audio, you can use the commands under menu **Edit** > **Range** > **Edit Range** > **Find Zero Crossings** to move the range boundaries to the zero crossings.

Command	Description	Keyboard shortcut
Beginning of Range -> 0	The range start is shifted to the right to the next zero crossing.	**Ctrl** + **Page Up**
Beginning of Range <- 0	The range start is shifted to the left to the next zero crossing.	**Shift** + **Page up**
End of Range -> 0	The range end is shifted to the right to the next zero crossing.	**Ctrl** + **Page Down**
Beginning of Range <- 0	The range end is shifted to the left to the next zero crossing.	**Shift** + **Page down**
0 -> Range <- 0	The range limits are shifted inward to the nearest zero crossings	

> ℹ️ *Regardless of which tracks the range selection includes, the zero crossings are sought in the object that is under the range boundary in the current track. If there is no object there, the search is made in the top track that contains an object at the corresponding time position. In any case, after the zero-crossing search, the range selection extends only over the track of the analyzed object.*

Other Menu Commands for Editing Ranges

In addition to the above commands for editing ranges, which come with keyboard shortcuts by default, there are other commands for range manipulation in the menu **Edit** > Range > **Edit range**.

Move range ends to object edges: With **Range start to left object edge** the range start is moved to the nearest object edge to the left. It is irrelevant on which track this object is located. With **Range end to the right object edge** the range end is moved to the nearest object edge to the right.

Extend range to start or end of project: With **Range to Beginning** the range start is moved to the start of the project, with **Range to end** the range end is moved to the end of the project.

Range length: With the commands at **Range length** you can set the range, starting at the play cursor, to the length of 1, 2, 4, 8 and 16 beats.

Marker manager: The ranges of the current project saved in range markers are displayed in the marker managerMarker manager (↗325). Keyboard shortcut: Ctrl + Alt + Shift + BKeyboard shortcut: **Ctrl** + **Alt** + **Shift** + **M**

Range editor: With the range editor you can numerically edit the beginning, end and length of the selected range in different units of measurement.

You open the Range editor via menu **Edit** > **More** > **Range editor...**

Adjustments to the parameters exhibit themselves as follows:

- Change start of range -> Range end remains constant

- Change range length -> Range start remains constant

- Change range end -> Range start remains constant

Copying, Pasting and Deleting ranges

Various edit operations can be performed with the selected range. The content of the selected range can be copied to the clipboard (also called "clip" here) or cut, deleted, overwritten with silence, or pasted via the clipboard elsewhere in the project.

> ℹ *Notes:*
>
> - *The menu commands and keyboard shortcuts for the edit operations apply to selected objects as well as to selected ranges. Depending on whether objects or a range is selected, the corresponding operations are performed. If both objects and a range are selected, at **Copy**, **Cut** and **Replace with silence** the range selection always has priority, at **Delete** the order of selection decides: The last thing selected is deleted, range or object.*
>
> - *The object modes (↗155) are also respected during cutting operations with ranges, i.e. if a corresponding object mode with rippling is set (e.g. **Link all tracks to the right**), subsequent objects will also be moved forward to fill the gap when cutting ranges.*

Copy

Command	Description	Keyboard shortcut

Copy	The selected range is copied to the clipboard. The previous contents of the clipboard will be lost.	**C** **Ctrl** + **C** **Ctrl** + **Ins**
Copy as...	This operation is available only for Wave projects. There you can save the selected range directly into a new wave file.	**Shift** + **C**
Copy to ClipStore	The selected range is copied and stored as an entry in the Clip manager (↗311). This allows you to perform several copy actions in succession and preserve the respective clipboard contents. This operation is not available for wave projects.	**Ctrl** + **Shift** + **C**
Copy + Clear	Copies the current range to the clipboard, removes it from the project, and replaces it with silence. ℹ️ *This command is very similar to the command **Cut**, but no rippling of subsequent audio material is performed here, even if an Object Mode with rippling is active.* This operation is not available for wave projects.	**Ctrl** + **Alt** + **C**

Cut

Command	Description	Keyboard shortcut
Cut	The selected range is copied to the clipboard and deleted from the project. In wave projects, the subsequent audio material is rippled, the audio file becomes shorter by the cut part. In virtual projects, the subsequent material is rippled or not according to the set Object Mode.	**X** **Ctrl** + **X** **Shift** + **Del**
Cut with time/ripple	The selected range is copied to the clipboard and deleted from the project. The subsequent material is rippled, even in virtual projects, regardless of the object mode set.	**Ctrl** + **Alt** + **X**
Cut to Clipstore	The selected range is deleted from the project and stored as an entry in the clip manager (↗311). This allows you to perform several cut actions in succession and preserve the respective clipboard contents. The subsequent material is rippled or not according to the set Object Mode. This operation is not available for wave projects.	**Ctrl** + **Shift** + **X**

Paste

Command	Description	Keyboard shortcut
Paste / Insert Clip	The data from the clip is inserted at the position of the play cursor, or, if there is a range selection, at the beginning of the range. The length of the selected range does not matter, the content of the clip will always be inserted completely. In virtual projects, the content of the clip is inserted starting from the selected track; if the clip contains multiple tracks, only as many tracks are inserted as there are target tracks down in the project available; the remaining tracks of the clip are skipped. In wave projects, the subsequent audio material is moved to the back, the audio file becomes longer by the inserted part. In virtual projects, according to the set Object Mode, the following material is moved to the back (rippled) or overwritten with the content of the clip. A new range is defined over the inserted data.	**V** **Ctrl** + **V** **Shift** + **Ins**
Paste from ClipStore	Clips selected in the Clip Manager are inserted at the play cursor position or at the beginning of the selected range. ℹ *Caution: For multitrack clips, only as many tracks of the clip are inserted as tracks are selected in the project. If you want to insert several tracks, select a range over several tracks in the project beforehand accordingly.*	**Ctrl** + **Umschalt** + **V**
Paste with time/ripple	Like **Paste/Insert Clip**, but the following material is always moved to the back (rippled), even in virtual projects, regardless of the Object Mode set.	**Ctrl** + **Alt** + **V**
Overwrite with clip	Like **Paste/Insert Clip**, but the subsequent material is always overwritten.	**Alt** + **V** **Ins**
Mix with clip	This operation is available only for Wave projects. The content of the clip will be mixed with the audio file. Since both components are added to the mix at 100% each, there is a risk of overdriving.	
Crossfade with clip	This operation is available only for Wave projects. The content of the range is crossfaded with the content of the clip. Here the position of the play cursor determines the end of the clip with which the audio file is to be crossfaded. The length of the clip sets the length of the crossfade.	

Delete

Command	Description	Keyboard shortcut
Delete	All audio data and automation data in virtual projects in the selected range will be deleted. In wave projects, the subsequent audio material is rippled, the audio file becomes shorter by the deleted part. In virtual projects, the subsequent material is rippled or not according to the set Object Mode.	**Del**
Ripple Delete	All audio data and automation data in virtual projects in the selected range will be deleted. The subsequent material is rippled, even in virtual projects, regardless of the object mode set.	**Ctrl** + **Del**
Extract	This command deletes all data of the project that *is not* within the selected range. In virtual projects, all tracks are preserved, even if they do not contain an object in the selected range. -ᑭ- *Single objects can be extracted with the* **Object** > **Edit** > **Trim Object** *(⬀162).*	

Silence

Command	Description	Keyboard shortcut
Insert Silence...	This command inserts silence - in other words, time - on all tracks from the current play cursor position or at the beginning of the selected range. If there are objects at this position, they are split and the corresponding parts are moved behind the silence together with the following data. If a range is selected, its length is used as a default for the time span to be inserted. The selected range remains after the operation. The project is extended by the length of the inserted object.	
Clear	All audio 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 function thus corresponds to a **Delete** without rippling, for virtual projects regardless of the selected object mode.	**Alt** + Del

Saving and Restoring Ranges

You can save the current range selection in a **Range marker** to restore this selection at a later time. Only the playback range is saved, not the track selection. Similar to the **marker track**, these range markers are displayed in a separate **range marker track**. For more information on marker and range marker tracks, read the section **Marker**Markers (↗112)

All saved range markers can be listed and managed in the **Marker manager** (↗325).

Quickly Saving and Restoring ranges with the Keyboard

The keyboard shortcuts for saving markersSetting Markers (↗113) (Shift + 1...0) can also be used to save ranges if a range was selected before the keyboard shortcut is applied.

For reasons of compatibility with older program versions, the keyboard shortcuts **Alt** + function keys (**F2...F10**) are also available specifically for saving/restoring ranges. You can restore these ranges using the keyboard shortcuts **Ctrl** + **F2...F10**.

> ⓘ **Note**: F4 is not used because Alt+F4 is the Windows keyboard shortcut for closing windows. Alt + F9 is also not available, because this keyboard shortcut is used for other functions. This leaves 8 quick accesses for range markers (F2,F3,F5...F8,F10)

With **Alt** + **F11** you can save range markers and name them.

> 💡 Restoring a range is also possible during playback. This allows you to switch between different ranges at the touch of a button to compare them by ear.

You can also use the keyboard shortcut **Shift** + **Ctrl** + **F2...F10** to simply restore the length of a saved area, which then starts at the current position of the play cursor and has the length of the saved area.

Range Mixdown

The Mixdown function lets you combine a selected range into a single audio object. The objects involved are then removed from the tracks. You can use the feature to free up resources as real-time effects are rendered, to indicate finished ranges as such, or just to simplify the project.

To perform a range mixdown, select **File** > **Advanced Export** > **Range Mixdown** from the menu.

If the bit depth of the objects in the project is more than 16 bits, you can decide in a query dialog whether the new file should be saved in 32-bit float or 16-bit format:

- **32 Bit (Float)**: The new file will be saved in 32 bit float format. Select this option if audio material in 24-bit or 32-bit float format is involved in the mixdown and the high resolution should be preserved.

- **16 bit (integer)**: The new file will be saved in 16 bit format. Select this option if 24-bit or 32-bit float objects are involved in the mixdown and the final product is to be used in the form of 16-bit audio (e.g. as an audio CD). In this case, the 24/32 bit files in the project are converted to 16 bit using dithering.

The object with mixdown file is placed on the active track. If this track was used in the project, i.e. contains volume, pan settings or effects, these now also affect the mixdown object. Since the mixdown should sound exactly like the corresponding area in the project before, it is recommended to create and select a new track before executing this function.

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 mixer channel (↗478). This allows you to adjust the volume of all objects on a track together, apply effects to them, and mute or solo them.

A new Sequoia project is created with a specified number of tracks. Using **project templates**, mixer and surround setups, you can already assign certain properties to the tracks, e.g. create a certain number of tracks as busses (↗495).

Inserting New Tracks

New tracks can also be inserted using the commands in the menu **Track** > **Insert new tracks**:

- **Append empty track**: With this function you create a new track and add it as the last track to the project.
- **Append multiple empty tracks...**: Use this command to add multiple empty tracks to the project. Enter the number in the dialog at **Number of new tracks** and confirm with **OK**.

  > ℹ️ *The maximum track count in Sequoia is limited to 999 stereo tracks.*

- **Insert empty track**: Use this command to insert an empty track after the selected track.
- **Insert multiple empty tracks...**: Use this command to insert multiple empty tracks after the selected track.

  > 💡 *To quickly insert a new track below the existing tracks, you can also double-click in the empty area below the track headers. Drag&Drop of an audio file from the File Manager/Explorer/Clipstore or an object from the project into the free area below the last track also creates a new track. If an object has been moved from within the project for this purpose, the track properties of the source track are adopted.*

You can also insert individual special tracks:

- **New MIDI track**: With this command you insert a MIDI track behind the selected track.

  ℹ️ *"MIDI" track just means that MIDI input recording is enabled for this track. A track can contain MIDI and audio objects and the recording mode can also be changed later.*

- **New Tempo Track**: see Tempo Track (↗362) in the Tempo editing section.

- **New video track**: see videoVideo (↗615).

- **New folder track**: see Folder tracks (↗135).

- **New surround folder**: This option is only available if a surround master exists, a new folder track is created whose subtracks are routed to the individual surround channels of the master accordingly.

- **New Submix Bus**/**New AUX Bus**: Use this command to insert new buses (↗495).

- **New Surround Bus**/**New Surround AUX Bus**: A surround bus corresponds to a normal submix bus with surround functionality. All tracks routed to a surround bus get 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.
  If there is no Surround master 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 hardware outputs that are specified for this purpose in the Surround Setup dialog (↗513). see Surround Sound (↗512).

- **New Surround Master**: You can also mix your project as a surround version afterwards. 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. Once you have created a surround master, you can also route only the output signal of individual objects to this surround bus and arrange them in the surround panorama independently of the track panorama settings. Read more on this in Working with multiple Masters (↗527).

Selecting and Moving Tracks

Individual tracks are selected by clicking on the track header (↗132) or into the track. When you select objects, the track of the last object you clicked on is also selected.

To switch between individual tracks using the keyboard, use the keyboard shortcuts **Alt + up arrow** and **Alt + down arrow** or menu **Track** > **More** > **Select tracks** > **Activate next/previous track**.

To add individual tracks to the selection, hold down the **Ctrl key** and click on the **track head**. To select multiple consecutive tracks, select the first track and click the track header of the last track with the **Shift** key held.

Use the keyboard shortcuts **Shift + Alt + up/down arrow** or the menu commands Menu **Track** > **More**> **Select tracks** > **Add next/previous track to selection** to add adjacent tracks to the track selection. Use the menu **Track** >

More > **Select tracks** > **Select all tracks** to select all tracks.In the mixer and project window, you can also use the keyboard shortcut **Ctrl + A**. In the project window, you must first click on a track name in the track header.

To select only the active track from a multiple selection of tracks and deselect all other tracks, use the menu **Track** > **More** > **Select tracks** > **Select only active track** or the keyboard shortcut **Shift + Alt + Enter**.

To move one or more selected tracks, click on the track name in the track header of a track and drag the mouse up or down to the desired position between the other tracks; possible insertion positions are highlighted.

Track Height

The height of a track can be adjusted individually for each track by pulling the lower edge of the track head. To set the track height of all tracks together, use the vertical scroll bar on the right (see also the topic Scrolling (↗108) above).

In the menu **Track** > **Track size** there are further possibilities:

- To maximize a track, select **Maximize track** (keyboard shortcut: **Alt + Enter**). When you run the command again, the previous track size is restored.

    ⓘ *Maximized here means sufficiently high for the volume and pan sliders to be visible in the track header.*

- The magnification of individual tracks is preserved when zooming with the scroll bar. To bring all tracks back to a uniform height, use the command **Minimize none**.

- If you activate the option **Maximize track automatically** in the menu, the selected track will be automatically maximized and minimized again when you select another track.

Deleting Tracks

To delete selected tracks, select Menu **Track** > **Delete Tracks**.

ⓘ *If other tracks are routed to these tracks, a warning is issued, also if the track still contains objects.*

Copying and Pasting Tracks

You can copy or cut tracks via the clipboard and paste them into the same or other projects.

To copy one or more selected tracks, choose Menu **Track** > **Copy Track(s)**. With menu **Track** > **More** > **Cut Track (s)** the tracks are copied to the clipboard and deleted from the project. All track properties and contents are copied. When inserting, you can specify which of them will be inserted.

With **Paste Track(s)...** you paste all tracks copied to the clipboard below a selected track.
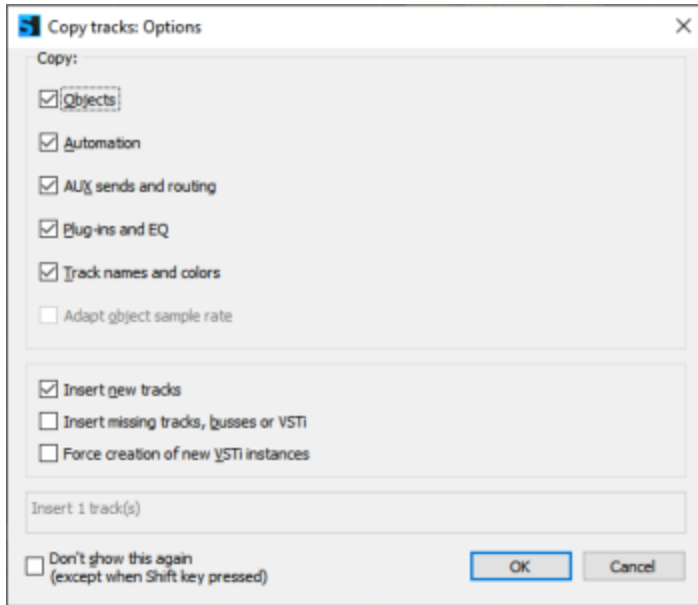
Copy: Here you can specify which track properties and contents are to be inserted.

- **Objects**: All objects including their effects and automations.

- **Automation**: All automation curves of the track. Note that always all automations are inserted. If you have automated plug-ins in the copied track but do not copy the plug-ins as well, you get automation curves without a target. These are displayed in the automation menu with **???**.

- **AUX sends and routing**: The value of the AUX send controls as well as the audio and MIDI input and output routing. Note also the **Insert missing tracks, busses and VSTi** option 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 insofar the assignment of the copied routing is based on the track names.

- **Adjust object sample rate**: This option is only available if tracks are to be inserted into a project whose sample rate differs from the sample rate of the objects of the copied tracks. In this case, the objects of the inserted tracks are adapted to the sample rate of the project using Object resampling. If you don't want this to occur, you can deactivate this option.

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.

> 💡 *Thus you can also use this function, for example, to transfer properties of one track such as plug-ins, automations or routings to another.*

If the clipboard contains more tracks than were selected in the target project, additional tracks are created if the option **Add missing tracks**, **busses and VSTi** is active.

 Add missing tracks, **busses 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 submix 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.

> ℹ️ **Attention**: *AUX and Submix busses are only created if there aren't already busses with the same name in the target project. Otherwise, the existing buses and their effect settings are used.*

Force creation of new VSTi instances: If this option is active, the aforementioned rule is overridden and a new instance of a VSTi is created, even though a VSTi with the same name already exists in the project into which the tracks are inserted.

> ℹ️ **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 buses are again routed to buses, those buses will not be inserted any more.*

The dialog remembers the settings, so for further insert operations with the selected settings you can skip opening the options dialog by setting the **Don't show this again** option in the lower right corner. If you need the dialog again, you can reopen it by holding down the **Shift** key while selecting the **Insert Track(s)...** menu command.

Selecting Multiple Tracks - Grouping Track Controls

To select multiple tracks, click on the track number or in the track name area while holding **Ctrl** key or **Shift** key.

This groups the track controls. The corresponding controls are changed on all tracks of the selection when one of the involved controls is changed.

Now, when you operate a button or control on one track, the corresponding controls of the other tracks in the group change with it. This applies to most of the track's parameters, only certain settings, such as plug-ins or the comment field, are not edited together.

> ℹ️ *This group behavior also applies in the mixer. In this case, the multitrack selection exists in addition to control groups Select and Group Multiple Controls (↗481) previously created in the mixer, if the same controls are included.*

When selecting audio inputs and outputs, all available inputs or outputs can optionally be distributed to the selected tracks.

1. Select multiple tracks.
2. Assign an output to one of the tracks, e.g. in the **Audio** section of the **Track Editor**.
3. Click on **OK** in the query dialog (Multi I/O Routing).

4. The output is assigned to the first selected track, subsequent tracks get the output with the next higher number.

> ℹ️ *If more tracks are selected than outputs are available, the assignment for the remaining tracks remains unchanged.*

To remove a track from an existing multi-track selection, **Ctrl**-click in the track name area.

As soon as you click on a track that is not part of the multiple selection, the multiple selection is resolved. If you always want to adjust certain tracks together, you can keep the multiple selection by grouping the tracks.

To do this, choose **Track Properties** > **Group Track Controls** from the **Track** menu or the Track Head context menu. The grouping of the tracks is now preserved, even if you select a different track. To ungroup, select **Track Properties** > **Ungroup Track Control**.

> ℹ️ **Note**: *Please note that a control element can only be in one control group at a time. Overlapping control groups cannot exist.*

Editing Track Properties with the Keyboard

Most of the properties of a track, such as recording status, recording readiness, mute or solo status, can be set with the mouse using the corresponding buttons in the Mixer, Track head or Track editor.

However, in the menu **Track** > **More** > **Track properties** there are also special menu items for all these properties to be able to set these properties with the keyboard.

> ℹ️ *For further information on the properties, also read the section* Channel strips *(↗482) in the Mixer chapter!*

Property	Description	Keyboard shortcut
Mute	Mutes selected tracks.	**Alt + M**
Mute/Inactive	Mutes and additionally disables the selected track.	**Ctrl + Alt + M**
Solo	Switches selected tracks solo.	**Alt + S**
Solo-exclusive	Exclusively solo selected tracks, all other tracks are muted	**Shift + Alt + S**
Record	Activates selected tracks for recording	**Alt + R**
Monitoring	Activates Manual monitoring for the selected track	**Ctrl + Alt + Shift + F**
Lock	The selected tracks are fixed (locked).	**Alt + L**

Rename track	Activates the text input field for the track name of the last selected track	**Ctrl + Shift + N**
Edit volume	Numerical input of the volume in dB.	**Ctrl + Shift + K**
Edit pan	Numerical input of the track panorama in dB (-100...100).	**Ctrl + Shift + P**
Track phase invert	Inverts the phase of the selected track	**H**

> ℹ️ *The menu contains even more entries up to the* Global Solo Modes *(↗505) to which no keyboard shortcuts are assigned by default, but which can be* assigned.

Hiding 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. To do this, open the **Track Manager** (↗324) (menu **View** > **Manager** > **Track Manager**) and deactivate the corresponding checkbox in the columns **Arrangement** and **Mixer** to hide the tracks.
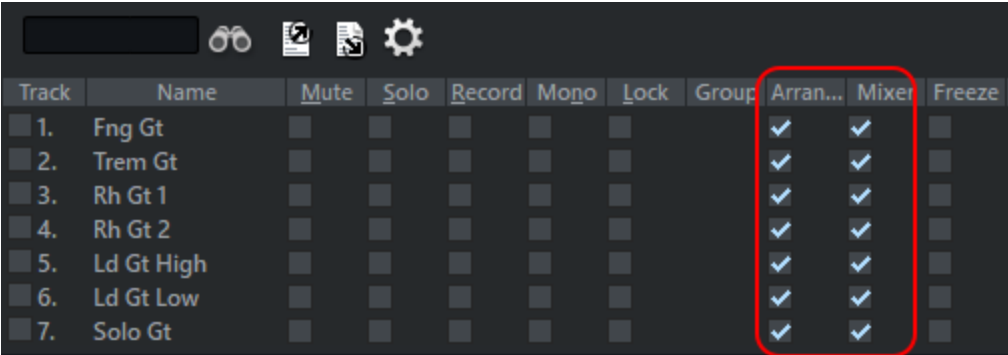
You can also select multiple tracks and hide them together using the **Ctrl** + left-click and **Shift** + left-click keyboard shortcuts.

By clicking the checkboxes in the Track Manager again, you can make the hidden tracks and channels visible again.

For more track visibility commands, see menu **Track** > **Track visibility (arranger or mixer)**. Select one or more tracks and then select one of the options:

- **Select all track routing targets**: The track selection is expanded to include the tracks (busses and masters) to which the selected tracks are routed.
- **Select all track routing sources**: The track selection (typically buses or masters) is extended by the tracks routed to these tracks.

You can execute these commands before the following ones, e.g. to hide tracks together with their buses or to display only the tracks that are routed to a certain bus.

- **Show only selected tracks**: All unselected tracks are hidden.

- **Hide selected tracks**

- **Show all tracks**

- **Synchronize Mixer and Arranger**: The visibility of the tracks in the mixer is matched to that in the arranger.

Folder Tracks

Folder tracks can help you keep a better overview of your arrangement by combining all tracks that belong together into one folder.

Combine tracks in a folder track

To combine tracks in a folder track, select the tracks, right-click on the track header and select **Move track(s) to folder track** > **Move track(s) to new folder track** from the context menu.

> 💡 Use **Move track(s) to new submix folder track** to create the folder track as a submix bus (see below).

You can drag additional tracks to a folder track by clicking on the track name or a free area in the track header and dragging the track to the track header of the folder track or by choosing **Move track(s) to folder track** in the context menu of the track and then the name of the folder track. Individual tracks can be removed from the expanded folder track again in the same way. Or you can select **Remove track(s) from folder** in the context menu.

When you delete a folder track, you can optionally delete all tracks contained in the folder as well.

> ℹ️ The context menu commands mentioned can also be found in the menu **Track** > **More** so that you can assign keyboard shortcuts to them.

Track header and track properties

The volume control is used to control the volume of all tracks together. The functions **Mute**, **Solo**, **Record Arm**, **Monitoring** and **Lock** affect all tracks contained in the folder track. By right-clicking on the Record Arm button, you can set further track properties for all tracks in the folder track collectively, up to the selection of the hardware input. This assigns multiple inputs to the tracks in ascending order, if available. Right-click on the track name to open the **Track Properties dialog**. It contains only a few options:

- **Color**: This sets the color of the frame around the tracks in the arranger and mixer.

- **Track type Submix Bus**: If you activate this option, the folder track additionally becomes a submix bus and the contained tracks are routed to this bus. This means:

  - The volume and pan controls of the folder track no longer control the individual tracks, but the properties of the bus.

- The folder track can also contain track effects, be frozen and so on, in short, it becomes a normal submix bus.
- It also has the properties of a folder track in that it can be expanded and collapsed and can display the objects of the tracks it contains.

- **Audio recording Interleaved**: This option can be used to record multi-channel files, e.g. in surround formats, into a single audio file. Conversely, when loading such multi-channel audio files, folder tracks can also be created automatically, with this option then active. In this use case, the folder track is not used to bundle tracks to facilitate shuffling, but as a marker for a group of tracks containing such objects.

Display

With the folder icon at the track header you can open and close a folder track.

For keyboard operation *of this function, use menu command* **Track** > **Track Size** > **Display subtracks**.

When **collapsed**, the tracks it contains are hidden below the folder track in the arranger.
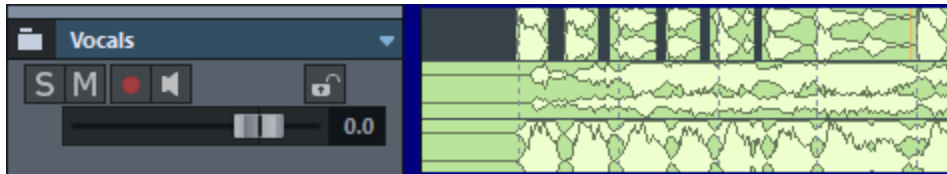
i *In the Mixer, the tracks are still displayed, but are indicated by the frame color of the folder track.*

When **expanded**, the contained tracks are displayed below the folder track. They get a frame with the track color of the folder track to indicate their belonging to the folder track.
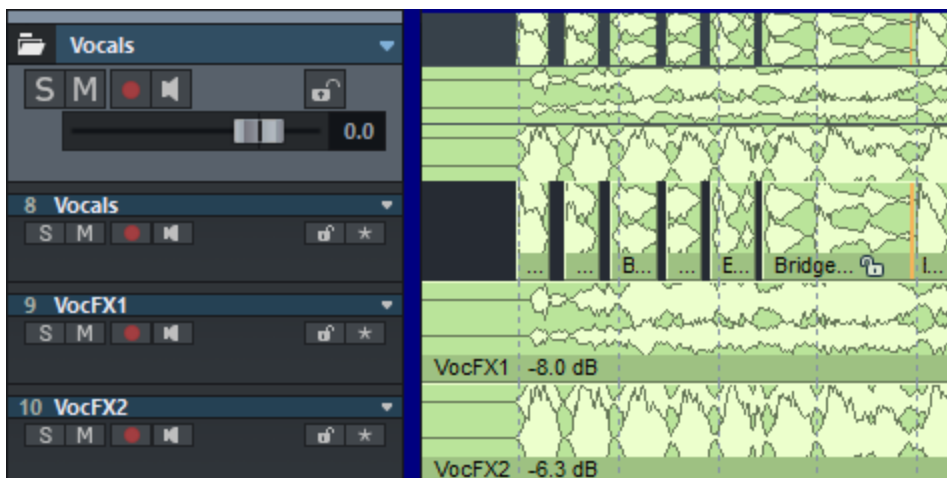
Cut operations in folder tracks

In the folder track itself, all objects of the contained tracks are displayed in a minimized view by default. You can select ranges in the folder track in the same way as in a simple track; this range selection then always spans over all the tracks it contains. Range-based editing operations in the folder track therefore always affect all tracks of the folder track.

You can select one of the tracks contained in the folder track as a **reference track** by clicking the triangle next to the track name.
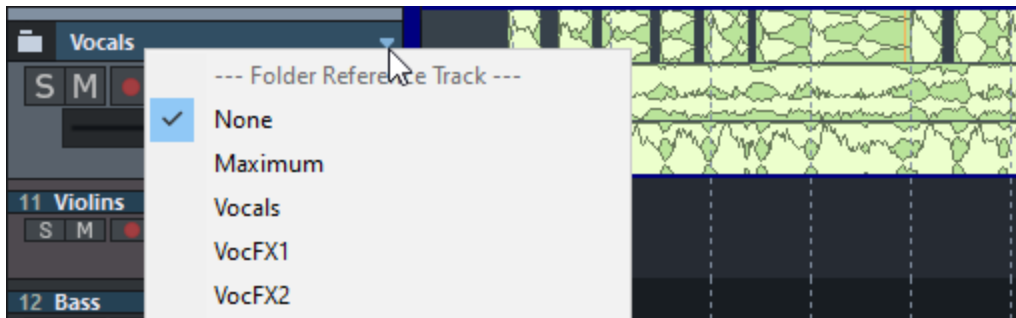
The objects on the reference track will then be displayed in the folder track and can be edited there in the usual way.
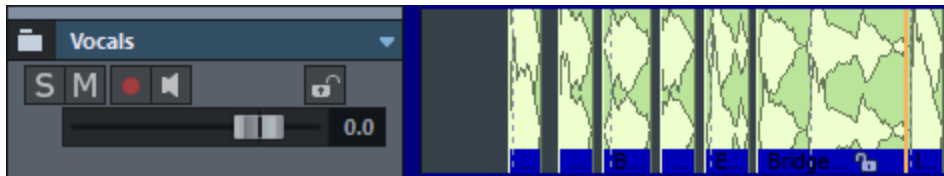
> **ⓘ** *However, please note: Even if only the objects of the reference track are displayed in the folder track, cutting operations with ranges still affect all tracks of the folder track!*

When selecting **Maximum**, no reference track is selected. Instead, a waveform display of the loudest signal of the tracks at a given time is displayed.

Revolver Tracks

Revolver tracks are variants of a track in terms of the objects placed on it. You can thus quickly compare different variants of recordings with each other.

✱ With this button at the track head or **Ctrl + right click** somewhere on the track head you open the revolver track context menu, in which the associated commands are located.

With **New revolver track (copy)** you create an initially identical copy of the existing track. Now you can edit the objects and thus create a new cut version of the track.

The revolver track button lights up to indicate that there are revolver tracks on the track. An asterisk appears in front of the track name.
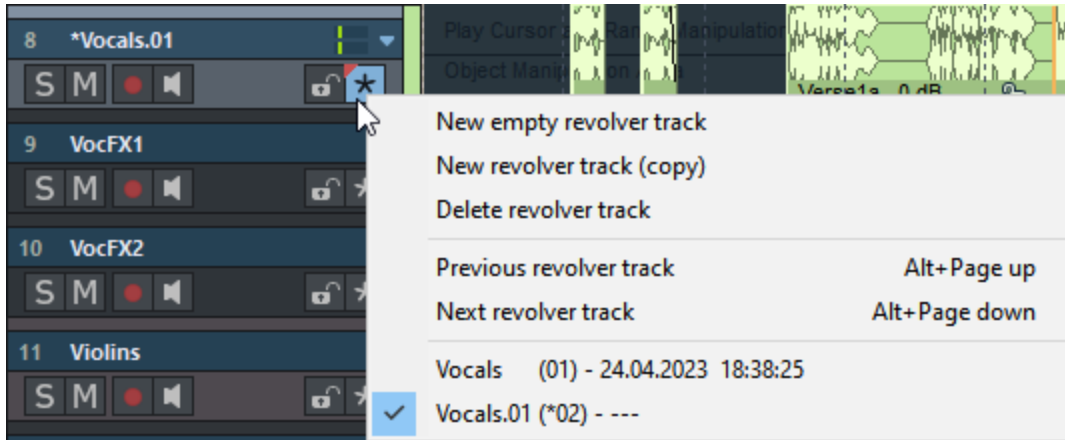
In the lower part of the menu you can directly select the different revolver tracks. Use the keyboard shortcuts **Alt + Page Up / Alt + Page Down** (Previous revolver track/Next revolver track) to switch between revolver tracks.

Of course, the context menu may also be used to create a new, empty revolver track or to delete the current one.

Take Lanes

> To understand the function of the Take Lanes, we recommend that you also read the introductory explanations on the topic **Takes** in the section Take Manager (↗328)!

The **Take Lanes** can be used to display the individual recording passes (takes) for the objects of a selected track one below the other. This allows you to combine the best parts of each take of an object to create the perfect shot. This function is also called "comping".
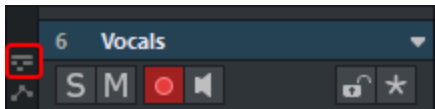
 You open the Take Lanes of a track with this symbol at the track header or with the keyboard shortcut **Ctrl + Shift + Enter**

In the Take Lanes, all existing takes of the objects of the track are displayed below the respective object.

The active take can then be selected by simply clicking on a take. There are two ways to combine sections from different takes:

■ In **Universal Mouse Mode** : Click and drag over a take to insert that section of the take into the track. The object in the track is split at the corresponding positions and the take is changed in the newly created object.

The sections of the takes used for each object are highlighted in the Take Lanes. Click in another Take Lane below or above a section to select a different take for that section.

- In **Range mouse mode** : the section of the take is also selected by clicking and dragging, but copying to the track is not done until you press **Shift + C** afterwards. Also, in this mouse mode it is possible to select ranges in takes and copy them to any place in the track using **Copy** and **Paste** (Ctrl + C / Ctrl + V).

The takes are always based on the original recording positions and are synchronized in time. If different timings occur in the takes (e.g. varying onset of a singer) and you move an object, this change also affects the other takes, so that the timing would have to be corrected again when you change the take again. So it's better to finish comping the takes first and correct the timing errors afterwards.

> ⓘ *With copy/paste in the range mouse mode you can also correct the timing already during comping, since in this case the audio material does not have to be inserted synchronously in time, but you should not change the takes afterwards either.*

Like changing complete takes in the Take Manager, comping in the Take Lanes works across multiple tracks: If there is a group of objects in multiple tracks after a multitrack recording, you can comp in one track in the Take Lanes, and the takes in the other tracks will also be swapped.

This is based on the object grouping. If you don't want to swap takes for one or more tracks or want to combine them differently, break up the group and create other object groups.
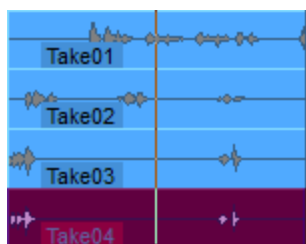
 If an object group does not have corresponding takes on all tracks, you can recognize this by the fact that these takes are displayed in a different color in the take lanes of the tracks in which they are present.

If you activate the Solo button at the take lanes **S** , you will play the complete take of the corresponding lane instead of the objects of the track.

Track Freeze

Use Track Freeze to render the selected track into a wave file that replaces all objects in the active track. The object and track effects are included so that the CPU is relieved of computationally intensive real-time effect calculations. The frozen track is stored in an internal VIP and can also be edited there. When "unfreezing", these edits are transferred to the source project.

To freeze a track, select **Track > Track freeze > Freeze track** from the menu, or right-click the track head and select **Track Freeze** from the context menu (keyboard shortcut: **Alt + Shift + F**).

The track is rendered into a wave file with the current volume, all effects and automations, and then replaced by a track in default setting containing a single object with this wave file.

> **ⓘ Note**: *The created object is slightly longer because the decay/reverberation time is included. The length of the reverberation time when freezing can be specified by the parameter* **Maximum reverb time for objects without fade out** *in* **Program preferences** > **System Options** > **Playback** *(↗702).*

With Menu **Track > Track Freeze > Edit Track Freeze** you open the internal freeze project where the frozen track is stored. You can make any changes there, just like in any other project, from object editing to effect calculations in object and track to inserting other audio files. When you save the Freeze project, it is rendered and the changes are applied to the source project.

With menu **Track > Track freeze > Unfreeze track** (keyboard shortcut: **Alt + Shift +U**) you unfreeze the track. The wave file created during the track freeze is removed and replaced by the track from the freeze VIP. Changes made in the freeze project are taken over to the track.

> **ⓘ Note**: *If you have added additional tracks in the Freeze project, you will not be able to run* **Unfreeze track**.

Freeze Track for AUX Busses/Submix Busses

Submix and AUX bus tracks can also be frozen.

Submix Bus

- Tracks that have been routed to the frozen submix bus remain unchanged.
- The Submix bus input is muted.
- The file created while freezing is inserted into the submix track as an audio object.
- The outputs of tracks that are routed to the submix bus are marked with red frames in the Track Editor and Mixer.
- Changes made to these tracks after the freeze will have no effect on the playback of the frozen submix bus.

AUX bus

- The AUX send settings of the tracks are taken into account during the freeze, but not the Object AUX sends.
- Tracks routed onto the frozen AUX bus remain unchanged.
- The AUX bus input is muted.
- The corresponding AUX send controls of the tracks are marked with red frames in Track Editor and Mixer.
- The file created while freezing is inserted into the AUX track as an audio object
- Changes made after freezing on tracks sending to the AUX bus will not affect playback of the frozen AUX bus.

If you have changed anything in tracks that are routed to a frozen Submix or AUX bus, you should "unfreeze" the bus with the command **Unfreeze track** and immediately freeze it again at the current state with Freeze track.

Freezing VST Instruments

You can also freeze tracks with software instruments. For a simple routing (↗372) (audio+MIDI in one track) you can simply freeze the track as described above. You can then use the function **Edit Track Freeze** to make changes to the track's MIDI object.

As soon as the VSTi uses multiple tracks for audio or MIDI and additional individual outputs are used, you must use the Freeze function in the **VSTi Manager** to ensure that all tracks involved are frozen.

To do this, select the instrument in the VSTi Manager and go to **Functions** and select **Freeze**. This replaces the audio signal of all audio return tracks with audio objects. The frozen instrument will no longer be controlled by tracks with MIDI files and will be disabled (↗382).

Unlike a simple track freeze with audio files or a VSTi hosted within a track, the MIDI data is not part of the freeze project. Therefore, to edit the freeze data of the individual outputs, you cannot use the function **Edit track freeze**. (There is no data at all on these tracks).

The MIDI objects are still in the project, but their contents are not used and later changes in them do not affect the instruments output. So if you want to make changes to the MIDI data of frozen VSTi, first select the **Unfreeze** option in the VSTi Manager, make the desired changes to the MIDI object, and then freeze the instrument again.

> ⚠ *You could also freeze the individual tracks of the instrument outputs this way, because tracks can be frozen even if they do not contain any objects. This creates an audio file from the audio of the single output and the effects in this track. Since the instrument and also the corresponding MIDI track are still active, you would then have to deactivate the relevant individual output directly in the instrument in order to avoid the track being played twice. For reasons of clarity, we do not recommend this procedure at all!*

Track Options Dialog

The dialog **Track Options** shows all settings of the selected track. Most of the settings here can now be adjusted more quickly in the Track head, Track editor or Mixer, it is in this respect a legacy dialog of older Sequoia versions with a simpler interface. But it also contains a few controls that make it easier to set certain track properties for several tracks together.

To display the Track Settings dialog, right-click Track Name or Track Number in the Track Head or Track Editor, or select Menu **Track** > **Track Options...** (keyboard shortcut: **Alt + I**).
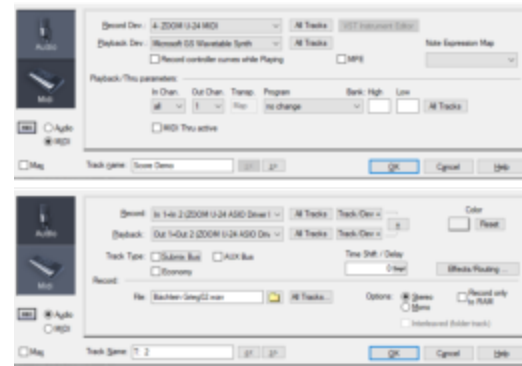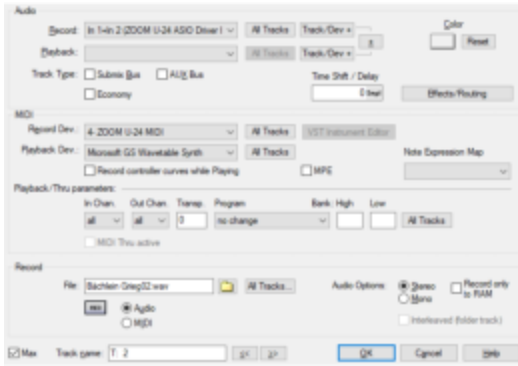
The **Max** option in the lower left corner of the dialog can be used to switch between two different views, where the settings for audio and MIDI are displayed either on two different views Audio and MIDI or together in one view.

In the field **Track name** at the bottom the track name can be displayed and changed, with the **<</>>** buttons you change the displayed track.

Audio

Here you make settings for recording and playing back audio on the tracks.

- **Record/Playback**: Recording and playback device for the track. Submix busses can also be selected for playback.

- **All tracks**: The recording or playback device selected for this track is used for all tracks.

- **Track/Dev +**: Use this button to assign successive input or output channels to successive tracks (channel pairs for stereo tracks). Clicking this button switches to the next track and simultaneously sets the next device for this track, e.g. from an output pair 1+2 to 3+4.
  The **+** - button next to the two **Track/Dev +** simultaneously switches to the next recording and playback device.

  > ℹ️ *If there are no further devices, the assignment for the next tracks is started again from the first device.*

- **Color**: Setting the track color and the color of objectsRenaming and coloring Objects (↗160) placed on it.

- **Track type**: This indicates if the track is an AUX or Submix bus (↗495). A single track can also be both.In addition, a track can also be marked as a Timecode track (↗602) or a Economy (↗694).

- **Timeshift/Delay**: Use this field to set a time offset for an entire track. Positive values cause the playback of this track to 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.

- **Effects/Routing**: This button opens the Effects routing dialogEffect Routing Dialog (↗220) for the corresponding track.

MIDI

Here you make settings for recording and playing back **MIDI** on the tracks.

- **Record Dev./Playback Dev.**: MIDI input and output device selection. The list of playback devices also includes all VST instruments.
- **All tracks**: The recording or playback device selected for this track is used for all tracks.
- **VST Instrument Editor**: Opens the graphical interface of an included VST instrument.
- **MPE**: Sets MIDI recording to MPE mode (↗467)
- **Note Expression Map**: Here you can select a VST3 Note Expression Map (↗469).

Record controller curves while playing: If this option is active, incoming MIDI CC events (controllers) are written to automation curves for these controller values during playback. This is a legacy option that was once used for MIDI remote control of digital mixer hardware. Attention, the function is independent of whether recording is active for a track.

Play/Thru parameter: For detailed information about these parameters see Track Editor > **MIDI**!

MIDI Thru active: Corresponds to activating monitoring (speaker symbol) in audio tracks, i.e. incoming MIDI data is forwarded to the output.

Recording

The section "Record" contains settings for recording audio and MIDI.

Rec corresponds to the recording readiness for the track, **Audio/MIDI** determines whether you want to record MIDI or audio.

File: File name of the audio file to record to on this track. Since these names are only used internally, as long as you do your production entirely within Sequoia, you can leave these names at the defaults. If you want to use the recorded raw material elsewhere or pass it on, you have the option of naming these files according to specific naming schemes for better identification. To do this, click **All Tracks** and choose an option from the menu:

- **Filename_track number**: Enter any filename in the field **File** and select this option to use this name on all other tracks, appending the respective track number, e.g. funky_01.wav, funky_02.wav..
- **Project name_track number**: (default) Uses the project name on all tracks, supplemented by the track number, e.g. demo_01.wav, demo_02.wav..
- **Track name**: Uses the track name of each track as file name, e.g. Drums.wav, Bass.wav...

- **Track number_Track name**: Uses 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.
- **Track number_take name** and other options with take name: These options correspond to the optional Recording setting (↗699) **Save each take in a new file**, but allow a bit more flexibility in naming the files.

The last two options are also in the menu, but they are not used to define the file name, but vice versa to define the track names based on the file or object names:

- **Set track name to first object name**: Tracks are named after the name of the first object in the track.
- **Set track name to first object file name**: Tracks are named after the audio file name of the first object in the track.

With these commands you can name the tracks according to the file names after importing several files one below the other, e.g. stems from a remix kit.

Audio options

- **Stereo/Mono** determines whether the recording on the track should be in **Stereo** or in **Mono**. This also determines whether you can select individual inputs or input pairs on the recording device.

  ℹ️ *In the track's* input routing*In (Input) (↗487) (right-click on* **Rec** *in the track head) you can also specify stereo processing for tracks recorded in mono (* **Mono In / Stereo FX***).*

- The option **Record only to RAM** allows recording to be done directly in the system RAM.
- **Interleaved (folder track)** is only available for folder tracks and activates a interleaved recording (↗101) for them.

OBJECTS

The audio data is represented in Sequoia projects by **objects** in the tracks. Objects are playable and editable entities in the project window of Sequoia, which can be moved, copied or cut on the tracks as desired.

An object is an representation of an audio file or part of it and contains references to the underlying audio files, which are visualized in a waveform display. An audio object accesses audio data and adds a variety of object-specific settings in real time: Volume, pan, object length, sound changes, fades, pitch changes, time stretching, AUX send taps, and effects. All these settings are only added to the original material afterwards, which itself remains untouched on the hard disk (non-destructive editing).

> ℹ️ *MIDI data is also represented by objects. MIDI objects contain playback instructions for MIDI devices or VST instruments in MIDI format. Unlike audio objects, MIDI data is stored together with the objects and is directly linked to the respective objects. For MIDI objects, read the chapter* MIDI *(↗417)!*
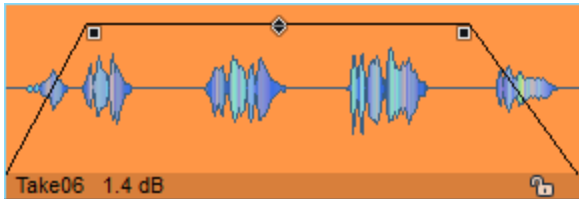
There are various **handles** (↗147) on the objects that can be used to directly change the most important object properties: start time, length, fade-in and fade-out, and object volume. Other object properties, such as panorama, AUX send, object effects or time stretching can be set in the **Object Editor** (↗183).

Objects that overlap in a track can be blended into each other. For more precise setting of such a crossfade there is a **Crossfade Editor**.

With the lock icon at the bottom you can lock an object and thus protect it against accidental changes.

> ℹ️ *The visual appearance of the objects can be configured very flexibly. You can access the display options via the menu* **View** > **VIP display** > **Project display...** *or with the keyboard shortcut* **Shift** + **Tab***. For detailed information on the project display options, see the section in the* **System options** >**Project display**.

Selecting Objects

Selecting/Unselecting Single Objects

Click on an object to select it. If an object is selected, it changes its background color and handles appear at the corners and in the middle of the object.
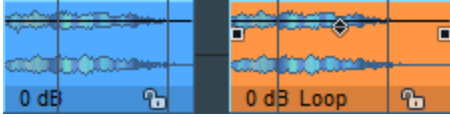
The selection can be canceled by clicking with the **Ctrl** key held down or by clicking next to the object.

Selecting Multiple Objects/Unselecting Individual Objects/Inverting Selection

- Select several individual objects by holding down the **Ctrl** key and clicking on them.

- To select several consecutive objects, click the first object and then, while holding **Shift** key, click the last object. This will also select all objects that are between the first and last selected objects.

- **Lasso selection**: To select all objects in a certain screen area, left-click in the lower part of a track on a free space and draw a selection frame (lasso) around the objects.

  > ℹ️ *The objects do not have to be enclosed by the lasso, it is enough if they are touched by the lasso.*

- If there is no free space between objects, it is sometimes difficult to drag a lasso without moving an object. In this case, select the menu command **Object** > **Select objects** > **Object lasso** (keyboard shortcut: **Ctrl + Alt + L**). After that, you can click and drag once in an object without moving the object as usual, but dragging a lasso to select the object. After the selection, the mouse resumes its old behavior: The next click and drag on an object has the usual effect again - the object is moved.

- To select all objects of a track, **double-click** in the lower part of a track on a free space.

- To deselect a single object of a multiple selection, click on it while holding the **Ctrl** key.

- To deselect all objects, click in a free area in the lower part of a track next to the objects. Also if you click on an object that has not been selected so far, the existing selection will be canceled and the new object will be selected.

> ℹ️ *Note: The phrase "into the lower part of the track" refers to working with the* Universal mouse mode. *In this mode, ranges are selected in the upper part of the track. In other mouse modes (object mode, object/curve mode) the entire track can be used.*

More Commands for Object Selection in the Menu Object > Select Objects

- **Select objects under play cursor/range**: All objects located under the play cursor or within a selected range will be selected.

- **Select all objects**: All objects in the project are selected.
  Keyboard shortcut: **Ctrl + A**

- **Select previous/next object**: With these commands you can navigate through the objects of a track with the keyboard. In conjunction with the keyboard commands for moving objects step-by-step (↗151) and changing object start, fade, and length, you can edit the objects completely keyboard-controlled.
  Keyboard shortcut: **< / >** or **Ctrl + Alt + Q** / **Ctrl + Alt + W**

- **Add previous object/next object to selection**: This adds the previous or next object of the track to the selection.
  Keyboard shortcut: **Ctrl + Alt + Shift + W** / **Ctrl + Alt + Shift + W**

- **Toggle selection**: An existing object selection can be reversed (inverted). This means that all unselected objects will be selected and all selected objects will be deselected.

- **Remove object selection**: This allows you to cancel the object selection without a mouse click.
  Keyboard shortcut: **Ctrl + Shift + A**

Object Handles

If an object has been selected, handles appear at the upper corners and at the center of the object. These handles allow you to set the most important object parameters directly in the project window.
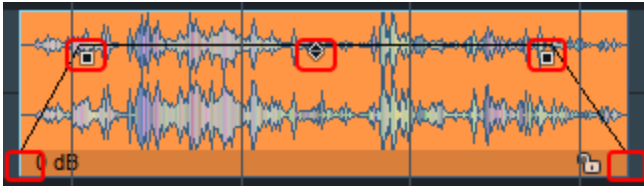
- To fade an object in and out, drag horizontally on the **fade handles** top left and top right. Pull vertically on the handles to adjust the curve shape. You can use the Object Editor to precisely adjust the curve shape.

- The **volume handle** at the top center of the object can be used to change the object volume.
  The volume handle can also be configured in the view options in a way that the height of the handle corresponds to the set object volume. To do this, deactivate the **Volume handles on top** option in the **Object** section.

  > 💡 *The object volume can also be changed at the numerical display by dragging the mouse vertically.*

- The **lock symbol** is used to lock objects (↗154).

- The **length handles** are at the bottom left and right corners. They can be used to change the length of an object and the start time in the referenced audio file.
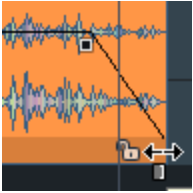
  By default, the length handles are hidden by the object footer. However, when you move the mouse over the bottom ends of the object, you can tell when you can use the length handles by the change in the shape of the mouse pointer.

> ℹ️ *Note: If two objects have a* crossfade *(↗169), the crossfade can be moved with the mouse at the leading edge of the second object. However, the length handle of the object is still available. It is located at the bottom of the object directly at the object edge. Pay attention to the change in the shape of the mouse pointer!*

Object Handles with Multiple Selections/Object Groups

If several objects or a group of objects have been selected, the following rules apply when moving the object handles:

- The volume handles are moved together, a change to one involved volume handle is applied to all objects of the selection/group.

- Object start, object end and the fades can be set separately for each object. To set the handles for all objects of a selection/group, press the **Alt** key before clicking the handle.

Grouping Objects

Multiple objects selected together can be edited together. However, such a multiple selection exists only as long as no other object is selected. In order to always select several related objects together, they can be combined into a **group**. When you select an object of an object group, all objects of the group are always selected as well. They can now only be moved as a group; the distances between the objects in the group are retained. All changes you make to an object are applied to all group members.

> ℹ️ *Note: The same restrictions apply here regarding the editing of object fades as for a* multiple selection of objects*Object Handles with Multiple Selections/Object Groups (↗148).*

Grouping and Ungrouping

To group objects, select all objects to be grouped and either click the "Group" icon in the toolbar or select **Object** > **Groups** > **Group objects** in the menu or in the context menu of an object.
Keyboard shortcut: **Ctrl + G**

To ungroup the group of objects, select it and either click the "Ungroup" icon in the toolbar or choose **Object** > **Groups** > **Ungroup objects** from the menu or the context menu of an object.
Keyboard shortcut: **Ctrl + U**

Each object group is assigned an internal number in sequence. This number is displayed in a dedicated column in the Object manager (↗321). Optionally, you can display this number on the object by activating the **Group number** option in the project view options (↗716) (keyboard shortcut: **Shift + Tab**) under **Objects** There you can also activate the option **Group colors** at **Waveform color**. Then the objects of each group get their own automatically assigned color. This will ensure that you can easily distinguish the groups from each other.

Temporarily Detaching Objects from a Group

Sometimes you want to edit the object position or volume of a single object of the group separately. Normally, you would have to ungroup, edit the individual object, and then regroup all objects. To make this tedious and error-prone procedure easier, there are some special commands:

- With menu **Object** > **Groups** > **Temporarily exclude object from group** (keyboard shortcut: **Ctrl + Shift + U** or Shift + click on the "Ungroup" button) you ungroup the last clicked object from an existing group. The object can now be selected and edited separately, but remembers that it belonged to a group. You can also edit other objects as you go along, and even detach other objects from this or any other group. If you select this object later and execute the function again, the object will be added to the group again.

- With menu **Object** > **Groups** > **Temporarily exclude all objects from groups** (Shift + Alt + click on the "Ungroup" button) you temporarily ungroup all objects from their groups. Now all group objects can be edited separately.

  ⓘ *To indicate this state, the button for "Ungroup" flashes.*

The groups are restored by calling the function again or by pressing the flashing button for "Ungroup".

ⓘ *When saving the project, the temporary states are ignored and the project is saved as if all objects are in their groups.*

You can also combine groups with other groups or objects to form higher-level groups. When such groups are ungrouped, the grouping status of their elements is retained. However, this only affects the top level. Further ungrouping of the groups always leads to individual objects, regardless of how deeply nested the group structure was before.

This allows you to combine objects that are grouped vertically after a multi-track recording into horizontal groups.This means you can also combine the objects that are grouped vertically after a multi-track recording into horizontal groups. When these groups are ungrouped, the vertical grouping is retained.

Moving Objects

Moving with the Mouse

Simple moving of objects with the mouse is possible in the mouse modes **Universal mode** , **Object mode** and **Object curve mode** : Click with the left mouse button on the object (in Universal mode: the lower half of the object) and drag it the desired place. You can also move an object to another track.

> -♡- *When moving objects to other tracks, the time position is preserved if you press the **Shift** key while moving.*

If you want to move an object beyond the visible section, first drag the object in the appropriate direction and move the mouse pointer onto the scroll bars at the window borders. The arrangement scrolls in the corresponding direction and you can move the object to the now visible tracks or time sections.

To avoid accidentally moving objects already when selecting them, activate the option **2nd click required to move object** in the **Program preferences** under **Keyboard/Menu/Mouse** > **Mouse**. Then objects have to be clicked again extra to move them, the first click only selects the object.

For multiple selections or groups, all involved objects are moved together when one involved object is moved. But be careful: If one of the objects is locked, no object can be moved!

> 🛈 *When moving objects, the* object modes (↗155) *also apply. This means that, for example, in mode, **Link one track to the right** all subsequent objects are moved as well.*

Moving to a Specified Position

The following options can be used to move an object to a specified position:

- Menu **Object** > **Move Objects** > **Move Object...**: This allows you to enter start position for a selected object numerically. With the unit field on the far right of the dialog you can select the unit of measurement used. For a multiple selection or group of objects, the foremost and topmost object of the selection is moved to the position and the other objects are moved by the same amount.

- In the **Object editor** you can also enter the desired position in the **Fades** view at **Object start**.

- In the **Object manager** you can set the object position in the column **Position**.

- Menu **Object** > **Move objects** > **Object start/object end to play cursor position**: The selected objects are moved to the play cursor position. For a multiple selection/group, the start of the foremost object is taken as the reference point, regardless of the track; for "Object end to play cursor position", the end of the rearmost object is used.

- **Object hotspot to play cursor position** moves an object so that its hotspot (↗154) is placed at the play cursor.
  Keyboard shortcut: **Ctrl + Alt + P**

  > ℹ️ *If multiple objects are selected, the hotspot of the foremost object is considered the reference point, even if other objects have hotspots that are further ahead. If the foremost object does not have a hotspot, its object start applies, even if other objects have hotspots.*

- Menu **Object** > **Move objects** > **Object start to end of previous object**/**Object end to start of next object**: A selected object is moved to the previous/next object so that it is directly adjacent to it. In the case of a multiple selection/group, the active object is the reference point.

- Menu **Object** > **Move objects** > **Arrange objects...**: This allows you to enter the pause between multiple objects numerically and set it to a uniform value.
  Keyboard shortcut: **Ctrl + Shift + Alt + A**

  > ⚠️ *Attention: the function works in such a way that the pauses before the selected objects are changed. This also affects the first selected object (exception: it is the first object on the track). So if you only want to change the distances between several objects, do not include the first object in the selection.*

Moving Objects Step by Step

In the menu **Object** > **Move Objects** > **Object Move Step 1/2** you will find a large number of commands that allow you to move objects, change start and end time, volume, fades and much more. This menu is not designed to be operated with the mouse, but contains a series of keyboard shortcuts that follow a logical principle:

Select one object for simple object editing. If you have selected two objects for editing, you can edit them both independently of each other as well as together.

Each of the commands below is available in two step sizes. The step sizes can be set with menu **Object** > **Move objects** > **Object/Fade step settings...**.

Step Width for Object Move Commands ×

Step Width 1

Samples:	4.410
Millisec.:	100 ms
SMPTE:	00:00:00:02
Bars:	000:00:076

Step Width 2

Samples:	8.820
Millisec.:	200 ms
SMPTE:	00:00:00:05
Bars:	000:00:153

OK

Cancel

Keyboard shortcuts with the **Ctrl** key edit the **left object**, with the **Alt** key edit the right object. The commands in the following table are executed with **Step size 1**. To execute the commands with **Step size 2**, additionally hold down the **Shift** key. Use the number keys **0...9** to determine the 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*
Move object(s) left	Ctrl + Alt + 1
Move object(s) right	Ctrl + Alt + 2
Object(s) track downwards	Ctrl + Alt + Shift + Arrow down
Object(s) track upwards	Ctrl + Alt + Shift + Arrow 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
Move crossfade left	Ctrl + Alt + 3
Move crossfade 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
Move Object(s) content right	Ctrl + Alt + 0
Additional key for step size 2	Shift key

*If only one object is selected, the commands for moving the second object (**Alt** key) also affect this object, but take the object mode into account, i.e. subsequent objects are moved if a corresponding object mode is set.

> 💡 *A simpler alternative for moving objects step by step can be found in the **Object Editor**. In the **Fades** section, in the **Position** area, you can click on the arrows next to the position values and thus change these positions step by step according to the step size set below.*

Original Position

The original position of objects is the time position at which they were originally recorded. It is stored in the audio files and is important for assigning audio material to recording takes and thus for the function of the Take Manager (↗328).

With menu **Object** > **Move objects** > **Object to original time position** or keyboard shortcut **Ctrl + Alt + O** the object is moved to the original recording position.

With menu **Object** > **Move objects** > **Set original time position** the current position is set as original position.

-💡- *Only objects created in Sequoia by a recording have an original position, objects created by a file import do not. So you can use this command to integrate imported files as takes into a recording session.*

With **Edit original time position...** you edit the original position numerically.

Moving Audio Material under the Object

If you also press the *right* **Ctrl key** when moving an object, it is not the object that is moved, but the audio material under the object. However, for this to work, audio must still be available outside the object boundaries, i.e. the object must be shorter than the audio file it refers to.

Similarly, dragging the length handles on the left or right edge of the object while holding down the right **Ctrl** key moves the object start or end along with the audio. In other words, the object is lengthened or shortened, but starts or ends at the same point in the audio.

If you additionally hold down the **Shift** key to the right **Ctrl** key, you move the opposite object edge together with the audio material.

Hotspot (snap point)

Sometimes the start of an object does not align with the beat in the arrangement, for example, if a recording includes an upbeat, an accidental, or a snare pickup. To be able to move such an object to fit the beat grid, you can use the **Set Hotspot** function to give the object a special snap point that serves as a reference point for the snap (↗79) instead of the leading edge.

Select an object and place the play cursor at the desired position over the object. Select **Object** > **Hotspot** > **Set Hotspot** in the menu or the keyboard shortcut **Shift + H** to set the hotspot within the object.

A vertical line represents this in the object. From now on, this object snaps to its hotspot at the respective snap positions.

Use **Delete hotspot** to delete the hotspots of the selected objects.

-💡- *Use the Object hotspot to play cursor position (↗150) (keyboard shortcut: Ctrl + Alt + P) function to move a selected object so that its hotspot is on the play cursor position.*

Locking Objects

By locking you can protect objects against unintentional changes.

To lock an object, click the lock icon at the bottom of an object or choose Menu **Object** > **Edit objects** > **Lock Objects**.

Clicking again on the lock icon or menu **Object** > **Edit objects** > **Unlock objects** unlocks the object again.

By default, locking prevents objects only from moving. In the program preferences for **locking objects** (↗708) (menu **Object** > **Lock objects** > **Lock definitions...**) you can define further object properties (deletion, change of length, fades or volume...) to be locked for locked objects.

 To lock all objects on a track, click the lock icon in the track head or track editor. Right-click on a lock symbol to assign the track to one of 16 lock groups. Membership of such a group can be identified by the different colored dots on the lock symbol . Clicking on the lock symbol in a lock group locks or unlocks all members of the lock group.

You can temporarily disable the lock function by holding down the **Alt** key while clicking objects.

Object Mode

The object mode determines whether subsequent or previous objects are moved as well when an object is moved or ranges are inserted or removed from the project ("Ripple").
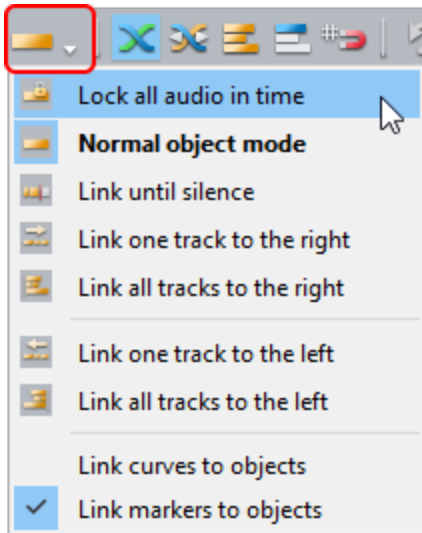
ℹ️ *You can also select the different object modes from the **Edit** > **Object Mode** menu.*

Normal Object Mode

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

Lock All Audio in Time

This mode locks all objects against moving and thus prevents objects from being moved accidentally. By simultaneously holding down the **Alt** key while clicking on an object, the lock can be temporarily removed.

> ℹ️ *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 *(↗708), you can specify against which changes exactly locked objects should be locked. You lock all objects of a track with the lock symbol at the track head.*

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 parts of a project, if separated by pauses, remain unaffected from moving.

Link One Track to the Right

In this mode, all objects on the selected 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.

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.

> ℹ️ *This object mode is an option, which means that it can be combined with other object modes.*

Link Markers to Objects

With this option, when you move objects in the top arranger track, you can also move the markers with them. Moving objects in object mode **Link all tracks to the right** then moves the markers independently of the track of the moved object.

Using the submenu **Connect markers** in the **Markers** menu you can specify in detail how existing and newly set markers behave when moving objects. More info under Markers (↗114).

Temporary Change to Other Object Modes

By pressing and holding certain keys, you can temporarily switch to another object mode. This way you can combine several modes without much clicking.

> 💡 In the **Program preferences** (↗709) in the section **Keyboard/menu/mouse**, under **Special keys**, you can specify other keys for switching or set additional temporary key functions for switching object modes.

With the **K** key the object mode **Link one track to the right** is temporarily activated, with **L** the object mode **Link all tracks to the right** is activated. So you could also work in a way that you only use the normal object mode and activate the rippling modes with the keyboard if needed.

The special keys also work in reverse: If, for example, the object mode **Link all tracks to the right** is active, the normal object mode can be temporarily set by pressing the **L** key.

Splitting Objects

Use this option to split an object into two independent objects. To split objects, use menu **Object** > **Edit objects** > **Split objects** or the keyboard shortcut **T**.

If additionally a range (↗118) is selected over the objects, the split occurs at the two range edges edges, if necessary.

The following rules apply:

- If there is a range selection, all objects located under the range edges will be split, regardless of the object selection.
- Without range selection, selected objects are split at the play cursor.
- If no object is selected under the play cursor, only the object on the current track will be split.
- All objects created by the object split will be selected.

*This way you can also use the function to cut off a part of an object at the end by placing the play cursor on the desired cut edge on the object and then pressing the **T** and the **Del** key.*

When **Auto crossfade mode** is active, a short crossfade is automatically created. When using the command **Split objects with alternative (linear) fade** alternative crossfade settings are used for this crossfade. These settings are specified in the Crossfade Editor Preset (↗178) **Default_Crossfade_Alt**.

Tip: The two new objects maintain their object effects. For example, you can add variety to a loop by first applying an effect to an object as a whole, then cutting it into individual parts, and then varying the effect parameters in the individual parts.

Splitting Objects Automatically

In the menu **Object** > **Edit objects** or menu **Edit** > **Split** you can find further functions to split objects:

- **Split objects on marker positions**: Selected objects are split at the project marker positions. The newly created objects are named after the markers that are at the beginning of these objects.

- **Split objects on track marker positions**: Selected objects are split at CD track indexes in the project.

- **Split objects on audio marker positions**: Selected objects are split at audio markers (↗116) in the object.

Healing/Unsplitting Objects

Use the command Menu **Object** > **Edit objects** > **Heal/Unsplit Object** to rejoin split objects without additional audio rendering. To do this, the objects in the VIP must be positioned exactly beside each another, and the audio material must be connected. Thus, the objects must refer to different but directly consecutive audio material of the same audio file. Without these conditions, the command will not be available. You can then use the command **Glue objects** for this, a new file will be created by this containing the contents of the two objects.

Typically these conditions are given directly after the splitting of an object, undoing unwanted object splits is therefore also a main purpose of this function.

Select the previously split objects and execute the command. All object properties of the second object, like fades and effects, will be discarded. The resulting object retains only the properties of the first object.

Copying, Pasting and Duplicating Objects

Copying and Pasting Objects via the Clipboard

Objects can be copied, cut and pasted via the clipboard, which is also called "clip" in Sequoia.

The menu entries (Menu **Edit** > **Copy**/**Cut**/**Paste**/...) and the corresponding keyboard shortcuts for the cut functions (**Ctrl + C**, **Ctrl + V**...) are the same as for copying and pasting ranges. Depending on whether objects or a range is selected, the corresponding operations are performed. If both objects and a range are selected, the range selection has priority with **Copy**, **Cut** and **Replace with silence**. At **Delete** the order of selection decides: The last selected is deleted, range or object.

> ℹ️ *There are also menu commands to explicitly copy, cut, paste and delete objects. They are located at Menu **Object** > **Edit objects**.*

The edit operations in the menu **Edit** work for objects in the same way as for ranges in virtual projects. In fact, the "clip" created when copying objects is nothing more than a range from the virtual project that includes all the objects involved. For detailed information on the different variants of copying, pasting, and deleting via the clipboard, therefore, read the section Copying and Pasting Ranges (↗123).

> ℹ️ *In Sequoia there is the Clip Manager ("Clipstore") (↗311) where all copied objects are kept.*

Duplicating Objects

Simple duplicate: Click and drag on selected objects while holding the **Ctrl** key to duplicate them.

The **New Object** command in the **Object** > **More** menu creates a new object by inserting the last selected audio object at the play cursor position of the selected track.

Duplicate and move: Use the command **Duplicate** in the menu **Object** > **Edit objects** to duplicate selected objects. The duplicates are placed behind the original objects at a distance of one grid unit. When object snap is active, the objects are placed directly behind the originals; with a multiple selection, the first duplicate is inserted at the point where the last original of the object selection ends.

Keyboard shortcut: **Ctrl + D**

Duplicate multiple: This function allows selected objects to be duplicated several times in succession. Enter the number of copies and the spacing or total length.
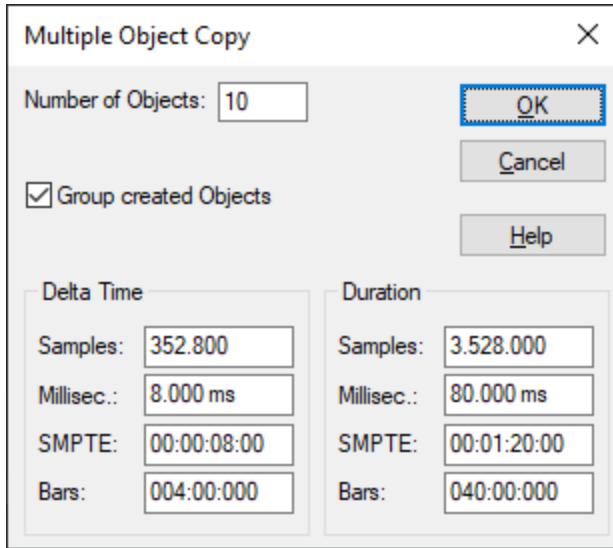
- **Number of objects**: Specify the number of duplicates here.

- **Group created objects**: The duplicated objects can be optionally grouped.

- **Delta Time**: Set the distance between each object's starting point here. The default length is set to that of the object you wish to duplicate. The objects follow each other seamlessly in this setting.

- **Duration**: As an alternative to the time difference, the total duration of original + duplicates can be selected here. The duration is the time between the start time of the original and the start time of the last copy.

Other Actions with Objects

Renaming and coloring Objects

Object name

Each object has a name. By default, this corresponds to the filename of the imported or recorded file. It is displayed by default at the bottom of the object.
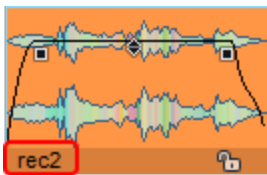
> ℹ️ For more on the display options for objects, see **System options** >**View options** (↗719).

To change the name, use menu **Object** > **Object name/color** > **Object name...**, the keyboard shortcut **Ctrl + N** or the Object editor (↗184).

Object Colors

Each object can be assigned a **foreground color** (coloring of the waveform) and a **background color**.

You have the following options in doing so:

- If no color is explicitly assigned to the objects, the default colors (↗714) are used.

- The **background color** of *selected* objects always corresponds to the default color defined for it.

- Use menu **Object** > **Object name/color** > **Object background color...** or **Object foreground color...** to explicitly assign colors.

- Set both colors by clicking objects in Color mouse mode (↗78).

- If you set a track color (**Track Head** or **Track Properties** dialog), the foreground and background color of all objects on that track will also be set to that color.

- With the corresponding option (**Use automatic sample colors for record**) in the color settings a random color is used for recorded objects.

- Set the background color in the object editor (↗184).

> ⚠ *The foreground color assigned in this way is used only if the option **Predefined color setting** is set at **Waveform color**. With other display options, the foreground color is determined for example by the track color, group membership or object content (WaveColor). For more on the various options for the **waveform color**, see **System options** >**View Options** (↗721).*

Looping Objects

When an object is "looped", the object or an object part is played repeatedly. Loop objects are suited, for example, to quick creation of full drum tracks from one single drum loop.

An object is normally played back between the object start and end. A looped object is also played from the object start, but after reaching the loop end point, the specified loop range between loop start and loop end is repeated until the object end.
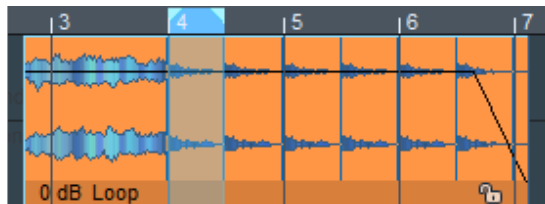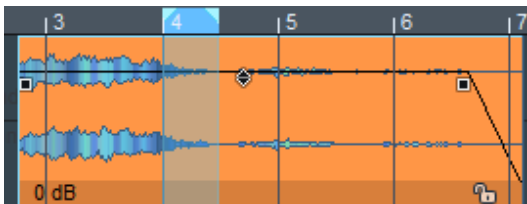
To loop an object, select the object and a range over the object to specify loop start and end points. If only the object is selected, but no range over the object is selected, the object start time will be the same as the loop start and the entire object will be looped.

Select Menu **Object** > **Edit objects** or in the context menu of an object **Edit objects** > **Build loop objects** or press **Ctrl + L**.

The object is now looped. The end of the object can be extended to the right as far as desired, repeating the audio material between the loop start and loop end. The loop area is marked in the object by vertical lines.

> 💡 *If you split a looped object in one of the loop passes and deactivate Loop in the newly created object, this object will finish playing exactly from the loop you started.*
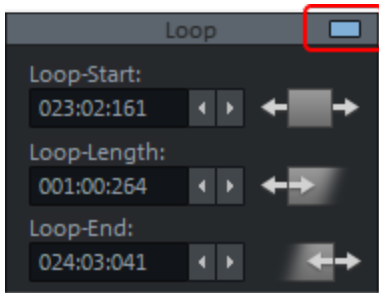
The loop can also be edited in detail with the **Object editor**. Activate the loop mode for selected objects by activating this button in the Object Editor, tab **Time/Pitch** in the section **Loop**. When Loop is activated, you can precisely set the loop start, loop length and loop end within the object individually.

Trimming Objects

Trimming objects cuts away the parts of objects that are outside a selected range.

> ℹ️ *The range does not have to be within the object boundaries. If the affected objects are completely within the range, the command has no function.*

1. Select the objects to be trimmed.
2. Then select a range. The range does not have to be selected directly over the objects, all selected objects will be trimmed to the length of the range regardless of their track.
3. Choose **Trim objects** from the menu **Object** > **Edit objects** or from the context menu of an object or the keyboard shortcut **Ctrl + T**.
4. All selected objects that have extended beyond the range boundaries now start and end at the range boundaries.

Muting Objects

Use the keyboard shortcut **Ctrl + M** or the "Mute" button in the object editor to mute selected objects.
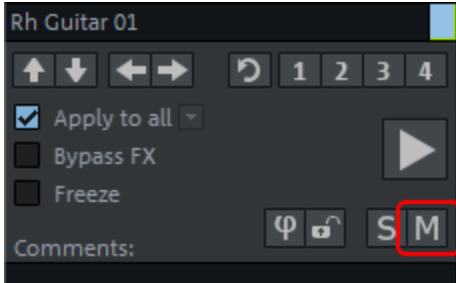
By right-clicking the button you can also mute only the left or right channel.

> ⓘ **Note**: *If you have selected multiple objects and want to mute them together using the object editor, check the **Apply to all** check box and then turn on mute.*

Freezing Objects

Similar to the Track Freeze (↗139) function, there is also an Object Freeze function. As with "frozen" tracks, all object effects are calculated into a wave file and do not require any further processing power. MIDI objects are replaced by an audio object that contains the audio output of the software instrument. For this, the audio signal must be routed to the same track that plays the MIDI object (see Routing of software instruments (↗372)).

To freeze an object, select Menu **Object** > **Freeze Objects** > **Freeze Objects** or the keyboard shortcut **Ctrl + Alt + F**.

The function **Edit object freeze** allows subsequent editing of the freeze's source material in the freeze VIP, as with tracks.

With **Unfreeze Objects** (keyboard shortcut: **Ctrl + Alt + U**) the original object can be brought back into the arranger window.

> ⚠ **Caution**: *This also applies to the object freeze: Do not create any new tracks in the freeze VIP, otherwise the "Unfreeze Objects" function will no longer be possible.*

Object unfreeze can also be switched on and off with the **Freeze** option in the Object editor (↗183).

> ⓘ *Notes:*
>
> ■ *If more than one object is selected, the **Freeze Objects** function will be applied to each individual object. To create one object from several objects, select the function **Glue objects**. In this case, however, you have no possibility of subsequent editing as with the object or track freeze.*
>
> ■ *Fade-in and fade-out of the object are not included, but transferred to the newly created object. Optionally, the object volume can also be excluded from the freeze. To do this, activate the option **Object freeze without object volume** at **Program preferences** > **Effects** > **Resampling/Freeze**. However, this only works if the default effect order (volume at the end of the effect chain) has not been changed, see also reverb tails in the object)*

Stereo Objects

Objects can reference mono or stereo files. Pairs of mono files, where a stereo source has been recorded on two mono tracks, can also be loaded and edited together as one object.

The files must be named the same and end with `_L` and `_R`. Then you can activate the option **Load L&R files as stereo** in the dialog Options for loading audio files (↗658). You can then select both files together when loading via the **file manager** or via menu **File** > **Load audio file...** and they will be loaded together into one object.
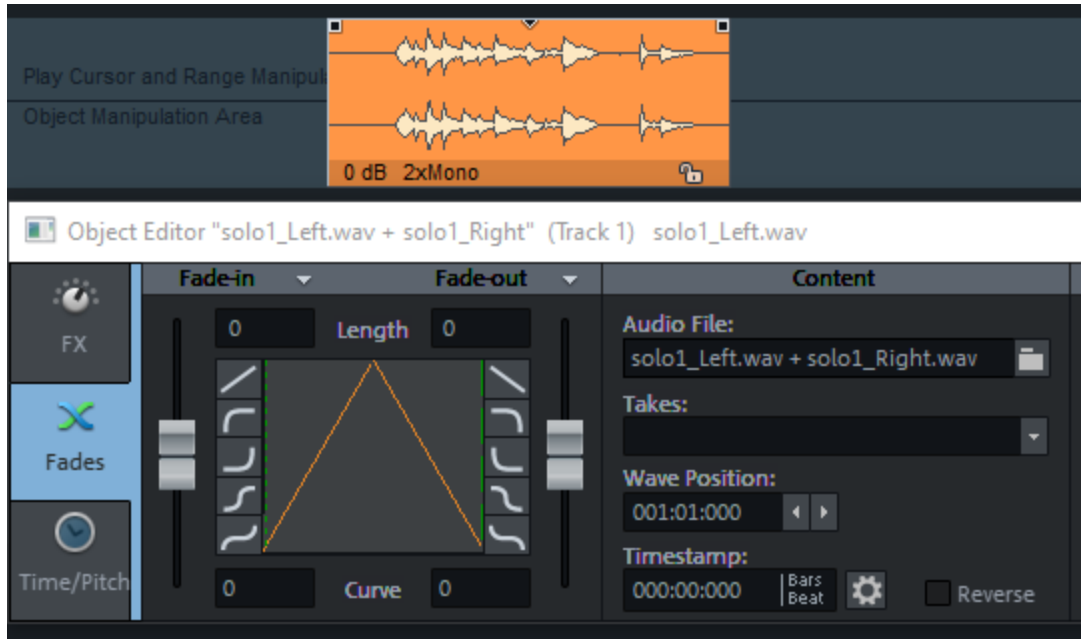
In the File manager you will find corresponding commands in the context menu of files to append `_L` or `_R` to the file name. You can also use the command **Add _L and _R to file name alternately** to apply the command to a series of files, provided that the files are in the correct order in the list. This can be achieved by having the recorded files contain the track number as part of the name and, as is usual when recording in mono pairs, having the mono tracks next to each other. In Sequoia you will find the corresponding options for this in the dialog Track settings (↗143) at **File name** at the button **All tracks**.

To convert mono objects into stereo objects in the project, select the two objects and use the command Menu **Object** > **Edit object** > **Stereo object from mono objects**. The objects do not have to have the same start or end time, nor do they have to be placed on directly adjacent tracks, but they should overlap in time. The stereo object replaces the upper object and preserves its object effects. It contains on the left channel the audio data of the upper object and on the right channel the audio data of the lower object.

⚠ *Attention: When the objects no longer overlap in time, the audio data of the lower object will be shifted.*

You can also combine two complete tracks with mono objects into stereo tracks with stereo objects. The tracks must be directly below each other and all objects must have the same start and end positions. To do this, select the command Menu **Track** > **Stereo object from mono objects (whole track)**.

Gluing Objects Together

You can glue two or more objects on a track together to form one object. This also executes an **object freeze**, which means that the object effects, except for the fade-in and fade-out of an object, are calculated into a new audio file and placed in the "FreezeData" folder of the project folder.

Unlike the object freeze with multiple objects selected, however, gluing together combines all objects into a single file, including all objects and gaps that lie on the track in between. If objects are selected on multiple tracks, one object is created per track.

To glue two or more objects together into one object, select both objects or, in case of multiple objects, the first and the last object, and choose **Glue objects** from the menu **Object** > **Edit objects** or from the context menu of the objects or press the keyboard shortcut **Ctrl + Alt + G**. If only one object was selected, it will be glued together with the object to the right of it.

> There is also a toolbar icon for the function, it is hidden but can be added to a toolbar via Edit toolbars (↗54).

> ⓘ *It is not possible to glue MIDI and audio objects together*

For the glued object, the **Freeze** option is also set in the Object Editor. If you deactivate the option or select **Unfreeze Objects** in the menu **Object** > **Freeze Objects** (keyboard shortcut: **Ctrl + Alt + U**), the object will be separated into its old components again. Likewise, subsequent editing of the glued-together objects is possible with **Edit Object Freeze**.

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

Reverberation Tails in Objects

If you use echo or reverb effects in the object, reverb tails at the end of an object usually do not decay. This is because, by default, effects are inserted into an object's effects chain before the volume. And at each end of the object, when editing operations with Auto Crossfade are made, short fades are applied, which then also fade out the reverb tail.

To change this behavior, open the **Effect Routing** dialog via the **FX** button in the Object Editor.
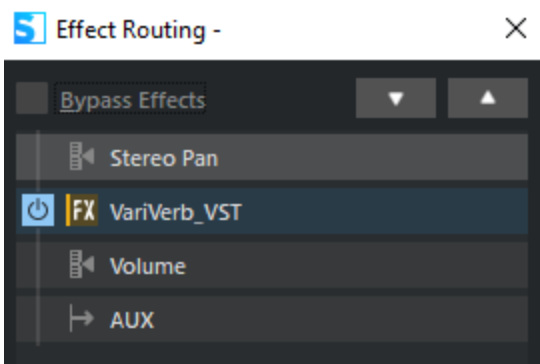
For a reverb tail to decay, change the effect order so that the reverb is placed after the volume.
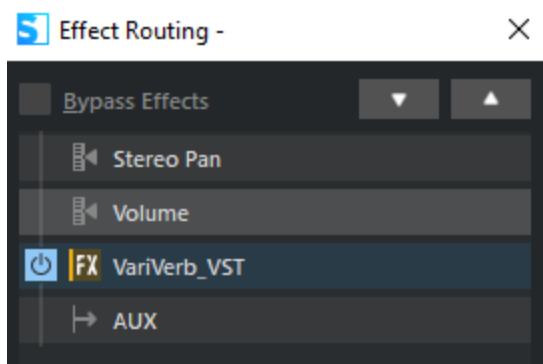
The decay time depends on the option **Maximum reverb time for objects without fade-out**, which is located in the Playback options (↗702).

Replacing an Audio File under the Object

Replacing an Audio File under the Object
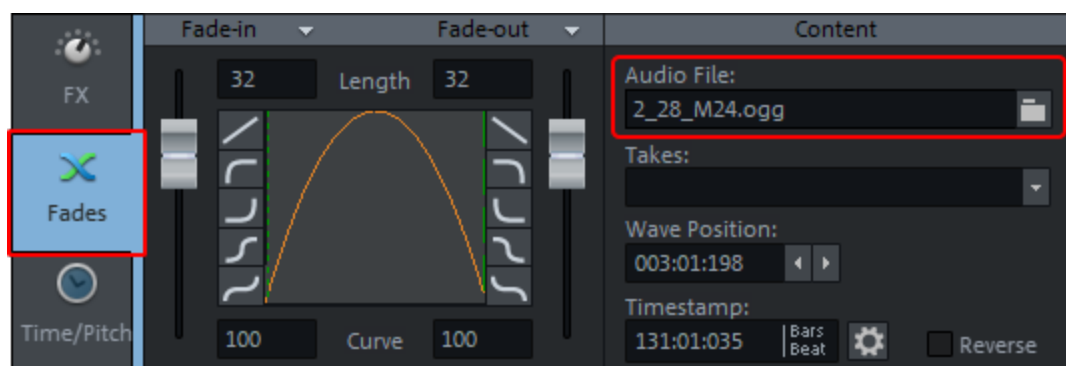
To exchange the audio file referenced by the object, open the **object editor** and switch to the **Fades** view. Click the folder icon to the right of the audio file name and select a new audio file.
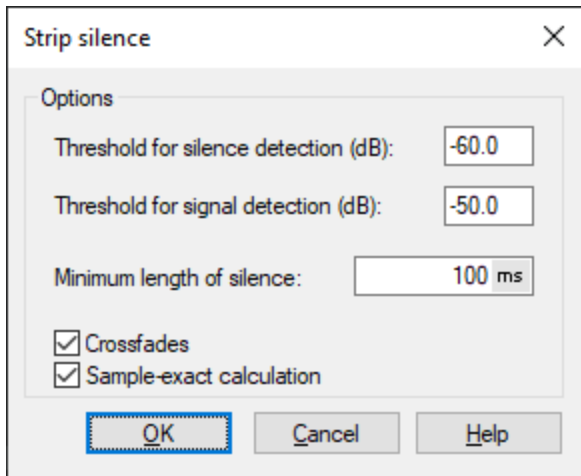
Copying or Renaming Audio Files under the Object

You can also specify a new name for the file that the object will use by typing the name at **Audio file** and pressing the **Enter** key. In the confirmation dialog you can specify whether the current file should be copied or renamed. If a copy is created, you can optionally specify whether all objects should use the copy or only the current object.

Removing Silence from Objects

Use this function to separate objects at the points where silence occurs and at the points where it ends. So after applying the command, the objects are divided into several objects containing either silence or no silence. The objects containing silence are also selected, so they can be deleted afterwards with the "Del" key.

Select the objects from which you want to remove silence and choose Menu **Edit** > **Silence** > **Strip Silence...**

You can specify a threshold value for both silence and wanted signal in the dialog.

Threshold for silence detection (dB): If the signal falls below this value, the object will be split at the corresponding position.

Threshold value for used signal (dB): If the signal rises above this value, the object will also be split.

The **Minimum length of silence** parameter specifies the minimum length that silent passages should last in order to be selected and thus enabled for deletion. This way, you can remove very short passages of silences from the cut.

With the **Crossfades** option, you create crossfades between at the split positions of the objects.

Sample-exact calculation: 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.

> ℹ️ *The ranges in the objects identified as silence are not deleted, but are only outside the object boundaries afterwards. You can therefore correct the result of the function later if too much has been cut out.*

Editing the Audio Data of the Object

Audio objects refer to audio files that are open in the background and contain work instructions on how to process these files. In Sequoia, most editing can be done directly on the objects in the project without changing the original audio.

However, for certain use cases it may be advantageous to edit the audio data directly. In doing so, you no longer edit only objects in the virtual project, but the audio files stored on the hard disk itself. This type of audio editing is also called "destructive" or "offline" editing.

To open an audio file of an object directly, double-click it while holding **Ctrl + Shift** key. Alternatively, you can also select the option **Wave Editing...** in the **Object** menu.

The range in the audio file used by the object is selected as range in the wave project.

If you edit an audio file that is used in several objects with **Wave editing...**, all objects in the VIP will then access the edited audio file and the edits will affect each of these objects.

If you want to edit the audio file only for a specific object, select **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 object now refers to the copy that has been created. After editing, the copy and the original audio are available as Take (↗328) for the object.

> ℹ️ *For more information on destructive editing of audio files, read the section* Editing audio files *(↗569)!*

Editing Objects Externally

You can edit the audio file of an object with external audio editing programs such as MAGIX SOUND FORGE Pro. After editing, it is then available in the VIP in its edited form.

To configure the transfer, use the dialog **External Tools** (↗732) in the menu **File** > **Program Preferences**. If everything is configured correctly, then there are entries at the very bottom of the context menu of objects for transferring them to the external programs.

Crossfading Objects

Only one object can be played in a track at a time. When one object is moved over another, it covers the one preceding it in time. The covered part of an object is not played. If you move the rear object back again, the invisible part of the front object can be made visible again and thus audible.

However, it is possible to crossfade two overlapping objects.
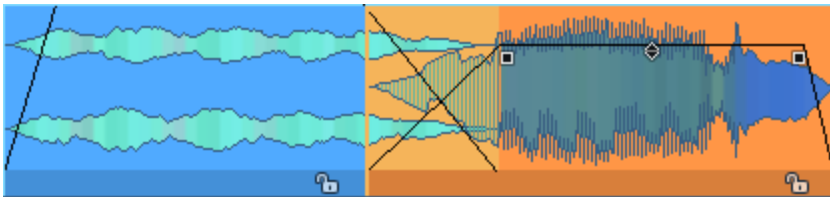
Auto Crossfade Mode

The auto crossfade mode is activated with this button on the upper toolbar or with **Auto-Crossfade active** in the menu **Edit** > **Crossfade**.

If **Auto-Crossfade** is active, all objects created by cuts (copy/paste of ranges or objects, split objects or trim) or by recording automatically receive a fade at the object start and at the object end or a crossfade.

> ℹ️ *Optionally, fades can also be applied to objects created by loading audio files, see the* **Create fades for new objects when auto-crossfade mode is active** *option in the* **Program Preferences** > **Program** > **General**.

Also, a crossfade is automatically created as soon as an object is moved over another object. Any possible fade-out on the front objects is hereby lost. If the trailing object has a fade-in, a crossfade shorter than or equal to the length of this fade-in is created, depending on the duration of the overlap; without a fade-in, the default crossfade is applied.

Editing objects with the handles in the arrangement changes slightly: the fade-out of the front object and fade-in of the back object are moved together when you change object start or object position of the back object. The fade-in handle of the rear object controls the length of the crossfade and thus also the fade-out length of the front object.

The default crossfade settings used with Auto-crossfade are stored in the Crossfade Editor preset (↗178) **Default_ CrossFade** and can be customized in the Crossfade Editor.

> 💡 *Crossfades can be edited in great detail with the* Crossfade editor (↗170) .

Creating Crossfades Manually

Sometimes it is not desired that a crossfade is created when moving objects over each other and that the fade-out and fade-in of the objects are coupled. In this case, disable the **Auto-Crossfade** option.

To manually create a crossfade between two objects, right-click the fade-in handle of the trailing object and choose **Crossfade active** from the context menu. You can also find the option as **Allow crossfade (to the left)** in the menu **Edit** > **Crossfade** and in the context menu of objects.

You can also remove a crossfade with auto crossfade mode active by right-clicking a crossfade and deselecting **Crossfade active** in the context menu.

Asymmetric Crossfades

The standard crossfade is symmetrical, which means that length, curve shape and position of the fade in the object (overlap) are coupled for fade-out and fade-in. Right-click the fade-in handle of the rear object and select **Asymmetric Crossfade** from the context menu to set fade-in and fade-out separately. You can also find the option in the menu **Edit** > **Crossfade** and in the context menu of objects.

Crossfade Editor

The Crossfade Editor provides you with a powerful tool for editing crossfades between objects. All functions can be executed with the mouse as well as with the keyboard.

In conjunction with the Source-Destination Editing (↗193), you achieve a completely cut-point-oriented mode of operation.

To edit a crossfade with the Crossfade Editor, select an object or select a region over a crossfade and open the Crossfade Editor.

The Crossfade Editor is opened with this button on the upper toolbar, via menu **Edit** > **Crossfade** > **Crossfade Editor** or with the keyboard shortcut **Shift + F**.

Above the crossfade editor, a special view of the project window is displayed, in which the visible section of the project is arranged around the crossfade. The crossfade is divided into two superimposed tracks, the upper one containing the fade-out, the lower one the fade-in. The audio data of the objects involved is displayed in full, i.e. also beyond the object boundaries, with the audio data that cannot be heard in the project being grayed out.

Crossfade Position Settings

The position of the **Fade Out** and **Fade In** reference point can be moved with the minus and plus buttons. The ***** button moves the respective point to the position of the play cursor.

> ℹ️ *The reference point can be at any position between the start of the fade and the end of the object, the exact position is determined by the **Overlap** (see below).*
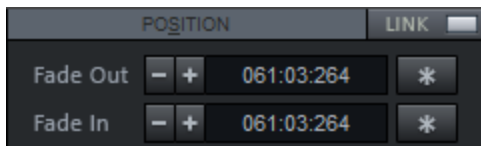
For the **+/-** buttons, 3 step sizes can be set, whose values can be changed in the Crossfade Editor Preferences.

Keyboard Shortcut:

Fade Out +/-	Num +/-

Fade In +/-	Ctrl + Num +/-
Step size 1,2,3	1,2,3
Fade out to play cursor	Num *
Fade In to play cursor	Ctrl + Num *

By double-clicking the corresponding input field, you can also edit the time position of the fade reference points directly. Keyboard shortcuts: **Shift + S** for the fade out position, **Ctrl + S** for the fade in position, **F** for the Point Offset.

When **Position Link** (keyboard shortcut: I) is active, fade-in and fade-out are moved together. By pressing the **Alt** key while editing a fade, this link can be temporarily deactivated.

The audio of the front object ("fade-out") is generally kept on the timeline to avoid changes to crossfades placed temporally in front of it.

The behavior of the fade-in and subsequent audio material when moving positions is also determined by the selected **Move Mode**:
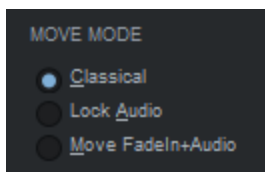
The **Classical** mode (keyboard shortcut: **C**) is optimized for the operations that occur during editing classical music, while the **Lock Audio** mode (keyboard shortcut: **A**) is optimized for synchronous editing (e.g. for film dubbing). The **Move Fade-In + Audio** mode (keyboard shortcut: **M**) is only used in special cases.

Depending on the selected combination of **Position Link** and **Move Mode**, the following behavior results:

Position link active

Moving the fade out

- **Classical mode**: The entire rear object (audio and fade) is moved as well.

- **Lock Audio**: The fade-in of the rear object is moved, its audio is not moved.

- **Move Fade-In + Audio**: The entire rear object (audio and fade) is moved.
  Since this Move mode is no different from Classical mode with Position Link active, you will always want to use it with Position Link inactive. For this, there's the Crossfade Editor Preference **Ignore position link status in "Audio with fade in" move mode**. When this is active, the position link state is ignored in this mode. In this case, the rear object remains unchanged and only the fade-out is moved. This makes the crossfade asymmetrical, i.e. the fade-in and -out positions are no longer in the same position.

lon Move the fade-in

- **Classical mode**: The audio of the rear object is moved "under" the object in the opposite direction, the fade-in and fade-out positions remain unchanged.
- **Lock Audio**: The fade-out of the front object is moved so the entire crossfade is moved over the objects.
- **Fade-In + Audio:** The entire rear object and crossfade (fade-out and fade-in) is moved. If the Crossfade Editor Preference **Ignore position link status in "Audio with fade in" move mode** is active, the fade-out remains unchanged. This makes the crossfade asymmetrical.

Position link inactive

Move the fade out

In all move modes the fade-out is moved relative to the audio, the fade-in remains unchanged.

Move the fade-in

- **Classical mode**: The rear object (audio material and fade-in) is moved.
- **Lock Audio**: The fade-in reference point is moved over the audio material.
- **Fade-In + Audio**: The rear object (audio and fade-in) is moved.

Moving with the Mouse
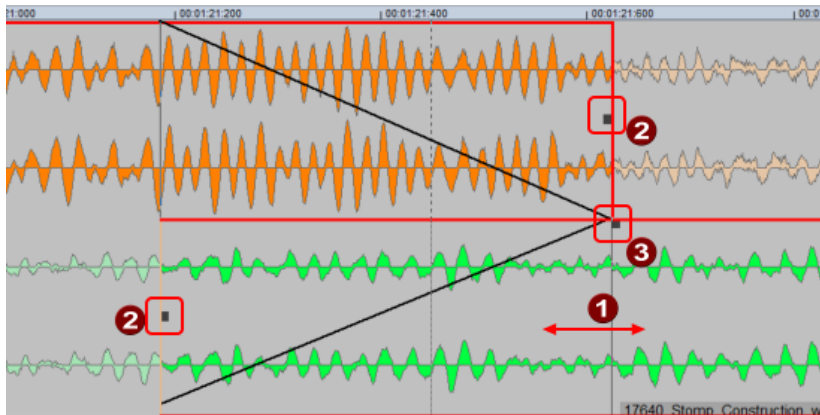
❶ By dragging on the audio material in use, you can move the entire fade-in object along with the crossfade in **Classical** and **Fade-In + Audio** mode. The "Fade-out" object cannot be moved.

❷ By dragging on the fade position handles you can change the fade position relative to the audio material (Overlap).

❸ Drag the fade handle of the fade-in object to set the fade length of the fade-in.

Crossfade Editor Object Mode

Because subsequent objects are moved when you work on cuts and crossfades in Classical mode, to protect subsequent edits you probably want to use the Object mode **Link one track** while you are working in the Crossfade Editor. By default, this mode is automatically loaded when the Crossfade Editor is opened. In the project itself, however, you may prefer to work in **Normal Object Mode**. Therefore, the Crossfade Editor has a separate setting for the object mode (↗155), which becomes active as soon as you open the Crossfade Editor.
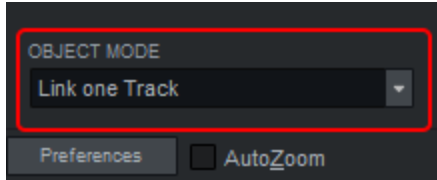
When you exit the Crossfade Editor, the mode that was set in the project is restored.

> ⚐- *If you want to use the same object mode as in the Arranger, select **Use Global Mode** in the Crossfade Editor preferences for **Preset Object Mode**.*

Fade Length/Overlap/Shape

Use the **Form Link** button (keyboard shortcut: **N**) to specify whether the shape of the fades can be set together or or independently.

The fades can be set always independently if you also press the **Alt** key while adjusting to temporarily disable the link state.

> ⓘ *Tip: **Shift + N** / **Shift + I** operates the position and shape links simultaneously.*

Fade Length

Set the fade length with the **Length** sliders. The faders are logarithmically scaled, which means that the shorter the fade length, the more precisely it can be adjusted. For a finer adjustment, additionally press the **Shift** key.

By double-clicking the corresponding input field, you can also edit the fade lengths directly. Keyboard shortcuts: fade-out length **L** or **Shift + L**, fade-in length **Ctrl + L**

In the project window, the fades can be changed using the fade length handles.

Overlap

The **Overlap** determines the location of the reference point within the fade, i.e. how much of the fade takes place inside and how much outside the object boundaries.

For the fade-out, you determine the object end of the fade-out object by positioning the reference point.

0% means that the fade-out occurs completely before the object's end - the fade is completely inside the object and to the left of the reference point. At **50%** the fade is 50% inside and 50% outside the object, at **100%** the fade is completely outside the object and to the right of the reference point.

Similarly, you determine the object start of the fade-in object by positioning the fade-in reference point.

0% means that the fade-in takes place completely behind the start position - the fade is completely inside the object and to the right of the reference point. At the value **50%** the fade is 50% inside and 50% outside the object, at **100%** the fade is completely outside the object and to the left of the reference point.

If you activate the option **Keep fade** (keyboard shortcut: **K**), use the **Overlap** faders to move the reference point relative to the crossfade. If the option is not active, you move the crossfade relative to the reference point.

With the **Align**- function you connect the overlaps of linked fades so that the values of fade-in and fade-out together always add up to 100%.

In the crossfade editor project window, drag the "Overlap" handle to set the fade relative to the reference point.

ℹ️ *This handle is not visible by default. To do this, activate the option **Fade handles for overlap** in the **Project display** settings (keyboard shortcut: Shift + Tab) in the section **Objects**.*
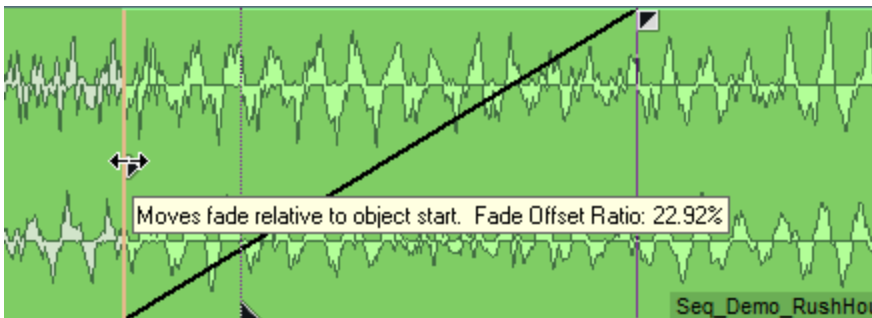
Right-click the **Keep Fade** button (keyboard shortcut: **Shift + K**) to open a context menu that lets you set the fade reference points to default values.

Fade Shape

The selection menu **Form** contains mathematical functions for the shape of the fades. In addition to the linear form, various sine/cosine as well as exponential and logarithmic curves are available.

With the shape faders you can change every fade curve from linear (0%) to its original form (100%). At a value of -100% the curve is inverted. Parameter values larger than 100% or smaller than -100% create additional distortions in the curve.

The **PowerLock** option couples the shape parameters when the link function is active. This function ensures a constant amplitude level in the parameter range between -100% and +100%. When you load a new curve shape, the complementary parameter value for the other curve is automatically used.
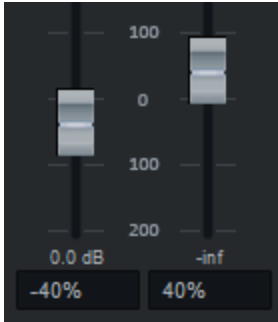
The dB display below the faders shows the relative level of the object at the fade reference point.

💡 ■ *Double-clicking a fader toggles between the current value and the default value. Default values are:*

   ● *Fade length 100ms*

   ● *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 can be edited numerically by double-clicking on the value display as well as by pressing the* **Enter** *key while the fader is selected. You can accept the changed values by pressing* **Enter** *and discard them by pressing* **Esc**.

Volume/Level

Use the level fader (keyboard shortcut: **V**) to set a relative change in the object levels of the fade-in objects.

Playback Functions

1 Playback in the pre-roll time until the end of the fade-out.

2 Playback from the beginning of the fade-in until the end of the post-roll time.

3 Playback of the entire crossfade, including pre- and post-roll time.

4 The **Play Pre/Post** option decides whether the playback of the crossfade including pre/post roll is performed when starting playback with the space bar. If it is disabled, playback will start from the current play cursor position. If it is activated, the play cursor is first moved in front of the crossfade according to the pre-roll time and then playback is started.

ℹ️ *All crossfade playback functions can also be found in the **Play/Rec** menu > **Play cut** (↗86).*

Keyboard control of the crossfade playback

■ **Spacebar** or **In+Out** Playback button: The crossfade is played back with pre- and post-roll time.

ℹ️ *The **Play Pre/Post** option must be active, otherwise playback will start from the playback marker.*

■ **Alt + Space**: The audio will be played from the playback marker.

■ **Shift + Space** or upper small play button: The audio material of the fade-out object is played with pre-roll time.

■ **Ctrl + Space** or lower small play button: The audio material of the fade-out object will be played back with post-roll time.

■ **F7**: The fade-out objects are played back without fade until the "fade-out" reference point ("out point").

■ **F6**: The fade-in objects are played back without fade from the "fade-in" reference point ("in point").

■ **F8**: The unused audio of the fade-out objects is played from the "Fade-out" reference point ("out point").

■ **F5**: The unused audio of the fade-in objects is played back until the "fade-in" reference point ("in point")

■ **Shift + Alt + Space**: The upper audio material (also called "tail") is played from the playback marker position.

■ **Ctrl + Alt + Space**: The lower audio material (also called "head") is played from the playback marker position.

Snapshots

You can save different variants for the crossfade currently being edited in the 8 snapshots. To do this, click the corresponding button on the left while holding down the **Shift** key. If there was no snapshot saved on the button before, a simple left-click is also sufficient. You can also save the snapshots using the **Ctrl** key + number pad keys 1 - 8. Snapshots can also be loaded, saved or deleted via the context menu of the buttons.
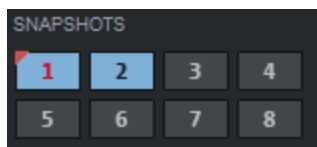
Used snapshots are highlighted in blue, the active snapshot is additionally marked with red text

You load saved snapshots by left-clicking the buttons. This allows you to quickly compare different crossfade variants.

As soon as you exit the Crossfade Editor or switch to the next or previous fade, the snapshots are lost. Running **Revert** does not delete the snapshots.

Crossfade Presets

The parameters of a crossfade (fade lengths, overlaps, point offset, curve shape, symmetry) can be saved in presets for further use. The saved presets are applied by selecting them from the **Presets** menu. The "fade-out" reference point of a crossfade and the reference points of a single fade-in or fade-out are retained.
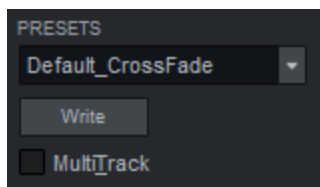
To save the current settings in the Crossfade Editor as a crossfade preset, click the **Write** button and assign a new name. The saved preset will now appear in the **Presets** list.

Standard Crossfade Presets

There are a number of standard crossfade presets that are applied at various points in the program during cutting operations when the **Auto Crossfade Mode** is activated. You can use the Crossfade Editor to adapt these default crossfades to your needs by loading them, modifying them accordingly, and saving them again under these default names.

Default_CrossFade is applied with all cut operations. It is a very short, symmetrical, linear crossfade designed to produce cuts that are as inaudible as possible. In cuts where a simple fade is created, the objects are "cropped", i.e.

the fade is completely inside. For cuts that create a crossfade (e.g. splitting an object), the fade in the front object is completely outside (100% overlap) and in the back object completely inside (0% overlap).

Default_CrossFade_Alt is linked to the menu command **Split objects with alternate (linear) crossfade** (↗157) (keyboard shortcut: **Shift + T**). It is also linear, slightly longer and symmetrical to the split point, but with an overlap of 50%.

Default_CrossFade_SD corresponds with the default application for source destination commands. This crossfade is optimized for fading similar signals, i.e. as a root cosine 50/50 fade. Adjust the standard length to suit your requirements.

Default_FadeFromBlack and **Default_FadeToBlack** are linked to the blackfade source destination commands. A standard fade-in and a standard fade-out are stored here, which can be applied contextually. Customize this preset to your most common use case.

> **ⓘ** *In* multi-user mode *(↗636) each user can use his own presets.*

Multitrack Crossfade

You can edit crossfades on multiple tracks together. For this purpose, the objects involved must be grouped (↗148) beforehand. Objects created in a multitrack recording are already grouped.

After opening the crossfade editor, you can zoom out in the integrated project view until you see all the tracks where the grouped objects with crossfades are located. If you now change a crossfade, the changes also affect the crossfades of the other tracks.
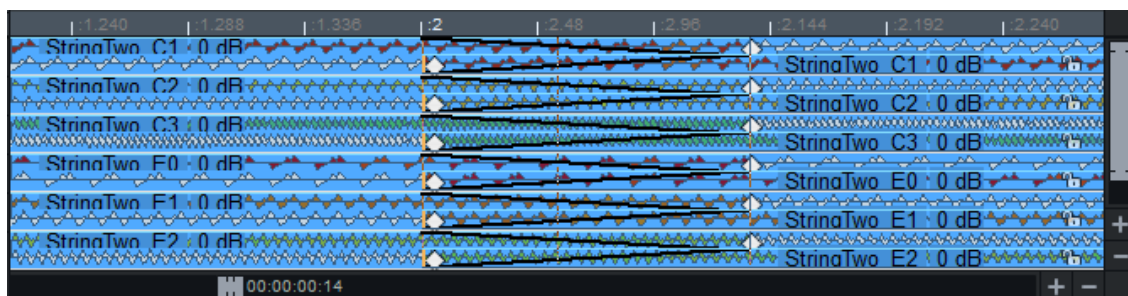
Lock tracks/crossfades

If you do not want to include the fades of a particular track in the edit, you can lock that track. Use the Lock button in the track head of the track to do so. The Crossfade Editor automatically switches the locked track into object mode **Lock all audio**, so that objects are not accidentally moved against each other.

Multitrack Presets
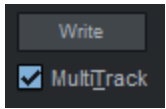
The **MultiTrack** option lets you specify when loading crossfade presets whether the preset should be loaded on multiple crossfades on multiple tracks or only on the selected crossfade of the active track. (Shortcut: T).

> ℹ️ **Note**: *The option for loading on all crossfades must be activated before loading the preset.*

Saving presets: Also when saving, you can specify with an active **MultiTrack** option that the preset should be saved as a multitrack crossfade.

If the option is active, the preset is saved as a multitrack crossfade preset. For example, for a two-track crossfade it gets the extension `MT02.cfp`, for a three-track `MT03.cfp` and so on.

For example, if you load the saved "Crossfade_MT03.cfp" preset on a two-track crossfade, you will get a message that the number of tracks does not match. You can still continue loading, it will load the crossfade settings for the two selected tracks only.

General Crossfade Functions

- **OK**: (Keyboard shortcut: **O**) confirms all changes to the current crossfade and closes the crossfade editor.
- **Cancel**: (Keyboard shortcut: **Esc**) discards all changes to the current crossfade and closes the crossfade editor.
- **Revert**: (Keyboard shortcut: **R**) discards all changes to the current crossfade and opens the crossfade again with its original parameters. Saved snapshots for the current crossfade are retained.
- **Back/Forward**: (Keyboard shortcut: **U** / **X**) accepts all changes to the current fade and reopens the crossfade editor for the previous or next fade in the current track, or for multitrack crossfades in the reference track displayed in the title bar of the crossfade editor.

Symmetrically adjusting the fades

The option "Symmetrical" **Sym.** causes an adjustment of the fade-in to the fade-out. **Position Link** and **Form Link** are switched on.
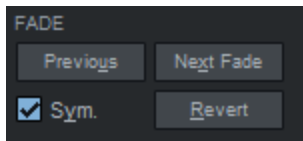
When opening the Crossfade Editor with a symmetrical fade, the option is active. The state of the option indicates whether the requirements for a symmetrical crossfade are still met. These are:

- Point offset: 0

- The length of Fade In/Out are the same

- Overlaps add up to 100%

AutoZoom

If **AutoZoom** is active, the visible section is adjusted after changes to position, fade lengths or overlap so that the crossfade is always placed in the center of the Crossfade Editor project window and is fully visible. The size of the display can be set in the Crossfade Editor Preferences (button **Preferences**).

When clicking the AutoZoom option with the **Shift** key held down, the visible section is moved to the crossfade position without changing the zoom factor, the AutoZoom function remains deactivated.

Crossfade Editor Preferences

Clicking the **Settings** button opens the Crossfade Editor Preferences in the program settings:

Position changes

Nudge steps (ms)

Step 1	1
Step 2	10
Step 3	100

☐ Remember last stepwidth

Default move Mode

Classical

Default object Mode

Link one track

☑ Default linked positions/parameters

Playback

☑ Default Pre/Post Play

Pre-roll time (sec)   2

Post-roll time (sec):   2

Display

☑ Default horizontal auto-zoom

Crossfade part of screen (%):   30

Minimal screen range (ms):   10

☐ Remember horizontal zoom width

☑ Vertical auto-zoom of wave form

☑ Dim unused audio

Operation

☑ Enable Undo

☑ Restrict VIP-Mouse-handling according to move modes

☐ No object movement with mouse in linked "Classical" move mode

☑ Ignore position link state in "Audio with FadeIn" move mode

☐ Ignore mixer keys (Arrows, PgUp, PgDn, Home)

☐ Keep focus

☑ Center crossfade for zooming

☐ Always center crossfade

Position changes

Nudge Steps: Select up to three step sizes for the crossfade editor. You can also enter fractions of milliseconds like 0.2 ms.

Remember last step width: If this option is active, then the last step size selected will be used with the next edit.

Default Move/Object Mode: When the crossfade editor is opened, the mode set here is used.

Default linked positions/parameters: With this setting, the crossfade editor opens preset with position link enabled for fade-out and fade-in.

Playback

Default Pre/Post Play: Here you decide whether option **Play Pre/Post** should be active for the playback button by default.

Pre-roll timePost-roll time: Here you determine the pre- and post-roll time in seconds for the corresponding playback functions in the crossfade editor and in the menu **Play/Rec** > **Play cut**.

Display

Default Horizontal Auto zoom: Set the status of the "Auto zoom" button when the Crossfade Editor is opened.

Crossfade part of screen (%): Specify how much of the screen the crossfade should occupy when the Crossfade Editor is opened or after Autozoom. If you enter the value 0, the value for the **Minimal screen range (ms)** will be used.

Remember horizontal zoom width:If you select this option, the next time you open the Crossfade Editor, the last set horizontal zoom level will be recalled.

Vertical auto zoom of waveform: This scales the graphical representation of the sampled data of each object independently to a maximum. This allows you to better detect transients, but the waveform data of the objects no longer reflects the level relationships between the objects.

Dim unused audio: You can disable the visual attenuation of the unused audio for better visual juxtaposition of the upper and lower objects here.

Operation

Enable undo: Here you turn on/off the "Undo" function.

Restrict mouse operations: If active, restrict mouse operations in the project window to actions that are executable according to the current position link state and move mode to avoid operating errors.

No object movement with mouse in linked "Classical" mode: This locks the movement of linked objects in "Classical" mode.

Ignore position link status in "Audio with fade-in" move mode: This turns off the link status in "Audio with fade-in" move mode. The background is that the Audio with Fade mode with active Position Link is not very useful and you will mostly disable the Link status for it. This option will save you a few clicks.

Ignore mixer keys: This disables the mixer key shortcuts Page up, Page down, Home and Arrow keys for a selected fader to be able to use these keys for inputs in the project window.

Keep focus: Activate this option so that the crossfade editor always remains in focus, even after mouse operations in the project window.

Center crossfade for zooming: If this option is active, the crossfade is always displayed centered when zooming.

Always center crossfade: If this option is active, the crossfade is always displayed centered when zooming or moving objects.

Object editor

The heart of object-oriented work is the Object editor, which can act as a "real-time channel strip" tool for single or multiple objects.

> ℹ **Note**: *There is an object editor for MIDI objects as well. For detailed information, see the section* MIDI object editor *(↗420).*
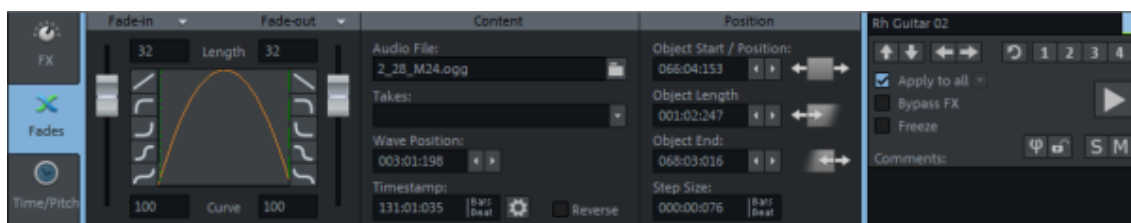
The object editor is divided into three sections - **FX**, **Fades**, and **Time/Pitch**. To edit objects, the object editor can always remain open, its display always adapts to the selected object, for a multiple selection, this is the object that was last added to the selection.

To open the object editor, select objects and choose Menu **Object** > **Object Editor...** or the keyboard shortcut **Ctrl + O** or double-click the object. The Object Editor opens in the Docker (↗57).

Basic Functions
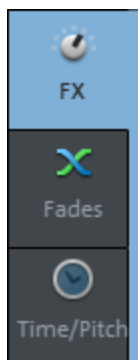
On the left there are buttons to switch between the three different views of the Object editor:

- **FX**: Setting of audio effects, e.g. gain, AUX sends, FX plug-ins, EQ, panorama, automation and volume.

- **Fades**: Settings for the object in the project and for the file, e.g. Fade-in, Fade-out, Wave file, Wave position, Object position

- **Time/Pitch**: Here you can set time stretch / pitch shift algorithms and loops.

> **ⓘ Note**: *By right-clicking in a free area, you can select different color schemes (skins) for the object editor display. The skins with the name suffix "Editor Max" display all three views together in one window.*

The right section is the same in all three views:

❶	This text field displays the object name, which can also be edited here.
❷	By clicking on the square next to it you can set the background color of the object.
❸	Use the **arrow buttons** to move between objects on the tracks. They are not available when multiple objects are selected. Keyboard shortcut: **< / >** or **Ctrl + Alt + Q** / **Ctrl + Alt + W**
❹	**Reset**: Resets the selected object to the original (neutral) setting.

⑤ **1-4**: **Snapshots for object editor settings**. These snapshots can be used on all objects in the project, so you can use them to transfer effect settings from one object to one (or more) other objects.
Click one of the buttons to save. To load the snapshot, click this button again. To overwrite a saved snapshot, click the button while holding down the **Shift** key. The active snapshot is marked by a bullet point.
In the context menu of the snapshots you can load and save snapshots and also delete them. There, the object settings can also be saved to a file or loaded from a file in order to use them across projects. The file names of the saved object settings are listed in the lower section of the menu.
In the **load options** you can define which settings should be loaded from the snapshots.

⑥ **Apply to all**: If you activate the "Apply to all" option, all selected objects will be updated when the object editor is open. All modified settings after the selection will be applied to all other selected objects.
In the menu under the small arrow ▼ you can specify which settings are to be transferred. There are two different options for the volume, **relative** or **absolute**.
Example: You have changed the level in an object from -7dB to -3dB, i.e. increased it by 4dB. If you transfer the volume to another object with volume -2dB **relatively**, it will have -2dB + 4dB = +2dB afterwards. If you transfer the volume **absolutely**, it will also have -3dB afterwards.
The **Plug-ins** option initially refers only to the fact that a specific plug-in is loaded into an object. The actual parameter settings of a plug-in are transferred only when the plug-in's interface is opened and closed. This can lead to a situation where you have adjusted a plug-in on one object and then accidentally open the same plug-in on another object, overwriting the settings. For this reason, there is an option in the menu **Transfer plug-in changes (open dialogs) now** to explicitly update the same plug-in in all other objects without closing the plug-in dialog. **Transfer all plug-ins now** transfers the parameters of all plug-ins to all selected objects.

⑦ **Bypass FX**: This deactivates all object effects, EQ/panorama settings, and AUX sends.

⑧ **Freeze**: Switches object freeze (↗163) on and off.

⑨ **Play/Stop**: This button corresponds to the normal playback function, except when **Solo** is active in the object editor (see below).

⑩ **Phi**: The Phi button causes a 180 degree phase rotation. By right-clicking on the Phi button you can invert the left or right channel individually.

⑪ **Lock**: Lock Object (↗154).

⑫ **Solo**: When playing with the play button in the object editor (keyboard shortcut: **spacebar** *with focus on object editor*), playback starts from the beginning of the selected object and mutes all other objects.

13   **Mute**: This button mutes the selected object. By right-clicking you can also mute the left and right channels individually.

14   **Comments**: You can enter comments relevant to the selected object in this field.

Object Effects

The **FX** view contains all the settings for the object effects.
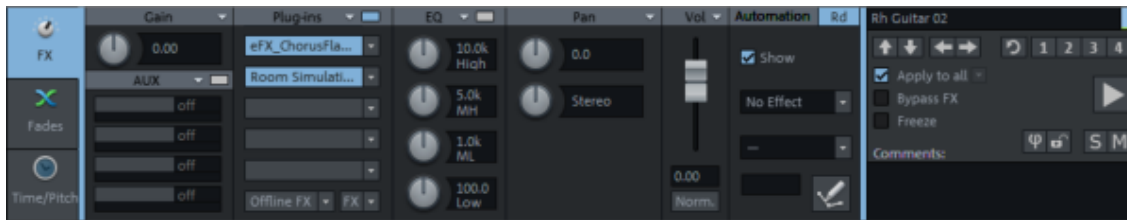
In addition to the effects in the tracks and in the master channel of the mixer, you can also apply effects in audio objects. This allows you to selectively apply effects that you only need at a certain point in the arrangement, and you don't have to set up an extra track for an AUX send effect or automate a track effect for it.

The effects " stick to" the objects, therefore all settings are preserved when moving or copying the objects. Even if you split an object (↗157), the new object inherits all the effects of the original object.

It is also possible to use the object effects via the **Effects** menu or via the context menu on the object.

> ⓘ *Level and pan of the object as well as the volume controls in the track are calculated by default after the object effects.*

The dialog is divided into different sections (Gain, AUX, Plug-ins, EQ, Pan, Vol, Automation). Some of the sections have a menu ▾ . From this menu you can:

- **copy** the settings of the section and **paste** them into the same section of another object,
- **reset** the settings of the section and
- **apply** the settings of the section to all selected objects.

> ⓘ *For transferring all settings to the other selected objects, see* **Apply to all** *(↗185)*

Gain and AUX Sends

Gain: Here you can set the gain for the object.

> 💡 *If you normalize an object with a very low level, the volume fader in the object is at its maximum value after that and the object volume can no longer be changed so easily with the fader. In such cases, it is a good idea to enter the numerical value from the object volume at Gain (simply by copying and pasting) and set the object volume to 0.*

AUX sends: AUX sends 1-4 are displayed in the object editor. Use the respective slider to set the send level.

ℹ️ *Object AUX sends are fixed to **Post**, i.e. after volume and pan, you cannot change this position.*

Any object can send to all available AUX busses. Right-click the small rectangular button in the header to open the extended AUX Send dialog where you can edit all AUX Send paths. Click this button on the left to bypass all AUX sends (bypass function).

Plug-ins and EQ

Use the Plug-in section to apply effect plug-ins to objects.

Clicking on the empty insert slot opens the plug-in browser (↗216), 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 plug-in's interface. In the menu at the insert slot ▾ you reach further functions, e.g. you can open the plug-in browser again to exchange or remove the plug-in.

The **Plug-ins** button at the top turns all effects in the respective effect chain on and off. A visual indicator (*) for plug-ins shows that they were previously active and will be reactivated the next time the **Plug-ins** button is clicked. The order of the plug-ins can be changed by dragging and dropping a filled slot to another position.

The **FX** button opens a simplified Effect routing dialog (↗220) for changing the plug-in order. Use the ▾ menu next to the "FX" button to save or load object effect settings to a file.

The button **Offline FX** opens a menu that contains roughly the same entries as the menu **Effects** in the main menu, with the difference that here the effects are applied offline (↗227) to the object's audio file. It is also possible to edit either the left or right channels individually.

At **EQ** you can directly adjust four frequency bands of the EQ116 Parametric Equalizer using the rotary controls. By clicking the small rectangular button in the header you switch the equalizer on/off. Right-click on the knobs or the button to open the interface of the EQ116, where you can edit all frequency bands.

Pan

Here you can edit the panorama position as well as the stereo width.Right-clicking on a knob opens the Panorama Editor dialog (↗492).

If surround busses or masters are present in the project, additional panels appear that let you apply track-based or object-based surround panning (↗516).
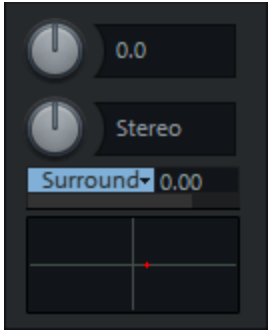
Right-click in the surround panorama box below to open the Surround editor (↗517).

Vol

Vol.: The volume can be controlled using this fader. Double-clicking on the fader sets it back to 0 dB.

Norm.: normalizes the maximum level of the object to 0dB.

Automation
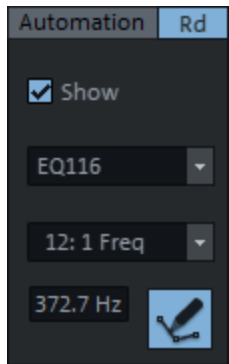
Here you can configure settings for object automation. Select the parameter to be automated in the menus. Use the **Show** option to make the parameter's automation curve visible in the objects. Activate the Draw automation mode button and draw the object automation curve into the object.

ⓘ *More information on object automation can be found in the section* Automation *(↗347)*

Fades

In this view you can set the Start, Length, End time positions and Fade parameters for the selected object. In the **Content** section you can change the audio file used by the object.
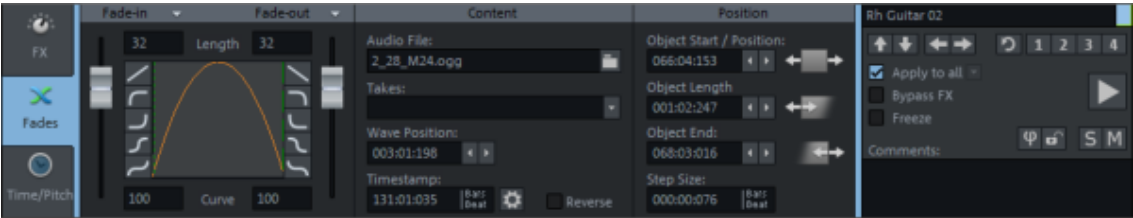
Fade In/Fade Out
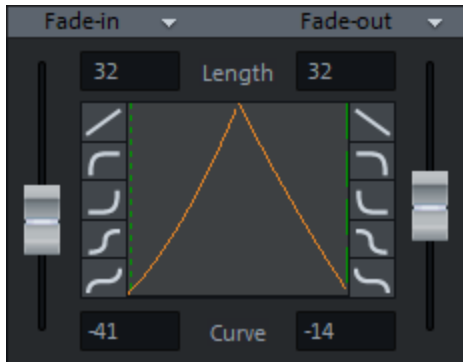
Fade in and fade out can be edited in this area. In the fields **Length** you can specify the fade lengths numerically in ms.

The sliders can be used to adjust the curve shape of the fades. Double-click the fader to switch the fade between **Linear** and the last setting. For the default curve shapes, there are corresponding buttons to the left and right of the graph. In the **Curve** fields you can specify the curve shape numerically (-100...100).

There are more options in the menus next to **Fade In** and **Fade Out** ▼ :

- **Reset** removes the fade completely.

- **Allow crossfade (to the left)** decides whether the object cuts off an object in front of it or whether it forms a crossfade (↗169) with it.

- **Asymmetric crossfade (to the left)** removes the coupling between the fade out of the previous object and fade in of the current object for such crossfades.

- **Linear**, **Exp.** (exponential), **Log.** (logarithmic), **Cos.** (Cosine) and **Sin.** (Sine) correspond to the preset waveforms of the buttons. With **Load...** you can apply additional waveforms from the crossfade editor (↗174) in the object editor.

- The following three options specify the proportion of the fade that should be inside or outside the original object boundaries, see Fade-Overlap (↗174) in the Crossfade Editor.

  - **Fade inside**: The fade is located completely within the object, overlap is 0%.

  - **Fade symmetrical**: The fade is symmetric to object borders, overlap is 50%.

  - **Fade outside**: The fade is completely outside of the original object edge, overlap is 100%.

  The object will be lengthened accordingly. The original object edge, which now serves as the "axis" of the fade, so to speak, is shown as a dashed line. For an overlap above 0%, audio must still be present outside the object

boundaries so that the object can still be faded in or out. For example, if the object start coincides exactly with the beginning of the audio file, it will no longer be possible to fade in when the overlap is greater than 0%.

- With **Get global crossfade** the crossfade settings are set according to the default settings for automatic crossfades. With **Set global crossfade** the current crossfade settings can be set as default values for automatic crossfades.

ⓘ **Note**: *For detailed crossfade editing, you can also use the* professional crossfade editor (↗170) *(Keyboard shortcut: **Shift + F**).*

Content

Audio file: Here the referenced audio file of the object can be exchanged, copied and renamed. For more information see Changing the audio file under the object (↗166)

Takes: Select here the desired take (↗328) that the object uses.

Wave Position: The start position of the object within the audio file. This can be moved with the arrow buttons, this corresponds to the function Moving audio material under the object (↗154).

Timestamp: The value corresponds to the original position (↗153).

Gear: The gear button opens the Broadcast Wave Manager (↗650), which allows you to assign metadata to the underlying audio files.

Reverse: With this option the object will be played backwards.

Position

With these controls you can enter the object parameters **object start/position**, **object length** or **object end** numerically.

With the arrow buttons behind the values you can change the positions step by step. In the field **step size** you specify the step size for the arrow buttons. The graphics behind the buttons clarify their function.

ⓘ *When changing the object length step by step, the object end is kept fixed und the object start moves in the object. If you want to change the length without moving the start, change the position of the object end instead. When entering the object length numerically, it is always the end of the object that is adjusted.*

By clicking on the unit of measurement to the right of the step size, the unit of measurement can be changed. In the lower part of the menu you will find presets for the respectively selected unit of measurement, e.g. for the unit of measurement bar/beats different musical step sizes (1/16, 1/8 …)

ⓘ *The step size here corresponds to step size 1 for the function* Move objects step by step (↗151)

Time/Pitch
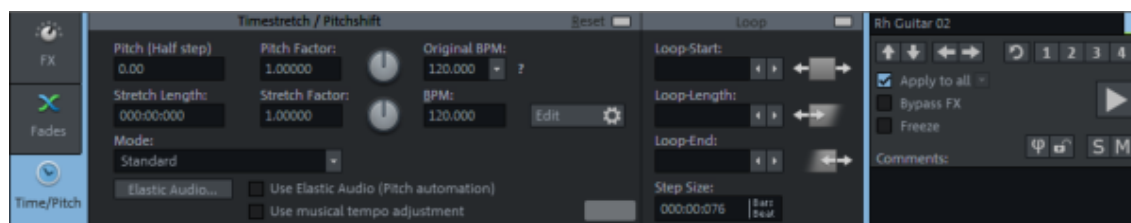
Timestretch/Pitchshift

In this view, objects can be changed in pitch (pitch shifting) and tempo (time stretching). Except in "Resampling" mode, this can be done simultaneously and independently.

The upper knob controls the pitch shifting (**Pitchshifting**). You can also enter the pitch change in the fields **Pitch** in semitones and cents or as **Pitch factor** relative to the initial value.

> 💡 *If you move the mouse over the field **Pitch**, you can set the pitch change in 10 cent steps with the **mouse wheel**, in cent steps with **Ctrl + mouse wheel** and in semitone steps with **Shift + mouse wheel**.*

The lower knob controls the tempo change (**Timestretching**). You can also enter the tempo change in the fields **Stretch Length** as a length change or **Stretch Factor** as a relative factor to the initial value, or enter the desired tempo in **BPM** (Beats Per Minute).

To calculate the stretch factor from the BPM input, the original tempo is needed.

 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.
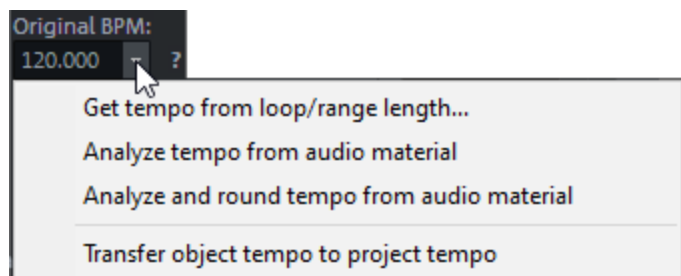
- **Get tempo from loop/range length**: A range section above the object is used to calculate tempo. Select the option and enter how many quarter notes should be included in the selection.

- **Analyze tempo from audio material**: Melodyne will calculate the tempo. Melodyne must be installed for this. More on this at Melodyne integration (↗222).

- **Analyze and round off tempo from 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**.

The button **Edit** opens an extended display of the timestretching/pitchshifting parameters.

> ℹ️ *Read more about the algorithms and the advanced parameters at* Resampling/Timestretching/Pitchshifting *(↗297) in the Effects section.*

The **Elastic Audio** (↗265) button opens the selected object in the Elastic Audio editor.

Elastic Audio(use Pitch Automation): If the option is active, pitch shifting is performed using Elastic Audio (↗265), the pitch shifting controls in the object editor are then disabled.

Use musical tempo adjustment: adjusts the playback tempo of the object to the project tempo (tempo map). For more information, see the section Musical Tempo Adjustment (↗364)

Loop

Here you will find the settings for looping objects. For more information, see section Looping objects (↗161).

SOURCE DESTINATION CUT

As an important editing function in **Sequoia**, source-destination cutting is particularly suitable for classical music productions, for montage work, and for current productions, whereby editing operations can be performed while a recording is still in progress.

The source-destination cutting method effectively and intuitively combines the advantages of a linear audio editing suite, consisting of a player and recording device, with those of a non-linear editor. In a special source-destination cut mode (↗202), two screen areas allow simultaneous and independent display and playback of the source tracks and the destination tracks. The lower monitor area (Source) displays the source material, i.e. all of the tracks recorded (including all of its takes).
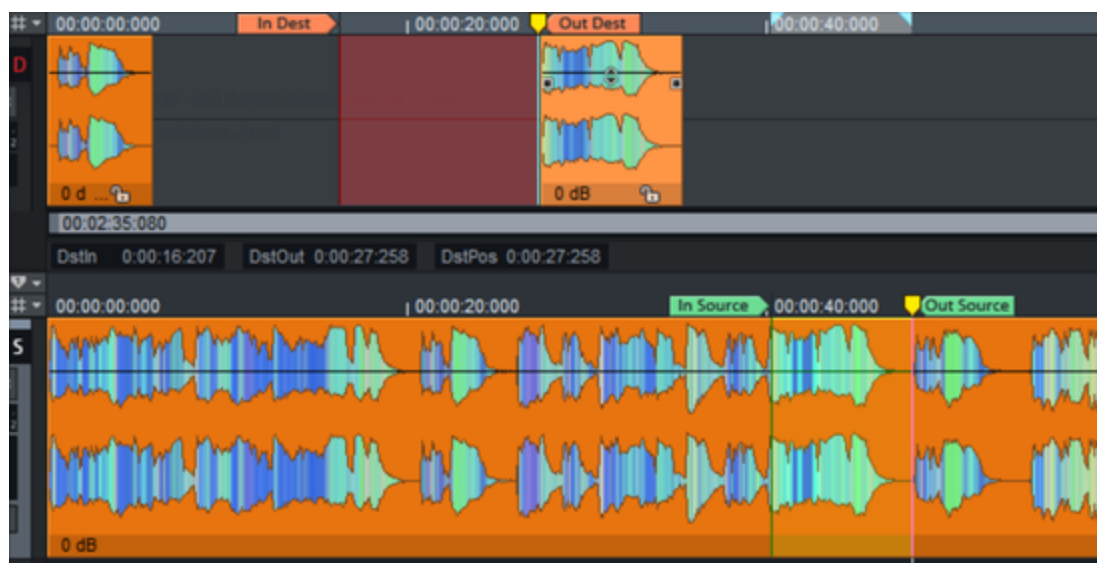
The upper area contains the target tracks (Destination). Here the edited material can be arranged using the mouse or the keyboard. You can move independently in both areas, scroll the section, zoom and play. You can move through the source while the destination is being played back to prepare for the next cut.

The actual editing takes place via markers. In and out markers in the source are used to define the material that is to be copied to a marker position in the destination by the cut command.

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 moved on the new cut borders synchronously over all tracks.

You can use the source-destination cut features in any multitrack project. There are also additional modes available, i.e. S/D Special Mode (↗202) and Multi-Source Session (↗201) and the Multi-Synchronous Cut (↗205).

Source and Destination Tracks

A **Source track** is a track with source material. In Source tracks, set the In Point (Source) and Out Point (Source) markers to select the audio material to be cut.

The audio material of the source track is compiled in the **Destination track**. Set here the In Point (Destination) and Out Point (Destination) markers to indicate the area where audio material will be inserted.

To define a track as a source or destination track, click the arrow to the right of the track name in **Track Editor** or **Track Head**.

These and other commands for this can also be found in the menu **Track** > **Source/Destination**:

- **Normal/Source/Destination**: Set the mode for the selected track and the tracks below it, depending on the number set at **track number**.
  Keyboard shortcut: **S** (Source) / **D** (Destination)

- **Source/Destination track number…**: Set the number of consecutive source-destination tracks in this dialog.

- **Swap source and destination**: With this command, source and destination tracks swap modes.

- **Shift source down/up**: The selected source track changes to normal mode and the respective track above or below it becomes the new source track.
  Keyboard shortcut: **Alt + Page up/down**

A track can also be both source and destination, e.g. to cut audio forward within a track from a later time period.

You can also define source and destination markers in different project types, for example, the destination markers in the current virtual project and the source markers in a wave project (audio file). A wave project can only be used as a source.

To perform cuts between different virtual projects, either create a multi-source session or define additional source projects via the Source Manager (↗330).

Commands for 4-point cut

All commands for the 4-point cut can be accessed via the menu **Edit** > **Source/destination cut**.

There are buttons on the upper toolbar for the most important commands for executing cuts and setting and deleting In and Out markers. These are turned off by default. They are displayed when you select the workspace (↗58) **SD editing**.

Set and delete In/Out points

An In Point marker defines the start point of a cut and an Out Point marker defines the end point of a cut.

The In and Out Point markers can be set with the keyboard or with buttons in the upper toolbar. To display the source-destination cut buttons, it is best to switch to the Workspace (↗58) **SD-Editing** (menu at the bottom left of the program window).

There are buttons for the most important commands for setting and deleting the source/destination markers.

Command	Keyboard shortcut
In Point (Source)	**Ctrl** + **Home**
Out Point (Source)	**Ctrl** + **End**
In Point (Dest.)	**Shift** + **Home**
Out Point (Dest.)	**Shift** + **End**
Delete source points	**Ctrl** + **Alt** + **Page Down**
Delete destination points	**Ctrl** + **Alt** + **Page Up**

All commands for setting and deleting the In/Out markers in Source and Destination are located in the menu **Edit** > **Source/Destination Cut** > **Set/Delete In/Out Point(s)**. For In and Out points there are commands to set and to delete, respectively for Source, Destination or unspecifically for Source or Destination. You can use the latter in **Source-destination cut mode**; the command then always takes effect depending on the focused project section. Source markers are set in the active source section and destination markers in the destination section.

> ℹ️ *The keyboard shortcuts for these commands are logically structured: **Home/End** determines **In/Out**, + **Ctrl** for the source points, + **Shift** for the destination points, + **Alt** to delete the respective points. The non-specific keyboard shortcuts are composed of **Home/End** for **In/Out**, with **Ctrl + Shift** for setting and **Alt** for deleting the points.*

Set In Point	Ctrl+Shift+Home
Set In Point (Dest.)	Shift+Home
Set In Point (Source)	Ctrl+Home
Set Out Point	Ctrl+Shift+End
Set Out Point (Dest.)	Shift+End
Set Out Point (Source)	Ctrl+End
Delete In Point	Alt+Home
Delete In Point (Dest.)	Alt+Shift+Home
Delete In Point (Source)	Ctrl+Alt+Home
DeleteOut Point	Alt+End
Delete Out Point (Dest.)	Alt+Shift+End
Delete Out Point (Source)	Ctrl+Alt+End
Delete Destination Points	Ctrl+Alt+Page up
Delete Source Points	Ctrl+Alt+Page down

You can delete both edit markers with the commands **Delete destination points** and **Delete source points**.

Cut Commands

Insert cut: This command executes the cut process. Audio material present in destination tracks is overwritten.

Keyboard shortcut: **F9**

Insert with ripple: This command also executes the cut process. Audio material present in destination tracks is moved forward.

Keyboard shortcut: **F10**

Delete and ripple: This command deletes the cut range in the destination tracks and closes the resulting gap by moving the following audio material.

Keyboard shortcut: **F11**

Delete silence: This command deletes the cut range in destination tracks. Following audio material is not moved.

Keyboard shortcut: **F12**

Cut commands with crossfades

If the Auto Crossfade mode is active, crossfades are created accordingly for the cut commands. For the shape of the crossfades, a special crossfade preset (↗178) **Default_CrossFade_SD** is used. The preset can be customized in the Crossfade Editor.

Every common cut command is also available in the variant **Blackfade**. Here, the beginning and end of an object are faded in or out. These blackfade cuts use the crossfade presets **Default_FadeFromBlack** and **Default_FadeToBlack**.

For keyboard shortcuts the keyboard shortcuts of the normal editing commands + **Shift** key are used

Blackfade insert	Shift + F9
Blackfade insert with ripple	Shift + F10
Blackfade Delete and Ripple	Shift + F11
Blackfade delete silence	Shift + F12

Other cut commands

Insert with time stretching: In this case, the source material is time-stretched between the destination in-point and destination out-point. Existing material in the destination tracks is replaced with the material from the source tracks.

This enables the timing of drum loops to be adjusted to the existing arrangement, without the need to apply complicated operations. The time-stretching is done non-destructively.

Keyboard shortcut: **Alt + F9**

Insert at Source Position: You can use this cut command in a dubbing cut. The original time position of the source material is used as the destination position. 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.

Keyboard shortcut: **Ctrl + Alt + F9**

Zoom commands for the 4-point cut

The commands **Jump to In Point/Out Point** move the play cursor to the Destination In or Out point. If no corresponding Destination point is set, the program jumps to the Source In/Out point.

Keyboard shortcut: **Ctrl + Shift + Page Up/Page Down**

The commands **Zoom to In Point/Out Point** move the play cursor to the Destination In or Out point and additionally adjust the zoom level according to the 4 point cut preferences (↗204). The preset value is a time of 2000 ms.

Keyboard shortcut: **Ctrl + Alt + Page Up/Page Down**

2 Point Cut

In a 2-point cut, one cut marker is set for the source track and one for the destination track. The source material will be transferred according to the set cut marker into the destination track. The 2-point cut offers you three different options:

1. You define the In Point (Source) and the In Point (Destination).
   With the cut, all audio from the source track is inserted after the In point at the In point of the destination track.
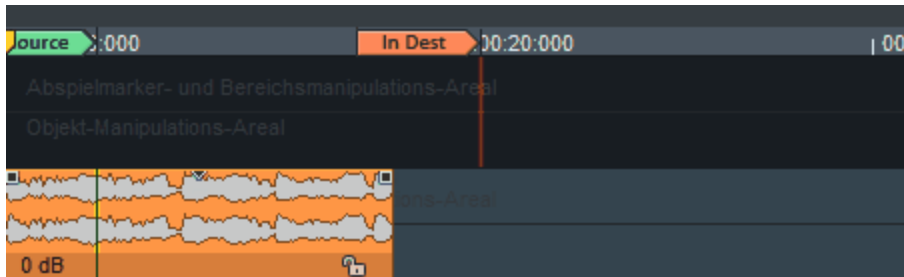
2. You define the Out Point (Source) and the Out Point (Destination).
   When cutting, all audio material on the source track before the out point is inserted before the out point of the target track, starting from the beginning of the target track.
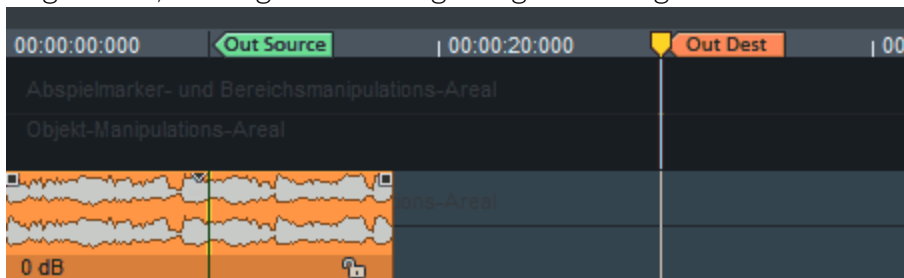
3. You define the In Point (Source) and the Out Point (Destination).
   When cutting, all source material behind the In Point is inserted, starting from the beginning of the target track to the Out Point of the target track.
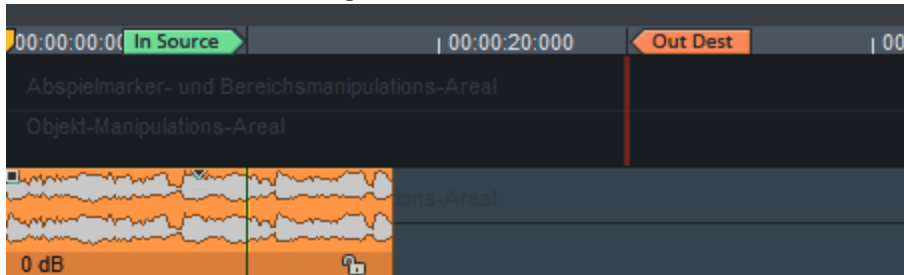
3 Point Cut

In a 3-point cut, a range is defined by setting an in and out point on either the source or destination track, while the third cut marker is set on the other track. This means that only one of the four section markers is not set. The cut is made with the length specified by the complete In/Out Point pair.

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 into the destination track from the In point (source) with the length specified by the destination In/Out points.
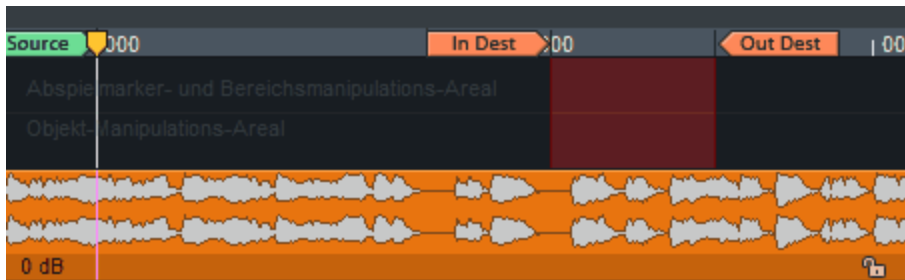
If you have set the In and Out points in the source track, but only the In point in the destination track, the source material specified by the In/Out points of the source will be cut into the destination track from the In point (destination) when cutting.

If you have set the In and Out points in the target track, but only the Out point in the source track, the source material is cut into the target track before the Out point (source) with the length specified by the target In/Out points.
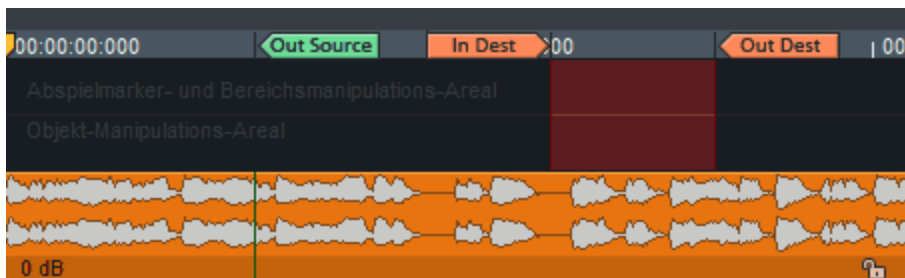
If you have set the In and Out points in the source track, but only the Out point in the destination track, the source material specified by the source In/Out points will be cut into the destination track before the Out point (destination).
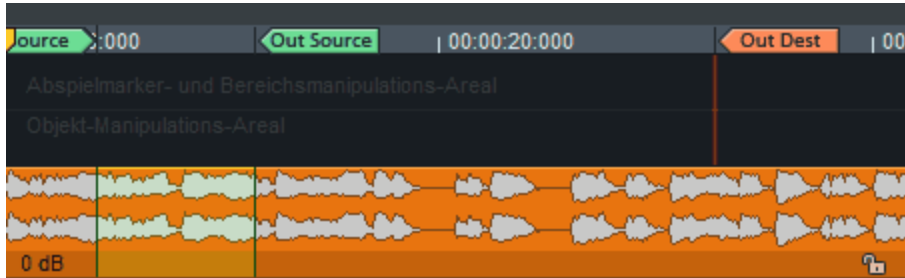
4 Point Cut

"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/Out point range of the source.
- The In/Out Point range of the destination will be replaced by the audio of the source.

If the source and destination ranges have different lengths:

- If the destination range is shorter than the source range, the audio following the out point (destination) is shifted to the right so that the source audio is completely fitted in and the following destination material is not overwritten.
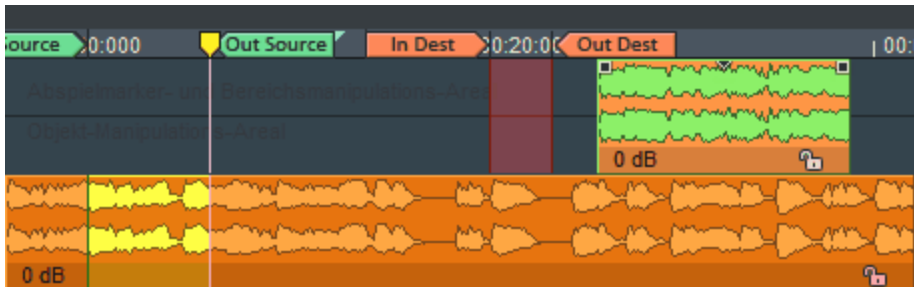
- If the destination range is longer than the source range, the source range is inserted and the audio material

following the out point (destination) is moved to the left so that it moves up to the inserted material.
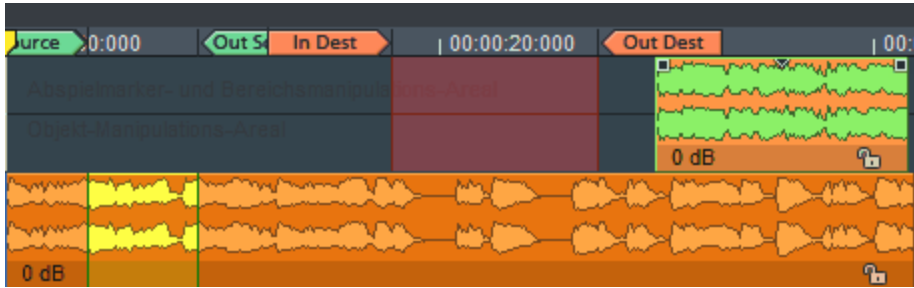
ℹ️ **Note**: *A 4 point cut can also be performed as a time stretch cut (menu **Edit** > **Source/Destination Cut** > **Insert with time stretching**; Shortcut: **Alt + F9**). In this process the contents of the source range are stretched or compressed to the length of the destination range. The main application of this is dubbing cut. Please note that time stretching has its limitations and can result in audio artifacts. Preset in **Sequoia** is a stretch factor range from 0.5 to 2.0. If you want to stretch or compress the pasted material more than that, you can do it in the Object Editor or with the Pitchshift/Timestretch mouse mode.*

Multi Source Session - S/D Cut with Multiple Projects

With source-destination editing with multiple projects (multi-source session), audio material from one project is transferred to another using the source-destination editing functions.

In a multi-source session, there are multiple projects, a destination project and at least one source project. Ranges from the source projects can be transferred to the destination project using the source-destination cut.

In a Multi-source session it is easier to sort audio material in context, such as source projects for cues, dialogs, MuSyC VIPs, or or multiple virtual projects for compiling a CD.

Multi-Source Session – Creating a New Project

In the **File** menu, select **New virtual project...**. Enable the **Multi Source Session** option and specify the number of source projects you want. Confirm this by clicking **OK**.

A destination project and the desired number of source projects are then created by **Sequoia**.
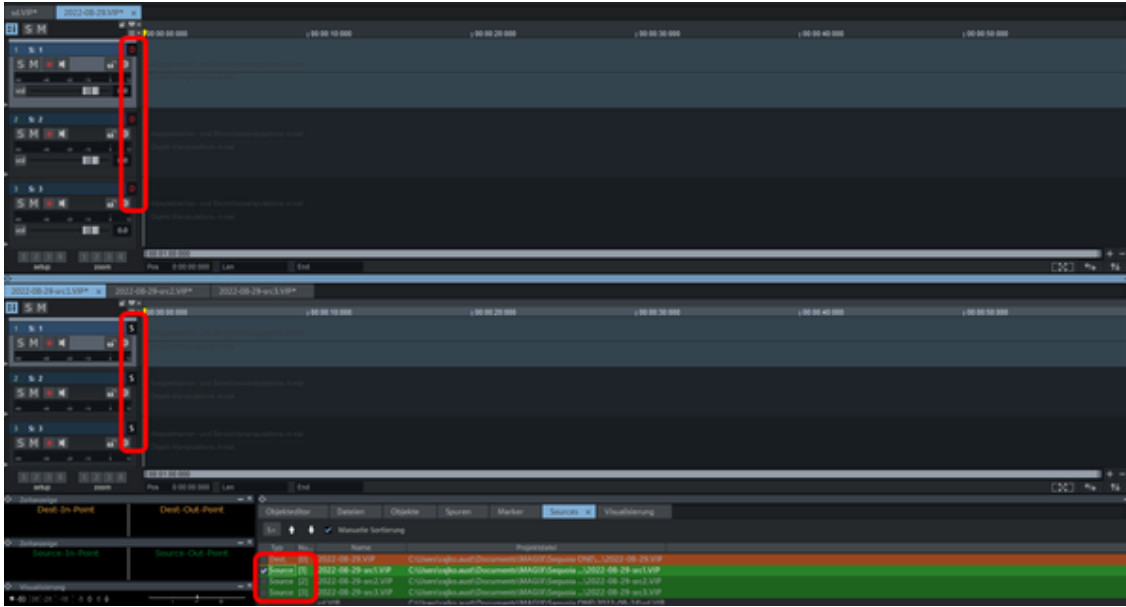
> ℹ **Notes**: *The individual windows can each be maximized. You can then continue to switch between the last activated source project and the destination project by using the **Page up** and **Page down** keys to switch. You can also use the tabs at the top of the project docker to switch between the individual source projects and the destination project.*

Opening a New Source Project or Creating one Afterwards

To subsequently integrate a new source project into an existing multi-source session, select the **Source** entry in this project from the context menu of the Track head or the Track editor. Assign the project to an opened destination project via Source list (↗330). On the source tracks of the project, you can now compile your cut in the usual way and transfer it to the destination project.

Selecting a Source for the Cut

Normally, you always cut from the last active source project. In all other projects, the source markers are grayed out. Activate the window of another source project to select it as the source. You can also select the source project in the source list (↗330).

Source Destination Cut mode (S/D special mode)

The **Source Destination Cut mode** function from the sets the project window to a special mode consisting of two separate track areas, whereby the destination tracks are 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. To specifically set an existing

audio track of your project as source or destination track, select this track as source or destination already before switching to the special mode.

You activate the source-destination cut mode with the button on the toolbar (WorkspaceWorkspaces (↗58) S/D-Editing) or menu **Edit** > **Source/destination cut** > **Source destination cut mode**
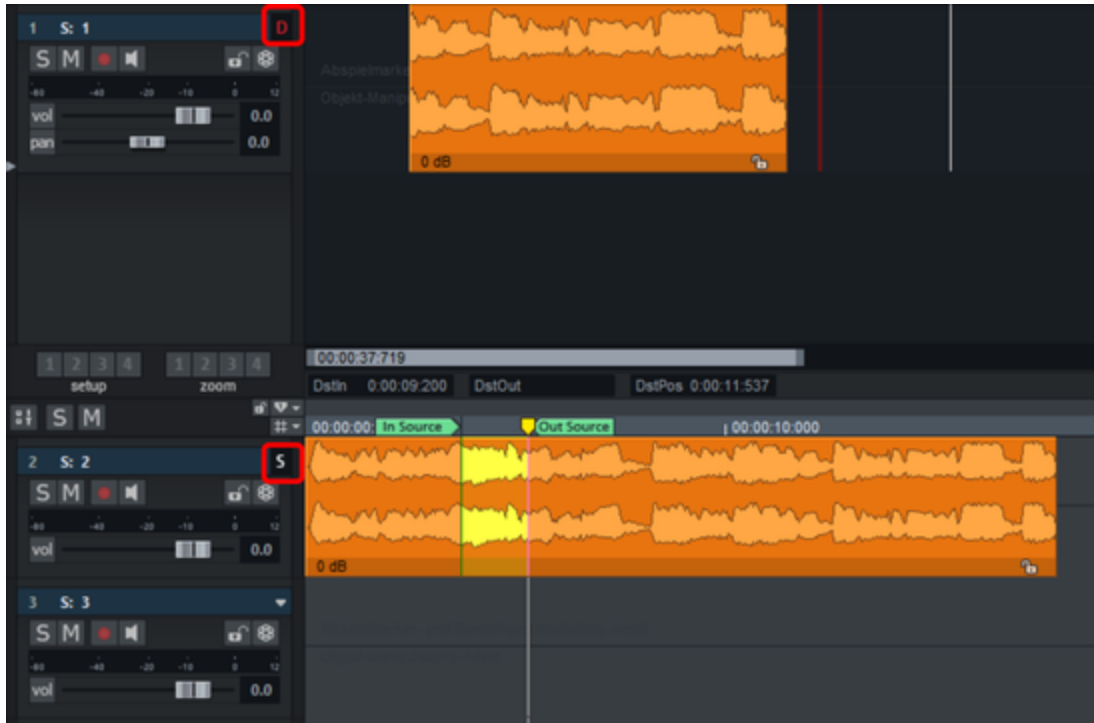
In both track areas, you can change visible section and zoom level independently from each other. The currently active track area is displayed highlighted. You can switch between source and destination area by simply clicking on a corresponding track or by using the buttons **Page up** and **Page down**.

In both track areas, the play cursor can be moved and ranges can be selected independently. Playback control always takes place in the active track area. However, you can change the track area without affecting the playback and make changes there.

Recording and loading audio files is always done in the source track area at the bottom.
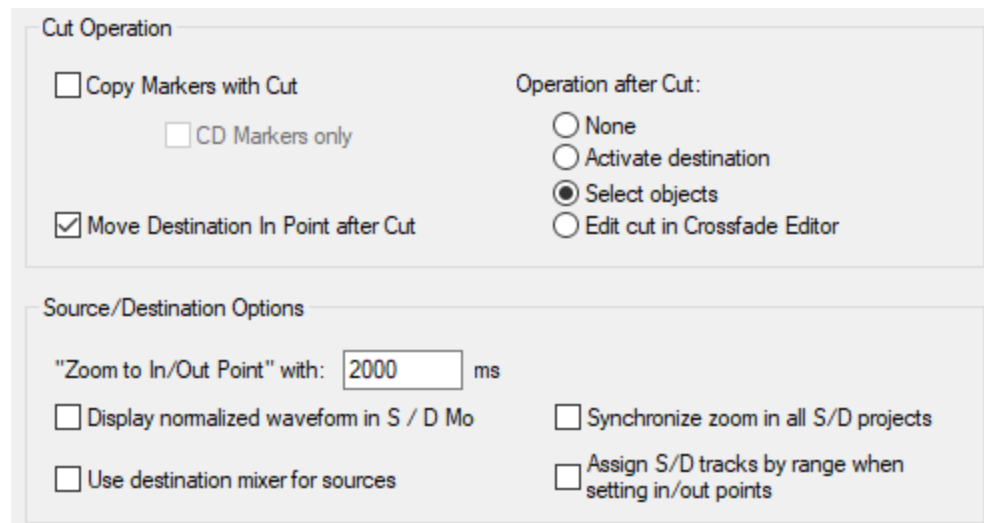
In the source-destination cut mode, proceed as follows:

- **Assembling raw material in the source**: The raw material can consist of several recording takes that are to be assembled. Thereby you can record directly into the source track or load already existing audio files into the source track.

- **Marking important spots in the raw material**: Mark the use of an instrument in the different recording takes to better compare the recordings. You can use project markers on the timeline of the source track and audio markers directly in the audio material for marking.

- **Cut commands/Setting In and Out points**: All cut commands (↗195) as well as the setting of the cut markers (↗198) work in the same way in S/D special mode like with normal source destination cut.

S/D Cut Settings

You can reach the settings dialog for the source-destination cut via the menu **Edit** > **Source/Destination cut** > **4-point cut preferences** or via the **Program Preferences** dialog > **SD cut/ MuSyC**.

Cut operations:

- **Copy markers with cut**: The set markers will be taken over to the destination during cutting.
- **CD markers only**: During editing, only CD markers are transferred to the destination.
- **Move Destination In Point destination after cut**: This option causes the In Point in the destination after each cut to be placed at the end of the current cut. You can then immediately cut more material without having to set the Destination In Point manually each time.

Operation after cut:

- **None**: With this option you avoid the change of focus to the destination and leave the focus in the source.
- **Activate destination**: This option activates the destination after the cut.
- **Select objects**: After a cut is made, the object cut into the destination is selected.
- **Edit cut in crossfade editor**: After a cut is made, the last object inserted into the destination opens in the Crossfade Editor (↗170).

Source Destination Options:

- The setting **Zoom to In/Out Point with ... ms** is related to the commands in menu **Edit** > **Source/Destination Cut** > **Zoom to In/Out Point** (↗197). Specify the time in milliseconds for the zoom display for the in/out-point. The keyboard shortcut Ctrl + Alt + Shift + Page up/Page down moves the play cursor in the currently selected track by the zoom factor to the in/out-point. The preset value is a time of 2000 ms.

- **Display normalized waveform in S/D mode**: This option activates the vertical auto zoom in Source Destination Cut mode. With this, the tracks are optimized vertically depending on the window size, i.e. the visible wave section is displayed normalized to 100%.

- **Synchronize zoom in all S/D projects**: If this option is active, the zoom factors are synchronized in all source destination projects and the MuSyC overview project.

- **Use destination mixer for source projects**: Source projects and destination project are played through the same mixer when this option is active. This mixer is also opened when the overview project is active in MuSyC. Folders and buses are ignored during the assignment, a uniform routing applies. Mute/Solo keyboard shortcuts and track head controls control this mixer in all projects.

- **Assign S/D tracks by range when setting in/out-points**: This option integrates tracks that have not been defined as source or destination track in the cutting process. If you drag a range in a track that was not previously defined as a source or destination track, the placement of a source or destination marker at the edge of a range will define this track automatically as source or destination track (depending on the marker), and this will be indicated accordingly in the track head.

MuSyC - Multi-Synchronous Cut

The **multi-synchronous cut** (MuSyC) facilitates the editing of multitrack recordings with a very large number of tracks and with several successive recording takes, as is often the case with classical music recordings.

For pop music or similar, recordings of multiple takes of the same section can be made *on top of each other* in the same section of the project. The tempo and musical position is clearly defined, often guided by a click track or pre-recorded guide track. Takes are created that can be exchanged on the object and conveniently selected and combined using Take Manager (↗328) and the Take lanes (↗138).

In classical recordings, working with takes in this way is impractical: a large number of tracks are often recorded, but they are never treated separately. The tempo is free, there is no guide track, and the interaction between orchestra and conductor should be as smooth as possible. An additional ongoing dialogue between the conductor or sound engineer would be required in order to synchronize the recording in the project with existing takes during the recording.

Therefore, the recording simply runs through during such recording sessions and after it the recording is available in different takes in one or more source projects *one after the other*.

If the recordings are later to be edited together with the source destination cut, this creates the problem of having to constantly jump back and forth in the project or switch between projects to compare takes and find the right sections. This is where multi-synchronous cut comes in handy:

Based on a content analysis of the audio material, different recording takes of the same passage of a piece of music are recognized and displayed one below the other in a special **overview project**. Each take of a multitrack recording is represented by a single reference object on its own track, with musical passages recognized as the same being displayed on top of each other.

The selection of the takes is done by selecting the track of the reference object and the selection of a time section (Source In/Out points). Without changing the project window or manually moving the play cursor, the takes can be compared and combined in a similar way to comping with the take lanes. "Visual time stretching"(**time alignment**) is used to synchronize passages that belong together musically, even with slightly different tempos: The waveforms are slightly compressed or stretched to synchronize the takes. However, this display only applies to the overview project; the audio material is displayed normally in the destination project.
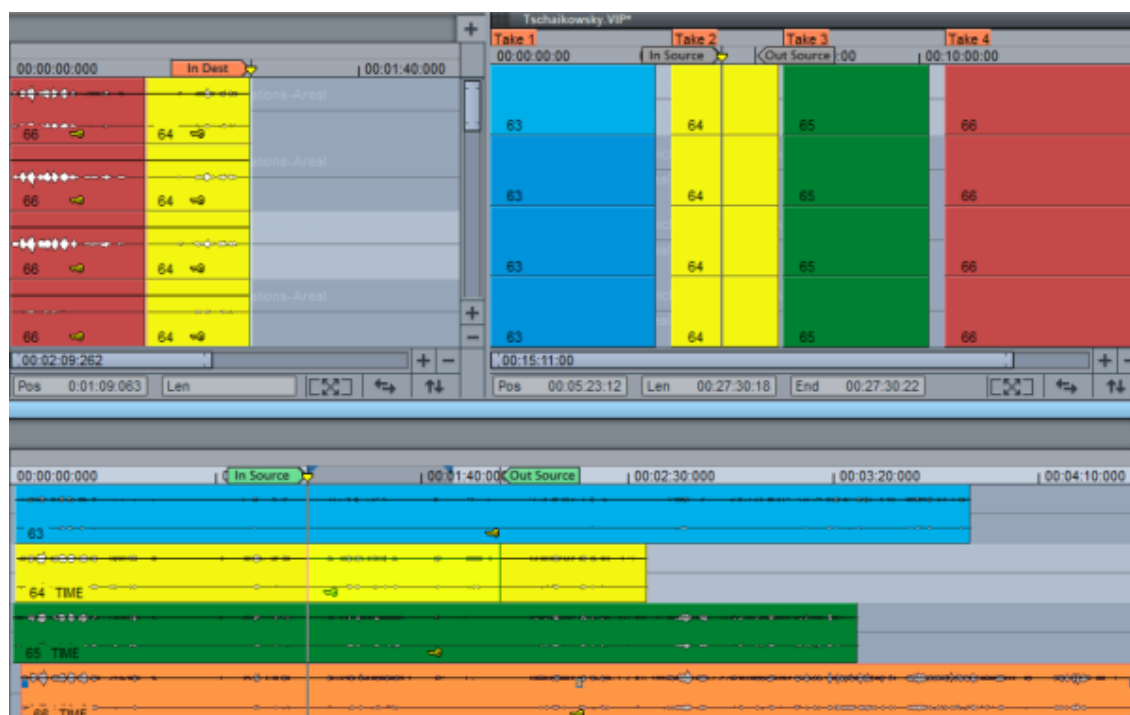
*In the overview project below the take is selected, the area to be used from the take is defined by the source markers. In the source project on the right, you can also see the source markers. When performing the cut, the source material from all tracks is cut into the Destination project at the position **Destination In**.*

MuSyC Preparation

Before performing multi-synchronous cut, the Sequoia session must be prepared accordingly. To do this, open the dialog **Multi-Synchronous Cut (MuSyC) - Preparation** with menu **Edit** > **Multi-synchronous cut** > **MuSyC Preparation...**.
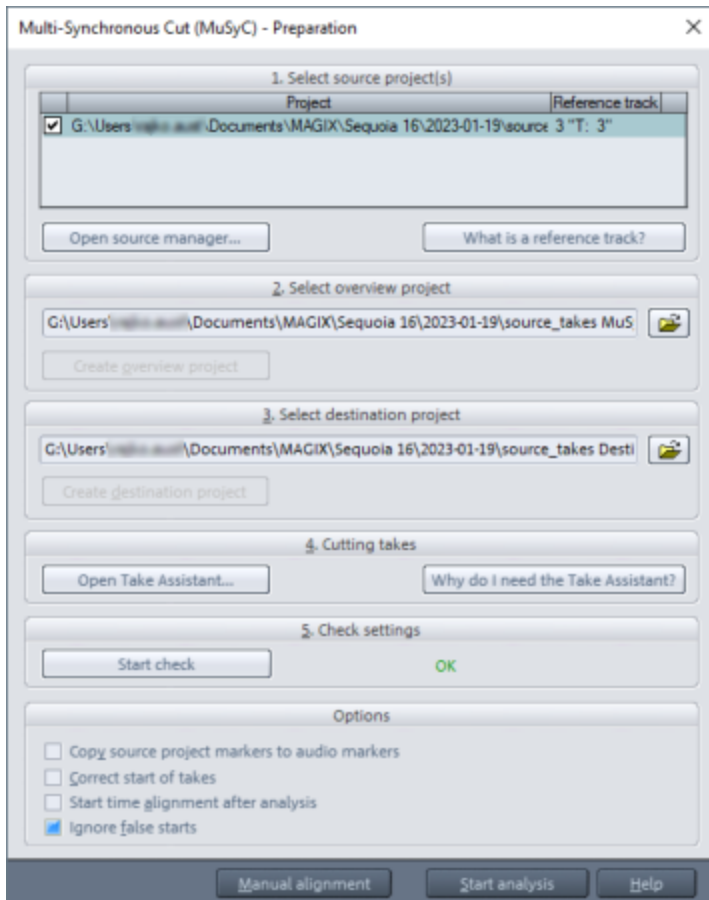
1. **Select source projects**: You are working with MuSyC in a multi source session (↗201). Source and destination project means that the project in which the cut version is created is different from the one in which the raw data was recorded. All source projects are displayed in the list.
   You use the Source manager (↗330), which can also be opened from here with the button, to specify which of the loaded projects act as source projects:

   a Load all the source projects you want to use.

   b Select the projects in the **Source Manager** and press the button **Add active source project to the list** (**S+**). Unused source projects can be deleted from the source list by selecting them and pressing the **Del** key.

   c Activate the source projects one by one and set the reference track (see below).

2. **Select overview project**: The overview project is displayed in the field, an existing overview project can be loaded via the folder button. With **Create overview project** a overview project (↗209) can be created for the current MuSyC session, if one does not already exist. The button opens the dialog **New project**.

3. **Select destination project**: In the same way, a destination project can be loaded here and, if necessary, a new one can be created.

> 💡 *When saving a MuSyC overview project, a **Sequoia** session file (`*.samx`) is suggested to be saved together with the project, which contains all opened projects and their layout. This allows you to load overview project, source and destination projects together at a later time.*

4. **Cutting takes**: For the analysis to work, it is essential that all takes in the source projects are in separate objects. The Take assistant (↗205) helps to find takes based on silence in the reference track of the source project and to cut the objects on all tracks at these locations to create new takes that MuSyC can use for analysis.

5. **Check settings**: Click **Start Check** to verify the current settings. If a green **OK** appears, everything is set correctly.

6. **Options**: Before starting the analysis you can set some more options:

   a **Copy source project markers to audio markers**: Use this option to convert source project markers into audio markers.

   b **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 adjustment at the beginning of the music from being inaccurate, the beginnings of the takes must not contain several seconds of silence or other sounds that do not belong to the music. You can find help with this task using the Take assistant (↗211).

   c **Start time alignment after analysis**: This option starts the time alignment automatically after analysis. If you leave this option off, you can check the overview project after the analysis and correct it if necessary before you start the time adjustment (**Edit** menu > **Multi-Synchronous Cut** > **Time Alignment** > **Align all objects**).

   d **Ignore false starts**: Use this option to ensure that false start takes are excluded from the analysis.

7. With **Start analysis** the analysis of the source material is started and the overview project is filled with the takes found: Takes recognized as the same are arranged one below the other, all other takes are arranged one behind the other on the first track.

8. Now you can edit your takes from within the overview project (↗209) using the source-destination edit.

MuSyC Reference Track

The reference track is the track in the source project that is used to identify the various takes in the source project that are to be displayed synchronized in the overview project. It is therefore the "master track" for the multi-synchronous cut.

Any mono or stereo track of the source project can serve as a reference track. Typically, you will use the track of the main microphones as a reference track. A rough mix can also be created in a new track and this can be selected as the reference track.

The reference track is the track that is selected in the source project at the moment when the analysis is started and the overview project is created. At the top of the **Multi-synchronous cut preparation** dialog, in the list of source projects, you can see which track is set as the reference track in the **Reference Track** column for each source project.
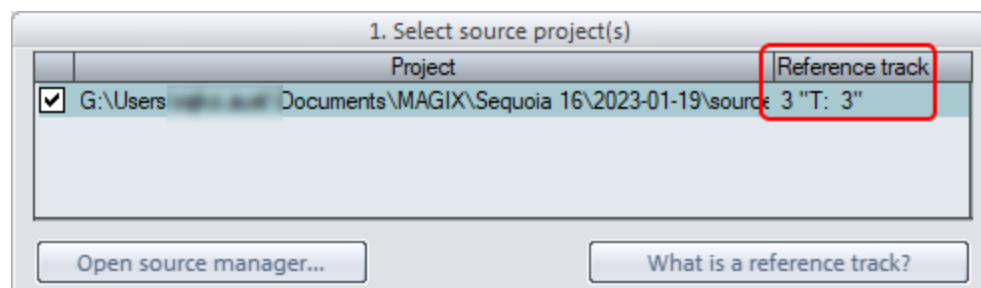
> ℹ **Note**: *Make sure that the reference track contains audio material as continuously as possible, i.e. that there are no long pauses.*

Cutting with the MuSyC Overview Project

After the analysis, the takes of the reference tracks of the source projects are arranged in the overview project. The overview project does not display the entire audio material in the chronological sequence of takes determined by the recording, as normal projects do. Instead, it presents an overview mode for better handling of all takes of all source projects collected, sorted by musical context.

The representation of the objects on the tracks is used to sort and select the takes: The takes - with all tracks on each - from the source projects are represented by a single reference object for each one. Matching takes are displayed synchronized in time one above the other, different takes one after the other. Small time deviations are compensated visually by the time alignment. The adjusted objects are marked with **TIME**.

The takes in the overview project are thus "proxies" for a particular take in a particular source project: the selection of the track and the positioning of the play cursor determine which take it is. Accordingly, the play cursor is also moved to the appropriate position in the associated source project.

In the overview project, therefore, only exactly one track can be active at a time, which thereby determines the active take. When you start playback, this take will be played back from the playback position. If you change the active track during playback, you can compare takes this way. Click in the track header or on an object to do this.

> ℹ *Normally the reference track of the source project is played. To play the entire source project, use **Edit** > **Multi-synchronous cut** > **Play source**. The default behavior can be changed in the system settings for MuSyC (see below).*

If you select a range in a take in the overview project or set a source-in or source-out marker, the markers are set to match the take or the corresponding range is selected in the corresponding source project. This connection between these time positions in an object of the overview project and those of the associated take in the source project is independent of the position of the object in the overview project. The positions in the overview project are meant to mark audio material as "simultaneous" that is present in the source project in succession. You can therefore also

move the objects in the overview project to correct errors in the analysis or to manually arrange the objects one below the other and thus use them as different recording takes of the same musical material.

> **ℹ** **Note**: *If, after analysis, you move objects that represent different takes of the same piece and for which you have performed a time alignment to visually align the takes, you must perform the time alignment again after moving it (menu **Edit** > **Multi-synchronous cut** > **Time alignment** > **Align selected objects**). The time reference is always the top object.*

So you use the overview project to select audio in the source project for an edit. Then, when you perform an edit, the audio is copied from the source project to the destination project; the overview project is not involved. You can perform the actual editing using the source-destination cut functions, as described in detail above. However, it is also possible to simply copy and paste a range selection from a take from the source to the destination project without SD functions.

> **ℹ** *To do this, the option **Command "Copy" copies source when called in overview** must remain active in the SD-Edit/MuSyC- options. If it is deactivated, the copied area contains only the content of the object from the overview project, i.e. only the contents of the reference track from the source project.*

MuSyC Menu Commands

The menu **Edit** > **Multi-synchronous cut** contains further commands for multi-synchronous cut:

- **Object Alignment:**
    - **Recalculate object position**: The analysis for arranging the takes is repeated. If you additionally select a range, only this range will be taken into account during analysis and reordering.
    - **Optimize project overview**: This command provides a more compact display of the overview project. Longer takes are preferably placed in the upper tracks. The longest take is moved to the top track, which is used as reference for the time adjustment.
    - **Find Audio...**: This function can be used to search for similar audio material across all loaded projects. The search uses the algorithms from multi-synchronous cut, but can also be used independently. Select a range in the audio material. The menu **Search in** can be used to limit the search to specific projects or the active project. Start searching. In the dialog, the search hits are listed, in which project they were found in which object at which position and how certain this search hit is (column **Relevance**).
- **Time alignment**: The commands in this submenu control the time alignment of the objects in the overview project, i.e. the visual adjustment of object starts and lengths to the take in the top track of the overview project.
    - With **Align all objects** the display of all objects is adjusted. Adjusted objects are marked with **TIME**.
    - **Align selected objects** adjusts only selected objects.

- With **Reset selected objects** you can reset the time adjustment.
- **Show true timing**: Use this option in the menu to toggle the option **Show true timings in overview** in the MuSyC system settings. For more information, see **MuSyC settings** (↗213).

■ **Fast insert**: You can use fast insert to speed up your work: This assumes that directly consecutive time ranges are combined one after the other from the different takes.

1. In the Overview project, select a range or place the In Point marker at the beginning and the Play marker at the end of a section in a take that you want to cut.
2. Run **Fast Insert**. (Again, keyboard shortcuts help)
3. The section is copied to the destination project in the normal way.
4. The In Point marker in the overview project is moved to the end of the area just inserted.
5. In the take that you want to cut the next section from, place the play cursor at the end of the section and perform the next fast insert cut.

Play source/Play destination: These commands can be used to play the respective projects, regardless of whether they are currently focused. For a meaningful use you should link them with keyboard shortcuts.

Activate overview (MuSyC): After executing a cut, the Destination project is activated. With this command you activate the overview project again. This command should also be linked to a keyboard shortcut.

4 Point Cut Preferences…: Opens the MuSyC settings (↗213) in the system settings.

MuSyC Take Assistant

For the analysis to work, it is essential that all takes in the source projects are in separate objects. The Take assistant helps to find takes based on silence in the reference track of the source project and to cut the objects on all tracks at these locations to create new takes that MuSyC can use for analysis.

And this is how you work with the Take assistant:

Place the play marker in front of the object you want to edit, which is typically an object on the reference track of a source project, and then open the assistant.
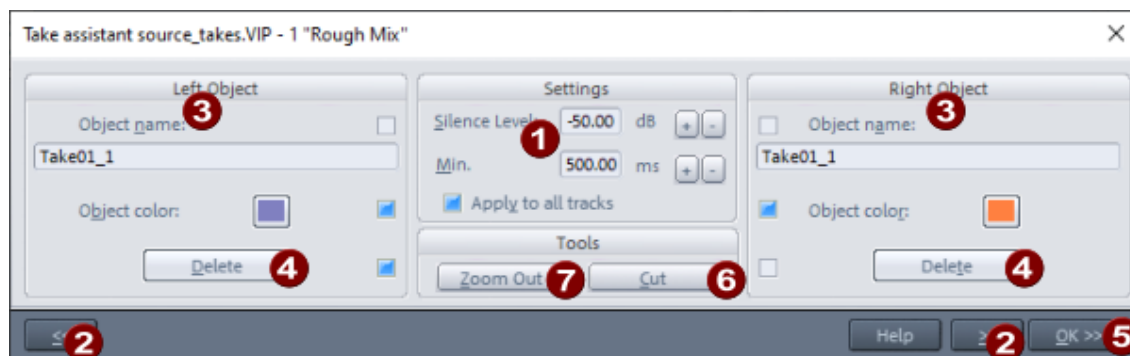
1 **Silence level**: This specifies a threshold value in dB below which the signal will be considered silence.
Min. duration: This value specifies how long the level in a part of the object must be at least below the specified silence level for its ends to be suggested as potential cutting positions.
+/- buttons: Use this to change these values step by step. To change the values directly, click in a field and drag the mouse up or down. Or enter the value numerically and press the Enter key to confirm.

2 Use the buttons **>>** and **<<** to move between the suggested cutting positions, i.e. to the start and end positions of the detected silence sections.

3 **Left/Right Object**: New objects are created when the objects are cut. In these areas, you can specify a new **Object name** (which acts as take name) and a new **Object color** for the objects to the right and left of the playback marker, which will be applied when the corresponding checkboxes are active and a cut is performed using the **OK>>** button.

4 **Delete**: Use this button to delete the objects to the left or right of the playback marker position. The deletion will also be executed when pressing **OK>>** if the corresponding checkbox next to the Delete button is enabled.

5 Pressing **OK>>** performs a cut at the play cursor position and applies all settings for **Right object** and **Left object** for which the checkbox is set.

6 **Set Cut** cuts the objects apart at the playback marker location without applying the changes for the right and left object.

7 **Overview**: With this button you zoom out the project for a better overview, so that it is completely visible.

The checkboxes in the Left and Right object sections are set automatically, according to a workflow where the **OK>>** button is used to step through the object and cut out the sections with silence one by one and create new takes:

- The checkbox for **Delete** is automatically activated if the new object created by the cut consists of silence, i.e. for the left object if the playback marker is at the end of a section with silence, for the right one correspondingly at the beginning of the silence section.

- If the new object does not consist of silence, i.e. a new take is created, the checkboxes for Object name/Object color are activated. If the name of the old object ends in a number, the number on the right is increased by 1 and the number on the left is decreased by 1 as the new name. The object color is set to a slightly different hue from the old object.

If the silence is detected correctly and you set the cuts one after the other with the OK>> key, this creates a series of new takes with ascending numbering and color gradient.

Manual Alignment

In certain cases, the automatic analysis to determine equal takes fails:

- There are similar or identical musical contents that cannot be logically distinguished by an algorithm (repetitions in music).

- Several independent pieces of music were recorded within one source project.

- Recording takes are not separated by object boundaries (continuous objects), e.g. for cyclic recordings of a passage in a single take/object.

- In the case of non-musical material or mixed material (e.g. speech or presentation)

Sometimes the analysis cannot be used in principle and it is still desired to combine different successive recordings, for example different variants of a recording of a jazz piece. The analysis would clearly classify the takes as different, since certainly there were not the same notes played in the solo parts. Still, it would be nice to use the best solos from the different takes.

The arrangement of the different takes below each other (and thus the definition as "same section") can also be done manually in these cases.

In the **MuSyC preparation** dialog select the option **Manual alignment** at the bottom. In this case, objects are created for all takes of the source projects (i.e. all objects in the respective reference tracks) one after the other on the first track of the overview project. The objects can now be manually moved on top of each other if you want them to serve as takes of the same piece.

> 💡 *If more tracks are needed for the takes than are initially available, a new track can be created automatically by dragging an object into the empty area below the last track.*

MuSyC System Settings

In the system settings in the section **SD Cut/MuSyC** (menu **Edit** > **Multi-synchronous cut** > **4 Point cut preferences…** there are some settings that impact working with MuSyC:

- **Automatic synchronization between projects**: These options concern the automatic alignment of certain positions between overview project and destination project. They are preset all on, so all positions are always kept equal. **Destination to overview play cursor + ranges** means, for example, that a change of the play cursor position and range selection in the Destination project is transferred to the Overview project. Similarly, the option works in the other direction and also the two other options for Source and Destination In markers. If necessary, you can deactivate individual synchronizations here.

- **Overview project settings**:
  - **Show true timing in overview**: 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 play cursor. This enables the actual timing of the takes to be observed.

  - **Play source project instead of overview project**: By default, only the reference track of the respective take is played when the overview project is played back. With menu **Edit** > **Multi-synchronous cut** > **Play source** the source project can be played if necessary. If the option is active, the source project is

always played.

- **Command "Copy" copies source when called from overview**: This option (active by default) causes that when copying a range in the overview project, the corresponding range from the source project is copied to the clipboard instead. This way you can also use the overview project to move audio data by using Copy and Paste without using the Source/Destination cut commands. If you explicitly want to copy the reference objects from the overview project, deactivate the option.

- **Automatic cut adaptation while cutting from overview to destination**: For cuts from the overview project into the destination project, the default crossfades are used and automatically placed so that the cuts are as inaudible as possible. For this purpose, another musical content analysis is performed in the close-up area around the cut to ensure perfect transitions.

EFFECTS

Sequoia contains a variety of effects for a wide range of applications. With a few exceptions, all effects can be used in real time, i.e. they are calculated during playback and the effect settings can be changed during playback and also automated (↗338). During recording, the recorded material is written to the hard disk without effects, but they can be heard during recording through software FX monitoring.

> **ℹ** *For more information on software FX monitoring, see the Program settings section at* Monitoring settings *(↗692)*

Real-time effects can be applied to different positions in a project:

- **Object effects** with the plug-in slots in the Object editor (↗186) or via the menu **Effects**
- **Track effects** with the plug-in slots in Track Editor, Track Header and Mixer or via menu **Track** > **More** > **Track effects**
- **Master effects** with the plug-in slots in the section **Master plug-ins** in the mixer

All real-time effects can also be used as offline effects (↗227). Some other effects are also only available as offline effects. In this case, the effects are calculated directly into the audio file. After that, no CPU resources are used for the effect during playback, but the parameters of the effects cannot be changed afterwards.

The effects contained in Sequoia can be roughly divided into the following categories, to which the top three folders in the tree structure of the plug-in browser (↗216) correspond:

- **Internal effects**: These are the oldest effects in Sequoia, they are integrated into the main program.
- **MAGIX Plug-ins**: These effects are VST plug-in effects from MAGIX.
- **VST plugins**: All other third-party VST effects.

Effect Slots

At each point where effects can be used (object, track, master) there are effect slots for inserting the effects.
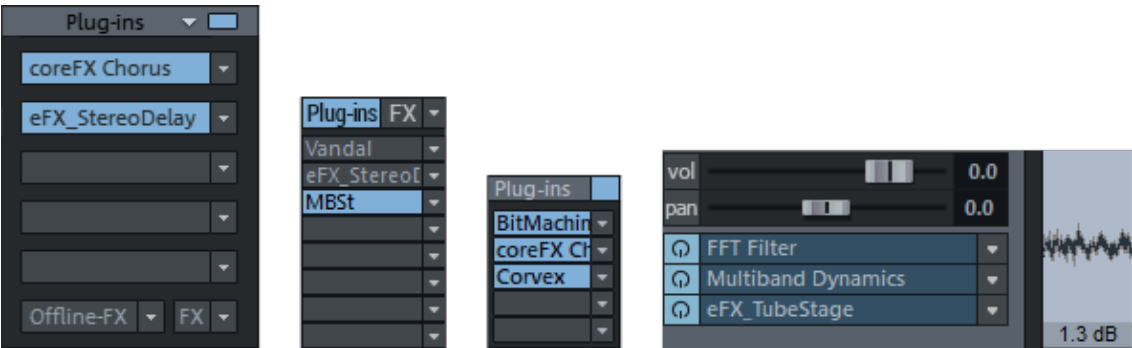
Effect slots in the object editor, track editor, mixer and track head

Clicking on the empty insert slot opens the plug-in browser (↗216), 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 plug-in's interface. In the menu at the insert slot ▼ you reach further functions, e.g. you can open the plug-in browser again to exchange or remove the plug-in.

The **Plug-ins** button at the top turns all effects in the respective effect chain on and off. A visual indicator (*) for plug-ins shows that they were previously active and will be reactivated the next time the **Plug-ins** button is clicked. The order of the plug-ins can be changed by dragging and dropping a filled slot to another position.

Plug-in Browser

The Plug-in Browser is used to load instruments and effects into the slots at objects, tracks, and the master. It offers centralized access to all available plug-ins and instruments.

The plug-in browser can be opened in two variants:

- By clicking on a **plug-in slot** in the object editor or in the track, you open it specifically for this slot as a "big" variant .

  🔆 *For this purpose, there is the program-wide keyboard shortcut **B**. This opens the plug-in browser for the track effects of the selected track.*

- Use the menu **View** > **Plug-in Browser** to open it as an independent, non-modal window that can remain permanently open floating or integrated into a dock at any location. It opens by default in the Manager/Docker at the bottom of the screen.

  ℹ️ *The plug-ins selected in this window will always be applied to the currently selected object or track when you add them by clicking on **Add** or by pressing **Enter**. The current selection is therefore displayed in the title bar of the window.*

  🔆 *You can also drag and drop effects from this window directly onto tracks and objects.*
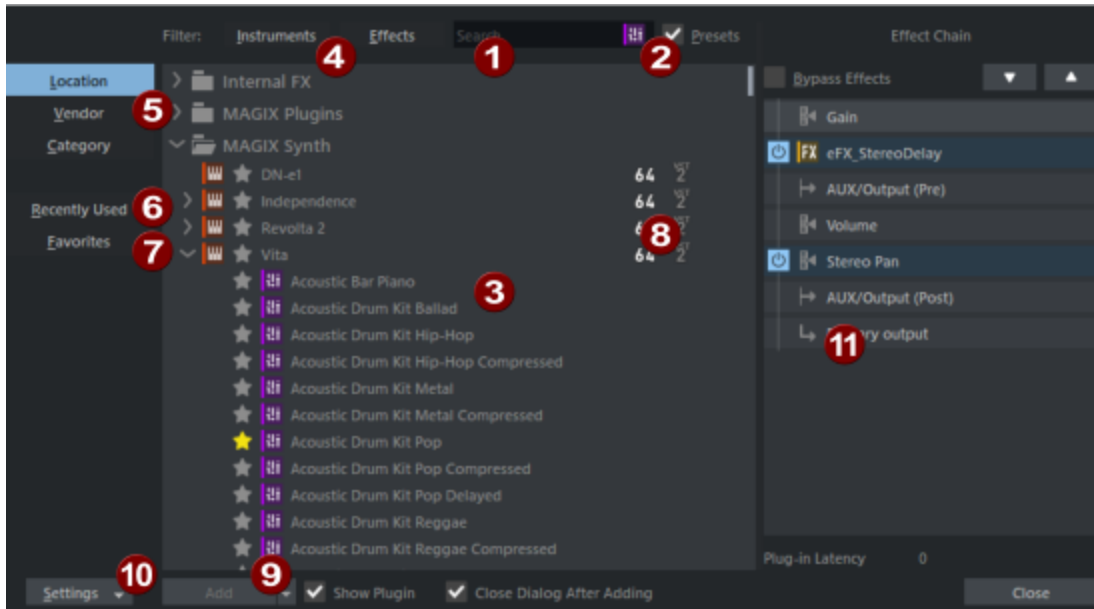
Plug-in browser at the plug-in slot
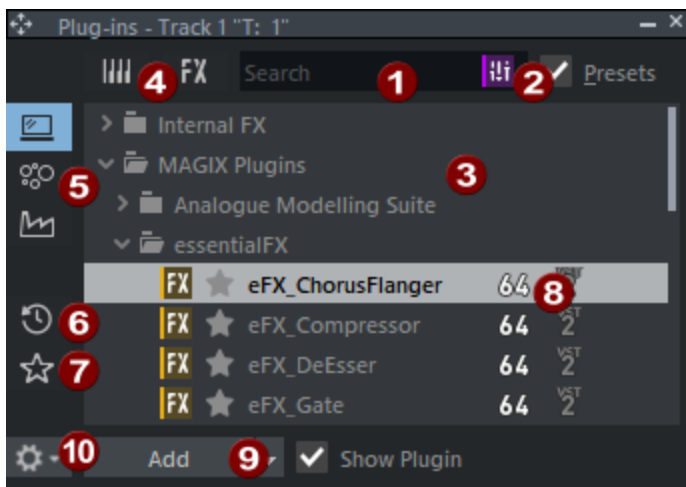
Plug-in browser in dock

1 **Full text search**: After opening the Plug-in 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. Then press **Enter** to load the selected plug-in into the slot.

2 **Presets**: If the **Presets** option is activated, all internal presets of a plug-in are listed in the plug-in list in a further level and can also be found with the full text search.

❸ Plug-in list: In the list, all plug-ins and optionally their presets are listed in a tree structure, depending on the setting at **(5)**. Use the arrow keys to move around the list to select an item. Use the **Enter key** to open and close a folder. **Ctrl + left arrow** collapses the entire tree, **Ctrl + right arrow** expands the entire tree.
If a plug-in or one of its presets is selected, you can add it by pressing **Enter** (see below). Alternatively, you can also move the mouse over a list entry and click on the plus symbol ➕ on the far right.

❹ Instruments/Effects: The display can switch between instruments and effects. Instruments are only used at track level and can therefore only appear there.

❺ Path/Manufacturer/Category: The tree structure can be sorted according to location on disk, plug-in manufacturer or plug-in category.

❻ Recently used: This view lists the most recently used plug-ins.

❼ Favorites: You can mark your favorite plug-ins and presets with a star. If Favorites is active, only your favorites will display in the list.
Double-click **New Folder** to create folders that can be renamed by double-clicking their names. 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.
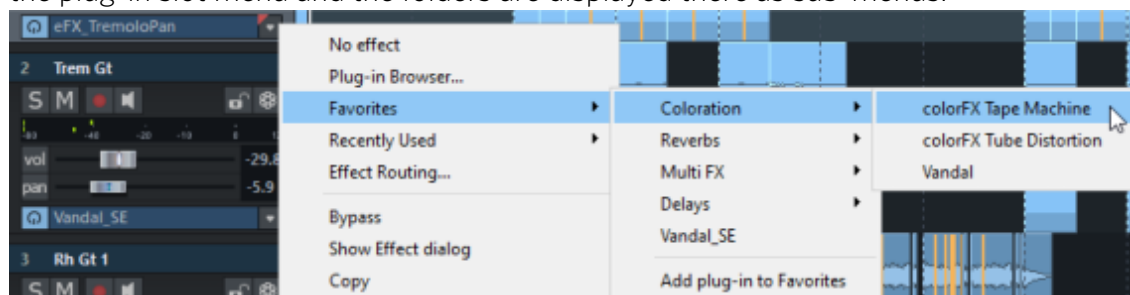
❽ Here you can see whether a plug-in is VST2 or VST3, and whether it's a 32-bit or 64-bit version.

❾ Add: Loads the selected effect or instrument into the effect chain, opens its interface and closes the plug-in browser. The opening of the effect dialog and the closing of the plug-in browser after adding the plug-in can be deactivated with the respective option.
In the menu at **Add** or by using additional buttons, you can also access two special forms of adding:

- **Alt + click on Add**: **Load as MIDI plug-in**. A VST instrument can be added as a MIDI plug-in MIDI Plug-ins (↗380).

- **Ctrl + click on Add**: **Use as VST 3 plug-in without ARA extension**. An ARA2 plug-inUsing Celemony Melodyne and Other ARA2 Plug-ins (↗222) is loaded without ARA functions, i.e. as a normal VST3 plug-in.

❿ Options: Here you can open a menu with settings. You can scan for new plug-ins or presets and open the VST settings (↗727).

11 **Effect routing**: The effect routing dialog (↗220) of the track, master or object is displayed here. In this you can change the effect order, display or delete effects.

> **ⓘ** *This section is not displayed in the dockable Plug-in Browser.*

Effect Dialog Windows

Most effects have their own settings dialog, which is opened by right-clicking on an effect slot. The following controls can be found on the upper edge of all internal effects and most MAGIX plug-ins:

> **ⓘ** *Note: Not all controls are available in all effect windows.*

❶ **Bypass**: This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

❷ **Automation mode**: Left-click to quickly switch the automation mode (↗342) between the last selected mode and Read. Right click opens the automation menu.

❸ **Presets**: Here you can load presets for the effect. Use the arrows next to the preset name to select the next/previous preset, the menu button ▾ opens the preset menu with the complete list of available presets.
Preset files from other folders can be accessed via **Load...** at the bottom of the menu. There you can also **delete** presets. With the **Save** button you can save the current settings as a preset. The self-created presets are also displayed in the selection menu.
All presets are stored in the folder C:\Program Data\MAGIX\Sequoia 17\fx-preset.
If you select the entry **Parameters when opening the dialog**, all changes made since opening the dialog will be undone.

❹ **A/B comparison**: This allows you to directly compare two alternative settings of a dialog. Use the arrow button in the center to copy a setting to the other memory location.

❺ **Reset**: This sets all the parameters of the effect to the default values.

❻ **Play** button: Starts and stops playback.

❼ **Help**: Opens the help for the effect. For some plug-in effects, the help button is located in the interface of the effect itself.

Below this bar, the actual effect window is displayed, where the settings for the effect are made.

When you close the dialog, the current settings are applied and, if necessary, transferred to the other objects or tracks of a multiple selection, see Applying an effect to multiple channels simultaneously (↗489) in the Mixer section.

To temporarily hide all open effect windows without closing them, choose **View** > **Windows** > **Hide/Show Effect Windows** from the menu. Select the command again to display them again.

Effect Chain Presets

In the **Track Editor** and mixer channel, right-click on the **FX** button or click the menu button next to it ▾ to access the track effects settings menu. There you will find menu commands to copy the entire "chain" of effects of a track together and paste them into other tracks. You can also reset all effects and save the effect settings as a preset to a file and load them from a file.

In the lower section of the menu there are, in submenus, the presets supplied and, after saving, your own presets. The track effect settings have the file extension .trk and are located in the default folder for the effect presets (`C:\Program Data\MAGIX\Sequoia 17\fx-preset`) in the subfolder `TrackFX`, for Master in the subfolder `MasterFX`. Subfolders in these folders correspond to the submenus in the menu **Track effect settings**. The subfolders can therefore be used to organize your own track effect settings in the menus. For multi-level menus, you can create folders inside the folders for this purpose.

Analogously, the storage option for effect chains is also available for object effects. The menu for the effect presets can be found in the object editor on the tab **FX** at the FX button below the effect slots. The presets are stored in the subfolder `ObjectFX`.

Effect Routing Dialog

The effect routing dialog lists all effects used in the respective context (track, object or master), as well as the possible positions for routing the signal to an AUX send/Submix bus, master or hardware output. This allows you to configure the order of effects and positioning of the track's outputs.

This dialog can be opened in the object editor and in a mixer channel/track with the **FX** button. Menu commands for this are also available in the menu **Object** > **Object Effects** > **Object Effect Routing...** and menu **Track** > **Track Effect Settings** > **Effect Routing...**
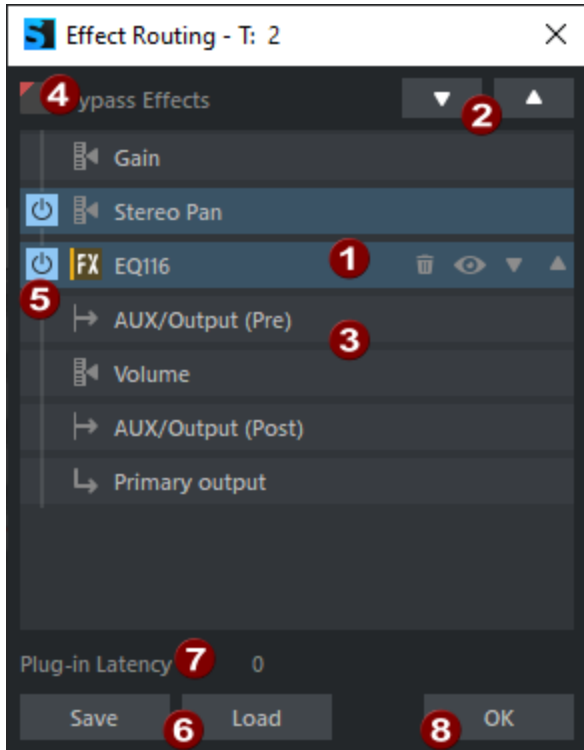
❶	When you move the mouse over an element, controls for that element become visible: 🗑 Deletes the entry from the effect chain. 👁 Displays the user interface of the effect.For AUX outputs, the AUX Send Routing dialog (↗498) is displayed, and for pan, the corresponding Panning dialog (↗492). ▼ ▲ The arrow keys move the element in the chain.
❷	Use the arrow keys to move selected effects in the chain.
❸	The entries **AUX (Direct/Pre/Post)** indicate the points in the effects chain where the signal for the AUX send busses is tapped. By default, **Pre** and **Post** are located directly before and after the volume, as with physical mixing consoles, but they can also be moved to a different position in the chain, just like the effects. You also define the positions for additional track output routings (↗487). Notes: ■ Exception: Object AUX sends always remain in the last position. ■ The AUX/Output (Direct) entry is not displayed until a corresponding routing has been created.
❹	**Bypass Effects**: The effects of the chain can be deactivated together here.

⑤ **On/Off**: This allows each effect to be turned off individually.

⑥ **Save/Load**: Use these buttons to save and load the settings of the complete effect chain (↗220). These settings can be accessed via the menu on the FX button at various points in the program.

⑦ **Plug-in latency**: Sequoia works with latency compensation for all plug-ins. The plug-ins report their latency to Sequoia, which converts this value into a time offset for the audio material. Here you can read the latency reported by the plug-in.

⑧ Click **OK** to close the FX routing dialog.

Using Celemony Melodyne and Other ARA2 Plug-ins

Melodyne is a software that lets you make detailed edits of pitch, timing and spectrum of audio material, as well as change the melody of individual instruments within a complete mixes. When used as a plug-in, the special program interface of Melodyne ARA2 is used (**ARA** = Audio Random Access). Unlike normal VST plugins, the plug-in has access not only to the currently processed audio data, but to all audio data that the object can access. Since Celemony has also made this technology available to other plug-in manufacturers, there are now a number of other ARA-compatible plug-ins. The following ARA2 plug-ins are supported by Sequoia:

- Celemony Melodyne
- Steinberg Spectralayers 11
- SyncroArts
    - VocAlign Pro 6
    - VocAlign Ultra,
    - VocAlign Project,
    - RePitch
    - revoiceProMonitor,
    - revoiceProLink,
- Acon Digital Acoustica
- Soundradix
    - Auto Align 2
    - Auto Align Post 2
- IRCAM

- ● ASAP Spectral Surface

- ● ASAP Pitches Brew

- ■ RipX Rip Link

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.

> ℹ️ *To avoid compatibility problems, we strongly recommend that you update Melodyne to the latest version.*

Edit Audio Objects with Melodyne

To edit one or multiple audio objects with Melodyne, select them, right click and select **Edit audio file in Melodyne** from the context menu.

This loads Melodyne for these objects as an object effect, analyzes the objects and opens the Melodyne interface. In this, the last object selected is displayed in the Melodyne-typical "blob" representation.
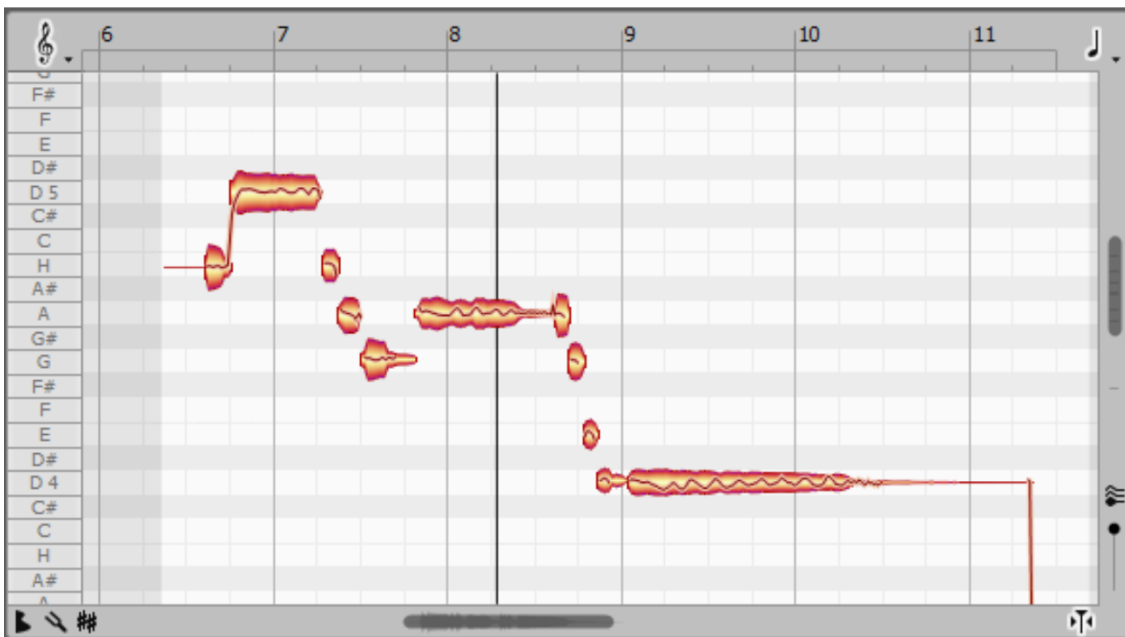
The blob view is a mix of piano roll and waveform display.

You can copy, move, lengthen, shorten and play back all notes here. The extent of the editing options available depends on the Melodyne edition installed, ranging from simple pitch and timing changes to detailed intonation control in polyphonic material and spectral editing, across multiple tracks.

> ℹ️ ***Tip***: *To help you exploit Melodyne to its full potential, we recommend watching the* video tutorials on the Celemony website.

If you select another object, the display in the Melodyne window will adjust accordingly.

Trumpets	🔗
Bass	

The "blobs" of the other objects can be displayed for reference purposes by activating this icon on the left of the Melodyne window.

If the objects originate from different tracks, corresponding tracks are also created in Melodyne (Melodyne *studio* only). The track names from **Sequoia** are adopted.

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.

You can use the **Audio to MIDI...** command from the object's context menu or the **Object** menu to create a new MIDI object from this data.
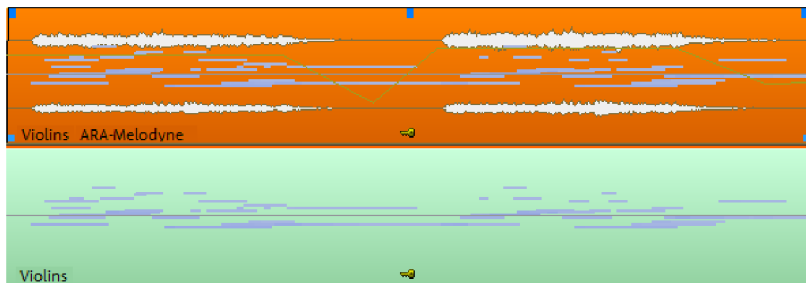
The MIDI object will be added to the track directly below the audio object. If there are already objects there, the MIDI object will be inserted into a newly created track.

Detecting Object Tempo

If Melodyne is installed, it can also be used to detect the tempo of objects. More about this in the section Object editor, Pitchshifting/Timestretching (↗191).
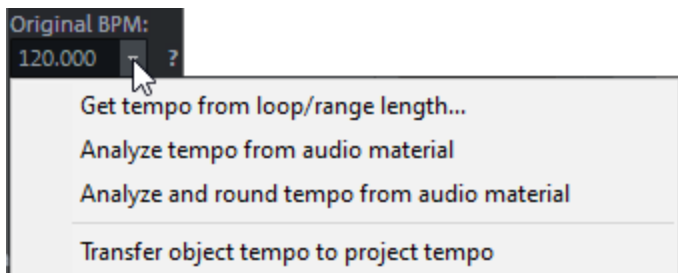
Notes

- The ARA integration requires the VST3 version of the Melodyne plug-in.

- Melodyne integration via ARA only works if you use Melodyne as an object effect. The fastest way to do this is to use the command from the context menu of objects. You can, however, load Melodyne via the Plug-in Browser into a slot in the Object Editor or via the menu **Effects**. When loading Melodyne (or another ARA plug-in) into the track, the plug-in is automatically inserted into every object on the track. This also works with objects added to the track later, for example if an object from another track is moved to the track.

- If Melodyne is loaded in an audio object, double-clicking on the object will open the Melodyne Editor, not the Object Editor. You can still access the object editor via the context menu, the **Object** menu or the keyboard shortcut **Ctrl + O**.

- Melodyne can't be used simultaneously with Sequoia tempo and pitch functions (Elastic Audio, Timestretching/Pitchshifting, musical tempo adjustment).

- When an audio object is split, Melodyne's edits are applied to both resulting objects, as they are when an object is duplicated using the commands **Duplicate objects multiple** or **Ctrl + Drag Mouse**. However, Melodyne edits are **not** transferred when you copy and paste an object onto the clipboard!

- The Object editor functions **Reverse** and **Loop** aren't possible.

External Hardware Effects Integration

In general, hardware effects could be integrated by routing the audio signal into the effects device via a dedicated output, and its output in turn via a dedicated input back into Sequoia. This could also be achieved with normal input and output routing. However, external effects can be integrated more conveniently with the special **External FX plug-in**. The advantages:

- You do not need to create additional tracks to use external effects. Additional routing is not required, you can use your external effects exactly as you are used to with plug-in effects.

- You can specify the latencies of the effects for consideration in latency compensation. These latencies can be measured in the plug-in.

- Presets can be saved and loaded in the plug-in. When you hard-assign your hardware effects to specific inputs and outputs on your audio hardware in the studio, you can use your hardware effects anywhere in the project with one click.

ⓘ *Note: The External FX plug-in only works if the* Hybrid Audio Engine *(↗692) is used.*

External FX Plug-in
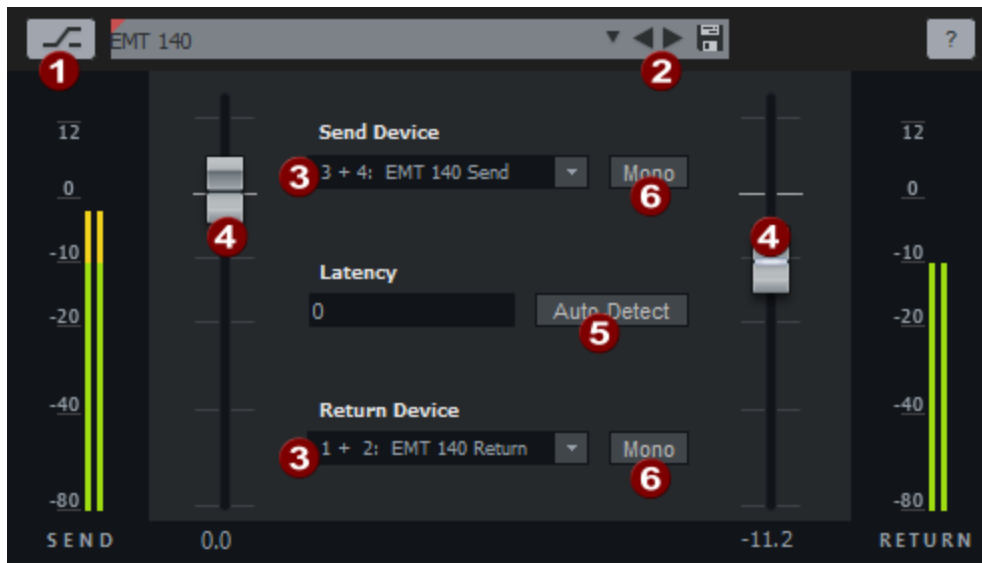
1 Bypass: Disables the effect.

2 Presets: Use the Save button to save the current settings of the plug-in to a preset. To load a preset, select it from the list box.
The current preset is displayed as the effect name in the plug-in slots.
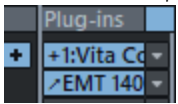
3 At **Send Device**, select the output to which your device is connected. At **Return Device**, select the input to which your device is connected.

4 Use **Send Gain** and **Return Gain** to trim the output and input levels.

5 With **Auto-Detect** you measure the total latency ("round-trip") of the effect, which consists of the latency of the inputs and outputs as well as the latency of the effect device itself. It is used for latency compensation during recording and playback, so that the effect is applied sample-accurately. When you click this button, a short pulse (ping) is sent to the effect device and the time delay with which it arrives at the effect input is measured.

6 **Mono**: This allows you to switch both the signal sent to the effect and the "return" signal to mono.

Export with External Effects

Since the effects processing does not take place within Sequoia, you must perform the final rendering in the export dialog using the **Realtime export** option. This plays back the project and writes the output to a file in real time.

Alternatively, you can play back the project and record the affected tracks into new tracks. Then you can still edit and re-export the project if the external hardware is not available (anymore):

1. If you have used the External FX plugin in multiple tracks or multiple external FX return tracks, first create a Submix bus and route all these tracks to the bus.
2. Create a new track and set as recording source this bus, if it concerns only one track, this track. To do this, right-click the Record button of the newly created track and select the bus from the menu at the very bottom at **tracks**.
3. Activate recording in the new track, click Record and record the complete track or the bus with the effect output.
4. You can now export the project using the Export dialog without having to activate **Realtime export**.

This approach also has the advantage that you can continue working on the project and no longer need to export the project in real time each time, as long as you don't make any changes that affect tracks that use external effects. Remember to then mute either the bus or track with the external effects, or the recorded track, as they will now both be played back in the project.

Applying Effects Offline

All real-time effects can also be used as **offline effects**. In this case, the effects are calculated directly into the audio file. After that, no CPU resources will be used for the effect during playback, but the parameters of the effects cannot be changed afterwards and the audio file will be permanently modified. We therefore sometimes speak of "destructive" effects.

To apply effects offline in the virtual project, select **Process effects offline** at the very bottom of the **Effects** menu. When editing wave projects (↗569) the option is always active, because there the effects, like all other edits, are always applied offline, i.e. destructively.

Certain effects marked with **(offline)** in the **Effects** menu are always calculated offline, regardless of this setting.

With the menu items **Edit left/right channel only** above it you can limit the editing to one channel of a stereo object or a stereo wave project.

Applying Offline Effects to Objects

To ensure the **Undo** function for effects applied to objects offline, the original, unprocessed audio material must be preserved. The effects are therefore calculated on a copy of the audio material and the object is adjusted to refer to the processed copy. You can decide on a case-by-case basis whether the Undo function is really needed, because to create the copy, the option **Create copy** must also be active in the effects dialog. You will save time and hard disk space if no copy of the audio file is created.

In the upper section of the dialog **Options for destructive effect calculation** (**Program settings** > **Effects**) you define how this copy should be created.

Destructive effect calculation in objects

File handling for effect calculation:
if "Create Copy" is active:

Write effect to a FX file (..._FX.wav)   ⌄

☐ Extended name for Effect Files
    (e.g. ..._FX_Dehisser_07-20-2003_11-30-00.wav)

○ Save Effect File in original file resolution
○ Save Effect File in 24 Bit resolution
◉ Save Effect File in highest resolution (32 Bit Float)

Destructive effect calculation in Wave Projects

Crossfade with original material at effect start
and effect end (length in ms):   `200`

Calculation includes samples before object
start and after object end: `0`

Preview time (seconds):   `4`

Maximum length of edit range for Spectral
Cleaning (seconds):   `30`
(memory critical)

- **Append effect to original file**: The audio data with the included effect will be appended to the original file. The data is saved in original resolution and no additional files are created. The disadvantage of this option is that the original audio file is extended, which can be a problem if it is to be used in other projects.

- **Write effect to an FX file (_FX.wav)**: All effects applied to a file are written to a separate file with the same name + suffix _FX so that the original audio file remains untouched. Additional effects are always added to the end of 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 can be given detailed names featuring effect descriptions and the date.

In addition, you can set here whether the effect file should be saved in the format of the output file, in 24-bit format or in 32-bit float format.

Destructive Effects Calculation in Wave Projects

When applying an effect "offline" in wave projects, the result of the effect calculation is always included in the audio file. The options of the offline effect calculation on VIP objects described above are not taken into account.

No permanent copies of the edited audio material are created, as it is desired that the original audio material is permanently changed when editing wave projects. Temporary files are created for the undo function during editing. To do this, the **Undo** option for audio files must be activated in **Program settings** under **Program** > **Undo**) and the **Create copy** option must be activated in the effect dialog for the respective effect processing.

If an effect is applied only to a range in the audio file, you can optionally add a crossfade between the effect and the original at the beginning and end of the range. At **Length of crossfade in milliseconds** you can specify its length.

Other options (virtual projects and wave projects)

Calculate additional samples before object start and after object end: Some effects require that they be calculated over a longer period of time than that ultimately used in the audio file or object. Here you can specify this additional buffer.

Preview time (seconds): Here you can set the amount of time that will be used for the preview function (**Play/Stop** button in the effect dialog) to preview the effects.

Maximum length of edit range for spectral cleaning (seconds): Enter how long (maximum) the audio material should be edited during spectral cleaning.

Help for the individual effects

In the following sections, the effects are described one after the other, sorted alphabetically. The effects not listed here are VST plug-ins, these have their own help file which you can open via the **?** button in the plug-ins' interface.

3D Reverb

3D Reverb is a unique and powerful algorithmic reverb effect. It creates the sound of a wide range of acoustic spaces, from small rooms to large concert halls, and 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, 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 reverberation that matches the

placement of the source signal to produce an impressive 3D sound. Late reflections are based on a feedback delay network with 16 delay lines (FDN16).

Decay times range from <100 ms to infinity and are independently adjustable for high and low frequencies. There is an attenuation filter for further processing of the room signal frequency spectrum. The early reflections can be diffused, the late reflections can be made more lively using modulation. On the output side, there is an equalizer and a Spatial Balance control for adjusting the spatial placement of the reverb signal.

Overview

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①   **Peak meter:** The peak meters for the input and output signal are located at the top left and right. See Peak Meter (⤢556)

②   **Visualization:** There are three different views for visualizing the generated reverb signal: impulse response, spectrum or waveform display. See Visualization (⤢232)

③   **Room:** This is where you'll find the controls for editing the early reflections, which include an equalizer for attenuating reflections and controls for room size and decay time. See Room (⤢233)

④   **Color:** Here you can access 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. See Color (⤢234)

⑤   **Output:** This is where you'll find the controls for processing the output signal: an equalizer, the Spatial Balance control for positioning in 3D space, and the controls for output level and mixing between dry and reverberated signals. See Output (⤢234)

⑥   **Presets and Settings:** In the Presets and settings area, you can load presets and also access additional settings and the bypass function. See Presets and Settings (⤢236)

Peak Meter

At the input and 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 0 dB) has occurred in a channel, the red clipping indicator lights up. The indicator can be reset with a single click.

Visualization

At the top center is a visualization area where the input and output signal can be displayed as a spectrum, waveform or impulse response of the reverb effect.
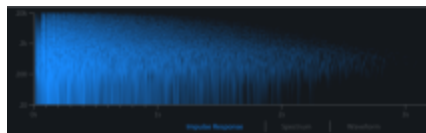
Impulse Response The graph shows the reflections generated by the reverb algorithm including all processing with reference to frequency and time.

> ℹ️ *If changes have been made to the parameters, it will take a moment for the display to update.*
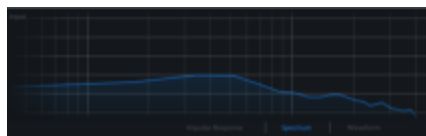
Spectrum: This displays the frequency distribution of the input signal (gray) and the output signal (blue) in real time.

Waveform: Waveform represents the envelope of the waveform of the input signal (gray) and the output signal (blue).

Room
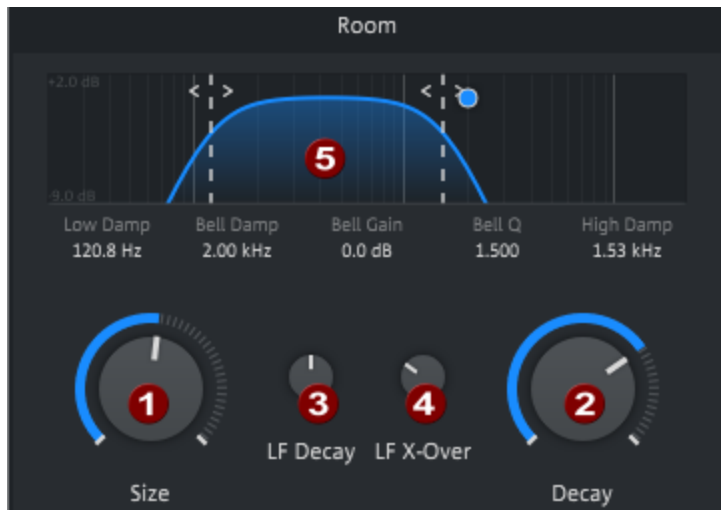
ℹ️ *The value can also be entered numerically by clicking on the value under a control. Use the Tab key to move to the next value.*

1 **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 by the listener.

> ℹ️ *In combination with the mixing ratio between early and late reflections, see the section on Color (↗234), which can be used 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 dashed 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
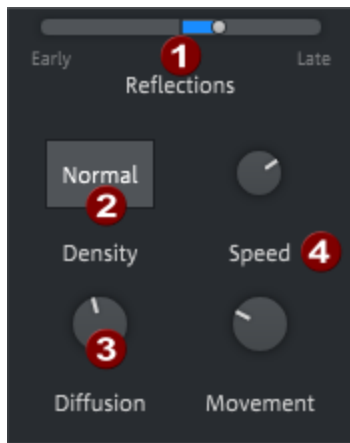
1 **Early/Late Reflections:** This control adjusts the mix ratio between early and late reflections.

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
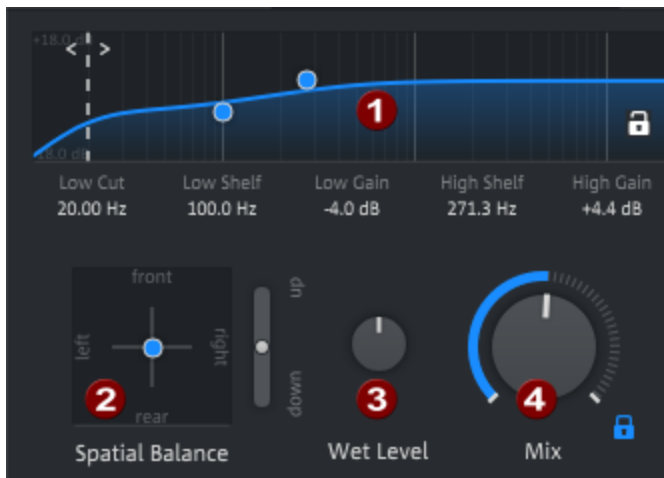
1 **Output EQ:** For the final sound shaping of the reverb signal there is an equalizer consisting of a low cut (high pass filter) for the complete removal of the low frequencies and a low and a high shelving filter for boosting and cutting the low and high frequencies. Drag the dashed line to adjust 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.

❹ **Mix:** Specifies the mix ratio of the original sound and the processed signal.
Use the lock symbol ⊟ to freeze the slider so that it won't change when you load a preset. This makes it easier to compare different presets within the context of a mix.

Presets and Settings

You can select presets, adjust view settings and temporarily disable the effect at the top of the plug-in window.

1	**Menu**: The menu contains various commands for managing user presets.	
2	**Previous/next preset**	
3	**Preset menu:** In the menu you can select the presets from different categories.	
4	**Settings:** Here, you can disable the tooltips. You can also switch labels for the controls. With **Value** the set value is always displayed, with **Dynamic** the value is only displayed when you move the mouse over it and otherwise its function. **Show tutorial** opens a tutorial that explains the function of the effects.	
5	**Bypass:** Use this button to temporarily disable the effect.	

User Presets

To save your own presets, select **Save as...** from the menu. You can set a category and a preset name. Once you have created your own presets, the Preset menu contains a second section entitled **USER**, where you can find your own presets, organized by category.

Changed Factory or User presets get an asterisk in front of the name.

With **Revert changes** you restore the settings of the preset, with **Save** you save the changes in the preset. Changes to Factory Presets cannot be saved, but you can save them as a User Preset using **Save as...**.

You can rename a user preset with **Rename...** and delete it with **Delete**.

You can also organize your user presets by using **Open user presets folder** to open the folder with the user presets. The categories in the Preset menu correspond to subfolders in this folder, the preset names to the file names of the presets. If you have moved or renamed files there, run **Reload user presets** so that these changes are applied to the preset menu.

Advanced Dynamics

All dynamic effects have a common working principle: the level of an audio signal is amplified, and the degree of amplification is simultaneously controlled by the level of this signal. When the level goes above or below a certain point (Threshold), the amplification changes to achieve a certain effect: For example, you get a compressor when the amplification is reduced above the threshold, or a noise gate when the amplification is reduced to 0 below the threshold, suppressing the signal.

The Advanced Dynamics effect processes the dynamics of a signal through a combination of several dynamics stages: a noise gate, two compressor/expander stages, and a limiter. The resulting characteristic curve of this dynamic effect can be edited graphically and parametrically.
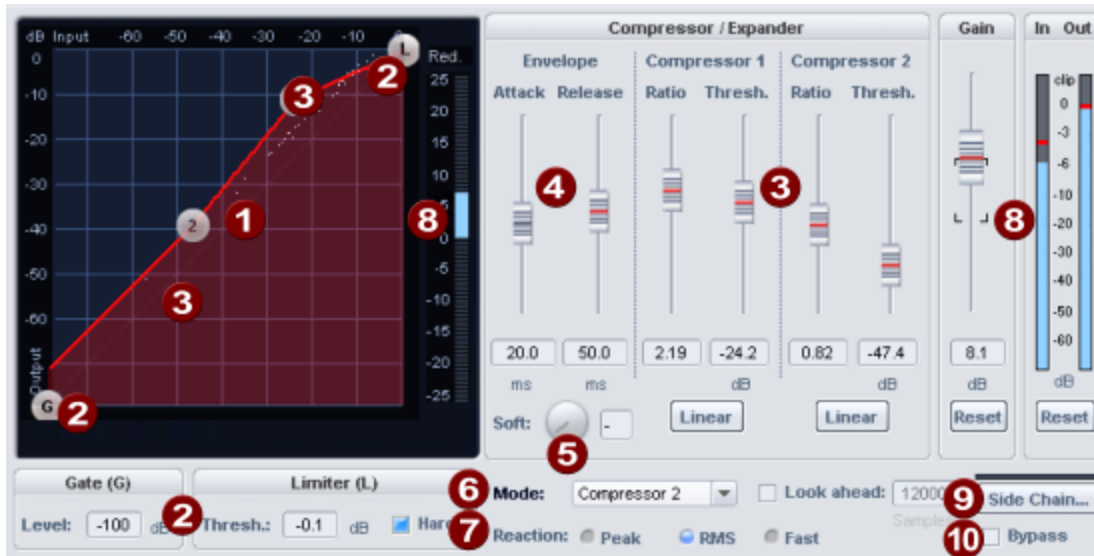
① Characteristic curve: The characteristic curve represents the transfer function from the input level to the output level. When you play back audio, you will see the current input and output levels of the signal as a white cloud of dots over the characteristic curve. With this **Dynamic Scope** the change of levels is constantly visualized by Advanced Dynamics.

You can edit the transfer characteristic graphically by moving the points on the curve or parametrically by changing the parameters of the characteristic sections with the sliders.

💡 *If you right-click on a point, you will get an input field in which you can directly enter the position of a point in the characteristic curve.*

❷ The control points **G** and **L** of the characteristic curve control the upper and lower limit of the characteristic curve , the **Gate** and the **Limiter**. Below the characteristic curve are the associated parameters.

- **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. The gate level can be set in the graphic by moving the point **G** horizontally.

Limiter (L):

- **Thresh.**: (Threshold) determines the maximum level at the output in dB.

- **Hard**: If **Hard** is active, the output signal is hard limited to the set threshold value. This means that no sample may exceed 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.

  > ⓘ *In the characteristic curve display, when hard limiting to levels below 0 dB, the area of the characteristic curve that is never reached as a result is marked with **Limited Zone**.*

  If **Hard** is not active, the algorithm controls according to the set transfer characteristic and and levels above 0dB can be present at the output. This allows you to use the limiter as an "analog" limiter for analog sound shaping.Then additionally use the dedicated "brickwall" limiter sMax11 (↗300), to prevent clipping.

❸ The points **1** and **2** control the two characteristic curve sections for the compressor/expander stages. The horizontal position determines the thresholds of the stages (**Thresholds**), the vertical positions result in different rises for the curve sections, which can also be expressed as **Ratio**:
With a ratio < 1 the stage works as an expander, with settings > 1 as a compressor.
If a stage works as a compressor, it means that the gain above the **threshold** is reduced: For example, with a set **ratio** of 2, this means that if the input level is increased by 3 dB, the output level will only be increased by 1.5 dB. High levels (above the threshold) are therefore amplified less than low levels, reducing the dynamic range and compressing the signal. The larger the ratio value, the stronger the compression.
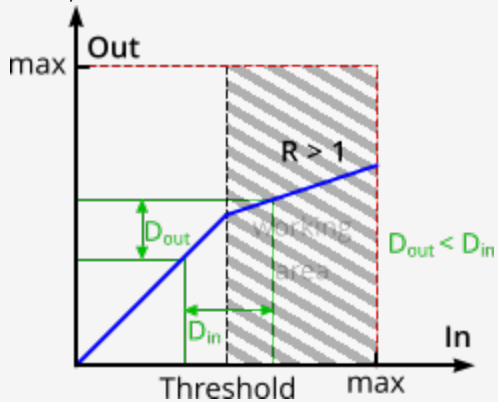
The dynamic range of an input signal (exemplarily marked by the green lines) is reduced by the compressor.

If a stage works as an expander, it means that the gain above the **threshold** is increased: For example, with a set **ratio** of 0.5, this means that increasing the input level by 3 dB will increase the output level by 6 dB. High levels (above the threshold) are thus amplified more than low levels, increasing the dynamic range and expanding the signal. The smaller the ratio value, the stronger the expansion.
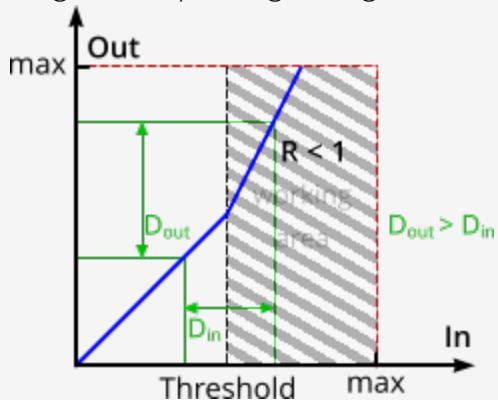
The dynamic range of an input signal (exemplarily marked by the green lines) is increased by the expander.

④ Envelope: The time constants regulate the sound characteristics significantly, so certain time constants can lead to distortion effects or to "pumping".

- **Attack**: Time span between crossing the threshold and the maximum impact of the effect in milliseconds.

- **Release**: Time span until the effect processing of the signal is completely withdrawn in milliseconds.

⑤ Soft: With this parameter you specify a rounding of the characteristic curve at the folding points. If the change between uncompressed and compressed signal is clearly apparent, i.e. the signal level fluctuates around the folding point, "Soft" achieves a smoother transition.

⑥ Mode: These are typical use cases for the **Advanced Dynamics**. They specify settings that make the graphical editing of the characteristic curve easier. The mode selection limits the number of applicable parameters, e.g. when used as a pure limiter.

⑦ Reaction:

- **Peak**: The peak levels of the signal are used for control, the effect reacts quickly to level peaks.

- **RMS (Root Mean Square)**: This mode uses the average loudness of the signal - this corresponds to the behavior of many analog dynamic effect devices. The time constant for the averaging is based on the length of the **Attack** parameter.

- **Fast**: Use this option if the dynamics are to be affected only slightly. The maximum level at the output never exceeds the threshold value of the limiter.

- **Preview:** When this option is active, the dynamics section works in preview mode. This means that not only the current input signal is analyzed, but also upcoming signals. The dynamic stage is not surprised by sudden level peaks, and attack phases that are too steep are smoothed out. To reproduce the sound behavior of analog dynamic effects, it is better to deactivate the preview. The length of the look-ahead may be entered in the **Samples** field.

⑧ **Level**:

- **Reduction display**: The average level reduction or level increase is displayed at the right edge of the characteristic curve.

- **In/Out**: The peak meters show the input and output level in dB. **Reset** on these displays resets the display of Reduction, In and Out.

- **Gain**: Use this fader to raise or lower the output level. In the characteristic curve, this is represented in such a way that the entire characteristic curve is shifted upwards or downwards.

⑨ **Sidechain**: If this option is activated, another signal from one or more other tracks is used for level control instead of the processed signal itself.

ⓘ *In Surround Master or in an object, Advanced Dynamics has no sidechain function.*

The **Sidechain...** button opens the Sidechain menu:

- **Sidechain In**: Enables and disables the sidechain function.

- **Side Chain Solo**: With this option you can listen to the sidechain signal. For this purpose, the compressor is switched to **Bypass** and only the sidechain signal is output. When closing the effect dialog, the **Side Chain Solo** function is automatically reset. **Side Chain Solo** is also ideal for acoustic control when applying the sidechain filter.

- **Side chain filter**: The sidechain signal can be filtered by a Parametric equalizer (⤴262).

The lower part of the menu lists all the tracks from which you can select one or more as sidechain sources. In the mixer channels used as sidechain signal, a separate sidechain bus is created that sends directly to the sidechain input. The signal tap of the sidechain send is preset to **Pre-Fader**. This causes the sidechain signal to remain independent of the volume setting of the sending channel.

⑩ **Bypass**: This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

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.

Section **REDUCTION**:

- **BITS**: This slider controls the resolution of the audio material. Turning the dial to the left results in 16-bit quantization, and thereby 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.
  In 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".

- **SAMPLERATE**: Das·Audiomaterial·wird·mit·diesem·Regler·"calculated down",·i.e.·the·internal·sampling rate·is·reduced. This creates a division ratio between the old and new rates; according to this ratio, samples are "dropped" at various points in the data stream.

ⓘ **Note**: *The two smaller controllers in this section are explained at* **MODULATION**.

Section **FILTER**:

The filter in the BitMachine is a digital model of one of the most well-known filters in music electronics. The filter in question is the 'Chamberlin 2-pole' filter, which was used in old Oberheim synthesizers. Filters of this type sound extremely musical. They can also be used in quite creative ways in the BitMachine and don't necessarily have to be used only to smoothen out artifacts.

The filter works in low pass mode – according to the settings, it lets through low frequency (or medium) material and dampens the highs and medium ranges.

- **FREQ**: With this control you determine the "cutoff frequency" of the filter. Filtering starts above this frequency.

- **RESO**: This allows the signal to be strongly boosted ("resonance") in a range around the cutoff frequency. This makes sharp, cutting sounds possible. 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. What happens is that the parameters of the internal workings of the filter interact with one another. So increasing **DRIVE** weakens the resonance, but at the same time, the signal receives more volume, more bass and becomes acoustically fuller.

> ⓘ **Note**: *The two smaller controllers in this section are explained at* **MODULATION**.

Section **MODULATION**:

By using the settings in the **MODULATION** section, you can automate your effects.

A low-frequency oscillator (LFO for short), whose tempo and waveform can be controlled, can be used to modulate the parameters in the sections **REDUCTION** and **FILTER**. The strength of the modulation is determined by the four small knobs under their parameters.

Note that the parameters must not be set to their maximum value, because then the modulation has no influence. The modulation is always added to the set value.

- As **waveforms** for the LFO sine, square or random values are possible.

- The LFO speed is specified with the **SPEED** dial. If **SYNC** is active, the LFO follows the song tempo and uses musical values (e.g. ¼ notes). Without SYNC you can set the tempo manually (in Hz).

- **Envelope-Follower**: With the lowest button at the LFO waveforms the volume of the signal is used for modulation, you can adjust the sensitivity with the **GAIN** knob.
  The **SPEED** knob is used to control the speed of the envelope follower (the display therefore changes to milliseconds). Short times result in a fast response to the input signal, longer times make the envelope rise and fall more slowly.

Brilliance Enhancer

With the Brilliance Enhancer you can compensate for sound losses in the high frequencies and give the audio material a silky sheen.

Such losses occur with MP3 compression or with old tape recordings. Unlike an equalizer, which only boosts existing frequencies, the Brilliance Enhancer calculates new harmonics from the existing signal and thus noticeably freshens up the sound. You can also use the Brilliance Enhancer well in pop or rock as a mastering effect if you want to increase transparency and brilliance of the sound.
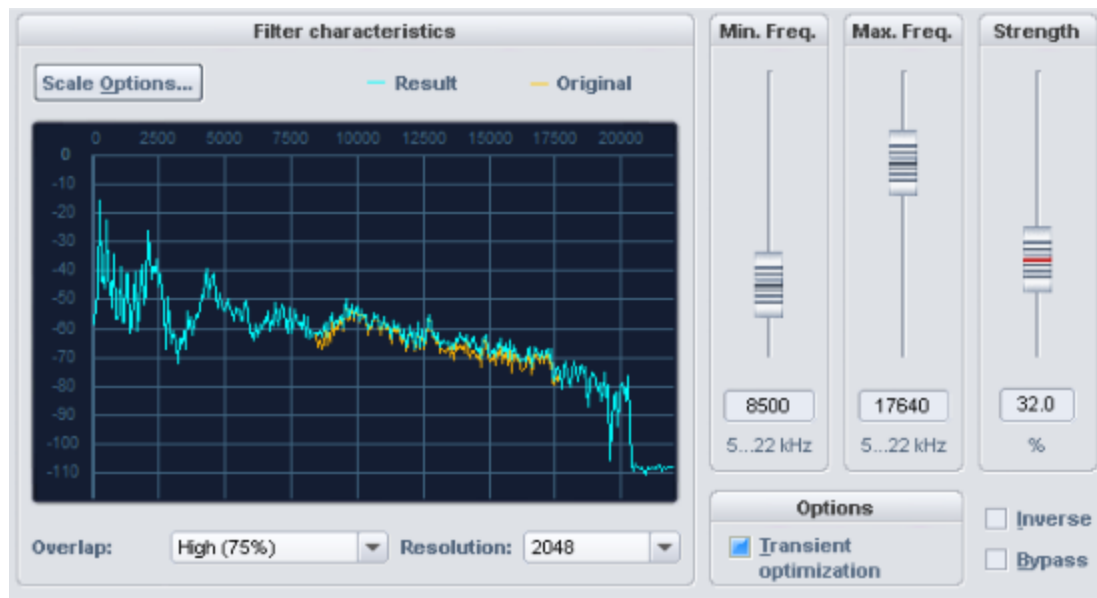
In the filter graph there are two curves, the orange curve for the original signal and the blue curve for the processed signal.

With the **scale options** you can adapt the signal display to your needs. The options correspond to those in the FFT filter (↗275)

Overlap: This parameter controls the overlap of the time windows for the calculations of the spectrum. Higher values improve the result, but also increase CPU load.

Resolution: With this parameter you can select the internal resolution of the algorithm. Higher resolutions do not automatically mean better results - the optimal setting depends on the audio signal.

Use **Minimum frequency** and **Maximum frequency** to set the lower limit and upper limit of the frequency range to be enriched with new harmonics. **Strength** determines the intensity with which the new harmonics are added.

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 suitable for tonal material, as odd harmonics can be generated.

- **Inverse**: If this switch is activated, you will only hear the newly generated harmonics that are added.

- **Bypass**: This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

Delay

> ℹ️ *This effect is deprecated. It is still included in* **Sequoia** *to ensure compatibility for old projects. For its purpose, however, there are now better and more flexible plugins such as the* **essentialFX Stereo Delay**, *Vintage FX Ecox or coreFX Delay .*
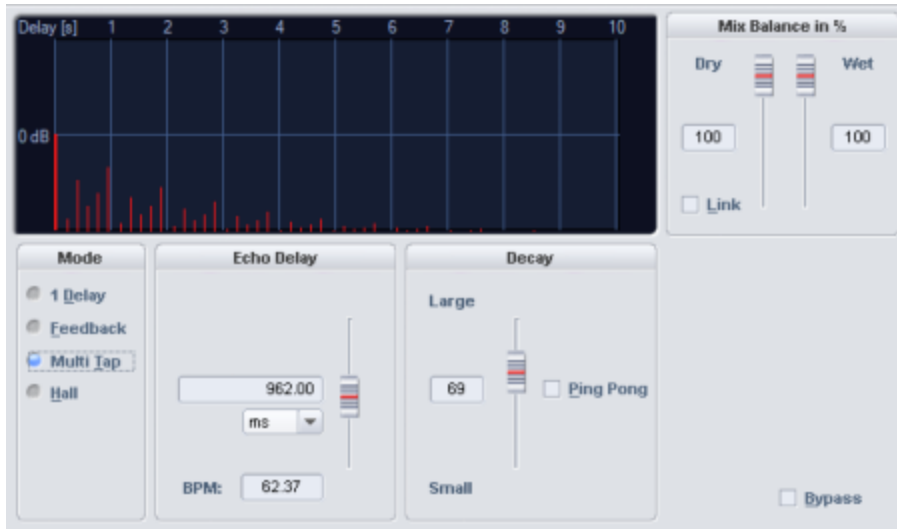
Mix Balance in %: Here you can enter the ratio of the original signal to the echo/reverb signal in percent. If you turn on the **Link** option, the values always add up to 100%.

Mode: With mode you can select the basic character of the effect.

Echo delay/reverb time: This controller sets the delay between the individual echoes or the original signal and the first echo in milliseconds. In the "Reverb" mode, the fader controls the reverb time.

BPM: Enter the current tempo in BPM, which automatically sets a delay value that corresponds to a quarter note in this tempo.

Decay / reverb coloration: Here you can set the decay value for simulating the size of the room and, in the case of reverb, the sound both numerically and using the fader. A ping-pong effect is also available for the delay modes.

Bypass: This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

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.

Settings and Controls

Parameters	Description
DeClicker Sensitivity	This parameter determines the DeClicker's sensitivity to noise.
DeClicker Mix	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 Mix	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, you will hear only the part of the signal that is removed by the algorithm. If the parameters are set optimally, the complete noise and just moderate components of the desired signal will be audible. If the parameters **Sensitivity** are set too high, parts of the useful signal are also filtered, which can then lead to tonal changes.
Bypass	This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

In the graphic, the processed audio material is continuously displayed as a **waveform** and as a **spectrogram** and you can see at which points the DeClicker (red) or the DeCrackler (blue) become active.
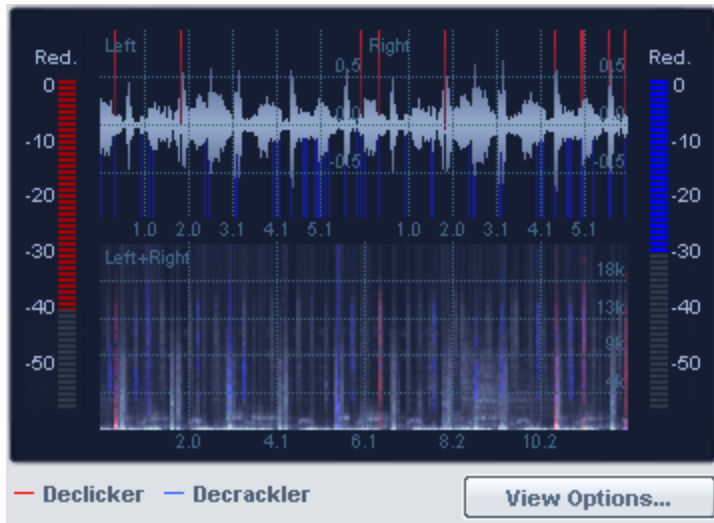
The left border of the graphic display features the reduction display for the DeClicker. It indicates by how many decibels a detected click has been reduced. The right border features the reduction display for the DeCrackler. This indicates how many decibels the crackling has been reduced.

View options: This opens a settings dialog to configure the signal display. Select the right and left channel or the mean value from both left and right (L+R) as a source for the waveform, spectrogram or the two reduction displays. The settings are applied by closing the dialog.

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 de-clipping algorithm is suitable for material featuring clearly audible overmodulation, e.g. distorted piano or vocals.
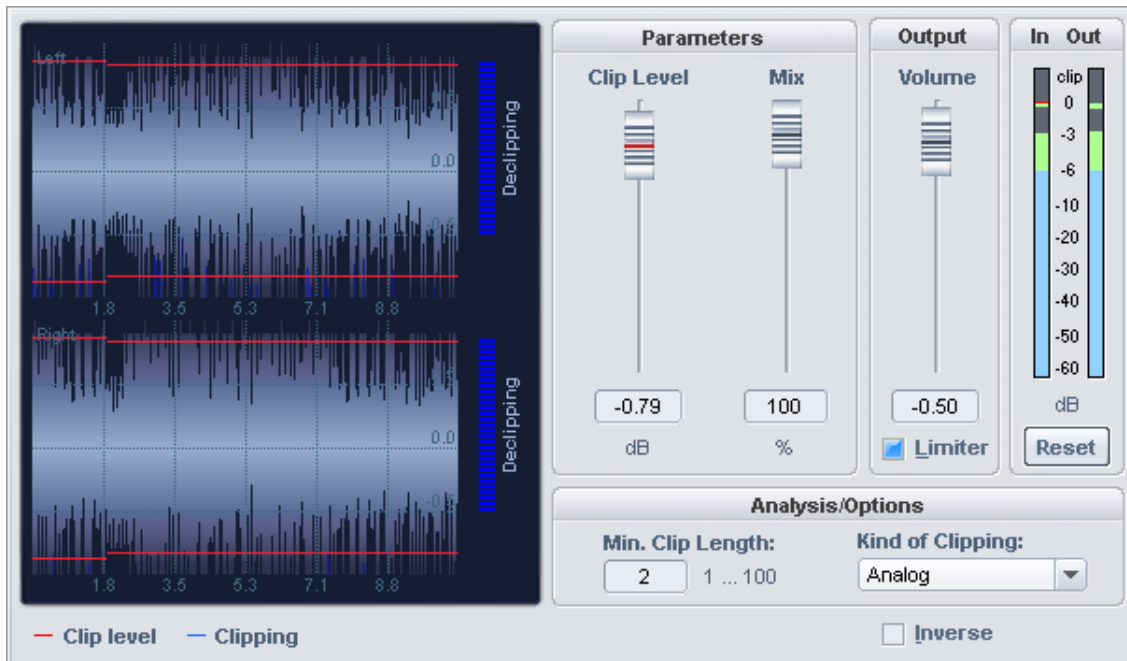
Signal display

The signal display shows you the edited material as a continuous waveform. The **Clip level** is shown as a red limiting line. You can see from the blue markings where the DeClipper has been applied. The meter to the right of the waveform display also shows the intervention of the effect.

Settings and Controls

Parameters	Feature
Clip level	Here you specify the level at which the algorithm considers samples to be overdriven and corrects them if necessary.
Mix	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. Level peaks higher than the maximum value can be caused by interpolation.
Reset	Set the peak meters to its original position here.

Analysis/Options

Parameters	Feature
Min. clip length	Set the minimum count of consecutive overmodulated samples for clipping detection.
Kind of clipping	Optimizes clipping detection by setting the applicable setting for the material – **Analog**, **Digital** or **Analog + Digital**.
Inverse	If is activated, only the part of the signal that is changed by the algorithm will be audible.
Bypass	This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

Possible 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, for example, -6 dB, all samples above half the control level are registered as overmodulated and recalculated. In this way, an analog damaged signal can be repaired.

DeHisser

The DeHisser is used to eliminate uniform, low-level white noise typically caused by microphone preamps or AD converters. Unlike for the DeNoiser, a Noise Sample is not required.
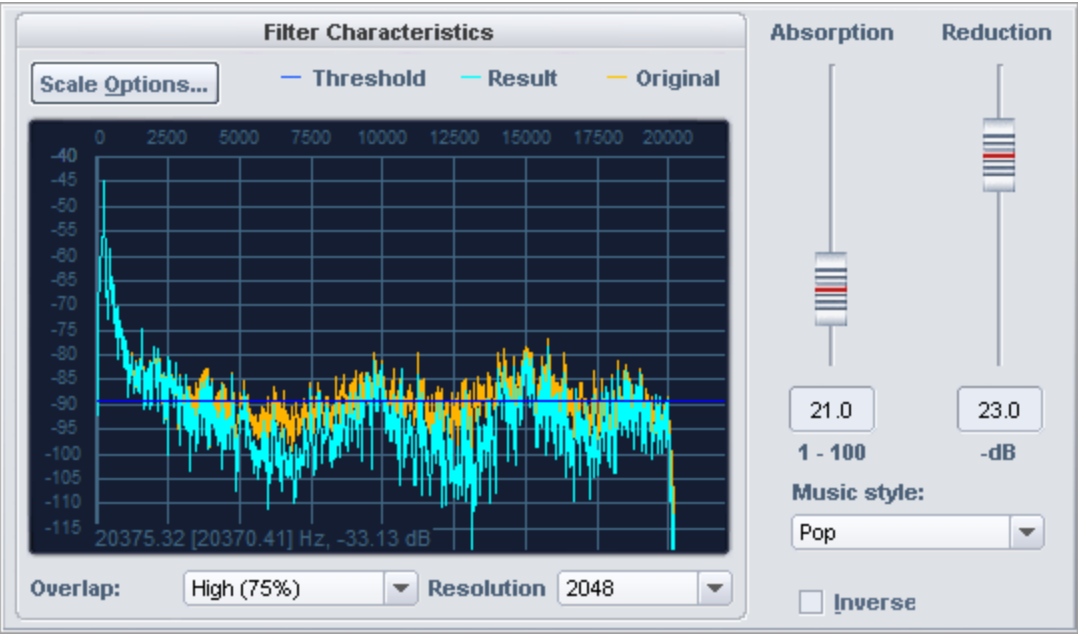
Graphic Display Description

The vertical axis to the left shows the level of the spectrum, while for the horizontal axis you may choose between displaying frequency or notes.

The yellow curve displays the original spectrum of the signal. Here, the spectral analysis is performed over the first second at the beginning of the selected audio section.

The light blue curve represents the spectrum after manipulation by the algorithm.

The dark blue line represents the threshold of the **Absorption** parameter.

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 for these frequencies.

Settings and Controls

Parameters	Description

|	
Absorption (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 a 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 audio, phase-like distortion or chirping modulation effects. The louder the noise, the more precisely your should try to set the DeHisser.
Reduction (0-30) dB	Set the noise reduction here. The highest possible reduction 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 a lower **Reduction** value (-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 signal 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. ⓘ *When **Inverse** is active, the **Reduction** parameter will have no influence.*
Scale options	The scale options correspond to those in the FFT filter.
Overlap	This parameter controls the overlap of the time windows for the calculations of the spectrum. Higher values improve the result, but also increase CPU load.
Resolution	This parameter allows you to select the internal resolution of the algorithm. Higher resolutions do not automatically mean better results - the optimal setting depends on the audio signal.

Operation

1. Find a critical place in the audio material to preview. Critical points are the quietest points, since the noise here has a relatively high level compared to the useful signal.
2. Set the **Reduction** parameter to the highest value (-30 dB).
3. Slowly increase the value of the **Absorption** setting. There are four stages in this process:
   a **Stage 1**: If the value is very low, no noise will be removed.
   b **Stage 2**: The noise is partially removed. Depending on the level of the noise signal, a small number of artifacts may be introduced.
   c **Stage 3**: The noise is completely removed.

d **Stage 4**: If the value is very high, not only will the noise be removed, but also a part of the signal. The audio material will lose its brilliance and sound duller. Click **Inverse**" to hear the removed parts of the audio signal.

e Optimal settings are usually available in **Stage 3**, where the noise signal will already be removed, but the audio signal is played back unaltered.

💡 *It often helps to listen to the critical parts with high volume and additionally boosted treble.*

4. If the noise cannot be removed without affecting the sound, now reduce the value for the parameter **Reduction** until you achieve the best possible compromise between reducing the noise on the one hand and still having an acceptable effect on the sound of the audio material on the other.

ⓘ *To remove humming noise, please use the DeNoiser.*

DeNoiser

With the DeNoiser function, you can effectively remove annoying background noise from audio material without its sound suffering to any significant extent. To do this, the algorithm requires a noise sample. The function is especially useful for eliminating consistent, long-lasting noise such as mains hum, fan noise, noise from inferior sound cards, tape noise or feedback.
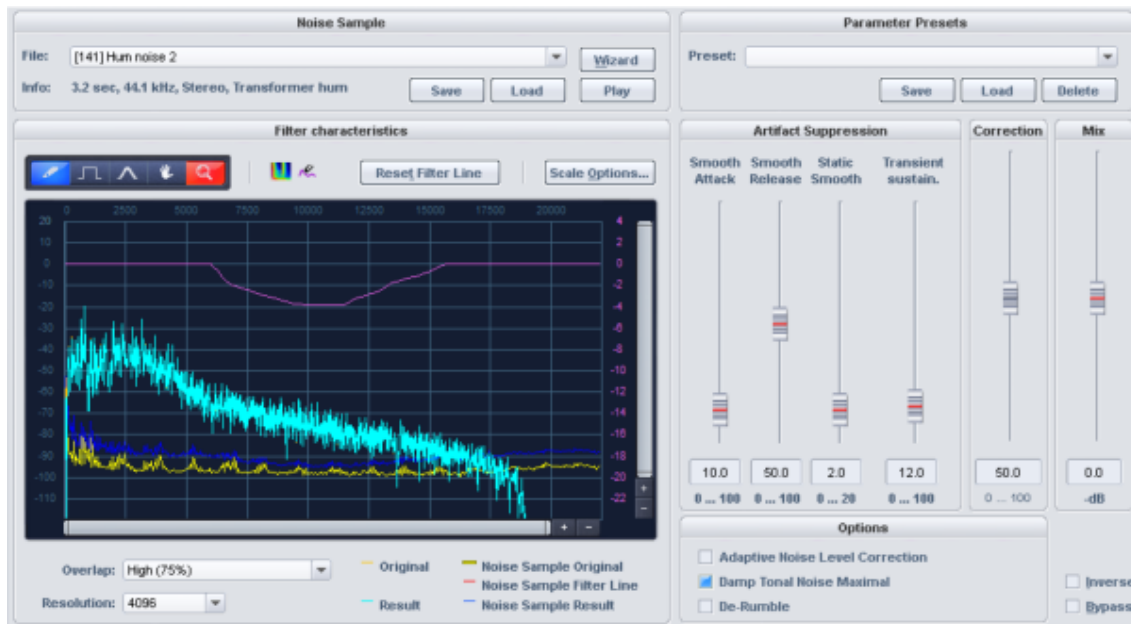
Noise sample

For noise removal, the DeNoiser needs a sample of the noise signal, the noise sample.

Before opening the Denoiser dialog for this, select an area in the object or wave project where only the noise can be heard and choose Menu **Effects** > **Restoration** > **Get Noise Sample**. Especially for hissing, as a rule, you will usually get better effects with longer noise samples.

At **File** the noise print just captured is displayed. In the selection list you will find more noise prints of typical noise such as camera noise, mains hum or tape hiss, which you can use if you cannot find a place in the audio material that contains only the noise. At **Info** you will find information about the length, type and format of the set noise sample. The **Save** button allows you to save the noise sample for later use. With **Load** you can import any wave file as a noise sample. With **Play** you can play the selected noise print.

Wizard opens the **Noise Print Wizard**, which can help you capture the noise sample.

The **Noise Print Assistant** offers two additional ways to create a noise sample in addition to the menu item **Get Noise Sample**.

Use **Set noise print length** to set the length of the noise print. With the option **Range length or internal default length**, if a range is selected, the range length is used. If no range is selected, a default length is used. With **User Length** you can explicitly set the length for the noise sample.

For **Create noise print from audio** there are two methods to create the noise print:

■ **Take noise print from range start or play cursor**: With the button **Pick!** the noise sample is picked from the beginning of an existing range selection, otherwise from the position of the play cursor.

■ **Take noise print from position with low level**: This scans the audio material for low level positions and extracts the noise sample at such a position. After setting the search options (see below), press the button

Start search. You can cancel the search by pressing "Esc".

After the search, you can switch between the found passages with the buttons **<<** and **>>** and listen to them with **Play**. With **OK** you accept the current position as Noise Print and close the wizard.

Search options

- **Start search in Object – Object start**: The search area starts at object start.

- **Start of the search in the object - Range start or play cursor**: The search starts at the start of the range if a range was previously defined. Otherwise the search starts at the play cursor.

- **Length of search in the object – To the end of the object**: The search for soft passages stretches until object end.

- **Search length in object – x minutes maximum**: Enter the search length in minutes here.

Filter graphic

The filter on the left can be used to additionally filter the noise sample to improve the results.

The graph shows the **original spectrum** of the noise as *yellow* curve and the corrected spectrum used by the algorithm to remove the noise as *blue* curve. If the effect is active (e.g. during playback or monitoring) you can also observe the input spectrum as *light blue* curve and the output spectrum (*orange* curve).

The *violet* curve can be drawn freely and is used for **filtering of the noise spectrum**. For filter parameters and drawing tools, please refer to FFT Filter (↗274).

Artifact suppression

The applied algorithm can leave a metallic chirping or twittering sound, the so-called artifact noise. The level of artifacts is lower than that of the original noise - usually in the order of -20 dB - but due to its synthetic character the ear reacts quite sensitively to it. The settings at **Artifact suppression** are used to suppress these artifacts. Please note that choosing high values can also worsen the quality of the playback result.

Smooth attack: This parameter controls the attack during noise reduction. At high delay, the artifacts are effectively suppressed. However, this can degrade the timing and impulsiveness of the audio material. 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. However, artifacts can again be amplified at high values. So find the most acceptable compromise between artifact suppression and impulse performance.

Static smooth: This smoothes the corrected spectrum of the noise that the noise removal algorithm uses. You can also see this in the graphical display: the blue curve is smoothed as the values increase. The artifacts are thus

reduced. For speech, singing or pop music, high values usually lead to good results. With orchestral instruments, on the other hand, high values for this parameter can lead to an unpleasant roughening of the sound.

Transient sustain: With this parameter you can influence the algorithm in such a way that the noise removal works more reservedly with transients. This leads to a significant improvement in noisy jazz or pop recordings, for example.

Correction

This parameter causes the noise sample spectrum to be raised or lowered (dark blue curve). Low settings may result in incomplete removal of the noise or increased occurrence of artifacts. High settings may cause the results to deteriorate. These phenomena depend mainly on the type of noise, and the nature of the original material also plays a role. Note that a higher level of noise does not automatically require a higher correction value.

Mix

Here you can set noise damping in decibels, between 0 dB and -40 dB.

In many cases it is most definitely advantageous to not completely remove noises: for example when working with gramophone recordings, it can be desirable to leave some of that "gramophone feeling" in. In reportage, a complete cancellation of background noise is also usually not desirable. In addition, the imperfect suppression of the noise also reduces artifacts that occur.

Options

Adaptive correction: This enables continuous automatic adjustment of the value for the parameter **Correction**. Using "Adaptive correction" makes most sense with distortions with variable noise levels.

Dampen tonal noise to maximum: If the material to be removed is tonal noise such as humming or camera noise, activate this switch. This means that the "Correction" parameter only affects the non-tonal signal components (noise). This can lead to better results, since attenuating tonal signal components produces fewer artifacts than attenuating non-tonal noise. The value for the **resolution** should be at least 4096.

De-Rumble: Frequencies below 40 Hz will be considerably dampened. This way impact noise like footsteps or rumbling on records can be removed.

Inverse: If this switch is activated, you will hear only the part of the signal that is removed by the algorithm. If the parameters are optimally tuned you will only hear all the noise and only a small portion of the used signal If the parameters are set insufficiently, larger components of the music or speech signal become audible.

Bypass: This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

Procedure for the adjustment

To find optimal settings for noise removal from your material, proceed as follows:

1. Search for a good **Correction** setting. The noise should just not be heard anymore. If artifacts occur, they should not be suppressed by a very high value for "Correction". The result could otherwise sound dull.

2. To suppress artifacts, increase the values for **Smooth Attack** and/or **Static Smooth**. Which parameter is the more appropriate depends on the audio material.

3. To improve the result, you can now slightly reduce **Correction**. In the process, the artifacts will reinforce themselves again. Now raise the **Smooth attack** and/or **Static smooth**parameters .

4. A high value for **Overlap** will further increase quality.

5. Increase the parameter **Transient sustain** until an increase in the disturbance is perceptible at the transients (with noise, this effect manifests itself as a kind of noise modulation). Now turn the fader back a little.

6. For difficult cases, we recommend the freehand draw **filter curve** for the noise sample. Here, problematic areas can be influenced by frequency-selective boosting or cutting of the noise spectrum.

7. If a convincing result cannot be achieved in very difficult cases, you can at least attenuate the noise by reducing the **Mix** value.

Distortion

ⓘ *This effect is obsolete. It is no longer offered in the plugin browser, since better and more flexible plugins for its purpose are now available, such as Vandal or colorFX Tube Distorsion. However, it is still available for old projects to ensure compatibility.*

Use this dialog to distort audio material using a nonlinear transfer characteristic. The signal becomes louder and overtones are added.
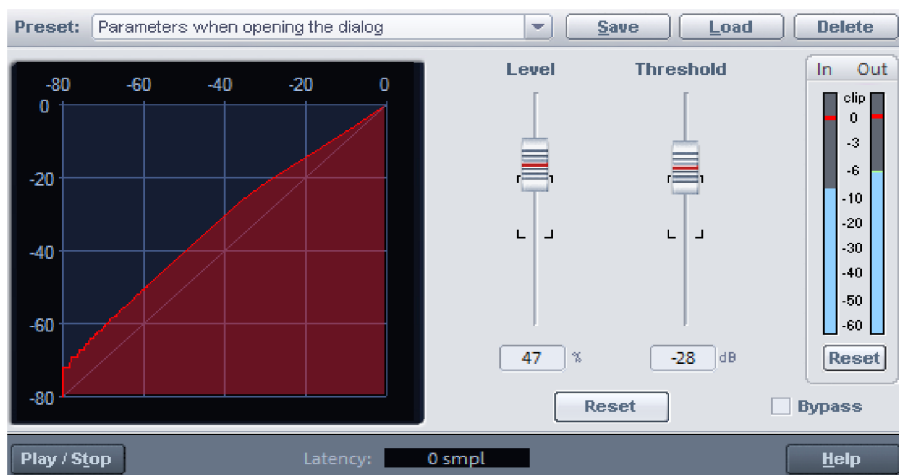
By adjusting the starting point of the distortion (**Threshold**), you can achieve a soft, analog-sounding distortion (overdrive), or a hard, digital-sounding distortion (threshold at 0 dB). The **Level** setting determines the strength of the distortion.

Dynamic Equalizer

With the Dynamic Equalizer, the frequency spectrum of a signal can be processed depending on the level. This means that the effect of a single filter band can be controlled depending on the strength of the occurrence of certain frequencies in the input signal. In other words, you remove an "interfering" frequency only where it occurs more strongly, but leave it unprocessed in other places.

Unlike the Multiband Dynamics (↗283) effect, where the signal is split into different bands by filters and then dynamics processing is applied to these bands separately, the Dynamic Equalizer processes the signal as a whole and the dynamics of each band are affected only by the frequency components that occur in that band. As a result, there are fewer phase changes between the frequency components in the signal, which means that fewer unwanted artifacts can occur.

❶	**Filter curves and filter symbols**: Each filter band is symbolized by a colored circle. The filter curve of the individual band is represented by the transparent colored area below the circle. The thick white line shows the resulting filter curve of all bands. In the background, the light gray line shows the resulting frequency spectrum of the signal in real time, and, somewhat paler, the original spectrum of the signal.
❷	**Band selection**: The Plus button can be used to add filter bands. With a single click on the colored area a band can be selected for editing, a double click on the colored area deletes the band. ⏻ The On/Off button can be used to deactivate the bands individually.

❸ **Filter Type**: Here you set the filter type of the selected band.

[icon] High Pass [icon] Low Shelving [icon] Band Pass

[icon] Peaking [icon] High Shelving [icon] Low Pass

The small number next to the symbol indicates the slope (slew rate) of the filter. By clicking on the 3 dots on the icon, you can change the slope for high, low and bandpass in the range between 1 (6dB/octave) and 4 (36dB per octave). The other filters are fixed at 2 (12dB/octave).

❹ **Filter parameters**: In this area the characteristic values of the filter band are set: Frequency for the cut-off frequency, Gain for the increase or decrease of the filter, Q for the bandwidth of the filter.
You can set these values graphically in the filter graph by moving the circle in the filter graph accordingly: Frequency and gain result from the position of the circle in the filter graph. The Q value of the filter (in the case of the peaking filter) is set with the mouse wheel.

❺ **Channel processing**: In this menu you determine whether the filter band affects both stereo channels, one channel at a time, or the mid or side signal. If one of the options Left/Right/Center/Side) is selected, the filter curve of the band is color-coded accordingly:

- Left: Green

- Right: Red

- Mid: Yellow

- Side: Blue

  ⓘ *This function is not available for multichannel files.*

❻ **Delete filter band**

❼ **Band Dynamics**: Click the On/Off icon to activate Band Dynamics for the selected band. The gain of a band is then affected by the corresponding frequency component of the input signal. The parameters correspond to those in a compressor or expander:

- **Attack/Release** determine as time constants how fast the dynamics react to level changes.

- **Threshold** sets the threshold at which the band dynamics affect the gain.

- **Ratio** determines the direction and strength of the gain correction: values <1 result in an expansion, values > 1 in a compression.

⑧ **Global settings** These settings affect all bands.

- ■ **Global Scale**: This control allows you to boost or attenuate the **Gain** values of all filter bands together. Negative values are also possible, reversing the effect of the EQ.

- ■ **Mono Below**: This is a global high pass for the side signal. Below the cut-off frequency, the side signal is attenuated. In the filter graphic, this additional filter is displayed with a green line.

- ■ **Stereo Width**: Controls the amount of side signal in all bands and thus the stereo width of the overall signal.

- ■ **Output Gain**: This allows you to compensate for level differences between the input and output signals.

⑨ **Scale**: With the +/- zoom buttons you can change the value range of the scale.

⑩ **Peak meter**

⑪ **Presets and Settings**

A ◄• B ⚙ ≡ ‹ ›   * DYN_Vocal_Lead   ˅ ⏻
   ❶   **❷** **❸** **❹**   **❺** **❻**
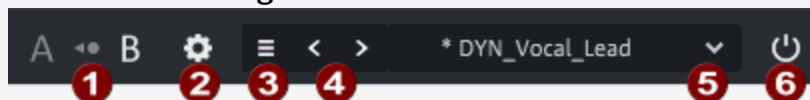
❶ **A/B comparison**: These are two temporary memory locations for A/B comparison of two settings. You can copy each selected setting to the other location using the arrow symbol. This way you can experiment with successful settings without losing them.

❷ **Settings**: The gear wheel opens the settings dialog (see below).

❸ **Menu**: The menu contains various commands for managing user presets (see below).

❹ **Previous/next preset**

❺ **Preset menu**: In the menu you can select the presets from different categories.

❻ **Bypass**: Use this button to temporarily disable the effect.

User Presets

To save your own presets, select **Save as...** from the menu. You can set a category and a preset name. Once you have created your own presets, the Preset menu contains a second section **USER**, where you can find your own presets, organized by category.

Changed factory or user presets get an asterisk in front of the name.

With **Revert changes** you restore the settings of the preset, with **Save** you save the changes in the preset. Changes to Factory Presets cannot be saved, but you can save them as a User Preset using **Save as...**.

You can rename a user preset with **Rename...** and delete it with **Delete**.

You can also organize your user presets by using **Open user presets folder** to open the folder with the user presets. The categories in the Preset menu correspond to subfolders in this folder, the preset names to the file names of the presets. If you have moved or renamed files there, run **Reload user presets** so that these changes are applied to the preset menu.

Settings

Show tooltips: Here you can disable the tooltips.

Disable dynamic labels: This allows you to switch the labels of the control elements: By default, a control will display the set value only when you hover the mouse over it, and its function otherwise. If the check mark is active, the set value is always displayed.

Disable FFT analyzer/ Disable peak meters/Enable OpenGL rendering/Frame Rate limiting: With these options you can accelerate the performance of graphical display in case of overload.

Oversampling: For a balance between available processing power and required quality, you can set the oversampling between off and 8x.

Zoom level: This allows you to display the plug-in interface in different sizes.

EQ116
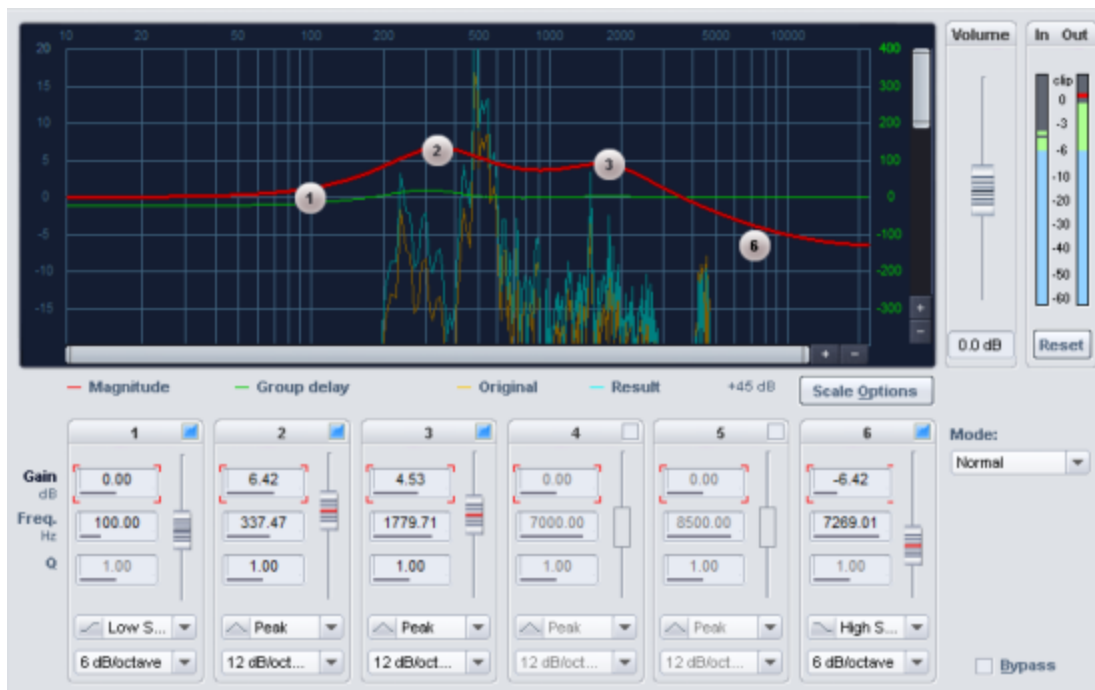
The EQ116 is a 6-channel fully parametric equalizer for influencing the frequency response of audio signals.

Graphic displays

The upper part of the dialog contains the **filter graphic**. This graphic provides you with several frequency-related information:

- **Magnitude**: The amplitude increase or decrease resulting from the individual band settings.
- **Group delay/Phase**: (switchable in the scale options). The group delay describes the frequency-dependent time delay when the signal passes through, while the phase shows the dependence of the phase from the frequency.
- **Original**: This curve shows the original frequency response in real time.
- **Result**: This curve shows the frequency response after processing by the effect.

You customize the display of the filter graph in the **Scale Options dialog** and by using the horizontal and vertical scroll bars.

- **Frequency axes**: The frequency can be plotted linearly or logarithmically. If you select the logarithmic display, either the frequencies or the musical pitch (notes) can be used for the frequency on the x-axis.
- **Value display**: At **Value display** you define the minimum and maximum value of the level values.
- **Curve display**: The curves can be displayed roughly as bars or as interpolated curves.
- **Hide real-time curves**: With this option you can hide the real-time curves.
- **Real-time offset**: Offset for displaying the real-time curves so that they do not obscure the amplitude response.

Volume: Use this control to adjust the level to compensate for level changes caused by the filter.

In / Out indicators: The peak meters show the input and output level in dB.

Reset (meter): Reset the in and out displays.

Editing the Characteristic Curve

Filter graphic: The parameters **Gain** and **Frequency** of a band can be changed in the graphic by moving the corresponding circle. The bandwidth (Q) can be adjusted with the mouse wheel.
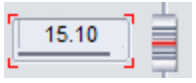
 In the input field you can change the values of the parameters by dragging the mouse vertically or enter numerical values.

A selected parameter can also be adjusted using the slider next to the parameters. You always select the parameter for all bands together.

Over the input field and the slider you can also use the mouse wheel for value changes. While doing so, hold the **Shift** key for finer changes.
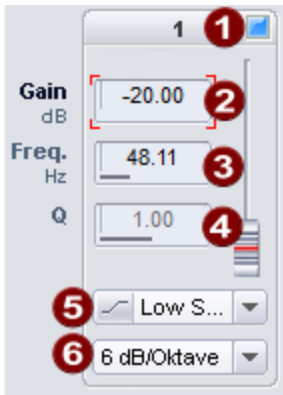
❶ Band On/Off: Switching the band on and off. In the filter graphic, the corresponding circle also disappears.

❷ Gain: These controllers allow you to raise or lower the filter. With a gain of 0dB the band is also deactivated (neutral setting).

❸ Freq. (Frequency): Here you set the cutoff frequency of the individual filters between 10 Hz and 24 kHz.

❹ Q (Bandwidth): Here you set the bandwidth of the individual filters between 0.10 and 10.

❺ Type: Here you set the filter type between **Peak**, **Low Shelving**, **High Shelving**, **High Pass** and **Low Pass**.

❻ Slew Rate (slope): For **High Pass** and **Low Pass** , choose between a slope of 6dB/octave, 12dB/octave, 24dB/octave or 36dB/octave. For **Low shelving** and **High shelving**, select between 6 dB/octave and 12 dB/octave. The filter type **Peak** is fixed at 12dB/octave.

Mode

With **Mode** you set the working mode of the **EQ116**.

- **Normal**: Minimum phase EQ, corresponds to the operation of the 4-band parametric equalizer of older Sequoia versions (compatibility mode).

- **Oversampling**: The effect internally works with a higher sample rate compared to the first mode. 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**: The equalizer works without frequency-dependent phase shifts. A fundamentally different algorithm is used than in the first two modes. The Linear phase mode sounds more neutral and less colorful,

and the amplitude of the overall signal is also less affected, since there are no amplification and cancellation effects due to the phase shift. However, the EQ requires more computing power in this mode. In addition, artifacts ("pre-ringing") can occur at higher frequencies when processing impulsive audio material.

ℹ️ *In Linear phase mode, the parameters of the EQ116 cannot be automated.*

Elastic Audio

Elastic Audio is a specialized editor that lets you selectively change the pitch of audio material. Automated resampling or pitch shifting as well as pitch detection for monophonic signals are used.

To open the Elastic Audio effect, use:

- Keyboard shortcut: **Ctrl + Shift + E**

- Menu **Effects** > **Time/Pitch** > **Elastic Audio**

- The button in the Time/Pitch tab of the object editor (↗191).

Editing modes

The Elastic Audio Editor can be used in **Relative mode** or in **Direct mode**.

💡 *If you want Elastic Audio to always open in **Direct** mode, select **Startup Mode Direct** in the Elastic Audio menu **View**.*

Relative Mode

Here you edit the pitch curve (automation of the change in pitch over time) as a relative pitch deviation. This is similar to the Pitch Bend controller progression for MIDI data.
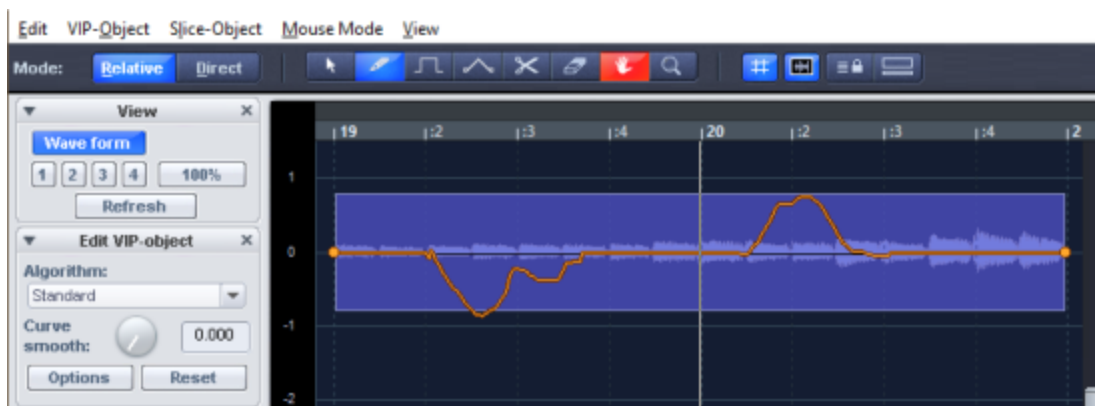
Editing can be done with a free-hand curve, quantized "step" curves, or using the Curve Bend tool. The "curve smooth" parameter enables smoothing of the automation curve. This avoids too steep changes in the pitch curve during playback.

Overview of the graphic display in the "Relative" mode:

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

Orange curve: Pitch automation curve as relative detuning of the original pitch.

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

Direct Mode

In Direct mode, the pitch is drawn directly, changes are absolute. To be able to adjust the pitch accordingly, the original pitch must be determined by a preceding pitch analysis of the audio material. This only produces meaningful results for tonal, monophonic material such as solo vocals, solo instruments, and speech.
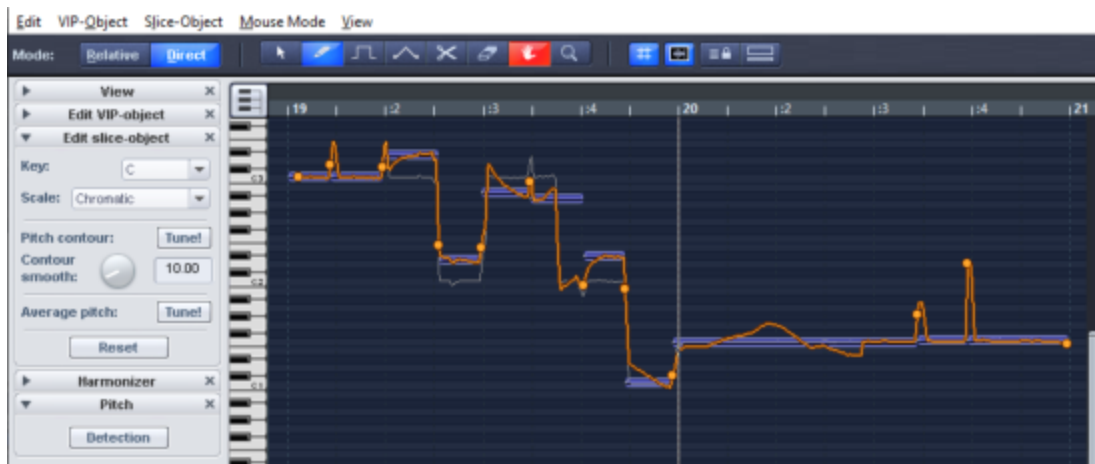
Start the analysis function with the pitch **Detection** button. After analysis the VIP objects are divided into individual slice objects according to the recognized pitches. The average pitch of a slice object determines its position in the graphical representation, independent of the concrete course of the pitch curve inside the slice object.

First, all slices are selected. Click in an empty area with the Selection tool to deselect all slices. Now you can select single slices by clicking on them, select multiple slices by additionally holding **Ctrl-** or **Shift-key**.

Two handles are created on the pitch curve at the boundaries 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 functions for automatic pitch correction (**Tune!**).

Overview of the graphic display in the "Direct" mode:
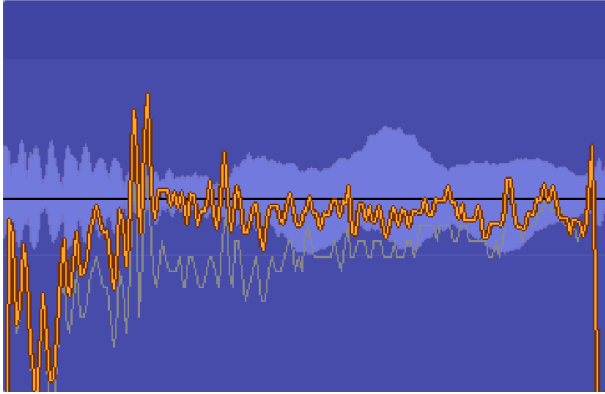
Orange curve: display of the editable pitch curve

Gray line: Original pitch curve

Y-axis: In the keyboard display on the left edge of the window you can see the pitches as notes. In the keyboard view you can deactivate individual pitches by clicking on them so that they aren't used in the automatic pitch correction or in quantized drawing. The corresponding keys will then be grayed out.

Selecting **Key** and **Scale** under **Edit slice object** disables all pitches on the keyboard that are not included in the corresponding scale.

Flexible Controls

On the right side of the dialog are a number of flexible controls that can be opened and closed individually.

> 💡 *In the lowest section of the **View** menu in the editor, you can define which control elements should be visible in the respective mode (Relative/Direct).*

View

Waveform: This turns the waveform display on and off.

1...4: Save the zoom depth and position of the current window view here. For this purpose, 4 memory locations are available via the buttons. Click a button to save the zoom depth and position. Click a button again to restore the saved zoom level. To save a new zoom level to a "occupied" button, click it with the **Shift** key held down.

Clicking on the button while holding down the **Ctrl** key restores only the horizontal position of the saved zoom level.

Keyboard Shortcut:

Save: Ctrl + numeric keypad 4 ... Ctrl + numeric keypad 6

Restore: Numeric keypad 4...6

Zoom 100%: Zoom to the entire audio material of the current track.

> ℹ **Note**: *The key combination **Ctrl + Alt + Mouse wheel** provides simultaneous horizontal and vertical zooming.*

Refresh: This updates the graphical display of the material. Length changes of objects (with the Resampling algorithm) are adjusted. When synchronization of project view and Elastic Audio is active (**View** menu > **Horizontal** > **Synchronize VIP** in the Elastic Audio window), the effects on the project are also displayed in the project window.

Edit VIP Object (Relative Mode)
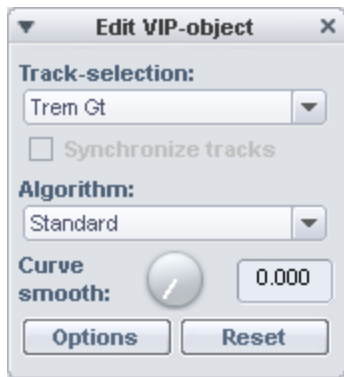
Track selection: If objects from multiple tracks were selected when you opened Elastic Audio, select the edited track here.

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

> ℹ *Detailed information about these algorithms can be found in* Resampling/Timestretching/Pitchshifting *(↗297).*

Curve smooth: This parameter causes the entered pitch curve to be smoothed by a time constant (specified in ms).

Options: Here you will find further setting options depending on the selected algorithm.

Reset: This will completely undo any changes made to automation curves or voices of selected objects.

Edit Slice Object (Direct Mode)

Key: Here's where you can set the key of the scale. This setting is not taken into account for the chromatic scale.

Scale: Here you select the type of scale (Major, Minor, Harmonic Minor, Pentatonic and Chromatic).

Pitch contour Tune!This button quantizes the pitch of the selected slices according to the selected scale.

Contour smooth: This function sets the quantization level (lower values quantize harder). As a result, the small pitch fluctuations (e.g. vibrato) that always occur with natural sound sources disappear, creating an unnatural mechanical sound.

Average Pitch Tune!: This quantizes the average slice pitch of the selected slices. The respective sections of the pitch curve of the slice objects are shifted as a whole, thus adjusting the average pitch. Pitch variations within a slice object are preserved.

> **ⓘ Notes:**
>
> - *Prerequisite for the automatic correction is the detection of the slice pitch (**Pitch** > **Detection**).*
>
> - *The calculated average pitch of slice objects with glissandi will often not correspond to the pitch that the human ear assigns to these slices. In this case, automatic correction of the mid pitches does not give satisfactory results. This can be remedied, for example, by removing the glissando passage by manually changing the slice boundaries (Ctrl key and curve bend tool) or by splitting the slice.*

Reset: This command resets the selected slices. The gray curve then corresponds to the orange one and the slice object is reset to the original mean pitch.

Harmonizer

Use the **Harmonizer** to add additional voices to the object

Chord: Here you can select the chord that the added voices will produce when the **Harmonic Adjustment** option is active.

If **Harmonic Adjustment** is not active, the voices will follow the key set in the **Edit Slice Object** section. The set chord then only serves as a guideline.

Humanize: Here you can set a "human touch" by detuning the individual voices more or less strongly against each other.

Create voices: Use this button to recreate parallel voices.

Pitch
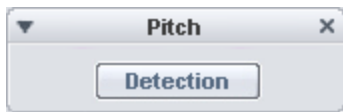
Detection: Use this button to start the pitch analysis. As a result, the absolute pitches are displayed graphically in the Elastic Audio Editor. This function is essential for further editing steps in Direct mode and must therefore always be executed first before you perform other editing operations.

Toolbar

Various tools are available for editing the slice objects and the pitch curve.

You can assign different tools to the left and right mouse buttons. The tool assigned to the left mouse button is displayed in blue, the tool assigned to the right mouse button in red. To assign, click on the desired button with the corresponding mouse button. Only the zoom tool automatically uses both mouse buttons.

❶	**Selection tool (arrow)**: Allows you to move slice objects, handles, or the pitch curve vertically. This modifies the pitch of slice objects as a whole. Slice objects and curve handles can also be selected. Selection frames and multiple selection with Ctrl or Shift key are possible.

❷	**Freehand drawing function**: You can use the pencil tool to draw the pitch curve freely.
❸	**Drawing pen for quantized drawing**: In quantized drawing, drawn line snaps to semitone steps (Relative mode) or to the notes of the scale selected at **Tune** (Direct mode).

💡 *With both drawing pens, press the **Shift** key additionally to draw a straight line. With the **Ctrl** key additionally held down, the slice objects are united while drawing.*

❹	**Curve Bend Tool:** You can use this tool to bend the pitch envelope between two neighboring handles. This changes the curve in the range between the curve handles, the handles themselves remain unchanged. The curve is bent at the clicked point. In combination with moving the curve handles at the slice object boundaries, you can thus bend the pitch curve while preserving the microtonal structures.
❺	**Scissors**: By simply clicking on the curve with the Scissors tool, you create additional slice objects and curve handles. To join slice objects back together, draw the curve over the slice objects using the Pen tool with the **Ctrl** key held down.
❻	**Eraser**: Use the eraser to reset the orange curve to the initial value at the clicked position.
❼	**Navigation tool**: Use the navigation tool to move the visible section vertically and horizontally.
❽	**Zoom tool**: Click with the left mouse button to zoom into the display; zoom out with the right mouse button. By clicking and dragging, you can span a selection frame, which is then zoomed into.

Numerical input of the pitch (Selection tool ❶ and Curve bend tool ❹): Double-clicking on a slice object, a handle or the curve opens a field in which you can enter the pitch numerically. Changes have a different effect depending on the tool; with the Selection tool active, the entire curve section is shifted; with the Curve bend tool, the curve is bent accordingly.

Playback Control
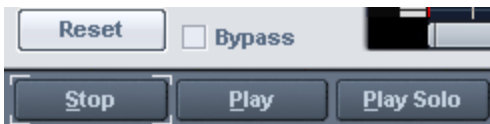
- **Reset**: All pitch curves are reset.
- **Bypass:** The project will be played back without the processing.
- **Stop/Play**: The playback of the project is started or stopped.
- **Play Solo**: Only the objects active in the Elastic Audio Editor are played.

Display Options

Grid: With this button you switch the display of the grid on or off.

Single Object mode: The maximum horizontal zoom level is limited to one object.

Synchronization mode for VIP and Elastic Audio Editor: The horizontal zoom levels and visible sections of both windows are synchronized.

Docked view mode: When this button is active, the project window is placed below the Elastic Audio Editor.

Applying Elastic Audio to Objects on Multiple Tracks.

Elastic Audio processes all the objects you have previously selected. The objects can also be from different tracks. In this case, use **Track selection** at **Edit VIP object** to select the track to be edited. The objects of the other tracks are displayed in gray in the background for control purposes.

If the option **Synchronize tracks** is active, in Relative mode the pitch changes of the object of the selected track are transferred to the objects of the other tracks.
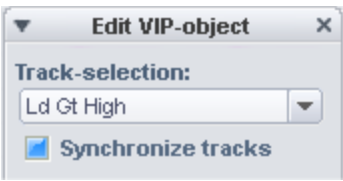
Keyboard shortcuts and Mouse Wheel

Navigating with the mouse wheel

Horizontal scrolling	Mouse wheel
Vertical scrolling	Ctrl + Shift + Mouse wheel
Horizontal zooming	Ctrl + Mouse wheel
Vertical zoom	Alt + Mouse wheel
Horizontal and vertical zooming	Ctrl + Alt + Mouse wheel

Keyboard shortcuts

Play solo/stop	Ctrl + Spacebar

Play/Stop	Spacebar
Select all	Ctrl + A
Refresh view	A
Undo	Ctrl + Z
Select tool for the left mouse button	Ctrl + 1-8
Select tools for the right mouse button.	Ctrl + Shift + 1-8
Show/Hide pitch curve	Shift + Alt + P
Show/hide objects from other tracks	Ctrl + Alt + T
Direct Mode	Ctrl + Up arrow
Relative Mode	Ctrl+Down Arrow
Play cursor to the left	Left arrow
Play cursor to the right	Right arrow
Horizontal zoom in	Ctrl + Left arrow
Horizontal zoom out	Ctrl + Right arrow
Change scale tuning (Frequency for "A" concert pitch).	Shift + R
Show/hide "View"	Shift + A
Show/hide "Edit VIP object"	Shift + O
Show/hide "Pitch"	Shift + F
Show/hide "Edit slice object"	Shift + S
Save zoom snapshot 1, 2, 3	Ctrl + Numeric keypad 4,5,6
Load zoom snapshot 1, 2, 3 without vertical zoom	Alt + Numeric keypad 4,5,6
Load zoom snapshot 1, 2, 3 without vertical zoom	Numeric keypad 4,5,6

FFT Filter

The FFT filter is used for precise, linear-phase filtering of signals.

The graphical view shows 3 curves, the vertical axis shows the volume in dB, while the horizontal axis shows the frequency in Hertz or the pitch in notes. The yellow curve represents the frequency response of the input signal, the blue curve the frequency response resulting from applying the red filter curve.

Drawing tools: You can edit the red filter curve in the graphical view with the drawing tools during playback and immediately hear how the sound of the edited audio signal changes. By clicking the left and right mouse buttons on the symbols, you select a drawing tool for each of these keys

❶	**Pen for free-hand drawing**: This tool allows you to draw any curve you need. Hold the "Shift" key down to draw straight lines.
❷	**Pen for quantized drawing**: Use this tool to draw stepped filter curves; the quantization steps depend upon the set resolution. If you hold down the **Shift** key, you will draw straight lines.
❸	**Bump tool**: Click with this tool above or below the curve to bend the curve/frequency area. The further away you place the bump tool from the curve when clicking, the wider the deformed curve area becomes. This effect can be intensified by simultaneously holding down the **Ctrl** key.
❹	**Navigation tool**: With the navigation tool you can move the visible section vertically and horizontally once you have zoomed into the display.

⑤ **Magnifying glass/zoom tool**: Click the mouse to zoom into the display. Click and drag to expand an area, which is then displayed as a zoom area.

Next to the drawing tools are buttons for actions with the filter curve as a whole:
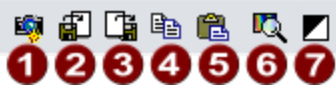
① **Snapshot**: The real-time display of the spectrum is paused and a still image is displayed.

② **Save**: Save the filter curve in a text file.

③ **Load**: Load the filter curve from a text file.

④ **Copy**: Copy the filter curve as text data to the clipboard (keyboard shortcut: **Ctrl + C**).

⑤ **Paste**: Paste the filter curve from the clipboard.
Use these commands to transfer a filter curve between different instances of the FFT filter.

⑥ **Analysis tool**: Creates a filter curve by spectral analysis. For this purpose, audio material is analyzed from the current track starting from the playback marker position. You can then save this filter curve and use it to filter the frequency response of another object.

⑦ **Invert current filter curve**: This inverts the current filter curve. This function allows you to correct the frequency response of rooms or speakers, for example.

Reset: This button resets the curve of the FFT filter.

Scale options: Options for displaying the filter curve.
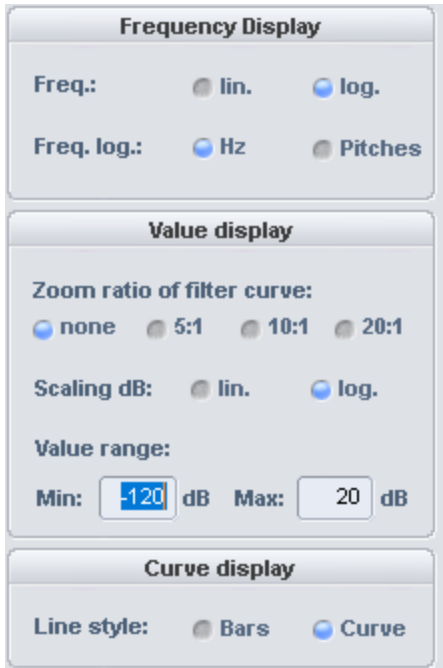
- **Frequency display**: The frequency can be plotted linearly or logarithmically. If you select the logarithmic display, either the frequencies or the musical pitch (notes) can be used for the frequency on the x-axis.

- **Value display**: The filter curve can be stretched on the y-axis in the ratio 5:1, 10:1 or 20:1. When the filter curve is stretched, a second dB scale is displayed on the right side of the graph. Note that then the display of the filter curve is no longer to scale with the display of the frequency responses.
  Also for the dB values a **linear** or the **logarithmic** representation is possible. At **Value range** you define the minimum and maximum value of the display.

- **Curve display**: The curves can be displayed roughly as bars or as interpolated curves.

Overlap: This parameter controls the overlap of the time windows for the calculations of the spectrum. Higher values improve the result, but also increase CPU load.

Resolution: With this parameter you can select the internal resolution of the algorithm. Higher resolutions do not automatically mean better results - the optimal setting depends on the audio signal.

Strength: This controller pinches or stretches the filter curve. This is especially necessary if the filter curve was created by analysis or the Sound Cloner function (see below). Then use this slider to control the amount of correction to the frequency response.

Vol.: With this control you can adjust the level behind the filter. Use the peak meters for **In** and **Out**.

Inverse: Reverses the effect of the filter, only the frequencies removed by the filter are then reproduced.

Bypass: This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

Sound Cloner Function

With the Sound Cloner feature you can determine the sound characteristic of a selected object and transfer it to another one. To do this, first determine a spectral image of the object that you want to use as a sound template and save it as a Sound Clone Preset. After that you can apply the Sound Clone to other audio material.

The filter curve of a Sound Clone preset is not taken directly into the FFT filter, but the target audio material is also analyzed and then a combined filter curve is generated that matches the spectrum of the target audio material to the spectrum of the Sound Clone preset.

1. Load audio material whose sound you want to use as a reference. Set the play cursor to a position representative of the desired sound spectrum.

2. Load the FFT filter as an object effect.

3. Now click on the button **Learn**.
   With the options **File** or **FX-Input** you can determine whether the unprocessed audio material of the audio file is analyzed or the input signal of the FFT filter is used. Effects that lie in the object before the FFT filter are included in the analysis in this case. This way you can further edit the sound before analyzing it, e.g. with an EQ.

4. After the analysis is complete, a save dialog opens. Save the newly created sound clone under a meaningful name (e.g. Late 50s Jazz).

5. Now select the object whose sound is to be adjusted and load the FFT filter here as well. Also place the play cursor at a position that is representative of the object's sound spectrum.

6. Now select the Sound Clone preset. The sound spectrum of the object is adjusted to the sound of the Sound Clone preset using the filter curve.

Use the **Strength** slider to control the intensity of the sound characteristic transmission.

The FFT Filter can also be operated as a track or master effect as a sound cloner. It is always important to remember: the play cursor and the selected track determine the audio material that is the basis for the analysis of the sound spectrum, both when creating the Sound Clone preset and when loading the Sound Clone.

Preset is 20 seconds of analysis, you can also set the length individually by selecting a range over audio material.

The menu **Presets** contains, in addition to the created Sound Clone presets, a number of universally usable filter settings.

Remove DC Offset (Offline)

This function removes the DC offset component from an audio file or object. DC offset can occur, for example, in a microphone recording due to A/D conversion. This component shows itself in a constant shift of the waveform around the zero line. The DC voltage component can lead to inaccurate results in subsequent effect calculations.

> ℹ️ *The effect processing is done destructively as an offline effect in the audio file of an object. See* Applying effects offline *(↗227)*

Options:

- You can specify a minimum **DC offset threshold** above which DC offset removal is to be performed.
- **Process stereo channels separately**: If you switch off the option, the left and right channels will be processed with the same value. This makes the calculation faster, especially with many long files.
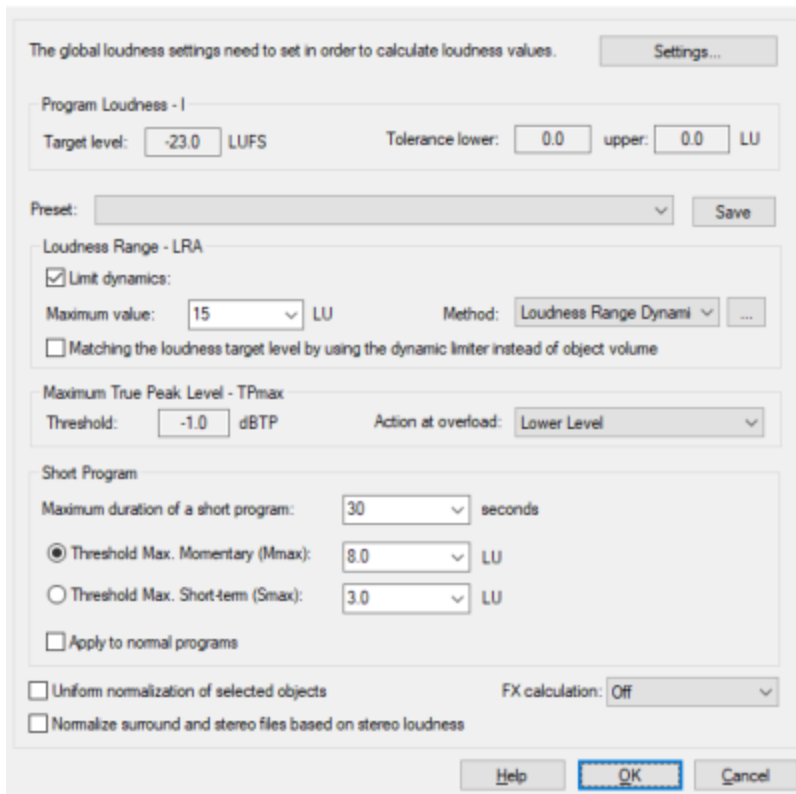
Swap channels

Applied as an object effect, the application of the effect corresponds to the function **Swap object channels** in the menu ▾ of the section **Pan** on the tab **FX** of the object editor (↗186).

You can also find a corresponding function for an object or track in the panorama editor (↗492), (right-click on the Pan slider) if you activate the options **Copy L > R** and **Copy R > L** at the same time.

When used offline, this function swaps the audio data of the right and left channels of a stereo file.

Loudness adjustment

Loudness normalization is performed in this dialog. In this process, the object level is adjusted so that its loudness corresponds to a loudness standard.

ℹ️ *For more information on loudness standards (e.g. EBU R128), refer to* Loudnessmeter (↗559).

The loudness calculation is based on the values defined in the **program settings** under **Effects** > **Loudness**. You can reach the settings via the button **Settings**.

Settings

Loudness standard: Select here the standard between **EBU R128** or **ITU-R BS.1770/1771** in its current version. By selecting **user** you can set your own measurement parameters (↗563) and your own **overload threshold** (Maximum True Peak Level).

Here you can also set the **Target value** in LUFS for the program loudness and specify the tolerance range. These values are used in the Loudness Adjustment dialog and in the Loudness meter.

You have the following **Save options**:

- **Write loudness values when saving to files (BWF - EBU Tech 3285v2)**: The loudness values are written to the BWF chunk.

- **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)
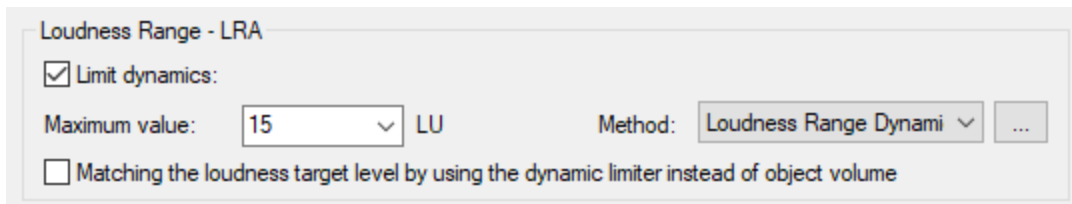
Dynamic limiting is a part of the dialog for loudness normalization and enables the limiting of object dynamics in order to even out loudness variation within the audio material.

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.

You can choose between two **procedures**:

■ **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 (↗183) and can be used to control the loudness range (LRA).
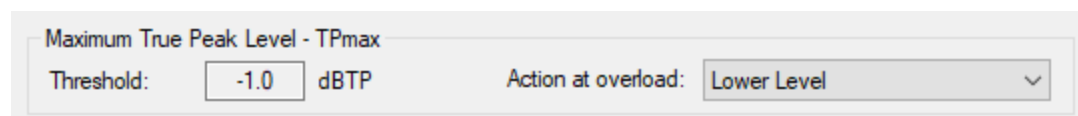
Loudness Range Dynamics in the Object Editor

- **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.

If the option **Matching the loudness target level by the using dynamic limiter instead of the object volume** is active, the adjustment of the loudness range to the target value is performed by the leveler or the loudness range dynamics. In this case the object volume is not changed.

Action when clipping occurs

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 Program: 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 loudness 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 loudness 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. Note that the loudest object co-determines the value for all others. This could cause the loudness of the other objects to be below the target value for the program loudness.

Effect calculation

- **Off**: Only the audio material of the object is used for loudness analysis.

- **Objects only**: This means that the object effects only are taken into account during the calculation of the loudness value.

- **Including master**: Object, track and master effects are included in the loudness calculation.

Multiband Dynamics

Multiband Dynamics lets you edit 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. Multi-band technology also lets you specifically edit individual frequency ranges.
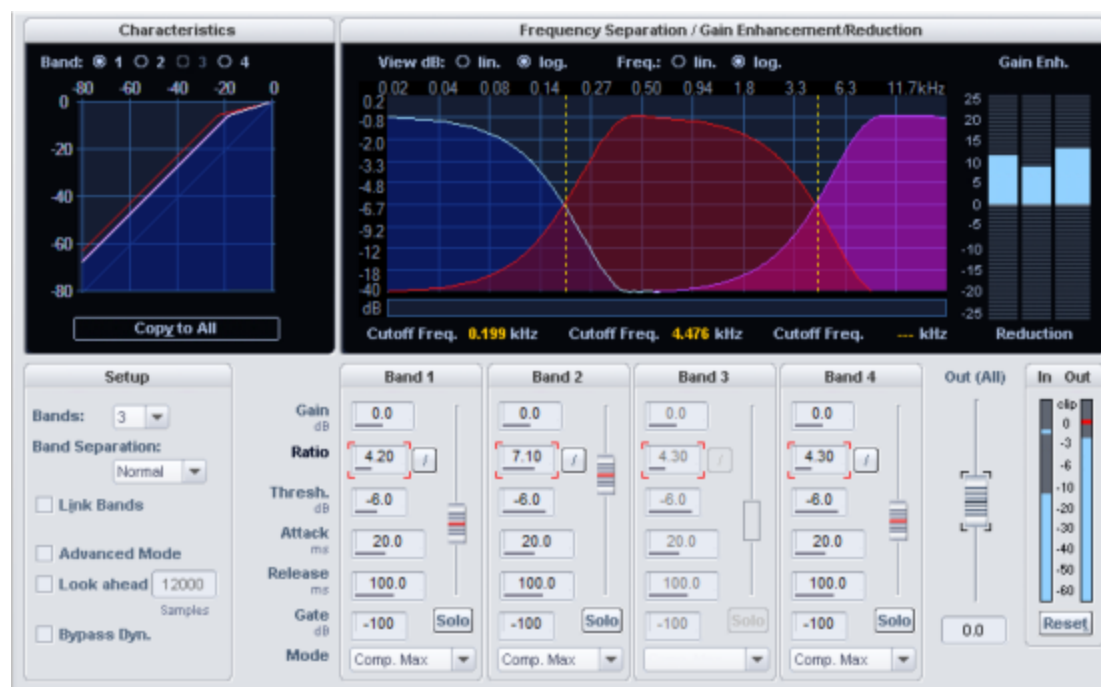
Frequency separation/Gain enhancement/Reduction

In the **Frequency separation / Gain Enhancement/Reduction**area, 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.
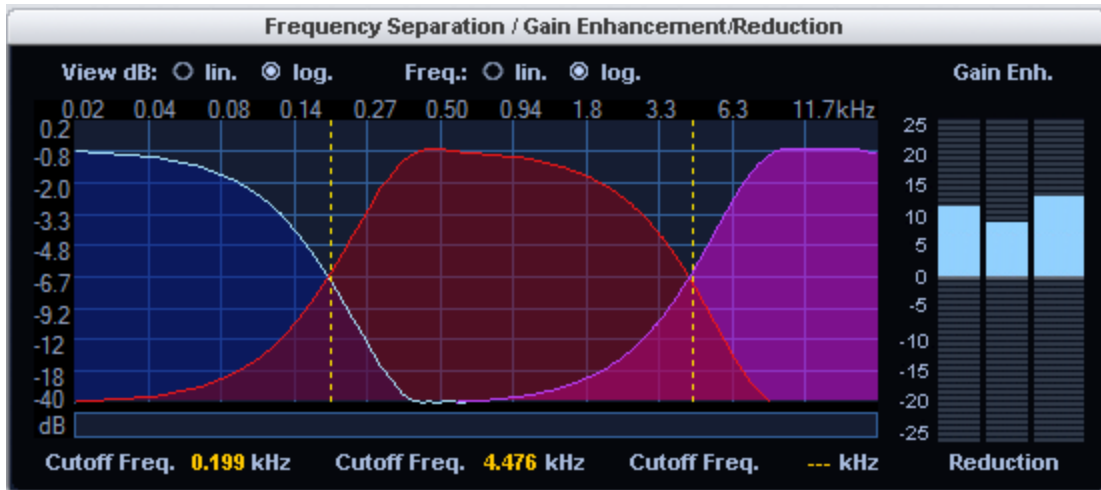
Separation frequencies

Drag the dashed lines in the graphical representation to set the crossover frequencies of the individual filter bands. The number of separation frequencies depends on the number of selected bands (**Bands** parameter in the **Setup** area). 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.

Characteristics

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).
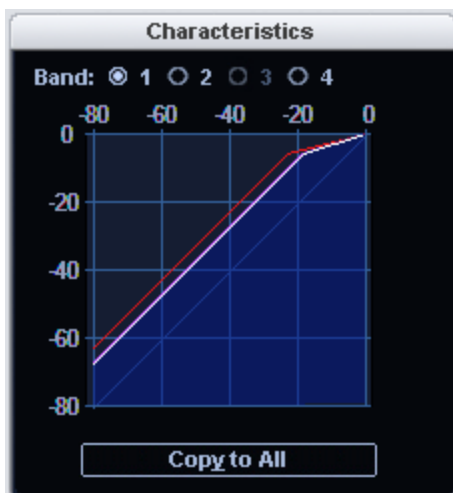
Band: You can select a band , the characteristics curve of the selected band is displayed in white, while the other bands appear as lines in their corresponding color. The band selection always follows the currently edited dynamic parameter.

Copy to All: After pressing this button the parameters of the selected band are applied to all bands. If you also have **Link bands** (in the **Setup** area) selected, the values for all bands remain coupled to one another.

Setup

Control	Description
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 a standard dynamics effect. The CPU load rises with increasing number of bands.
Band separation	With this parameter, different properties of the filters are influenced simultaneously, so that the "selectivity" is increased. With a higher setting for the band separation, the slope of the filter curves increases, i.e. the transition range between two bands becomes smaller and the attenuation in the stop band increases.
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. For rough adjustment at the beginning, it is often appropriate to adjust the parameters together at first. If you have activated **Link bands**, it does not mean that all parameters in all bands will have the same values. Initially, only the parameters that were changed after switching on "Link bands" are the same. If you want all bands to have the same settings, click **Copy to All** below the characteristic curve.
Advanced Mode	If this option is selected, the processing routines of the Advanced Dynamics (↗237) effect are used internally.
Look ahead	When this option is selected, the dynamics section works with "look-ahead". This setting affects all bands.

Bypass dyn. (bypass dynamics):	The dynamic processing of the individual bands is deactivated. This function serves to compare the editing results with the original.
Bypass all	This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

Parameters for the dynamic sections
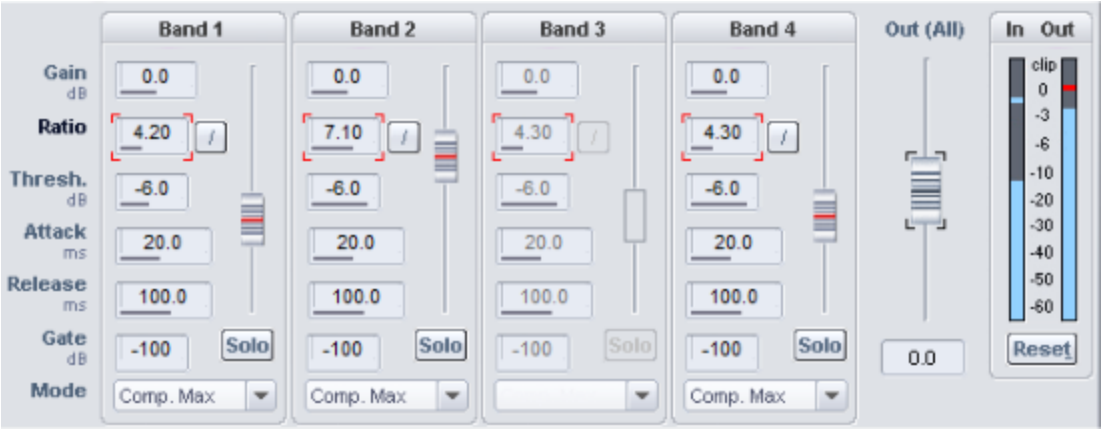
💡 Click in an input field (Gain, Ratio, Threshold, Attack, Release or Gate) to control the parameter with the slider next to the input fields.

Control	Description
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 too high, overdriving may occur - in this case the characteristic curve would have to continue upwards beyond the limit of the graphic display.
Ratio (factor)	This parameter controls the strength of the respective effect. 1.0 means no effect. Clicking the button beside it resets the ratio to 1.00.
Thresh. (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	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.
Out (All)	Here you can set the overall output of the entire effect. The graphic display does not reflect this setting. Use the fader to balance the level difference caused by the dynamics processing.

Dynamics Mode

Dynamics Mode	Description
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. This setting is often used to increase the loudness.
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.
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.
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.
	💡 *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.*

Strategies for coping with the flood of parameters

A multiband compressor like Multiband Dynamics naturally has a lot of parameters. Here are a few tips on how to adjust:

- If you need to alter the **dynamics of all bands**, rather than one individual band, the following approach may be useful:

    1. Common setting for all bands

        a Select the mode that corresponds the most with your needs. If you want to increase the volume of the audio material, select **Comp. max**. If you want to achieve an improvement of the sound characteristics, an increase of the "transparency", an improvement of the speech intelligibility, a refreshing of old recordings, crisper basses, etc., without increasing the loudness, use the **Compressor**. The selection of this mode only applies to the selected band.

        b Click **Copy to all bands**. Selecting this mode ensures that all bands feature the applied setting.

        c Activate **Link bands**. Any changes you perform in the current band are automatically mirrored on the other bands.

        d Change the parameters of all bands until the audio sounds right.

    2. Fine adjustment for the individual bands

        a Turn off **Link bands**.

        b Activate **Solo** mode for a band. Individual bands may now be isolated, making the task of optimizing settings for each band easier. The optimal time constants usually have lower values in the high frequency bands because the waveforms are shorter here.

        c If you can't find satisfactory settings for a band, 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 switch off the Solo mode and compare the original signal with the processed using **Bypass all**.

- **Presets**: Compare the presets provided and use a suitable preset as a starting point for finding the optimal 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.

- Compare at same volume: Activate and deactivate the effect with **Bypass** and adjust the volume with the slider **Volume** until the audio level with processing equals the level without processing. Use this method to compare the sound differences without allowing the dynamics processing to alter the volume levels.

Multiband Stereo Enhancer
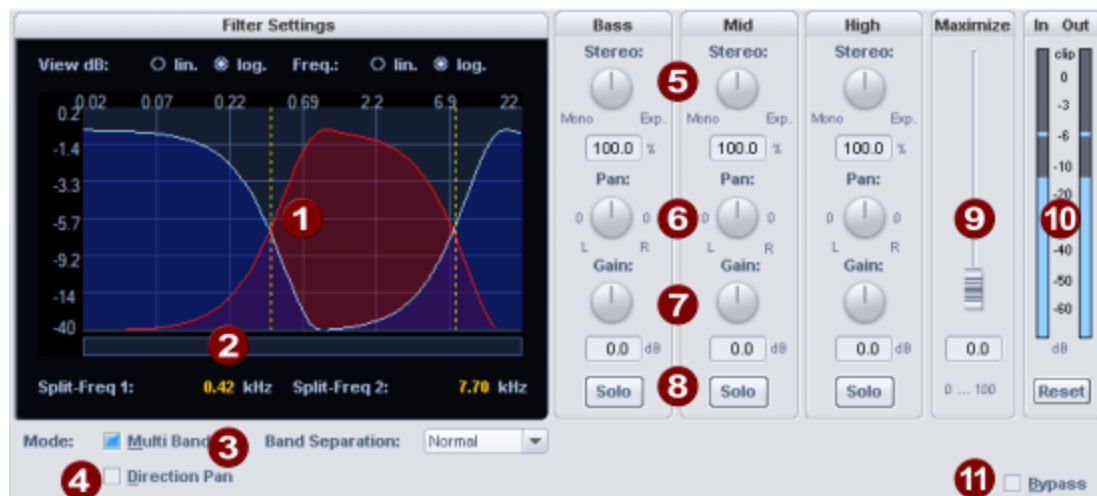
The Multiband Stereo Enhancer allows manipulation and correction of the stereo image in three independent frequency bands.

❶ **Filter graphic**: The graphic shows you the frequency responses of the individual bands. The left axis label displays damping in -dB, the top axis label - frequencies in kHz. The labeling can be either linear or logarithmic

❷ **Crossover frequencies**: Drag the dashed vertical lines to change the crossover frequencies of the three filter bands. The displayed values in kHz correspond to the intersection points of the adjacent frequency curves.
Band Separation: This parameter influences different properties of the filters simultaneously, so that the "separation sharpness " increases. With a higher setting for the band separation, the slope of the filter curves increases, i.e. the transition range between two bands becomes smaller and the attenuation in the stop band increases.

❸ **Multiband**: If this is active, the stereo enhancer works in three frequency bands.

❹ **Direction Pan**: Here you can choose between two modes for the operation of the pan controls: If Directional Pan is off, the pan controls operate conventionally. The entire stereo signal is shifted in the panorama. If the mode is active, only the mono part (the center signal) is considered in the panorama adjustment. 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 component (the side signal), which contains the sound sources that are located off center, remains unchanged.

Under Bass/Mid/High there are controls for stereo manipulation of the corresponding band:

❺ **Base width control**: Here you set the base width between 0 and 200. 0 means mono, 100 corresponds to unchanged base width (stereo) and a value of 200 means maximum base width (differential signal). Depending on the correlation between left and right, there may be an increase in level when the base width is reduced. In the extreme case of maximum correlation, which occurs when the left and right channels are identical, and a set base width of 0 (mono), a level increase of 3 dB occurs.
When the base width is increased (values above 100), the mono compatibility decreases. If, on the other hand, you reduce the base width, mono compatibility is maintained.

❻ **Panorama controller:** Here you set the panorama for the individual bands. Next to the slider the attenuation for left and right is displayed in dB. If the **direction pan** mode is active, the controllers operate as direction mixers. In this case, only the mono component (mid signal) is taken into account.

❼ **Gain**: Adjusts the level of the respective band.

❽ **Solo**: With **Solo** the frequency bands can be listened to individually. This option makes it easier to set the filter parameters. As an example, you can isolate a specific frequency range of a mix, and change the aspect of the stereo image for the range.

⑨ **Maximize**: This control compresses the side signal, increasing stereo transparency without affecting mono compatibility. In multiband mode, this setting affects the middle band.

⑩ **In/Out**: The peak meters show the input and output level in dB. **Reset** on these displays resets the display of the maximum level of In and Out.

⑪ **Bypass**: This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

Normalize

The **Normalize** function raises the level of an audio object to the maximum possible level without clipping the material. This is done by searching for the largest signal peak in the audio material and raising the level of the object so that this point is set to exactly 0 dB (or any other value between 1% and 400% of the level).

Keyboard shortcut: **Shift + N**
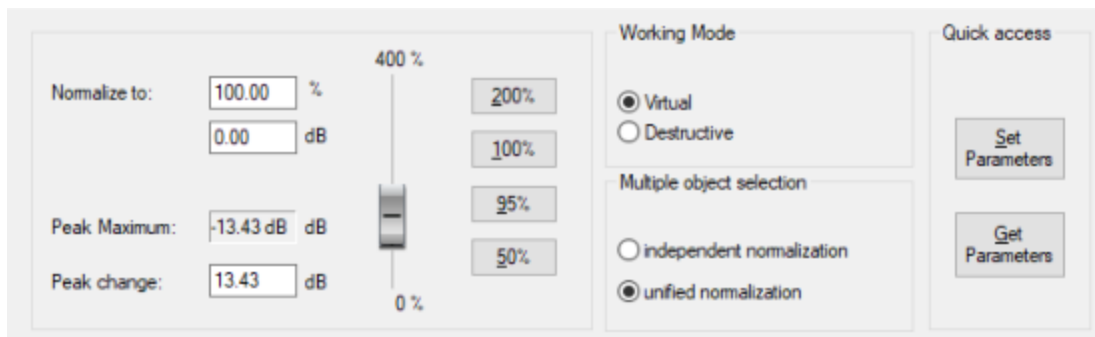
Normalize to: Here you define the value to which the audio material should be normalized by entering it in the input field in % or dB, adjusting it at the slider or selecting one of the preset values (50, 95, 100 or 200%).

Peak Maximum: Displays the maximum level determined for the selected range or object.

Peak Change: Displays the level change to be applied according to the selected normalization level and the detected peak maximum.

Working Mode

- **Virtual**: No sample data is changed, only the object level is adjusted.

  💡 *You can also apply this function to an object by clicking the* **Norm.** *button below the volume fader in the* **Object Editor**. *This normalizes to 0dB.*

- **Destructive**: Normalization is performed directly in the audio file, changing the stored data on the hard disk. In the dialog there is now an option **Create Copy** (on by default). The copy enables you to undo the normalization.

Multiple object selection

- **Independent normalization:** Each object is normalized according to its own maximum (peak) level.
- **Unified normalization**: (default) The maximum level of all selected objects is determined and each object is normalized based on this value.

Quick access: Use the button **Set parameters** to set the current settings of the dialog as parameters for the function **Normalize (Quick access)**.

Keyboard shortcut: **N**

Use the Get **Parameters** button to set the current Quick Access parameters in the dialog.

Invert phase

As a real-time effect in tracks and objects, this function corresponds to the phase switch in the object editor, mixer channel and track head.You will also find a corresponding function for objects and tracks in the panorama editor (↗492), (right-click on the Pan slider).

For audio files, this function inverts the data of the sample within the marked area along the zero line, i.e. its phase is inverted. Negative values become positive and vice versa.

Room Simulator

The Room Simulator is a reverb effect that simulates the reverberation of any room using its impulse response ("reverb tail"). A room impulse response is the reverberation of a very short, impulsive sound, such as a bang. You can additionally manipulate the impulse response with the various parameters in the dialog. This way you edit the reverb properties (reverberation time, damping…) similar to algorithmic, digital reverb effects. In addition, however, you have the option of fundamentally determining the reverb character by selecting the impulse response.

A graphic representation of the impulse response and the envelope allows an overview of the impulse response manipulation using the Room Simulator parameters dialog.
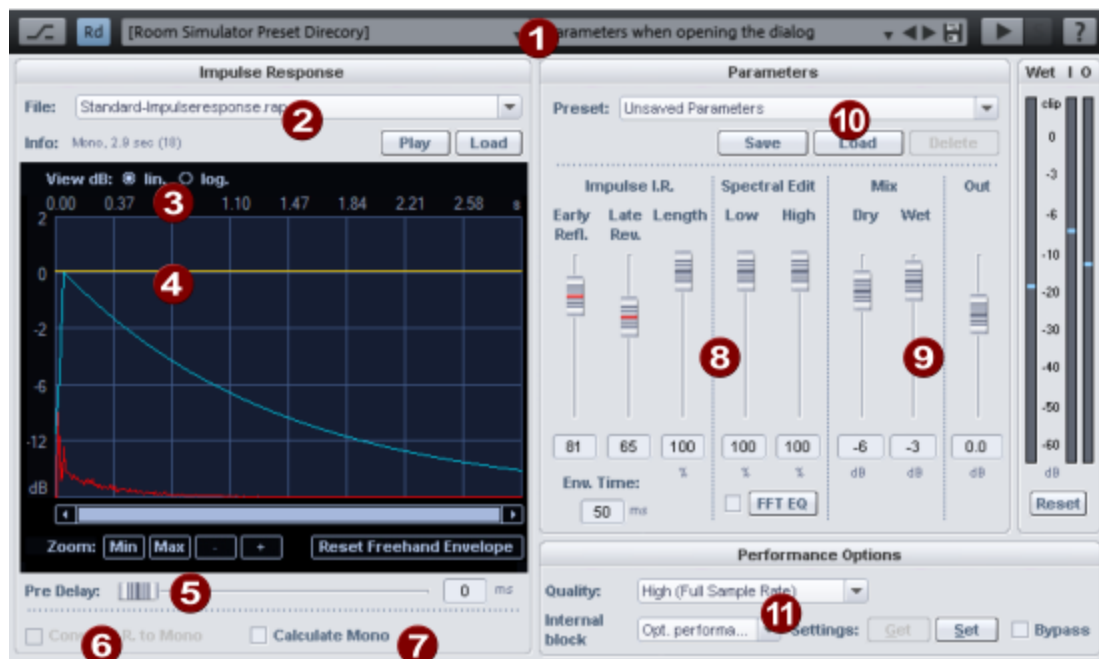
❶ **Banks and presets**: In the header of the dialog you can choose from a series of banks and corresponding presets.

ⓘ *The default installation of* Sequoia *only installs a small selection of presets and impulse responses for the room simulator. You can get the rest of the presets and impulse responses by installing additional content via menu* **Help** >**Download instruments and sounds...**.

The presets contain both the parameter settings of the dialog and the reference to a specific impulse response. The impulse response can be any sample, supplied impulse responses have the file extension ***.IMR**.

❷ **Impulse Response**:

■ **File**: Initially, the impulse response defined in the selected preset is selected here. From the menu you can select another impulse response from the folder of the loaded bank.

ⓘ *To select the audio files opened in the program here as the impulse response, select the preset bank* **Loaded projects**.

■ At **Info** you get information about the length of the impulse response and whether it is in mono or stereo.

■ With the button **Play** the impulse response is played.

■ With **Load** any wave file can be loaded for use as impulse response.

❸ **View dB lin./log.**: The graphic display with the impulse response has linear amplitude scaling. This settings corresponds to the common display of samples. The representation in logarithmic amplitude scaling corresponds to the human loudness perception.

❹ **Freehand envelope**: The amplitude curve of the impulse response can be edited by drawing the yellow freehand envelope into the graphic with the mouse. You can use this to attenuate or cancel individual early reflections of the impulse response.
With the zoom controls (**Min**, **Max**, **+**, **-**) you can zoom into the time course of the impulse response to find problematic reflections and edit them exactly.
By clicking on the **Reset Freehand Envelope** button you can reset the yellow envelope curve.

❺ **Pre-Delay**: The **Pre-Delay** value delays the entire impulse response by a period of 1ms to 100 ms.

❻ **Convert I. R. to Mono**: If you select the **Convert I.A. to Mono** option, stereo impulse responses will be converted to mono. The convolution is calculated in stereo - in the surround case in n channels corresponding to the number of group channels.

❼ **Calculate mono**: If active, the entire reverb is calculated in mono only, which reduces the required computing power. The input signal and the impulse response, if stereo, are converted to mono before convolution.For surround application, all channels of the group are summed to a mono signal.

⑧ **Envelope I.R./Equalizer**:
Use these controls to influence the duration and sound of the reverb by editing the impulse response.
Envelope I.A.: The envelope of the impulse response (light blue curve in the graphic) allows to fade in and out the impulse response. The edits made by the envelope are shown in the graph of the impulse response (red curve).

- By fading in the impulse response you influence the proportion of early reflections in the reverb signal, **Env. Time** defines the length of the early reflections, **Early Refl.** defines how much of it is faded in.

- At **Late rev.** you can fade out the reverberation by fading out the end of the impulse response. With **Length** you can reduce the duration of the Hall effect by shortening the impulse response. This will be cut off, so that an unnatural decay can occur, therefore combine the shortening with the fading by the parameter **Late Rev.**.

Equalizer:

- **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 signal can be post-processed with an additional **FFT filter** (↗274).

⑨ **Mix/Out**: With these sliders you adjust the mixing ratio of the **Dry** and **Wet** signals as well as with **Out** the overall level.
Wet/I/O: The level meters show the generated reverb signal (Wet) as well as the input and output signal level. **Reset** resets the markings of the highest levels of the level indicators.

⑩ **Parameter Presets**: Here you can select, save, load and delete presets, which are composed of the parameters for the envelopes, frequency response and mix. The presets can thus be combined with different impulse responses.

⑪ **Performance/Options**: Room simulation by convolution is relatively computationally expensive. With the settings at **Performance Options** you can set different qualities for the effect with different levels of computing effort.

- **Quality**: In the two modes **Normal** and **Normal Advanced** the room simulation is only calculated with half 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). The two **Normal** modes differ in the quality of the resampling applied, with the resampling quality applied in the "Normal/Advanced" mode being 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.

- **Block length**: This parameter determines the block length used for calculation. Short block lengths raises the number of the required calculation operations causing CPU load to increase. Long blocks lead to irregular CPU loads. The parameter has no effect on the calculation results. The **Opt. Latency** settings 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 **Opt. performance** setting sets the internal block length so that the performance is as good as possible without letting the latency become too large.

  > ℹ️ **Note**: 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 with the "high quality" setting.

- **Get/Set Settings**: This allows you to save and load quality options. When you have found a fitting setting, click on **Set**. This saves the setting. If you want to reload the saved setting, click the button **Get**.

Bypass: This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

Tips and tricks

- By applying effects to the impulse response you can influence the reverb character in many ways beyond the possibilities offered by the room simulator:

- If you apply menu **Effects** > **Sample manipulation** > **Reverse** to an impulse response, you get a reverse reverb.
- The application of time-stretching (↗297) to the impulse response allows the room size to be changed without altering the resonant behavior of the room.

■ Apply an pulse-like decaying envelope to any shorter samples and use the result as an impulse response - this creates the most exotic reverb effects.

■ Interesting reverb sounds are also created when you use samples of percussion instruments as impulse responses.

■ Via **Menu Object** > **Impulse response extraction...** (↗618) you open an assistant that supports you in creating your own impulse responses for use in the room simulator.

Reverse

In offline editing, this function reverses the sample's data in the wave project within the selected range along the timeline so that it plays from back to front. For objects, this is equivalent to setting the **Backwards** option in the **Fades** view of the Object Editor.

⚠ *Note that when using it in the object, some functions (Musical Tempo Adjustment, AudioWarping, Elastic Audio, Melodyne) are no longer available.*

Resampling/Timestretching/Pitchshifting

In this dialog, under Algorithm, some advanced settings can be made for the Timestretching/Pitchshifting in objects (↗191) as well as the settings when applying this effect offline.

Algorithm:

- **Resample**: Samplers and PCM synthesizers use this method when transposing samples. 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.

  > ⚠️ *By default, resampling is used for objects created when you load audio files whose sample rate differs from the project sample rate. Once objects are resampled in real time, some other effects cannot be applied. Read the notes at* Audio files with different sample rate

- **élastique Pro/élastique Efficient**: This Élastique algorithm, licensed from the zPlane company, is the default algorithm for timestretching and pitchshifting. If the CPU load threatens to become too high due to the real-time use of **élastique Pro**, you could reduce the computing load by using **élastique Efficient**, at the price of lower sound quality.

- Monophonic Voice: This is a special timestretching and pitchshifting for monophonic vocals, speech or solo instruments. The audio material should not contain any background noise, even strong reverb can reduce the quality.

Correction formant factor: (not with algorithm élastique Efficient) With this option, the formants that determine the basic character of a voice are preserved during pitch changes. However, the formants can also be shifted by +/- 12 semitones to achieve intentional voice alterations.

Use stereo M/S mode: (only with élastique algorithm) With this option the audio signal is interpreted as M/S signal, that is, the left channel is processed as center signal and the right channel as side signal. This may give you better results with stereo signals. To convert to an M/S signal use the MS <-> LR option in the stereo editor of the object.

The parameters **Time factor** and **Pitch (semitones)** are used to set pitch and tempo changes.

> 💡 *In* Pitchshift/Timestretch mouse mode *these parameters can be set directly on the object using the handles.*

In the area **Time factor calculation** you can calculate the required time factor from the original length and original tempo as well as the desired new length or tempo. Unlike the parameters in the object editor, the length is specified here as the resulting length and not as a change in length.

With **OK** you apply the effect offline, for objects the settings are transferred to the object editor.

Change Sample Rate (offline)

> ℹ️ *The effect processing is done destructively as an offline effect in the audio file of an object. See* Applying effects offline

This feature lets you change the sample rate of an audio file. This process is also called "Resampling".

Resampling changes the number of recorded samples per second. If the sample rate is increased, the change is made almost completely without loss. Extra samples are interpolated and the file size increases. When changing to a lower sample rate, some samples are removed and the file size decreases. This results in a loss of overtones.
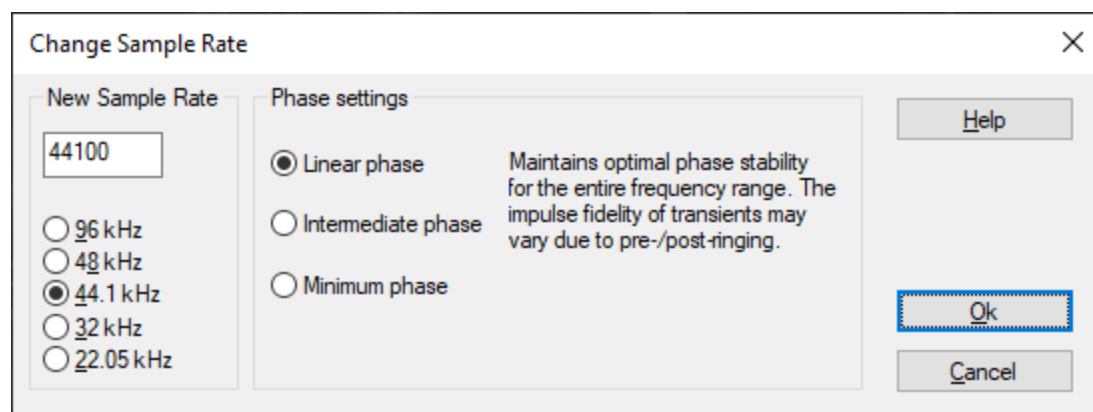
For the phase setting, you can choose between **Linear Phase**, **Intermediate Phase** and **Minimum Phase**. For most use cases, the **Intermediate** setting delivers best results. When you want to have a better preservation of the phases of the audio at all frequencies, you can chose **Linear**, for the price of the introduction of pre-/post-ringing at the transients. With the setting **Minimal** the transients are best preserved, but then the fluctuations of the phases in the lower frequencies are larger.

> ℹ️ **Note**: *Sample rate adjustment can also take place during recording or playback, for example during object resampling, when using audio files with a sample rate different from the project sample rate, or when using* Varipitch *(⬈89).You set the quality of the resampling used in this process in the* **System settings** > **Resampling quality options...** *(⬈726).*

sMax11
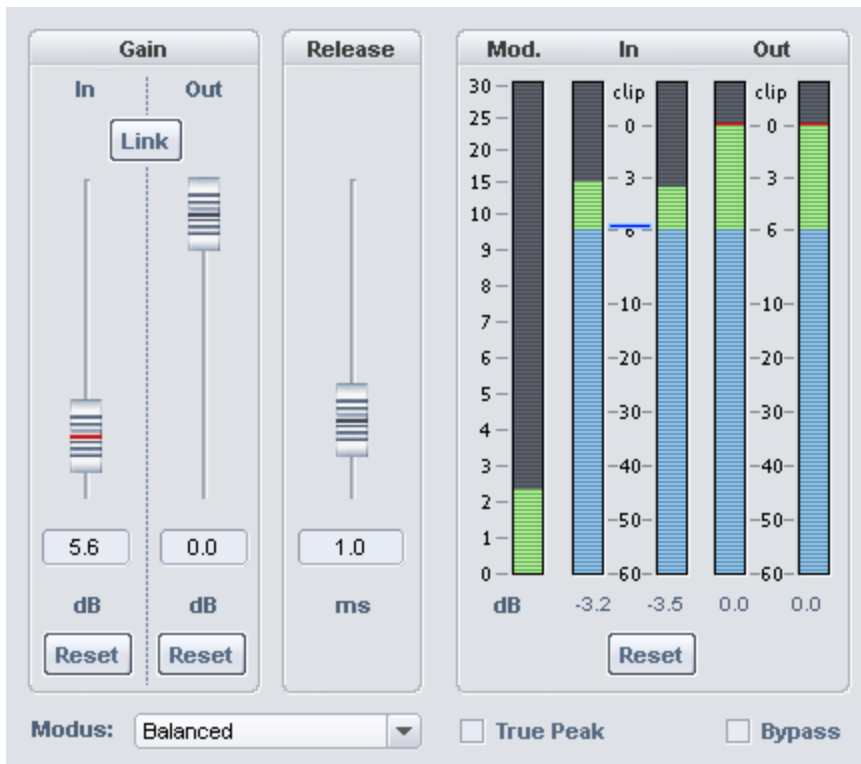
The Maximizer **sMax11** is a hard (brickwall) limiter with input gain that allows you to increase the loudness of your audio signal. Specify an input gain (**Gain In**), the signal will be amplified by this level. At the same time, the sMax11 ensures that the signal does not exceed the output level (**Gain Out**). With the parameters **Mode** and **Release** time you can, depending on the audio material, influence the necessary control processes.

Gain In: This value specifies how much the signal should be increased (maximal). **Reset (Gain In)**: Resets the "Gain-in" parameter.

Gain Out: Maximum output level. **Reset (Gain Out)**: Resets the "Gain-out" parameter.

Link: Gain In and Out are coupled, i.e. the Gain Out is reduced by the same value as the Gain In value is increased. For example, with the Link option turned on, you can increase the gain-in 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: Time span until the effect processing of the signal is completely withdrawn in milliseconds.

Mode: Control behavior of the effect. 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. For signals with dominant percussive components, the hardness of the beats is preserved. Distortions are masked by the beat.

- **Hard clipper**: In this mode the regulation is switched off. Level peaks are simply cut off, which can lead to strong distortions. This mode is appropriate for example to accentuate the transients of individual beats even stronger.

Displays:

- **Mod.**: Modification of the signal in dB. This does not take the input gain component into account. Only the level of processing necessary to intercept the level peaks is shown.

- **In/Out**: The peak meters show the input and output level in dB.
  Reset resets the peak meters.

True Peak : As the level above which the signal is considered overdriven (clip), you can also select the maximum peak level (Maximum True Peak Level), which you set in the loudness adjustment (↗279). When limiting the signal, inter-sample peaks are then also taken into account to prevent distortion during D/A conversion at the analog output.

Instructions:

1. First, turn on the **Link** option. This way, volume will stay constant and you can better compare your changes.
2. Now, raise the input amplification **Gain In** until the sound changes become unacceptable.
3. Now turn down the **Gain In** a little bit.
4. You can also minimize distortions by raising the **Release** time. However, this will reduce the compression effect and the increase in loudness.
5. Switch **Bypass** on once in a while.
6. When you have found the optimal setting, turn off the **Link** option and set the **Out Gain** to 0 dB.

Spectral Cleaning

With Spectral Cleaning you can remove distortions (like coughing, whistling, or singular claps) from a recording without influencing the wanted signal.

ℹ️ *The effect processing is done destructively as an offline effect in the audio file of an object. See* Applying effects offline *(↗227)*

The music is represented in the Spectral Cleaning Editor by 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.
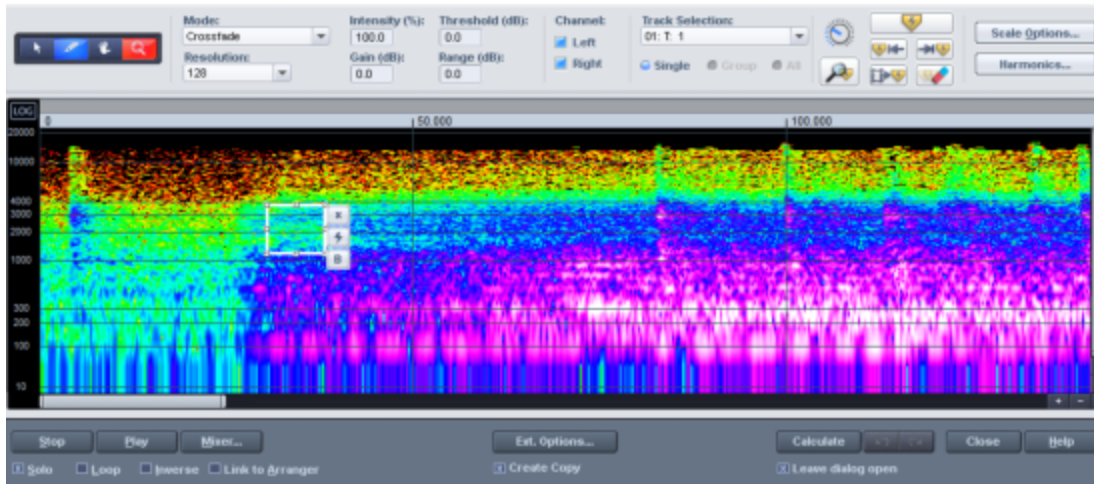
Audible distortion that is louder than the desired signal are usually limited to a certain frequency spectrum. They are highlighted with colors in the spectrogram. A sustained tone is represented by a pattern of horizontal lines corresponding to the tonal components and overtones of the tone. You will recognize a pulse-like disturbance as a vertical peak.

In Spectral Cleaning, you mark disturbances in the frequency spectrum with the drawing tool to remove them. To avoid an audible gap, components of the original frequency spectrum that have been removed are supplemented from the signal that surrounds the interference.
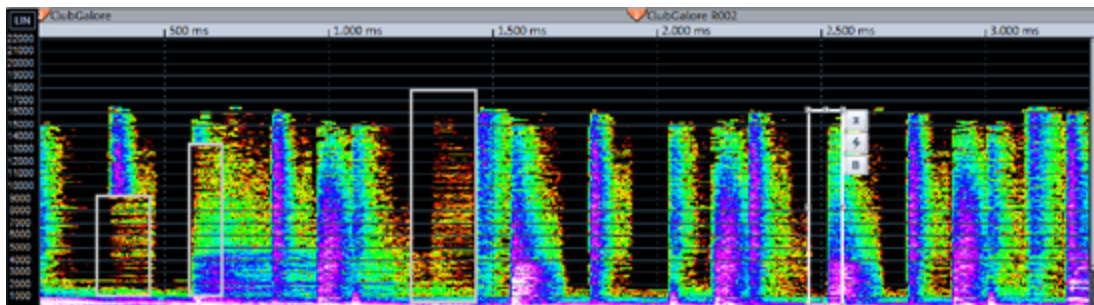
Spectral Cleaning is suitable for the removal of short, impulsive noises, such as crackling, coughing or short drop-outs. To eliminate permanent noise such as humming or static, please use the DeHisser (↗250) or the DeNoiser (↗253).

Operation

1. Select the object and place the play cursor approximately at the location of the disturbance. Then open the Spectral Cleaning Editor via the menu **Effects** > **Restoration**.

2. Select the drawing tool. The mouse pointer will turn into a pen icon. Then mark the interference by selecting a rectangle around the interference signal. You can also highlight and remove multiple interferences in the editing window.

You can change the selection with the handles on the frame. The Selection tool and the Drawing tool can be used to move a selection. With the **Alt** key held down, it can only be moved horizontally (time), with the **Shift** key held down, it can only be moved vertically (frequency).
Use the keyboard shortcuts **Ctrl + X**, **Ctrl + C** and **Ctrl + V** to cut, copy and paste markings. This only copies the settings of the rectangle, and not any audio material.

> ℹ️ *Double-clicking on a marking expands it vertically to cover the entire frequency spectrum.*

3. 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.

4. Three buttons are displayed at each marking after the marking is selected:

   🔲 **Delete**: If you click this button, the marking will be removed.

   ⚡ **Original**: With the flash button you can display the original signal containing the disturbance 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.

5. Sometimes it will also be necessary to modify the effect parameters (see below under "Editing Modes").

6. When you are satisfied with the result, click the **Calculate** button. The editing is included in the audio material.

> ℹ️ *There are a number of keyboard shortcuts for navigation and playback, you can also use them to save and load custom zoom levels. For a list, open **Program Settings** > **Keyboard, Menu, and Mouse** > **Spectral Cleaning Shortcuts**.*

Toolbar

Assign a tool to the left and right mouse buttons by left-clicking or right-clicking the button. The tool for the left mouse button is displayed in blue and the right mouse button in red.

①	**Selector tool**	This tool allows you to modify existing selection frames. The selection size can be adjusted at the edges, and by dragging in the center it can be moved as a whole.
②	**Drawing tool**	With this tool you can create new selection frames and also change the selection size and position the selection frame.
③	**Navigation tool**	If you are working at a high zoom level so that you cannot see the entire contents of the selected object, use this tool to move the displayed window section.
④	**Zoom tool**	Zoom into the display with this tool by left-clicking. Click and drag to select a zoom area.

Editing Modes

With the **Mode** setting you can specify how you want to remove the distortion. All marked disturbances are processed with the selected mode depending on the **Intensity** controller.

- **Crossfade**: In this preset mode, the noise is replaced by the areas immediately adjacent to the marking.

- **Crossfade (hard)**: This mode behaves like "Crossfade", but its effect is somewhat stronger. Thus, more severe disturbances can be better suppressed. It is especially important to mark the interference precisely when doing this.

- **Crossfade (from left)/(from right)**: In this mode, the noise is replaced only by the area immediately adjacent to the left or right of the marking.

- **Gap:** This mode is especially useful for very short dropouts.

- **Damping**: This mode dampens the selected noise. The surrounding audio material is not included. In this way no artifacts will be created, but there may be unexpected silence or even a drop-out. Use this mode only when the noise is present in a very narrow frequency range or when the surrounding signal is very quiet.

- **Fade In**: This mode is similar to the "Damping" mode. The signal will not, however, be uniformly dampened, but rather faded in. This means that frequency-selective fade-ins can be created.

- **Fade out**: In this mode, the marked interference signal is faded out over time. Use "Fade out" e.g. in case of disturbances at the end of a song.

Intensity: With this parameter you can adjust the strength of the processing.

> ℹ **Note**: The "Mode" and "Intensity" settings can be set individually for each marking. Select multiple markings (click with held **Ctrl** key) to change them for these markings together.

Resolution: Determines the resolution of the applied FFT (Fast Fourier Transformation). Lower values often produce a more effective removal of the disturbance, but also 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.

> ℹ **Note**: The setting **Resolution** applies to all edits and cannot be customized for each selection.

Gain: This allows you to apply a level boost to compensate for a drop in volume caused by the interpolation. This is particularly suitable for removing gaps when editing multitrack recordings.

Threshold: With this setting you can limit interpolation to certain amplitudes in the spectrum. Thus, louder disturbances can be removed from the spectrum without affecting background noise.

> **ⓘ Note**: *This function will only be effective with a set **Range** and should be used in the editing mode **Damping** where possible.*

Range: The range defines the volume window in which the interpolation is performed around the set **threshold value**. Quieter or louder signal components outside this range 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: If you have selected several objects from different tracks, determine here for which track the calculation should be performed.

> **ⓘ Note**: *To be able to perform a multiple selection, please make sure that the selected objects are of the same length and have the same start time.*

Single, Group, All: With these options you determine whether the calculation should be applied only to the selected single track, to all objects of a group or to all tracks.

Click Marker

Use the **click marker** buttons to mark specific locations with click markers and quickly jump to different markers in the spectrogram.

❶ This allows you to set new click markers. Click markers appear as a "C" in the marker bar of the edit window.

❸ With **jump to the following marker** (keyboard shortcut: **+**) and **jump to the preceding marker** (keyboard shortcut: **-**)you can quickly navigate between the markers in the spectrogram. If you only want to consider click markers, but not existing audio or project markers, additionally hold down the **Ctrl** key while clicking.

❸ Use the button **Apply marking to all click markers** to copy the current selection frame to all positions where there is a click marker. This allows you to use the settings of the current marking on all click marker positions.

❹ This allows you to delete a click marker at the play cursor position.

❺ With the button **Automatically set click markers** you can search for clicks in the entire project and have them marked automatically. Use the slider to set the sensitivity of the search.

> 💡 *The project markers are also displayed on the marker bar of the editing window. You can therefore mark conspicuous places in the project window before you open the Spectral Cleaning dialog. To convert a project marker into a click marker in the spectrogram, click the corresponding marker in the Spectral Cleaning marker bar and click the **Set Click Marker** button.*

Harmonics

This function allows you to remove individual noises, e.g. 50 Hz hum together with their harmonic overtones. To do this, select a single tone in the editing window with as little margin as possible. If the calculation for harmonics is activated, the corresponding harmonics are automatically provided with an editing rectangle above the selected fundamental in the editing window.

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 **Continued harmonics** it is even possible to have the harmonics above the 32nd harmonic removed as well.
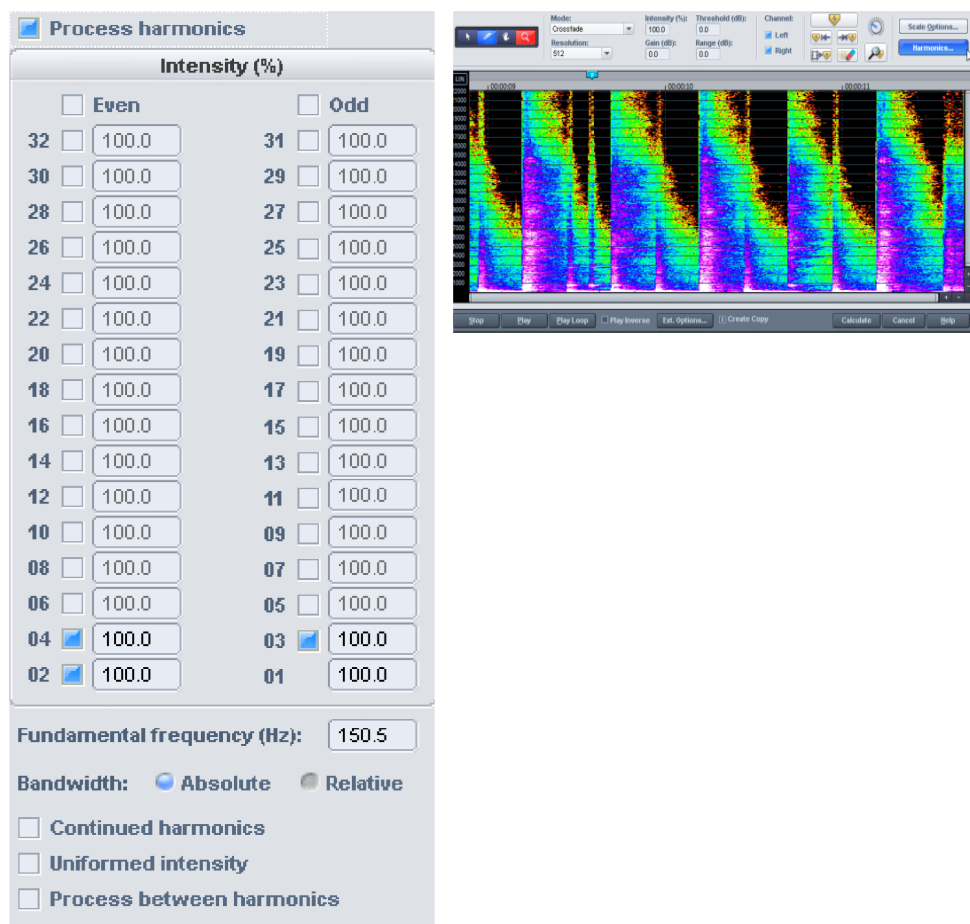
Process harmonics: If active, the harmonics are included in the editing.

Intensity (%): Customized selection of which harmonics are to be observed during calculation. For each harmonic, you can specify the strength to be used. You can adjust the settings for the root (1 harmonic) to up to 32 harmonics. In addition you can use the "Continued harmonics" option to include harmonics in the calculation.

Even/Odd: Selection of even and odd 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.

Bandwidth absolute/relative: With the setting **Absolute** the height, i.e. the frequency range of the harmonics is equal to that of the selected fundamental. At **Relative**, the frequency range increases at the higher frequency harmonics to better capture the acoustically relevant areas.

Continued 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 intensity: This means that the level set for 1 harmonic is also used for all other harmonics.

Process between harmonics: If you activate this option, Spectral Cleaning will treat the areas between harmonics. If you deactivate one or more harmonics, their ranges will be processed up to the next active harmonics.

Scale options

Here you can make settings for the display.

Color scheme: Select a color scheme for display here.

Show Grid shades the display of the grid in the spectrogram.

Frequency scaling: Here you can switch the spectrum display from linear to logarithmic. This is recommend for editing low frequency ranges. This results in the low frequencies being triggered higher up whilst still displaying the entire frequency spectrum. With **Zoom** the influence of the logarithmic display can be adjusted even more precisely to your individual needs.

> ℹ️ **Note:** *Frequency scaling can be switched between linear and logarithmic using a button in the editing window.*

Display resolution: This setting allows you to display the spectrogram with a different resolution, thus achieving either a better temporal or a better spectral resolution for the selection of the contained interferences.

> ℹ️ **Note**: *The calculation is performed independently of the selected display resolution.*

Contrast, Brightness: Adjust contrast and brightness of the spectral display, e.g. to make very quiet audio material more visible.

Playback Section

At the bottom of the editing window are controls for playback control, offline effect calculation, and undo.

- **Stop**: Stops playback (keyboard shortcut: **space**)
- **Play**: Plays from the play marker position (keyboard shortcut: **space**)
- **Solo**: Plays the track to be edited solo (keyboard shortcut: S)
- **Loop**: Repeats playback within a selected range (keyboard shortcut: **L**)

- **Play Inverse**: Plays the difference signal - i.e. the removed sound components (keyboard shortcut: **I**)

- **Mixer**: Opens the mixer (keyboard shortcut: **M**)

- **Ext. options**: Opens the settings dialog for offline effect calculation

- **Create copy**: Creates a copy of the existing file during calculation as a prerequisite for a later undo in the virtual project.

- **Calculate**: Executes the calculation and closes the dialog (keyboard shortcut: **Enter** key)

- With the option **Leave dialog open** you can keep the dialog open after the calculation and make further edits on the same object. In this case you can use the arrow buttons next to Calculate to undo and redo the calculations.

Vocoder
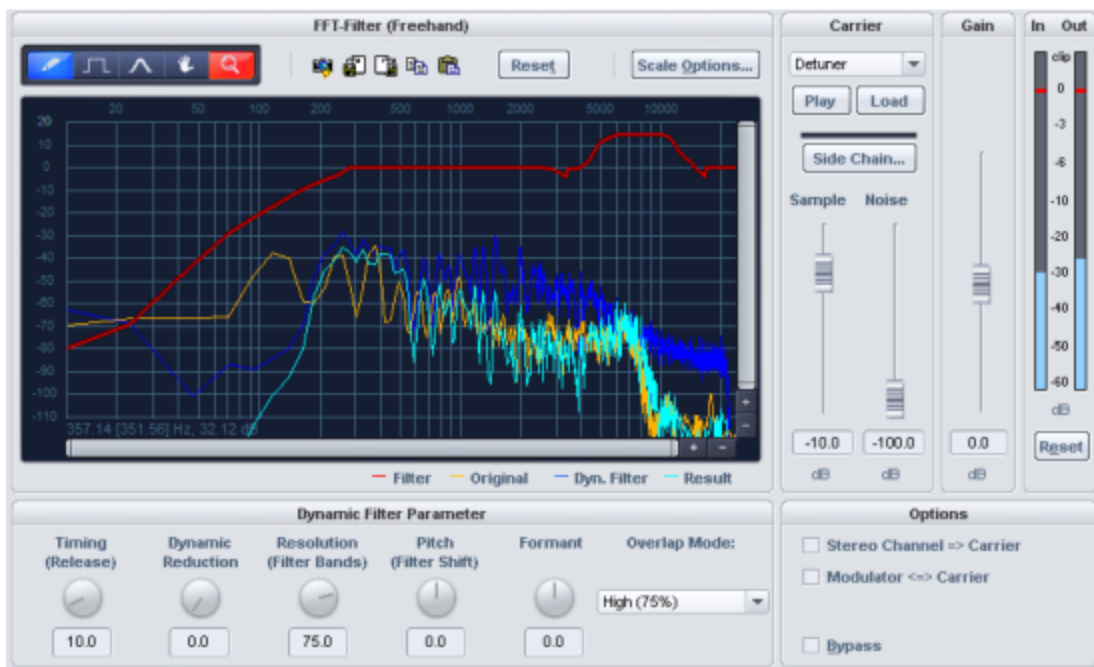
The vocoder works as follows: Carrier material (e.g. a string instrument pad or synth chord) is affected by a modulator (e.g. language or singing) to give the impression that the pad sound is "speaking" or "singing". Additionally, rhythmic pads can be created by modulating an area with a drum loop.

This works by transmitting the modulator's frequency characteristics (language) to the carrier (chord).

A vocoder thus always has two inputs and one output. Since Sequoia effects usually have only one input, the carrier signal is obtained within the effect, continuously mixable from white noise and any wave file. However, it is also possible to use the signal of another track as a carrier via the **sidechain** function. In addition, carrier and modulator can be swapped, using the input signal of the vocoder effect as carrier and the sidechain signal as modulator. Finally,

it is also possible without sidechain by using one stereo channel of the input signal as a carrier and the other as a modulator.

The Vocoder also offers a real-time FFT filter for editing of the Vocoder signal.

Filter Curves

The yellow line corresponds to the frequency response of the modulator, the light blue line represents the carrier signal. The red line is a freely drawable FFT filter (↗274). You can draw in the frequency response to optimize the results of the vocoder. The dark blue line is the resulting filter curve of the vocoder.

Reset: This button resets the red filter curve to its initial state.

Dynamic Filter

Timing (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.

As an alternative to using dynamic reduction directly in the vocoder, it is possible to adjust the signal dynamics of the modulator signal or the resulting signal using 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 resolution.

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.

Carrier signal

Sample selection list: The desired carrier signal can be selected here. All open audio files appear in the selection list, as well as special carrier samples loaded from the "Vocoder" folder. Preferred carrier samples consist of material

with even frequencies e.g. orchestral chords, broad synthesizer sweeps, sound of the wind, etc.

Sidechain: The sidechain option is available for the vocoder when you operate the effect as a track or master effect and the track is not a surround master.

Sample (dB): This fader adjusts the proportion of the carrier sample.

Noise (dB): This slider allows white noise to be mixed into the carrier. This is above all useful if the carrier material can't be modulated well or sounds too uneven. Whispering voices can also be produced in this fashion.

Volume (dB): Adjusts the vocoder output level.

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 is no longer used as the carrier signal, but a channel of the input signal. The other channel will continue acting as a modulator.

Bypass: This allows you to temporarily disable the effect in order to compare the unprocessed signal with the processed signal.

MANAGERS

The managers combine frequently required management and control functions, for example for markers, objects or tracks, in one window.

Managers can be opened within the Docker, but can also be displayed as separate windows. Multiple managers in various different views can be opened simultaneously and independently of one another.

You can open the Docker providing quick access to all managers and other windows via menu **View** > **Manager/Docker**; you can also access individual managers via the **Manager** submenu below.

The following manager windows are available:

Manager	Keyboard shortcut
ClipStore	Ctrl + Shift + Alt + C
File Manager	Ctrl + Shift + B
Object Manager	Ctrl + Shift + O
Track Manager	Ctrl + Shift + S
Marker Manager	Ctrl + Shift + Alt + M
Take Manager	Ctrl + Shift + Alt + T
Source Manager	Ctrl + Shift + Alt + S
VST Instruments Manager	Ctrl + Shift + I
Routing Manager	Ctrl + Shift + Alt + R
Soundpool Manager	
Info manager	

Clip Manager ("Clipstore")
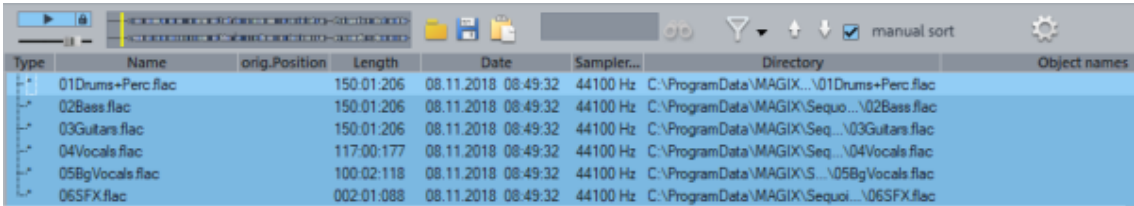
Sequoia contains a Clip Manager in which all audio files of all loaded projects are displayed together with "clips" from these projects in a common list for cross-project use.

A clip is a selection of single or multiple objects or a range selection from a project that can be stored in the Clipstore for further use in the same or other projects.

You can think of it as an extensive functional extension of the clipboard. After you have copied parts of a project to the clipboard (or in Sequoia briefly called "Clip") with Ctrl+C, you can now (but only now!) paste them into the same or another open project. The 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.

> ℹ️ Note: Technically, the clips are small stand-alone project files. They are stored in the `ClipStore` subfolder next to the project from which they originate. As a project, 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.

To display the Clip Manager, choose **View** > **Manager** > **Clipstore** or keyboard shortcut **Ctrl + Shift + Alt + C**. The manager opens either as a separate window or in the Docker (↗57).

Creating Clips

To create clips in the clip list, you can:

- ■ Select single or multiple objects or a range in the project and drag and drop them from the project window into the clipstore or

- ■ Select single or multiple objects, or select a range across one or more tracks in the project, and use menu **Edit** > **Copy** > **Copy to ClipStore** (keyboard shortcut: **Ctrl + Shift + C**) or **Cut to ClipStore** (keyboard shortcut: **Ctrl + Shift + X**).

You can now name the clip. The default is the name of the object. If multiple objects were selected, you can create individual clips for each selected object if the option **Create one clip for each selected object** is enabled, otherwise a clip with multiple objects (folder clip) is inserted. A folder clip is always created from a selected range over several tracks.

> 💡 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.

If a folder is selected in the clip list, the new clips are created in that folder. The clips are saved together with the project and are shown again in the clipstore when the project is reopened.

Importing Audio Files into the Clipstore

If you want to import audio files into your project but are not yet sure exactly how and where you want to use them, you can also use the clipstore as a collection of materials by first collecting the audio files in the clipstore and then loading them into the project from there later. 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 files from clipboard**.

- In the dialog **Load audio file...** in the menu **File** > **Import** activate the option **Load files only to the clipstore**, then the files are not loaded into the project, but only into the clip list. Also if you open an entry from a database connected to **Sequoia** such as DigaSystem using drag & drop, the selected file will only appear in the Clip list, not the current project.

Display of the Clip List

The clip list contains:

- The audio files of all open projects,

- other audio files that have been imported into the clipstore and

- the clips created by copying (↗312) from a project or loaded via clipstore presetSave and load the Clip List as Clipstore Preset (↗318)

A star in the **Type** column in front of the name indicates that the audio file is used in the current project. The label **Clip** in the column **Type** shows that it is a clip.

Colored entries in the clipstore belong to the current project. This means that they are either audio files referenced by the current project or imported into the clipstore, or clips generated from this project. Entries that are not highlighted belong to other projects that are open at the same time or are audio files opened independently of the projects.

These clipstore elements assigned to the active project are saved with the project and are available again in the clipstore the next time you open it. To assign a clip from the list from another open project to the active project, right-click on an entry and select **Assign selected clips to current VIP**. As a clip can only ever be assigned to one project, a copy of the clip is created.

Clips containing multiple objects can be expanded by clicking the **+** icon in front of the clip name or by pressing **Ctrl + Enter** to display the individual objects.

To view the file location of each item in the Clip List, select **Show Clip in File Browser** from the item's context menu. The Sequoia file manager opens and the file is displayed.

Folders can also be created in the clipstore for a better overview. To do so, right-click in the manager and select **Add new folder...** Specify a name for the folder and confirm with **OK**. You can now move clip list entries to this folder using 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 Options

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:
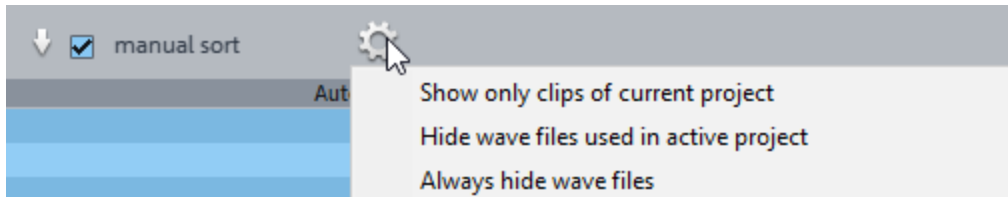
- **Show only clips from the current project**: All entries that are not highlighted in color, i.e. files from projects other than the current project, are hidden.

- **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 project 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. The file manager is opened (if necessary) 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 categorized. 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.

> ℹ️ *If the column **Tags** is not displayed, right-click on the list header and select **Tags**.*

You can now assign the tag to any other list item by selecting an entry 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 **Edit tag list...** context menu item of the clip list, you can create, edit and remove tags in a separate dialog.
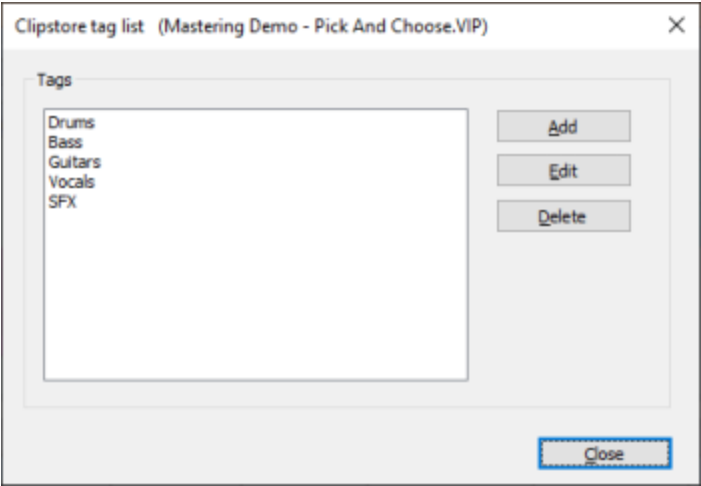
The list can also be sorted by tags by clicking on the **tags** column header.
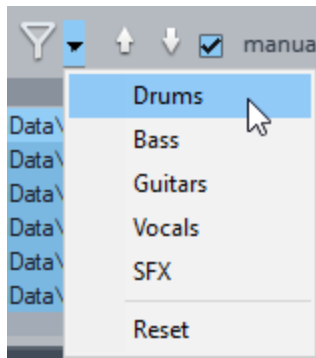
By clicking on the Filter icon in the Clip Manager toolbar, you can filter the list by a specific tag.
To set the tag, click on the small triangle and select the tag from the menu.

Sorting the Clip List

By clicking a property in the list header, you can sort the clips in ascending or descending order by the property you clicked. Any previous manual sorting is lost in the process.
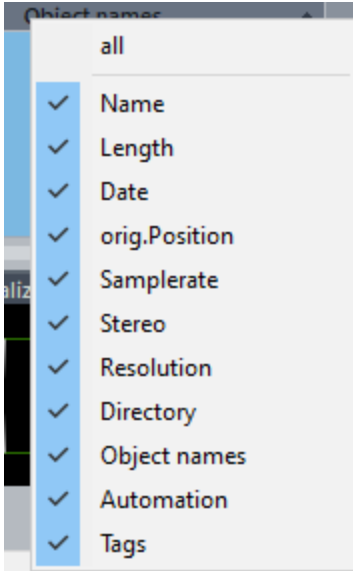
Right-click on the list header to select which properties to display. Besides name, length, date and information about the sample format, there are some special attributes:

- **Original Position**: This is the time position of the first object in a clip that it has in the project from which it was copied.

- **Object names**: The names of the objects contained in the clip are listed.

- **Automation**: Indicates whether a clip contains track or object automation.

- **Tags**: The tags assigned to an entry in the list. (See above).

If the **Manual sorting** option is active in the clipstore toolbar, you can sort the entries in the list manually, i.e. you can move clips to the desired position with the mouse. Or you can use the arrow symbol buttons to move selected entries up or down in the list.

ℹ️ *Moving selected clips with the arrow keys enables, sorting by property in the list header disables the* **Manual sorting** *option.*

Play Clips

A selected entry can be played back using the playback button on the ClipStore toolbar. If the "Auto play" button (lock symbol) is activated, all selected list entries will be immediately played back. This way it is possible to go through the file list using the arrow keys and listen to each file immediately. You can stop playback by clicking on the playback button again.

By clicking in the waveform preview you can jump to a specific location in longer files.

To minimize the loading time at playback start, you can preload all clips in the clipstore using the **Preload All Clips option** in the Options menu to start them without delay. Also the Clip List entries that are not yet part of a project are also opened in the background for this.

ℹ️ **Note***: Playback is not routed through the Mixer or internal effects. The audio device selected under* **Global playback device** *in the* **playback parameters** *(keyboard shortcut:* **P***) is used for playback.*

Insert audio files and clips into projects

To insert clip list elements into a project, you have several options:

- Select entries and drag them from the Clip Manager into the project at the desired position.

- To insert clips at the playback marker position, select them in the clipstore and choose Menu **Edit** > **Insert** > **Insert from Clipstore** (keyboard shortcut: **Ctrl + Shift + V**). Single selected clips can be inserted by double-clicking or by pressing the Enter key (only when the Clip Manager is in focus).

- Right-click on the clip in the Clip Manager > **Load clip into project** or **Insert clip at play cursor position**.

ⓘ *Tip: When inserting multiple entries, you should sort the entries in the Clip List beforehand, because by default the list order determines the order of insertion.*

If a multitrack clip contains more tracks than exist in the project below the track on which it is to be inserted, you can optionally either create additional tracks or insert only as many tracks of the clip as the project can accommodate.

If the clip contains track automations, you can specify when loading whether the automations should be transferred to the track, discarded or merged with existing track automations:

- Select **Replace Automation** to load automation from the clip. An existing automation on the tracks will be overwritten.

- Select **Keep Automation** to not load the automation of the clip. Existing automation is retained.

- Select **Merge Automation** to combine the automation of the clip and the track.

More options for positioning in the project can be found in the **Load Options** in the Settings menu (gear icon on the Clipstore toolbar).

Loading options for audio files and virtual clips
- ⦿ Use Clipstore order
- ◯ Load in alphabetical order
- ◯ Use sync positions (timestamp) of waves or clips

Loading options for audio files only
- ⦿ Load files to the selected track (in a row)
- ◯ Load files to different tracks (one below the other)

- ☐ Load L&R files as stereo
- ☐ Automatically group loaded objects
- ☑ Load multichannel files into a folder track

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 one below the other: The files are now sorted in vertical order from the selected track to the next one. If necessary, an additional track is added.

Automatically load L&R files as stereo: Mono files with the extensions _L/_R can be loaded into a stereo object with this option, see Stereo objectsStereo Objects (↗164).

Automatically 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 track (↗135) with the required number of tracks is created and routed to the surround master (that is recreated 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 selected track and the tracks below it.

Save and load the Clip List as Clipstore Preset

You can save the clip list using 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 manager toolbar.

The `fx preset` folder from Sequoia is used as default when saving and loading. You can also choose a different storage location if you want. The clipstore preset is saved with the extension `*.clp`. It is a text file that you can view and edit with a text editor.

When loading clipstore presets, the current clipstore content is preserved and the content from the loaded clipstore preset is added to the list. The list entries of the clipstore preset are also assigned to the current project and become part of the project after saving. They will be available in the clipstore the next time you load it.

File Manager

Sequoia contains a file manager that can be used to preview files and load them directly into projects. The file manager allows you to create favorite folders and access the recently opened folders.

To display the file manager, select menu **View** > **Manager** > **File Manager**.

Keyboard Shortcut: Ctrl + Shift + B
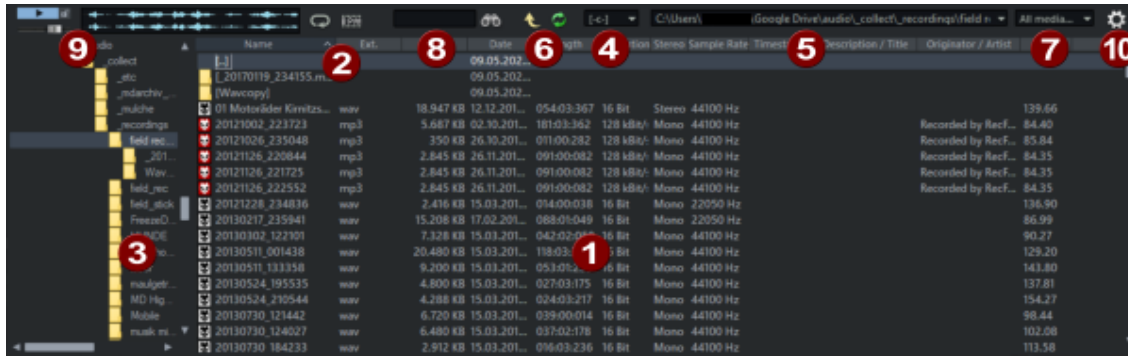
①   The file manager works in a similar way to Windows Explorer. Folder and files are listed with folder and file icons.

②   The list can be sorted by clicking on the respective column header. Another click switches between ascending and descending sorting. In the context menu of the column header you can select the displayed columns.

③   On the right side is the **folder list**, which works just like the one in Windows Explorer. It can be hidden in the file manager options.

④   The **drive selection menu** allows you to quickly switch between all available storage media.

⑤   Use the **Favorites menu** to switch to frequently used folders. It initially contains various default folders of the system, the current project folder and all recently visited folders. With **Add current folder as favorite...** you add the current folder to the list, you can specify a name.

⑥   Use the buttons to switch to the parent folder and re-read the file list.

⑦   The **display filter** allows you to limit the display to certain media types, e.g. only WAV, MIDI or project files. The default setting is "All media".

⑧   The **search field** allows you to search the current folder for folders or files. Enter a search term in the input field and click the binoculars icon or press the Enter key.

⑨   File preview controls

⑩   File Manager Options

Preview Audio Files

A selected audio file can be previewed using the Play button. With activated **Auto-Play** (lock symbol) the audio file will be played as soon as it is selected. This way it is possible to go through the file list using the arrow keys and

listen to each file immediately. You can stop the playback by clicking the Play button again. Use the slider to adjust the volume.
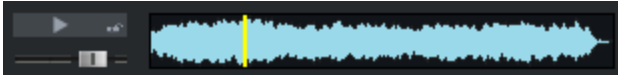
Next to the playback control a small preview of the file will be displayed, in the waveform you can move the preview playback position to any place in the file.

Playback is not performed via the mixer or internal effects, but 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.

Looped Preview: Previewing the selected file happens in a loop.

BPM Sync: This option allows you to preview the selected file in the project tempo. The original BPM is specified in this case by the length of the file. During playback, double-clicking loops inserts them into your project as loop objects at the next bar start.

> ℹ️ **Note**: *To preview certain media data, e.g. in WMA format, you have to convert these files to .wav format. Conversion is carried out automatically in the background.*

Load Files into Projects from the File Manager

To load selected media files from the file manager into a project:

- ■ Drag and drop them to the desired position.

- ■ Right-click on it and select **Load file**. If the File Manager window is selected, you can also press the **Enter** key.

- ■ Double-click on the file. (Not with a multiple selection).

The file(s) will be inserted into the last selected project beginning from the selected track at the play cursor position.

The loading options (↗321), in which you can specify whether the file should be copied to the project folder, how to proceed with multiple files, and others, first appear. Use **Do not show dialog again, settings always apply** to prevent this dialog from being displayed every time.

Insert file into new track: The keyboard shortcut **Alt + Enter** creates a new track into which the selected file is inserted. The track that is created is named according to the file.

Open the file as wave project: To do this, drag and drop the audio file from the list to the tab bar of the project docker or to the title bar of the program window.

File Manager Context Menu

In the context menu of a file in the file list you have access to further file operations. These are on the one hand the common standard actions for folders and files: **Select All**, **Rename**, **Copy**, **Cut**, **Paste** and **Delete**.

ⓘ *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.*

In addition, you con open the files via the default application (**Open in linked program**, show it in the Windows explorer (**Open in Windows Explorer** and copy the path to the files into the clipboard (**Copy path to clipboard**). Also you can display the respective file's **properties**.

Calculate loudness values: You can have the loudness values calculated (↗609), for BWF WAV files the loudness values are written into the file as metadata (compliant with EBU 3344).

Analyze audio: When analyzing audio, Sequoia also calculates loudness values for all selected files and just displays the result in a report window.

File Manager Options

⚙ You can access the options via the cogwheel icon on the far right.

Show directory tree: To hide the folder structure on the left side of the file manager, uncheck this option.

Read format information of files: The file manager lists folder contents and search results faster if you disable this option.

Load options: Here you set the options for loading files.

ⓘ *The options correspond to those in the **Load audio file** (↗658) dialog.*

Extended File Search: Additional criteria for the advanced file search are available here.

Search in subfolders: You can limit the search to files in the current folder by disabling this option.

Load (selected) search results into the clipstore: This option is only available if a search has been performed. You can then load all or the selected list entries into the clipstore (↗311) by selecting this option.

Waveform display: Select here how the waveform (↗721) should be displayed in the preview player.

Object Manager

The Object manager lists all objects contained in the current project and makes it possible to edit selected object parameters.

To show the Object manager, choose **View** > **Manager** > **Object Manager** from the menu or keyboard shortcut **Ctrl + Shift + O**.

❶ The objects are displayed in a tree structure sorted by tracks. Tracks are only listed if they also actually contain objects.

To expand and collapse a track, click the plus symbol in front of the track number. If you drag and drop objects from the list to a position in the arranger, they will be moved to that position. If you hold down the **Ctrl** key while doing this, you will copy the objects

‑💡‑ *In the context menu of a list entry you can find commands to edit objects directly from the Object Manager, to open the Object Editor for this object and to duplicate or delete the object.*

❷ The list can be sorted by clicking on the respective column header. Another click switches between ascending and descending sorting. In the context menu of the column header you can select the displayed columns.

‑💡‑ *If you select **No tree structure** in the Object Manager options, you can sort the list across all tracks, otherwise it will only sort within the respective track.*

❸ The parameters displayed in the respective columns (object name, start time, length,…) can be edited directly in the object manager:

- Double-click on a parameter and enter a new value.

- Numerical values can also be changed by dragging with the mouse, additionally press **Ctrl** for larger value change. The tab key advances to the next editable value.
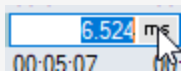
  For time specifications, you can change the unit of measurement used to the right of the number.

- In the **Object name** column you can switch between the text fields with arrow up/down.

④ **Selecting objects**: Select objects with the checkbox in front of the object name. This is also immediately visible in the project window.

With **Shift + Click** all displayed objects are selected.

If an object that is part of a group is selected, the other objects of the group will also be selected in the project window and object manager.

> 💡 *You can recognize grouped objects by the number of the object group in the "Group" column.*

⑤ **Search objects**: Enter a search term in the input field and press Enter or click the binoculars button to search for object or audio file names. The list of objects is filtered by the search term.

⑥ 🖼️ You can export the Object manager information as a text file. The following information will be saved:

- Project name and Project path
- Track and object name
- Start position in the project
- Path of source file

> 💡 *Using the menu under the Options button, you can open the **List Export Options** dialog, where you can customize the saved information.*

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

> ℹ️ *You can also execute the list export via the menu **File** > **Advanced export** > **List export** without having to open the Manager.*

⑦ 🔲🔳 Use the plus/minus buttons on the toolbar to expand and collapse the track object display throughout the window.

⑧ ⚙️ **Options**: In addition to the already mentioned options for the object manager, there are some more options in the menu below the button:

- **Show only objects from active track**: The list will be limited to the objects of the active track.
- **Scroll list during playback**: During playback, the currently playing object is highlighted. If this option is active, the list scrolls during playback to keep this highlighting visible.

Track Manager

The track manager displays all tracks contained in the current project and allows direct access to various track properties such as "Solo", "Mute" and "Record" as well as hiding tracks in the project window or mixer.

To display the Track Manager, select **View** > **Manager** > **Track Manager** from the menu or keyboard shortcut **Ctrl + Shift + S**.
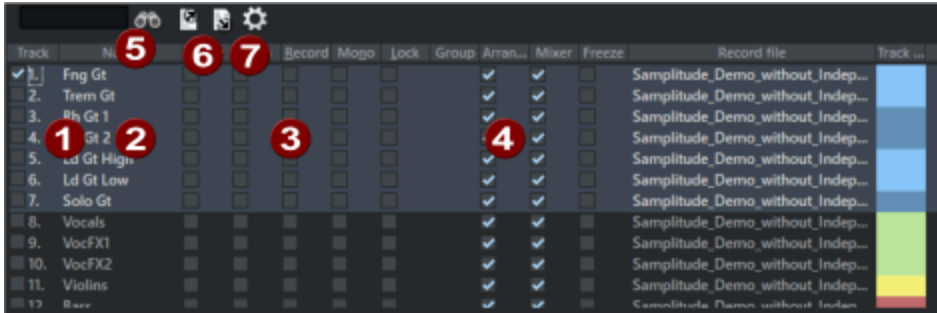
❶	The track manager contains all of the tracks in the current project. To select tracks in the project window, activate the checkbox at the very front or double-click the track number. If the track is outside the visible section, the project window scrolls to this position.
❷	You can rename tracks by double-clicking on the track names and entering a new name. The **Tab** key or **Arrow up/down** switches to the next name.
❸	Various track properties are listed next to a track. To select multiple tracks in the Track Manager to edit these properties (Mute, Solo, Record...) together, click on the track name while holding **Ctrl** key or **Shift** key. 💡 *In this way, you can also select multiple tracks in the project window by first selecting them in the Track Manager and then clicking the checkbox at the very front.* **Reorder tracks in the track manager**: Grab them with the mouse and drag them vertically to the desired position. **Delete tracks**: Select one or more tracks and press the **Del** key. **Insert new tracks**: Right-click a track entry and select **Insert New Track**. You can also add or insert folder tracks, submix buses, AUX buses, or surround buses. The **Track Options** (↗141) are also made available by right-clicking.

④ **Show and hide tracks**: In the column **Arrangement** or **Mixer**, deactivate the checkbox to hide a track in the Project window or Mixer.

> ℹ *Hidden tracks will continue to play and objects contained on them will continue to be processed (e.g. moved) if they are part of a group that also contains objects on non-hidden tracks.*

⑤ **Search for tracks**: Enter a track name or part of a track name in the input field and press Enter or click on the binoculars to search for tracks. The tracks found are selected and can be worked on together.

⑥ **Export text file**: Click the **Export text** button and Sequoia will create a text file with a listing of the tracks used in your project. With the button next to it you can import a track list from a text file.

> ℹ *You can also execute the list export via the menu **File** > **Advanced export** > **List export** without having to open the Manager.*

⑦ ⚙ **Options**:

- **Hide sub-tracks of folder tracks/surround folder tracks**: Tracks contained in Folder tracks/Surround folder tracks are not displayed in the list when you enable these options.

Marker manager

The Marker manager lists all markers contained in the current project and allows you to jump to or play them directly from the list.

To display the Marker Manager, choose **View** > **Manager** > **Marker Manager** from the menu or use the keyboard shortcut **Ctrl + Shift + Alt + M**.
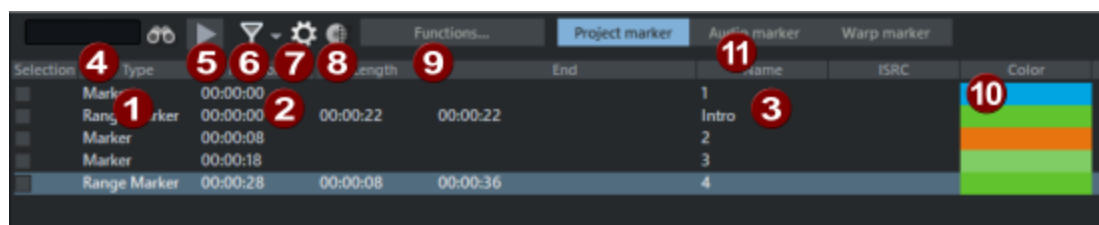
❶ In the column **Type** all markers of the project are listed. Use the checkbox in the **Selection** column to select markers to move them, color them or delete them together. Double-clicking the checkbox moves the play cursor to the marker position.

In the context menu of a marker in the list, you can redefine selected markers as CD Track Index, CD Subindex, CD Pause, CD End, or Position markers, insert new markers, rename marker names, edit marker positions (numerically enter absolute position), move markers (numerically enter a relative position change), and delete markers.

If you select two markers, you can use the context menu to set a **Range between selected markers**.

❷ In the **Position** column, you can edit the marker position directly in the Marker Manager:

- ■ Double-click on a parameter and enter a new value.

- ■ Numerical values can also be changed by dragging with the mouse, additionally press **Ctrl** for larger value changes. Pressing the Tab key moves the cursor to the next editable value.
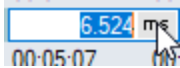
   For time specifications, you can change the unit of measurement used to the right of the number.

For range markers, you can edit the end of the range in the column **End**.

❸ **Marker name**: To rename a marker, double-click it and enter a new name. You can switch between the text fields with the arrow up/down keys.

❹ **Search for markers**: Enter a marker name or part of a marker name in the input field and press Enter or click on the binoculars to search for markers. The list of markers is filtered by the search term.

❺ **Play marker**: Select a marker and click on this button . Playback is started 2 seconds before the marker, the standard lead time, see **Program settings** > **Playback** (↗702), and stopped 2 seconds after the marker. If loop playback is active, the time period around the marker is looped. No playback area must be selected in the project for this to work.

❻ **Filter marker view**: To display only certain marker types, you can filter the view with this button. You can determine which types are to be displayed in the menu under the arrow on the right of the button. All marker types in the menu that are not selected are filtered out.

⑦ ![gear icon] **Options**

- **Scroll list during playback**: During playback, the marker that the play cursor has passed is highlighted. If this option is active, the list scrolls during playback to keep this highlighting visible.
- **List export options**: This allows you to customize the information that is stored during marker list export (under **Functions**).

⑧ This button opens the CD-Index-Manager (↗581).

⑨ **Functions**: Besides the self-explanatory functions **Insert new marker**, **Delete marker** and **Delete all markers** there are two more functions:

- **Export marker**: You can export Marker manager information as a text file. To do this, click 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 track indexes

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

  ℹ️ *You can also execute the list export via the menu* **File** *>* **Advanced export** *>* **List export** *without having to open the Manager.*

- **Copy selected markers into selected object(s)**: This allows you to transfer markers from the project to the audio file of an object. This corresponds to the function Copy VIP markers to audio markers (↗117), with the difference that here you can select the markers to be transferred. Select the markers to be transferred and the objects under the markers into which the markers are to be transferred before you execute the function.

⑩ **Color**: Click on the color field to select another marker color.
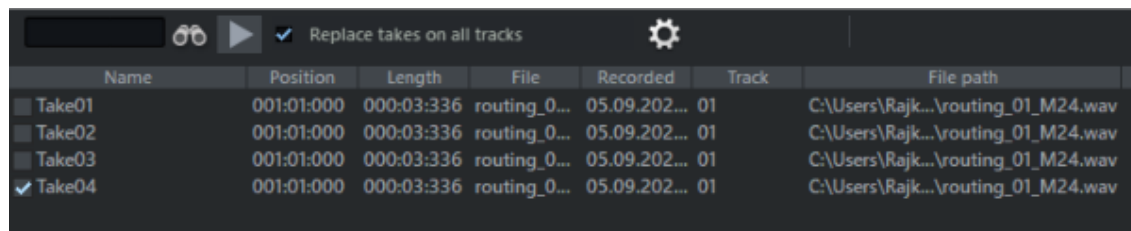
⑪ **Project markers/audio markers/warp markers**: Switch the display between displaying the project markers and the Audio markers (↗116) or Warp markers (↗366)

Take Manager

The take manager offers you a convenient option for selecting and organizing recording takes, loop recordings (↗102), and offline object edits (↗168).

Menu: **View** > **Manager** > **Take Manager**

Keyboard Shortcut: Ctrl + Shift + Alt + T

Name	Position	Length	File	Recorded	Track	File path
Take01	001:01:000	000:03:336	routing_0...	05.09.202...	01	C:\Users\Rajk...\routing_01_M24.wav
Take02	001:01:000	000:03:336	routing_0...	05.09.202...	01	C:\Users\Rajk...\routing_01_M24.wav
Take03	001:01:000	000:03:336	routing_0...	05.09.202...	01	C:\Users\Rajk...\routing_01_M24.wav
✔ Take04	001:01:000	000:03:336	routing_0...	05.09.202...	01	C:\Users\Rajk...\routing_01_M24.wav

When graphically comping with the Take Lanes (↗138), you cut together the perfect take from different take sections.

> ℹ️ **Note**: *Also after offline effect calculations, a new take is created and displayed in the Take Manager. To do this, the option **Create copy** must be active in the offline effect dialog.*

Take Manager - Basics

Sequoia stores in each recorded file not only audio and MIDI data, but also additional information about at which time position and in which track in the arranger the file was recorded (original position). This data is stored in the audio file and in the MIDI object as take information. When working with multitrack recordings the takes also contain information like, for example, which other tracks were involved in the recording. This allows all recorded material to be assigned to a time period and thus identified as a take of a particular recording.

So you can record multiple takes on the same tracks at the same time position on top of each other. The existing data is not overwritten, but, depending on the Recording options (↗699), the new recordings are appended to the recording file or are saved in a new file.

Now, when you select an object and open the Take Manager, all the audio in the project is searched for takes from the same track and with the same original position, and the matches are displayed as takes.

> ℹ️ **Note**: *The Take Manager lists only objects created by recording and not objects created by importing audio files.*

Selecting Takes

In the Take Manager, takes matching the selected object are displayed in a list. With default settings, this refers to all the takes from the same track and original position. The current object take can be identified by the checkmark preceding the name in the take list.

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.
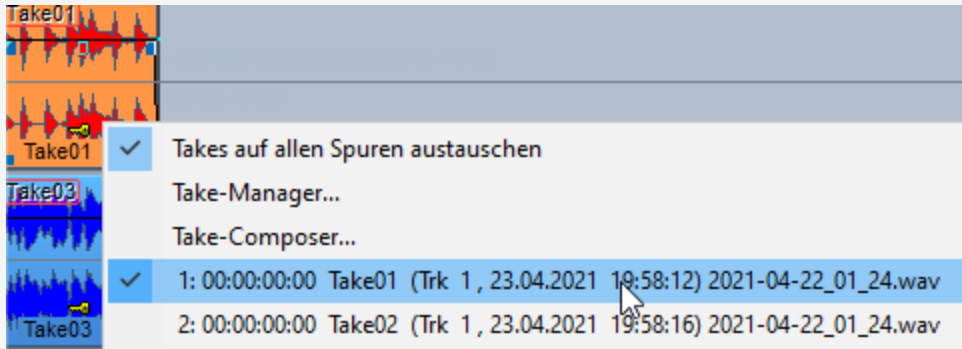
Rename take: Using the context menu of the take, you can rename it and edit the recording position. Multiple selection is also possible here.

Edit recording position: This option enables you to make changes in the position field of the take manager.

Delete Take: You can use the context menu of the take to delete it in the Take Manager. Multiple selection is also possible here. When you have deleted all takes except the last one, you can decide by query whether you want to delete also the referenced file in addition to the last take. However, make sure that the audio file you are about to delete is not used in other projects.

ℹ️ **Note**: If the audio file is no longer referenced by objects but contains multiple takes, you don't have to delete each take first to remove the file from disk entirely. Instead, use the **Remove Unused Samples** (↗684) function from the **File > Clean up & Backup** menu.

ℹ️ **Note**: For a better overview, convenient take naming options are already available during recording in the recording window (↗95).

Take Manager Options

The options can be used to filter the take list.

Filter record position: Only those takes are displayed that have the same original time position as the selected object. This option should be disabled only in exceptional cases.

Filter too short takes: Only takes that are at least as long as the currently selected take are shown. This allows you to filter out recording takes that were canceled earlier due to errors.

Show all multi-track takes: Takes of all tracks will be shown. This option should only be activated in exceptional cases, e.g. to find "lost" takes.

Takes in Multitrack Recordings

In multitrack recordings, each recording creates takes on each recording track that are classified as belonging together. Simultaneously recorded objects are grouped and also given the same take name.
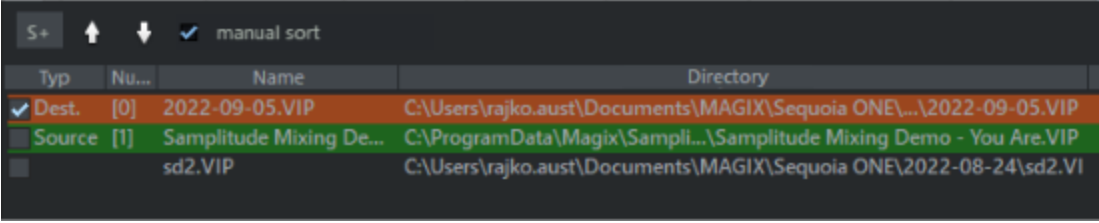
Replace takes on all tracks: If this option is enabled, you replace all takes of a multitrack recording in one go when you replace the take in one object of a track.

> ℹ **Note**: *Since the "Take Manager" display always refers to the last object clicked, you can change the reference track by selecting an object on another track.*

Source Manager

The Source 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** > **Source manager...**.

Keyboard Shortcut: **Ctrl + Shift + Alt + S**

To add a project to the source list, click the **S+** button ("Add active source project to list") in the source list toolbar. However, the source list only shows an entry if there is at least one project with a destination track 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.

> ℹ **Tip**: *You can remove projects from the source list and close them simultaneously by selecting the project in the list and pressing the **Del** key. A confirmation request prevents deleting a project that has not yet been saved.*

> ℹ *For more information, see the chapter* Source-destination cut (↗201).

VST Instrument 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 shortcut: **Ctrl + Shift + I**
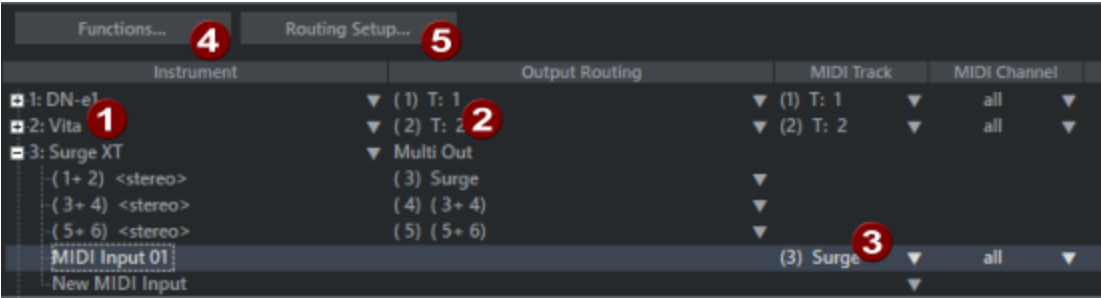
❶	**Instrument**: All software instruments (VST/Rewire) present in the project are listed. Right-clicking on an instrument opens the instrument's plug-in dialog. Press the **Del** key to remove a selected instrument from the project. To display the individual outputs and MIDI inputs with their assigned tracks, click the plus sign next to the instrument.
❷	**Output Routing**: By right-clicking on an output or clicking on the triangle, you can select a track to be supplied with the respective individual output. The assigned output then appears in the plug-in slot of the target track. Select an output channel and press the **Del** key to cancel the routing to a track.
❸	**MIDI track/channel**: These columns list the tracks for the instrument's MIDI input and the channels used. By right-clicking on an entry at MIDI Track or clicking on the triangle, you can select a track whose MIDI output is used by the instrument. **Insert MIDI track** creates a new track in MIDI recording mode and routes its MIDI output to the instrument. **Remove MIDI input** removes a MIDI routing to the instrument. The MIDI channel for a MIDI track can be set under **MIDI channel**.
❹	With **Functions** you can access the Freeze and Unfreeze (↗141) functions for the loaded VST instruments. Here you can also create new tracks for individual outputs and delete or reset selected instruments/outputs. With **Rename VSTi** you can rename a VSTi instance. If you use multiple instances of the same VSTi, this will help you distinguish them in the mixer and project window.
❺	Opens the Routing settings for multi channel software instruments (↗372).

Routing Manager

The Routing manager provides you with a matrix display of the inputs, outputs, AUX sends and VCA groups of all tracks.

Menu: View > Manager > Routing Manager

Keyboard Shortcut: Ctrl + Alt + Shift + R

The Routing Manager is divided into four sections, Input, Output, AUX and VCA for assigning hardware inputs, outputs (hardware, submix busses, master), AUX sends and assigning VCA groups (↗490) in the mixer.

Input

In the rows all tracks are displayed, in the columns the available inputs.

Assign the available inputs to the tracks of your project by simply clicking in the matrix fields. Another click cancels an existing routing.

Output

In the rows all tracks are displayed, in the columns the hardware outputs, followed by the tracks and the master. In the above illustration, all tracks are routed to the master.

Assign the available outputs (hardware outputs, masters, or busses) to the tracks of your project by clicking in the matrix fields. Another click cancels an existing routing.

By clicking the corresponding matrix field behind the available outputs (in the Tracks section) you route the output of the track to another track. In doing so, you convert this track into a submix bus.

In the context menu of a matrix field, you can define the selected output as "Direct Out" (controller display red), "Pre-Fader-Out" (controller display yellow) or as "Post-Fader-Out" (controller display orange). There you can also open the Stereo panorama dialog () for panorama setting of the respective output.

Assigning Multiple Consecutive Tracks to the Same Input or Output:

1. Click in the matrix field for the respective input or output of the first track you want to assign.
2. With the **Shift** key held down, click the matrix field for the same input or output of the last track you want to assign.

This assigns these tracks and all tracks in between to the same input or output.

Assigning Multiple Consecutive Tracks to Consecutive Inputs/Outputs (Diagonally):

1. Click in the matrix field for the respective input or output of the first track you want to assign.
2. With the Shift key held down, following the diagonal in the matrix, click the matrix field for the corresponding input or output of the last track you want to assign.

This assigns these tracks and all tracks in between to corresponding successive inputs or outputs.

AUX

Similarly, you can also route the individual tracks to AUX buses (↗498). Clicking a field sets the AUX send of the track to 0dB. By default, the AUX Send is created as "Post". If routing is to be made to a track that is not an AUX bus, you can optionally have the track converted to an AUX bus.

> ℹ️ *Since the purpose of AUX busses is to send variable signal components, the manager is primarily used here to provide an overview of which tracks have sends to the AUX busses at all.*

In the context menu of a field you can switch the selected AUX send to "Pre-Fader Send" (field highlighted in yellow) or "Direct Out" (field highlighted in red).

In the AUX view, existing sidechain sends (↗378) are also displayed (**>** symbol). In the context menu you can open the Stereo panorama (↗492) dialog for the respective AUX bus.

VCA

Here you can select each track as master of a VCA Group (↗490) and assign it as many tracks as you like.

In the above example, track 3 is set as the master of a 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". Volume faders 2...4 are now controlled by the VCA group in track 3. You can recognize this by the additional "ghost" fader knobs above the faders.

Soundpool Manager

In the Soundpool Manager you can integrate you Soundpool loop content in a clear form. You can access the Soundpools via the clearly laid out database view, which lets you arrange the display of the loops according to styles, instruments and pitch.

A Soundpool consists of one or more styles. Styles are sound libraries that belong together and cover a certain musical style. The sounds (sample or MIDI loops) of one style all have a certain tempo. You can mix loops from different styles, and the tempos will be adjusted accordingly. Within a style, loops are ordered according to instruments, and one instrument folder contains different sounds. Each sound can have a different pitch (except for drums and effects sounds).

❶ The Soundpool display consists of several lists: First, all styles available in the database are shown. The second list contains the instruments. The third list, "Name", contains the list of the sounds found. The respective name, tempo, length in bars (1, 2, or 4 beats), and type is listed for them. The different pitches are displayed above that (if available).

The list of samples found is created after entry selection in the first two lists. With **Ctrl** + click you can reduce or expand a selection. No selection (**Ctrl** + click on a single selected element) shows all entries from this category.

> 💡 *If you select an instrument, e.g. "Drums" and "Percussion" and no style, then all drums and percussion samples in the whole database will be displayed.*

❷ Loops can be loaded by double clicking, drag & drop, or by double clicking the corresponding pitch. The objects will be inserted directly behind one another so that complete accompanying tracks can be compiled quickly.

❸ **Full text search**: Above right in the search field you can search the list of samples found for a specific sound file name.

❹ ⚙ Options:

- **Add styles**: This option allows you to add new MAGIX Sound Pools. To do this, select the folder or disk where the sound pools are stored.

- Normally, loops can be previewed just by clicking them, even during project playback. If **Automatic playback** is deactivated, then use the playback button in the manager ❺ .

- The option **Hide instruments without available loops** completely hides instrument groups in the Soundpool manager for which there are no loops present in a certain style (instead of being grayed out).

- **Reset Soundpool** will reset the Soundpool database so you can re-import the Soundpools later.

- Every detected Soundpool (on CD/DVD or on the hard drive) is added to the database and displayed there, even if the corresponding medium is currently not in the drive or if the Soundpool has been deleted or moved. The option **Clean up Soundpool** removes these entries from the database.

❻ **Type**: There are audio and MIDI loops. Audio loops (.wav or .ogg) are normal (looped) audio objects that can be used anywhere in the project. MIDI loops (MIDI objects) contain MIDI data + the controlled synthesizer (Vita or Revolta). This is automatically loaded to the track where you place the MIDI loop. This means that you will need to use a new track for each additional MIDI loop which controls a different sound. However, the same sound in another pitch is possible.

Info manager

In the Info Manager, you can add text comments for the selected object, the selected track or the entire project.
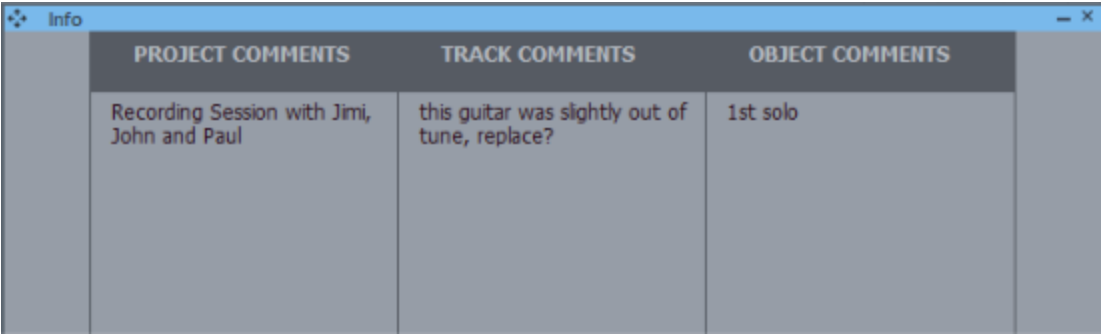
The project comments are also displayed in the Project information (↗648). They are also saved in the project folder in a separate text file in ASCII format (*.asc) so that they can also be accessed without opening the project. The track comments correspond to the comment field at the bottom of the Track Editor, the comments of an object correspond to those in the field on the right in the Object Editor.

AUTOMATION

When mixer settings such as volume, pan or effect settings are to change during the course of the project, for example when fading in tracks, moving audio signals in the panorama or changing parameters of effects, we speak of automation.

The display and recording of this automation is done using automation curves. An unlimited number of automation curves are available on each track. You can record automation movements on track and master level during playback in automation curves. All automation curves (track, master and object automation) can be edited or directly drawn in.

- At track level, dynamic automation of volume, pan, surround, AUX sends, effects and MIDI controllers is available.

- At the object level, you can automate the volume, (surround) panorama, AUX sends and fx parameters.

- At master level, you can automate the volume, EQs, VST plug-ins, and MIDI controllers.

Creating Automation Curves

The most important automation curves volume and pan can be activated for the selected track with the **vol** and **pan** buttons at the track header (↗63) or via keyboard shortcut:

Volume: Alt + K

Pan: Alt + P

If other parameters are to be automated, you can set the parameter either

- in the Automation context menu or menu Automation or

- in the track head of an Automation Lane or

- via keyboard shortcut **Ctrl** + **Alt** + Control element in the Mixer or the interface of a plug-in or

- in the VST parameters dialog.

Automation Context Menu or Menu Automation

In the **Automation** menu or Automation context menu (click on the parameter field over the Automation slider at the very bottom of the **Main** section of the **Track Editor**) you can select an automation parameter.

> ℹ️ *The Automation menu and the Automation context menu contain the same entries, so in the following we will refer to this as **Automation menu** in general.*

At the bottom of the menu, depending on the loaded plug-ins and effects, all available parameters are listed in submenus. Select the parameter you want.

All existing active or inactive automation curves are listed at the top of the menu. There you can select a curve that you can control during playback using the automation slider in the track editor. Also all commands in submenu **Edit selected curve...** refer to this curve.

The curve of the selected parameter is displayed in a new **Automation Lane**. The curves can also be displayed directly in the track, for details see Display of automation curves in the project (↗341).

Ctrl + Alt + Control Element

This is the fastest and smoothest way to automate a plug-in parameter. In the mixer or track editor, click a control (volume, pan, aux sends, EQ) or in the interface of a plug-in, click the control of a parameter while holding down the **Ctrl** and **Alt** keys. In the stopped state, a corresponding curve is only created; during playback, a change in the parameter is recorded directly, regardless of the selected automation mode (↗342).

> ℹ️ *Note: This works only with tracks, and not for the automation of* object effects (↗347).

With plug-ins, this function can also be carried out by first selecting the menu item **Automate next parameter** in the menu **Plug-in** of the VST plug-in dialog (↗375) and then moving a control element.

Automating the Master Track

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. To do this, in the context menu of the automation button of the master in the mixer window, select **Show master track in arranger**. An additional track called "Master" is now displayed in the arranger.
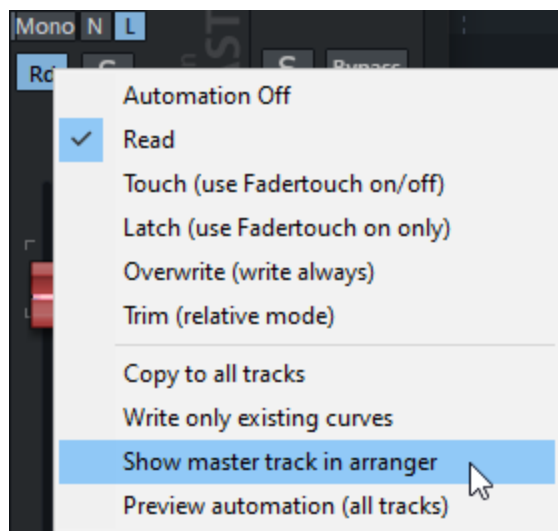
You can also make the master track visible in the project window via the track manager (↗324): To do this, activate the track **Master** in the column **Arrangement** at the very bottom.

Display of the Automation Curves in the Project

There are different options in which form and to which extent automation curves are displayed in the project. The automations can be displayed either individually in separate **Automation Lanes** or above the objects in the track.

Display in Automation Lanes

Click the symbol at the bottom left of the track header or select **Expand Automation Lanes** in the Automation menu (keyboard shortcut: **Ctrl + Shift + Enter**) to expand and collapse the automation lanes.

ⓘ *Also if an automation curve has been created using the methods described above, the automation lanes are expanded.*

If there are no automation curves, an empty Automation Lane is displayed.
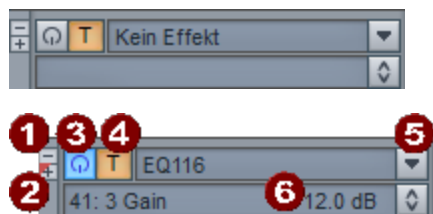
Automation Lane track head

❶ Click the **-** button to hide the Automation Lane. The curve remains active.

❷ Click on the **+** button to create a new lane below the last automation lane.

❸ **Automation on/off**: Used to activate/deactivate the respective curve.

❹ **Automation mode**: Automation mode of the track

❺ **Parameter Menu**: Click on the triangular button on the right to open a menu where in the upper section, you can choose an automation curve to display in the automation lane. In the lower section of the menu, you can also select any other parameter to create a new automation curve that will be displayed in the lane.

❻ **Parameter field**: The current parameter value at the play cursor position will be displayed here. In stopped state, this value can be modified by vertically dragging the button on the right. You can also edit the curve numerically using this method.

Display in the Tracks

Use the button **Track Auto** ⁄**Lanes + Obj. Auto** in the **Track Editor** (➚59) to switch the display of the automation curves in the track:

- In **Track Auto** mode, track automations are displayed in the track over the objects, object automations are hidden.

- In the **Lanes + Obj. Auto** mode, the object automations are displayed in the track over the objects, the track automations are displayed in lanes (➚340).

When you collapse the Automation Lanes in the **Track Auto** mode, the automation curves are displayed directly above the objects in the track.

Since it can quickly become confusing with many automation curves in the track, there are various options for how the automation curves are displayed in the track. Choose from the following options in the Automation menu:

- **Show Only Selected Curve**. Only the selected curve is displayed.
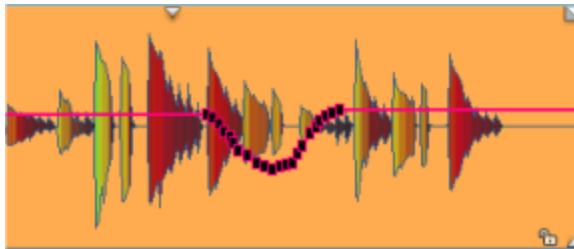
- **Show all curves (not selectable)**. All automation curves are displayed. The unselected automation curves are displayed paler and cannot be selected with the mouse.
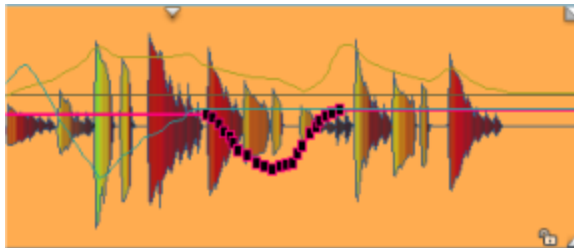
- **Show all curves (selectable)**. Again, the unselected automation curves are displayed somewhat paler, but can be selected by clicking on them with the mouse in Universal Mode, (Object) Curve Mode, and Draw

Automation Mode.
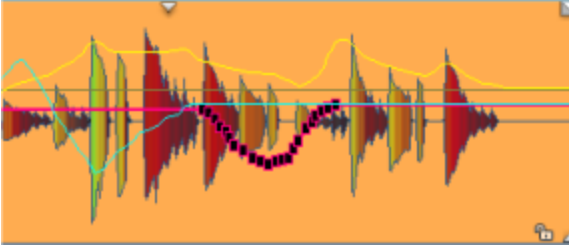
When displaying the automation curves in lanes, you can use the submenu **Show lanes** to restrict the display of the automation lanes to certain types of automation: **Volume**, **Panorama**, **AUX Sends**, **EQ** or **Plug-in parameters**. **Show lanes for all curves** resets this filter.

To hide the automation curves completely, select **Hide Automation** in the Automation menu. The automations remain active.

Deactivate Automation Curves

Use the On/Off button in the track head of the automation lanes or **Edit selected curve** > **Inactive** in the Automation menu to deactivate an automation curve without deleting it, to temporarily bypass an automation. The automation curve is shown as a dashed line in Track and Automation Lane.
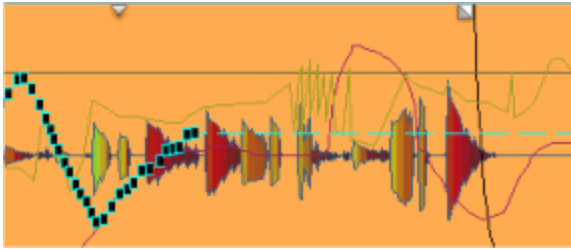
> 🔅 *To deactivate all automations of a track , select the automation mode **Off***. *To disable all automation in all tracks, select the automation mode **Off** in the* **Automation Panel** *(↗345).*

Automation Modes

The automation modes define how automation data should be saved or displayed. You can set the automation mode for each track. The **Automation Panel** can be used to set the automation modes for all tracks simultaneously.

You set the automation mode:

- In the automation menu at **Automation mode**
- By right-clicking on the automation button in the mixer channel or track editor. A left click on the button

switches between the last activated mode and the Read mode

- In the Automation Panel (↗345)

Off: In Off mode, all automation functions are disabled.

Read: In Read mode, all recorded automations are played back. This is the default automation mode of a track.

Touch (use Fadertouch on/off): In Touch mode, automations are recorded. Recording starts as soon as you touch a control element of the track with the mouse or from your external controller. As soon as you release the control, the automation recording stops. The parameter moves back to the value previously recorded at this point. You can set the duration of the return time in the **Program settings** (keyboard shortcut: **Y**) at **Effects > Automation > Automation Release Time**.

Latch (use Fadertouch on only): Also in Latch mode, automation data is recorded from the first touch of the control. However, if you release the control, the automation will continue with the last value until you stop playback or switch to another mode.

Overwrite mode (write always): In Overwrite mode, automation data is recorded as soon as playback starts, regardless of whether you touch the control or not. Automation continues until you stop playback or switch to another mode.

> ⓘ *Even if other curves are active, only the selected curve is recorded immediately after playback starts, as it does not make sense to overwrite all automations at once. For the other active curves, the mode works like the latch mode, the automation curve is overwritten as soon as the parameter is changed.*

Trim (relative adjustment): Trim mode is a special Touch mode that can only be used for **volume automation**.

Trim mode allows you to edit existing volume automations relatively. Use Trim mode when you want to increase or decrease the volume of a track without changing any existing automation.

The volume knob is no longer used for direct volume control in Trim mode, but to increase or decrease the current volume. It is therefore always initially set to 0 dB and does not move with the current automation data during playback as it would otherwise.

Drag it during playback to increase or decrease the volume of the existing automation. The difference value in dB is displayed on the fader. After release it jumps back to 0dB.

Trim mode can also be used when playback is stopped. All points of the automation curve are changed simultaneously then. If an range is selected, the change will only be applied within the range. This inevitably results in value jumps at the range edges.

> ⓘ **Note**: *For other automation parameters, Trim mode behaves like Touch mode.*

The **Copy to all tracks** menu option sets the automation mode of the last selected track globally for all tracks.

Write only existing curves: This menu option prevents a new automation curve from being created in tracks in Touch, Latch or Overwrite mode for each parameter change.

This is especially helpful if you want to change effect parameters while recording automations, but do not want automation curves to be created for these changes.

> 💡 *In the* Automation Panel *(↗345) the curves to be recorded can be filtered by volume, pan, aux sends and plug-ins.*

Preview Automation (All Tracks)

Suppose 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 an already automated parameter because it is controlled by the automation curve. However, if a writing automation mode (Touch, Latch or Overwrite) is active, when you adjust a parameter such as the volume, for example, the value is written to the curve. But this also creates a gradual transition between the last old curve point and the new value and the existing curve is changed, because this parameter value should actually remain constant until the end of the old section. Furthermore, if you change parameters that have not been automated so far, new curves will be created, which means these parameters will receive the new value also in the previous section of the project and thus be changed.

In such a case, you can use **Automation preview** in a writing automation mode to temporarily override the linking of the track parameters to the curves in order to cleanly separate the two parts with automations. To do this, proceed as follows:

1. Activate a writing automation mode (Touch, Latch, Overwrite)
2. Enable the **Preview automation** option via the Automation menu or the **PREVIEW** button in the Automation panel (↗345).
3. Start playback. You can now change all settings and effect parameters in the tracks without anything being recorded. If parameters are already automated, their automation curves are not changed. However, if you change a parameter that was not previously automated, a new automation curve is created. A point is created on this curve at the time position at which the change took place, which retains the old value from the previous section.
4. Stop playback, the play cursor jumps back to the beginning of the new section.
5. Then select **Set parameter value(s)** > **Jump to current value** in the automation menu or use the button in the Automation Panel (↗345).
6. This ensures that a subsequent recording of automation data does not affect any parameters in the previous section.
7. You can now deactivate the option **Preview automation** and continue working with the automation modes as normal.

Automation Recording in Read Mode

Another way to insert new automations without affecting the parameters in the rest of the project through the curves is to record in **Read mode**.

Actually, automated parameters cannot be changed in Read mode. But if you hold down the key combination **Ctrl + Alt** during playback while changing a control of the mixer or an effect, this movement will be recorded. As long this key combination is held down, you can also automate the selected parameters in Read Mode.

When stopped, clicking a Mixer or FX control with the same Ctrl + Alt key combination creates an inactive curve, which you can then edit further in "Draw Automation" mouse mode.

Set Parameter Value(s)

This command from the Automation menu creates new automation points in all active and inactive automation curves of the selected or all tracks according to the currently set parameters. This allows, in connection with the option **Automation Preview** (↗344), 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 commands correspond to the buttons in the Automation Panel (↗345) at **MANUAL WRITE** and are applied to the automation curves according to the filter options selected there.

- **Jump to current value**: New points are inserted as jumps in all automation curves.
- **Glide to current value**: New points are inserted in all automation curves as a transition to the previous value.
- **Write to start/Write to end**: The current value of the automated parameters is continued to the project start or end. In other words, an automation point is created at the play cursor position and all other automation points before or after it are removed.

Shortcut: **Ctrl + Alt + #**

A popup menu appears at the mouse pointer, select the corresponding option with another keystroke (underlined letter of the option).

> 💡 *Tip: These functions can be controlled more flexibly in the Automation Panel!*

Automation Panel

The Automation Panel allows you to edit the automation modes of all tracks together. It is also possible to write automation values manually.

The aim of this window is to be able to switch as elegantly as possible between the mere adjusting of parameters and automating them.

⚠️ *Note: To understand this dialog, we strongly recommend that you also read the sections* Automation Modes *(↗342)* *and* Preview Automation (All Tracks) *(↗344)!*
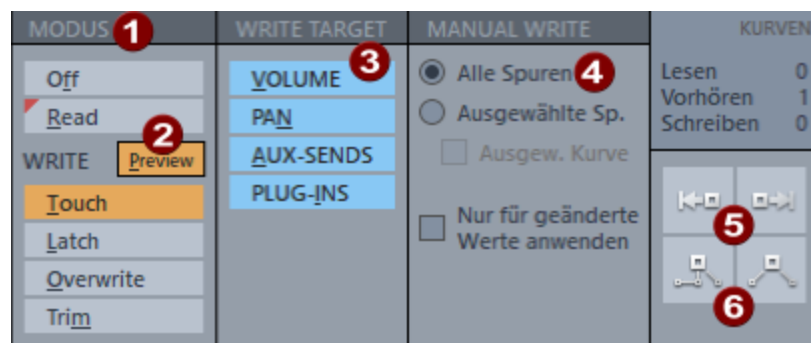
① **MODE**: These buttons allow you to switch automation modes for all tracks together. Regardless of this, it is still possible to select a different automation mode for individual tracks, for example, so as not to accidentally overwrite them. In this case, several buttons are active and the number of tracks in the respective mode is displayed on the button.

② **PREVIEW** : Activates the mode **Automation preview** where parameters can be changed but no recording takes place. Use this mode also to write automation values manually.

③ **PARAMETERS:** These filters allow you to disable automation writing for certain types of parameters (volume, pan, AUX sends, plug-in parameters). This allows you to protect their automation curves in tracks from changes, even though these tracks are in a writing automation mode or when values are written manually (see below). The parameters disabled at **Parameters** are controlled by their curves in Read mode, regardless of the selected automation mode.

④ In the area **MANUAL WRITE** it is possible to explicitly write values into the automation curves without having to edit them with the mouse or record an automation. The following filter options can be used to limit the selection of parameters / curves for which manual writing is to be performed.

- **All Tracks/Selected Tracks/Selected Curve**: Depending on the selection, the functions **Write to Start/End** and **Jump/Glide to Current Value** create new values in all tracks, only in the selected tracks or only in the selected curve.

- **Only apply to changed values**: If this option is active, automation points are only created in the curves where the current parameter value is different from the curve during manual writing.

⑤ **Write to start/Write to end**: The current value of the automated parameters is continued to the project start or end. In other words, an automation point is created at the play cursor position and all other automation points before or after it are removed.

⑥ **Jump/Glide to current value**: These buttons correspond to the menu commands in the **Automation > Set Parameter Values** menu, but the filter options make them more flexible to use.
These functions are only available in the Preview mode (Preview automation) (↗344).

Object Automation

In objects you can automate the volume, pan, AUX send level and effect parameters of the object effects.

In the Track Editor, select **Lanes+Obj. Auto** or select **Show Lanes and Object Automation** in the Automation menu. All track automation curves are displayed in the Automation Lanes and you can now draw and edit the object automation curves on the objects.

In the Track Editor, select **Track Auto** or the **Show Track Automation** option in the Automation menu to toggle the display again.

ⓘ *Recording object automation during playback as with tracks is not possible, you can only draw in automations as curves and edit these curves.*

For automating object parameters, select an object and open the Object editor (↗183).
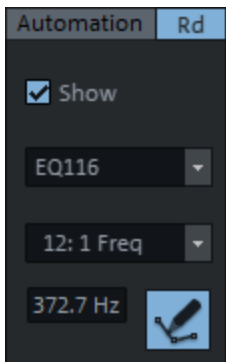
In the lower section of the menu ▾ , select the parameter to be automated.

ⓘ *Both drop-down menus open the automation menu. When automating effect parameters, the upper menu displays the effect and the lower menu displays the parameter.*

Now switch to the mouse mode Draw Automation (↗70) using the button below and draw the parameter changes directly onto the object with the pen.

◌ *To switch between displaying track and object automations, you can also use the **Show** checkbox at the top of the **Automation** section.*

To transfer volume and panorama automation from a track to an object, select the option **Edit selected curve** > **Convert track to object automation** in the Automation menu. Conversely, object automation can be transferred to the track using the command **Convert object to track automation**.

Editing Automation Curves

The automation curves can be edited in different ways. In addition to editing with the mouse in the project window, there are various commands for editing in the automation menu at **Edit selected curve...**.

Selecting a Curve for Editing

■ In the **Automation Lanes**, select a curve for editing by clicking directly on the curve or in the according the Automation Lane track header. If the curve is not displayed in any Automation Lane, select the corresponding parameter from the Automation menu.

■ In the **track** select the curve in the upper part of the automation menu. If the display option (↗341) **Show all curves (selectable)** is active, you can select the curve by clicking on it.

> 💡 *Sometimes you want to explicitly have no curve selected, but still have the curves visible in the track, for example, to edit the underlying objects. To do this, select **Show all curves (not selectable)** and **No effect (track)** in the Automation menu.*

■ In the automation menu, you can switch from curve to curve using the menu commands **Select next curve** and **Select previous curve**.

■ For object automations, the curve is selected in the automation menu of the object editor.

> ⚠️ *Attention: To edit object curves you have to use the automation menu in the object editor, the menus in main menu and track editor refer to track automation.*

Editing the Automation Curves with the Mouse

For this purpose you can use the Universal mode, besides there are some other mouse modes especially for editing the curves.

The complete details about editing the curves in the Universal mode and in the other modes can be found in the section Mouse Modes (↗70). Here is just a brief overview and the main differences of the modes:

■ In **Universal mouse mode** (↗73) 🖱 , you can create and edit individual curve points by double clicking. Individual points, curve segments (the line between two curve points) and all curve points in a range selection can be moved with the mouse.

> 💡 *If you select the option **Use snap for automation curve points** in (**System options** > **Program** > **General**), you can move the automation points only to the time positions set in the grid.*

- With the **Curve mode** ▨ or the combined **Object/Curve** mode you specifically edit curves. Selecting multiple curve points with a selection frame is easier and curve points can be created with a single click. In return, no curve segments can be moved.

- With **Draw automation** ▨ mode you can freely draw (↗76) curves. This allows more complex waveforms to be created more accurately. Straight lines can be created with additionally pressed **Shift** key. In return, you cannot move curve points in this mode.

> ⓘ *The Draw volume mode ▨ as a special case of the Draw Automation mode always draws a volume curve in the track, no matter which automation curve is displayed in the track.*

To facilitate the editing of small automation values of the volume curve, you can zoom this curve towards the zero line. The commands for this are located in the menu **View** > **Vertical**.

Command	Description	Keyboard shortcut
Zoom In Volume Automation	Increases the volume curve.	**Ctrl + Shift + Arrow up**
Zoom Out Volume Automation	Decreases the volume curve.	**Ctrl + Shift + Arrow down**
Zoom default volume automation	Restores the default zoom value, the entire automation curve is displayed.	

Editing the Curve Shape

Right-click on an automation curve point to open the dialog for numerical editing of the value and editing of the curve shape.

The dialog can also be opened with the command **Edit next node at cursor position** in the Automation menu or the keyboard shortcut **Alt + A**. The dialog will open at the point of the selected curve of the selected track that is closest to the current playback position.

In the input field you can enter the desired value numerically or change it with the mouse wheel. With the **Shift** key held down at the same time, change the value in smaller steps (factor 1/10) with the **Ctrl** key held down at the same time, change the value in larger steps (factor 10).
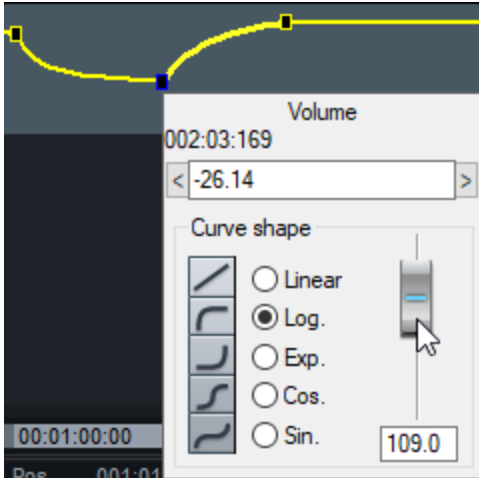
As long as the dialog is open, you can use the **Tab** key to jump to the next curve point. Close the entry field by pressing the Enter key.

With **Curve shape** you can define how the curve goes from the selected curve point to the next. The possible curve shapes correspond to the curves for objects fade in and outFade In/Fade Out (↗189). The slider can be used to fine-tune the waveform.

Curve Generator

The Curve Generator creates an automation curve from simple waveforms in sync with the clock grid or depending on the object volume level in the track.

To open the curve generator, select **Edit Selected Curve** > **Create...** in the Automation menu.
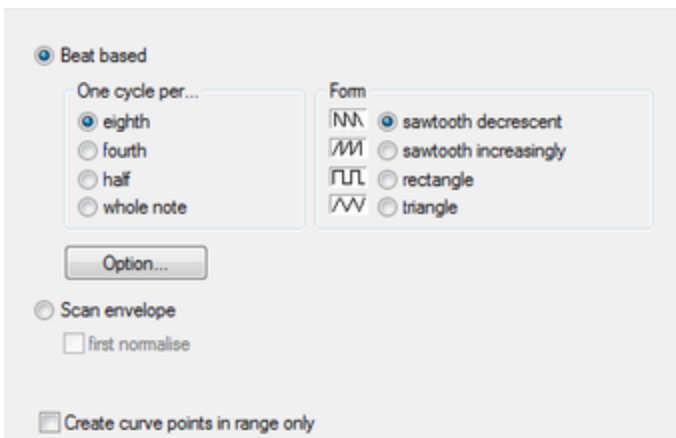
With the **Beat-based** option, the shape of the automation curve is based on the beat of the arrangement. At **Shape** you can choose between 4 different basic shapes for the automation curve: 2 sawtooth shapes, rectangle and triangle shape. With **One cycle per...** you can specify whether this waveform should be cycled once per eighth, quarter, half or whole note.

Click on **Options** to open an additional dialog to control the style and intensity of the influence of the beat on the automation curve.

- **Limitation** determines the minimum and maximum value of the envelope.

- **Delay at minimum/maximum in %**: Set here how long the maximum/minimum value is kept. This allows variations of the fundamental waveform, such as trapezoidal shapes for triangular waves or different pulse widths for square waves.
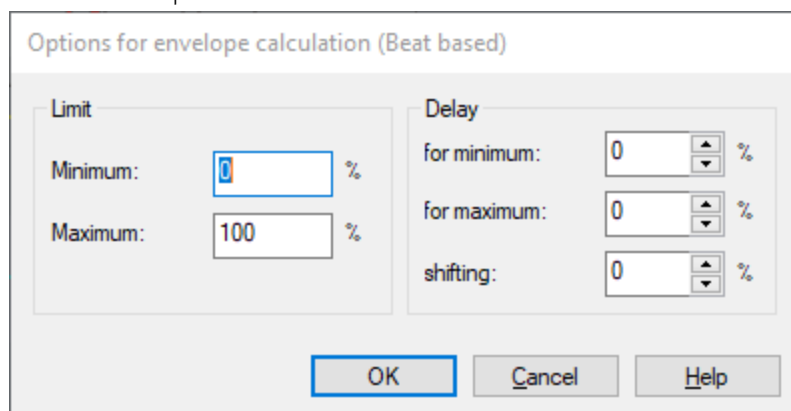
- **Shift in %** shifts the entire curve in the beat grid.

Scan Envelope: Use this option to create an automation curve from the volume envelope of the track's objects.
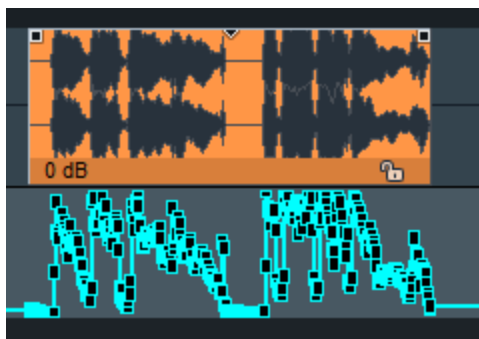
Normalize first: This function normalizes the level of the audio object before the object is scanned to create an automation curve.

Generate curve points in selected range only: If this option is active, curve points are generated only in the selected range.

Copying and Pasting Automation Curves

The commands **Copy**, **Paste** and **Paste in selected tracks** in the Automation menu can be used to copy automation curves partially or completely and paste them into the same or other automation curves.

Copy copies the selected curve completely. When a range is selected, only the section of the curve in the range is copied. Curve points selected individually or by selection frame can also be copied and are considered as a copied range.

Paste inserts the copied curve points into a selected curve. This works both with the same curve in the track, a curve for the same parameter on another track and even with any other parameters.

> ⓘ *The curve must be present and selected on the target track. So if you copy a volume curve from one track, then select another track and choose Paste, it will only work if you activate the volume curve on that track first. (Keyboard shortcut: Alt + K)*

If the curve was copied completely (without range selection), it will completely replace the automation on the target curve. A copied curve range is pasted from the position of the play cursor.

With **Paste in selected tracks** you can transfer a copied curve to several other tracks simultaneously. The curve is always transferred to the same parameter in the target tracks. A corresponding automation curve must be present in the target track, otherwise no curve points will be inserted.

Other Menu Commands for Editing Automation Curves

- **Create new point at play cursor position**: A new point is created on the curve at the playback position. It also opens the editor for the value and waveform.
  Keyboard shortcut: **Shift + A**

- **Invert**: The curve will be inverted. The smallest values become the largest and vice versa, so the curve is mirrored at the 50% line.

- **Thin out**: When recording automation curves, the automation points are set at very small intervals. This function reduces the number of automation points, whereby the course of the automation is still accurately displayed and reproduced afterwards.

- **Thin out automatically**: If this option is active, the command **Thin out** is automatically executed after each recording or drawing of automation curves.

- **Curve color...**: Use the command to open the color selection dialog.This allows you to assign a different color to the active curve. The 16 colors shown below at **Custom Colors** correspond to the palette you can select in Color Mouse mode (↗78).

- **Logarithmic**: This option lets you switch volume curves to logarithmic display. Thus the metric of the curve, i.e. the assignment of the curve height to the volume levels, corresponds better to the control behavior of real mixing consoles. When changing the display, you can optionally scale the values on the curve. This allows you to correct values that were drawn in the respective other display mode.

- **Delete**: The selected automation curve will be deleted.

■ **Delete all curves**: All curves of all selected tracks or of the selected object will be deleted.

Converting Object to Track Automation and vice versa

With **Convert track to object automation** you transfer the volume or panorama automation in the track to selected objects, with **Convert object to track automation** vice versa the object automations to the track.

If automations already exist in affected objects or tracks, you will be asked whether you want to overwrite the existing automation curves or overlay them with the new curve. Overwrite replaces the curve, when overlaying the values of the curves are added. The initial curves in the track or object are set to 0, so that nothing changes in the resulting volume or panorama curve.

Moving Automation Curves with Objects

If you want to move objects on the timeline and move the automations to the new position as well, activate the option **Link curves to objects** in the menu **Edit** > **Object mode** or in the menu at the button for the Object mode (↗155) on the upper toolbar.

When moving objects, all curve points from the object start to the next object are now moved. When you move an object to the left, existing curve points before the object are deleted, they are also not restored when you move the object back. In the same way, when you move the curve to the right, the following curve points are deleted when you reach the next object.

If you want to keep these automations, do not move the object to the new position, but select it and copy it. Then place the play cursor at the desired target position and paste the object there.

If you have automated an effect on a track and move an object from that track to another track, the automations will be reassigned correctly there if the effect is also present on that track. If the effect is missing from the target track, such automations are cached in a hidden automation curve so that the automation curve can be reassigned once you move the object back to a track where the effect is present. Such curves are marked with ??? in the automation menu.

> 🔆 *You can also create the effect on the target track afterwards. To reassign the automation, copy the "orphaned" curve, select the effect parameter in the Automation menu, and paste the curve.*

PROJECT TEMPO

Specifying a tempo in the project establishes the link between the project time and the musical time. Musical time indications (bars, quarter notes...) are needed at various places in the program:

- For the Beat grid (↗79) as a basis for beat-accurate edits

- For MIDI events, whose time positions are always musically defined

- For effects such as delays or modulation effects, when delay times or modulation frequencies are to be defined in musical time units.

- For VST instruments like drum samplers with their own sequencer

- For the grid display with the unit bars & beats

- For sending MIDI synchronization data (↗599).

The project tempo can be constant over the duration of the project or it can vary, controlled by tempo markers or a tempo track.

Constant Project Tempo

By default, a **Sequoia** project has a constant project tempo. This tempo is a project property and is displayed in the transport console.

Double-click on the BMP value to enter a new tempo, use the small arrow to open a menu to select a tempo and open a tap tempo dialog where you can tap in the tempo using the **T** key.

In the Project Properties dialog (↗645) (keyboard shortcut: **I**), you can determine the tempo: First, select a rangeRanges (↗118) that should correspond to a certain number of beats, and then click the **Get BPM from sel. range** button.

Tempo Changes

You can define changes of tempo in the course of the project by inserting **tempo markers** on the marker bar.

The menu commands for inserting tempo markers can be found in the marker menu (right-click on the marker bar or via the marker button) in the submenu **Tempo marker** and in the menu **Edit** > **Tempo**.

> ℹ *With variable tempo, the current values for tempo (BPM) and time signature at the current play cursor position are displayed in the transport console.*

There are two different ways to define tempo changes:

BPM Markers

If your arrangement requires an explicit tempo change at a specific time position in the project, set the new tempo in BPM by entering a **BPM marker** (red). The beat grid adjusts to the new tempo accordingly.
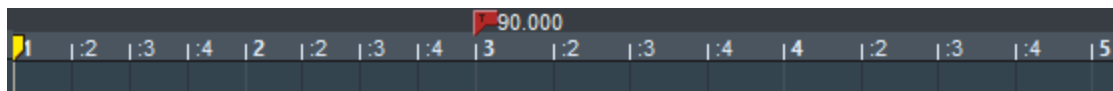
Place the play cursor where you want to insert the tempo change and right-click in the marker bar to open the marker menu and select **tempo marker** > **Set tempo change**.

The dialog **Tempo and time signature** opens with the properties of the tempo marker, enter the desired tempo there in BPM (beats per minute) and confirm with **OK**.

If you enable the **Interpolate tempo to preceding marker** option, the old tempo will be changed steadily starting from the previous tempo marker over the time span to the new tempo marker and the new tempo value will be reached at the current tempo marker. This way you realize tempo changes (musically "ritardando" or "accelerando").

The tempo markers are set to the next snap point near the playback position when the snap is active. Drag the tempo marker while holding down the **Alt** key to temporarily bypass the snap function.

> ℹ **Note**: *For audio objects in the project to adapt to tempo changes by tempo markers,* Musical Tempo Adjustment *(↗364) must be active in the Object Editor.*

Grid Position Markers

If you use audio material in the project in which the tempo varies, you can adjust the beat grid to match the audio material by reflecting these variations with appropriate small tempo changes in the project. But it would not be practical to directly control this changing tempo with all its fluctuations and enter corresponding BPM markers. Instead, you take an indirect route by entering **Grid position markers** (green).

Grid position markers assign a specific musical position to a specific time position. In this way, you define the tempo indirectly , because the tempo before this marker is adjusted in such a way that the desired musical position is reached exactly at the marker position.

In the tempo track (↗362), you can see the effects of the beat grid change on the tempo. In this example, there are quite drastic tempo variations for better illustration.

In practice, you use grid position markers in such a way that you insert markers according to the perceived tempo at regular intervals (for example, by "tapping in" (↗356)) and then move them in the project exactly to the corresponding transients in the audio material, thus adjusting the beat grid to the fluctuating tempo.

To insert grid position markers, right-click in the Marker bar to open the Marker menu and choose **Tempo Markers** > **Set Grid Position marker**.

> ⓘ *The grid position marker is set without opening the **Tempo and Time signature dialog**. The option **Interpolate tempo to previous marker** in the dialog is enabled by default, since it is assumed that continuous tempo variations will be captured with the input of grid position markers.*

Grid Tapping

You can also set grid position markers "live" during playback. This allows you to adjust a beat grid to music with a fluctuating tempo, for example, to synchronize a drum computer to the music in a remix:

1. To do so, assign the menu command **Set grid position marker** a keyboard shortcut of your choice, e.g."+".
2. Switch the tempo map to grid fit mode by clicking the marker bar on the right and selecting **Tempo marker** > **Tempo map grid fit mode**.
3. Now press "+" in time with the rhythm during playback and tap in the beat grid. At the moment the key is pressed, setting the grid position marker moves the nearest exact bar position in the beat grid to the current playback position.

4. This creates a beat grid that perfectly matches the tempo of the music. Small inaccuracies can be compensated by moving the markers with the mouse.

    💡 *First, tap once per measure at the whole measure boundaries to roughly capture the tempo. Then, in a second pass, work on finer tempo fluctuations in the "beats" grid.*
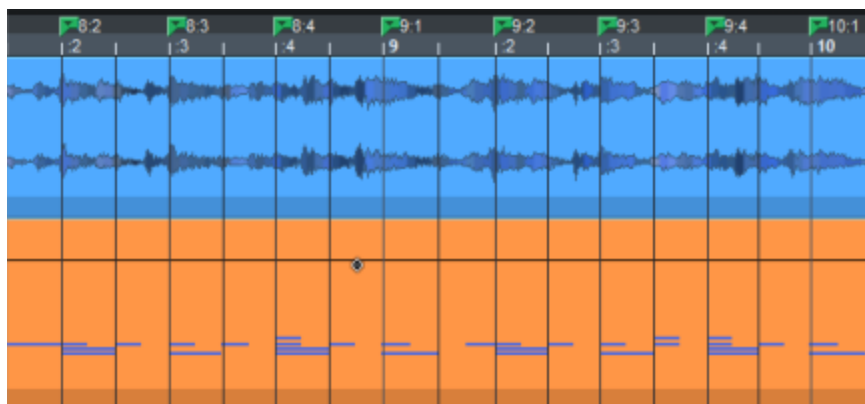
Creating and Editing Tempo Markers with the Mouse

In Pitchshift/Timestretch mouse mode (↗75) 🔵 you can create and edit tempo markers on the marker bar with the mouse.

Select the **Pitchshift/Timestretch mode** from the Mouse Mode menu on the top toolbar or menu **Edit** > **Mouse Mode**.

- **Create BPM markers**: **Shift + click on grid bar**. If you keep holding the mouse button and drag the mouse vertically, you can set the tempo right away.

- **Setting the tempo of the BPM marker**: **Shift + click on BPM marker + drag vertically**.

- **Create grid position markers**: **Alt + click on grid bar**.

    ℹ️ *The Grid Position markers are immediately converted to BPM markers in BPM mode. In raster mode, you can create only raster position markers with the mouse.*

- **Moving tempo markers without adjusting time positions**: **Alt + click on BPM marker + drag horizontally**. This allows you to manipulate the beat grid and adapt it to existing audio events.

Change Tempo Globally

If you have inserted tempo markers in a project and subsequently want to change the tempo of the project as a whole, you would have to adjust each individual tempo change accordingly. To simplify this, you can use the command **Tempo** > **Change tempo globally...** in the menu **Edit** to open a dialog in which you can change the tempo of the project as a whole by a certain factor. The entered factor is applied to each tempo marker.

Time Signature Change

To change the time signature (e.g. from 4/4 to 3/4) from a certain musical position, insert a **time signature marker** (blue). To do this, select **Set time signature marker** in the menu **Edit** > **Tempo** or in the marker menu > **Tempo marker**.

The time signature has no influence on the project tempo. It determines the display of subdivisions in the grid (↗81) and the grid length (↗79) with the setting **Beats** and the note display in the note editor.
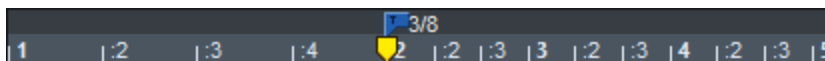
The first measure is in 4/4, with 4 divisions, from measure 2 in 3/8 with only three divisions in 1/8 steps

Time signature markers can only be inserted at whole bar boundaries and, when moved, can also only be positioned at the beginnings of bars.

Tempo Map Dialog

The Tempo Map dialog contains the list of all tempo markers, their properties and a number of options on how to handle the markers. In the dialog, additional markers can also be created and selected or all tempo markers can be deleted.

Open the dialog with **Tempo Map Dialog...** in the marker menu (right-click on the marker bar) or in the **Edit** > **Tempo** menu.
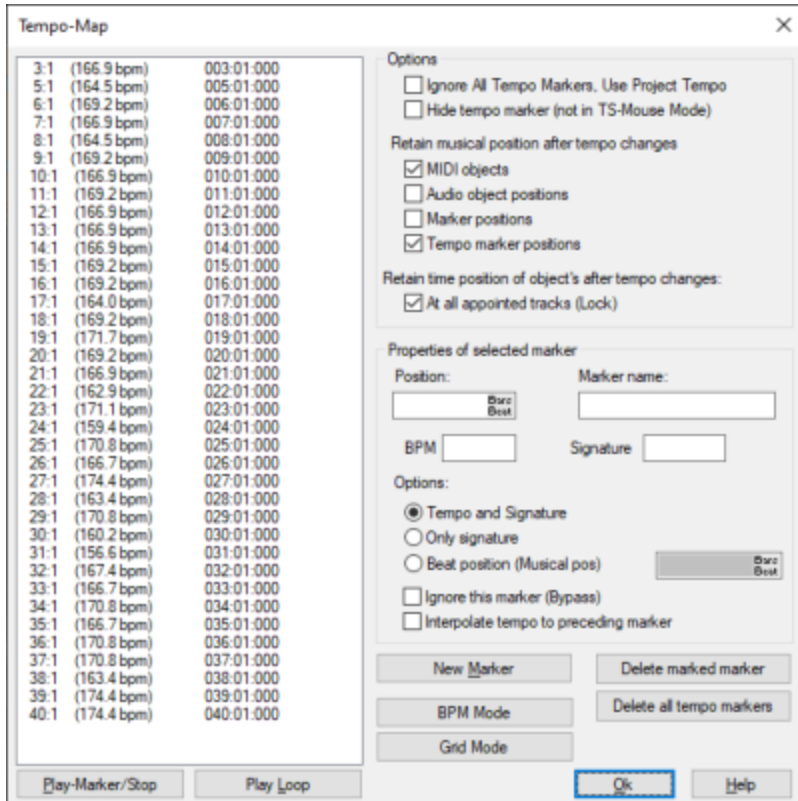
In the list on the left, all tempo markers are listed with their musical position, time signature and BPM value. With **Play Marker/Stop** the project can be played from the position of a marker, with **Play Loop** the area around the marker is played in a loop.

You can choose between two modes for displaying and editing all tempo markers: **BPM mode** and **grid mode**. In the respective modes all tempo markers are converted into the corresponding format, i.e. in **BPM mode** all tempo markers are displayed as BPM markers, in **Grid mode** as grid position markers.

Using the buttons, you can switch between BPM markers and Grid position markers at any time. The tempo markers remain the same, only the type of display changes. For example, first adjust the beat 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.

> ℹ️ *You can also switch between the modes using the commands in the Markers menu > **Tempo marker** or in the **Edit** > **Tempo** menu.*

Options in the Tempo Map Dialog

Ignore all tempo markers, use project tempo: This option ignores all tempo markers in the project, so that only the global project tempo is relevant.

> 💡 *This option can also be switched on and off via menu commands and at the track head of the tempo track.*

Hide tempo markers: Enable this option to hide the tempo markers in the arranger. However, they are always displayed when you select the Pitchshift/Timestretch mouse mode, because in this mode there are special mouse actions for editing tempo markers (↗357).

Retain musical position after tempo changes

If a BPM marker is changed or a Grid position marker (↗356) is moved, this influences the following markers and audio/MIDI in the project. There are two options for this: Either the time position is preserved or the musical position.

The time position is the absolute position, i.e. a certain time position corresponds to different musical bar positions at different tempos.

> **ⓘ** *Example: At 115 BPM, bar 5:1 is at 8347 ms, at 140 BPM this position is at bar 5:04:183.*

The musical position is flexible and depends on the time signature. At different tempos, the same musical position corresponds to different time positions.

> **ⓘ** *Example: At 115 BPM, bar 5:1 is at 8347 ms; at 140 BPM, this musical position is already reached at 6847 ms.*

If the musical position is to be preserved with changed tempo, the positions of the objects or markers must be adjusted to the changed beat grid, thus changing their time position in the project. If the time position is maintained, their musical position changes accordingly.

The behavior of markers, audio and MIDI objects when changing tempo markers can be set individually: By default, MIDI objects and tempo markers keep the musical position, all other markers and audio objects keep the time position.
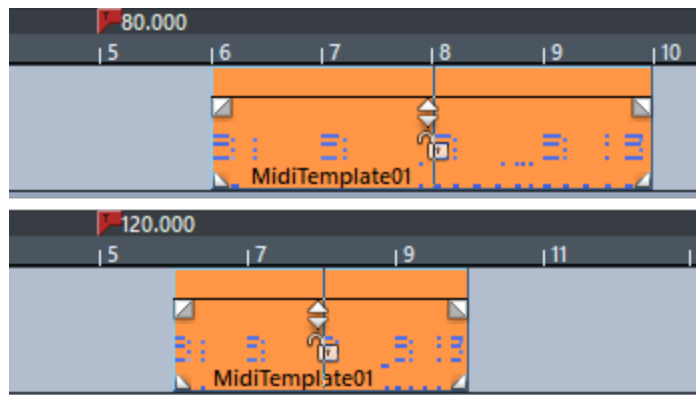
Example: A MIDI object is placed at bar 6:1 at tempo 80 BPM. When the tempo becomes 120 BPM, the object is moved back and is thus still placed at 6:1.

> ⓘ **Notes**:
>
> - *Audio objects for which the* Musical tempo adjustment *(↗364) is active behave like MIDI objects and are also moved.*
>
> - *If you drag a tempo marker in* Timestretch mouse mode *(↗357) while holding down the* **Alt** *key, the time position of an object is always preserved.*

For all locked tracks: If you want to preserve the time position of all objects in a track during tempo changes, select this option and activate the button 🔒 for locking all track objects in the respective track head.

Properties of Selected Markers

For the tempo marker selected in the list on the left, the properties are displayed here. If you have created a new marker at the play cursor position directly in the Tempo Map dialog using **New Marker**, set the properties of this marker here.

Position: Time position of the selected marker

Marker name: You can also assign names for tempo markers

BPM: Tempo in BPM When entering a grid position marker, the value cannot be edited. It updates to the tempo at that marker from the **Beat position** currently assigned to the grid position marker when you click into it.

Time signature: A time signature can be assigned to each tempo marker. If you switch to **Only signature** at **Options**, the current tempo marker becomes a Time signature marker.

With **Beat position (musical pos.)** you convert a selected BPM marker into a Grid position marker (↗356). You can edit the musical position of the grid position marker in the editing field.

Ignore this marker: Ignored markers have no effect. This allows you to cancel the effect of a marker without deleting it. This happens automatically with grid position markers if nonsensical bar positions are created by moving or

entering in the Tempo Map dialog, e.g. if grid position markers are swapped (bar 20 before bar 19) or moved in such a way that this bar position can no longer be reached by tempo interpolation from a previous tempo marker.

Interpolate tempo to previous marker: Activate this option if you want the tempo to change steadily starting from the previous tempo marker over the time span to the new tempo marker and reach the new tempo value at the current tempo marker. This way you realize gradually tempo changes (musically "ritardando" or "accelerando").

> **ⓘ** **Note**: *These settings apply to the marker currently selected in the marker list. So you can set the marker properties, such as the type or tempo, only after creating a new marker.*

Properties of a single marker can also be edited without opening the **Tempo Map dialog**. Double-click on an existing tempo marker or right-click on a curve point in the tempo track to open the **Tempo and Signature** dialog for that marker, which displays the properties of a single marker.

Tempo Track

With the tempo track you can display the course of the tempo as a curve in a separate track. The Tempo Track is a graphic display of the project's tempo map (↗358) as an automation curve (↗338). Each automation point of the curve corresponds to a tempo marker. Changes to tempo markers are automatically transferred to the Tempo Track and vice versa.
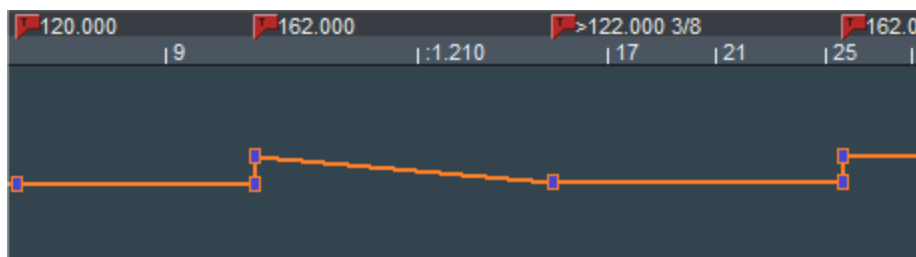
BPM markers, a tempo ramp (>) and a time signature change (3/8) and the tempo track below.

The Tempo Track head 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. However, you will then be missing the special controls in the track head of the tempo track. When you create a separate tempo track, the tempo automation curve of the other track is removed.*

Creating Tempo Tracks

To create a Tempo Track, select **Track** > **Insert new Tracks** > **New Tempo Track**.
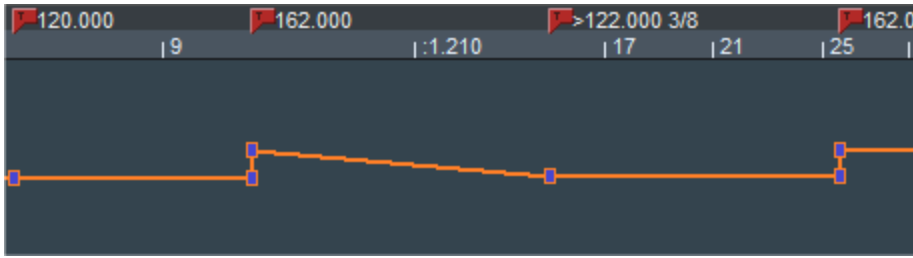
Tempo automation behaves somewhat differently than the other track automations: For example, in the case of a leaping tempo change, the two automation points (old and new tempo) are coupled together, and the automation drawing mode creates new automation points only by clicking, not clicking and dragging.

> ℹ️ **Note**: *In the tempo track the beat grid is constantly active (regardless of the system option **Program** > **General** > **Use snap for automation curve points**). The grid can be temporarily deactivated with the **Alt** key.*

Editing the Tempo Track

> ℹ️ *The tempo track can only be edited in BPM mode, in snap mode it serves solely as a display.*

Insert Tempo Changes

Double-click in the Universal mode ▶️ (single click in Curve mode 〰️) on the desired position in the tempo curve. With the new curve point, a new tempo marker also appears in the marker bar.

Drag the newly created curve point up or down to change the tempo. You can see that there are actually two automation points created at the same time position, one is connected to the previous point and can only be moved horizontally to adjust the time position of the tempo change. The other can only be moved vertically and is used to set the tempo.

> 💡 *If the tempo change is very small, it may happen that the points are on top of each other and you can only reach the point for the time position with the mouse. For the tempo change, you can therefore also drag the curve behind the point for the tempo.*
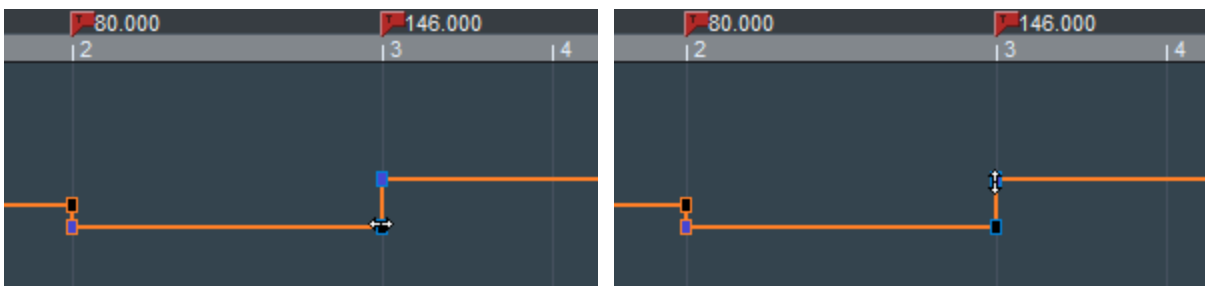
If you hold down the **Shift** key while moving a point, only the vertical position (tempo) can be moved, whereby the tempo can then be set more precisely.

Right click on a curve point opens the **Tempo and Signature dialog** (↗355), which allows you to change the properties of a tempo marker and the values numerically.

Insert Tempo Gradient

To insert a tempo gradient:

- Holding down the **Shift** key while double-clicking (Universal mode) or single-clicking (in Curve mode/Automation Draw mode) to create a curve point will create a tempo gradient to the previous BPM marker. Only one point is created, which can be moved in both directions (position and tempo).

- If you hold down the **Alt** key while clicking and moving a curve point, you convert tempo leaps to both adjacent tempo markers into tempo gradients. The two points are combined into one point that can be moved freely.

- Hold down the **Ctrl** and **Shift** keys and click a curve point to toggle between tempo gradient from the previous tempo marker and a tempo leap.

Tempo Track Head
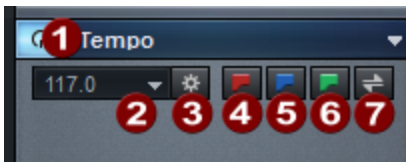
1. **Tempo automation on/off**: You can disable tempo automation entirely with this button. This corresponds to the option **Ignore all tempo markers, use project tempo** in the Tempo Map dialog (↗358).
2. **Current tempo**: The dropdown menu under the arrow corresponds to the one in the transport console and contains a selection of preset tempos, as well as the tap tempo feature.
3. Open **Tempo Map dialog**
4. Insert new **tempo marker**
5. Insert new **time signature change**
6. Insert new **grid position marker**
7. Switch between Grid mode/BPM mode (↗358).

Musical Tempo Adjustment

Musical tempo adjustment automatically adjusts audio objects to match tempo changes in the project. This means that if the project tempo changes, the tempo of the audio objects and their start position adapts to these tempo

changes. In order to maintain the musical position (bar position), the start position of the objects is moved and time stretching is used so that the object tempo is adapted to the current project tempo at all times.
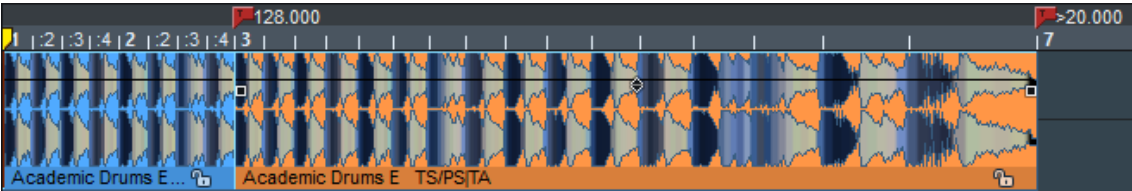
Tempo adjustment even works with tempo gradients.

To use the musical tempo adjustment, in the **Time/Pitch** view of the Object editor, select **Use musical tempo adjustment**.
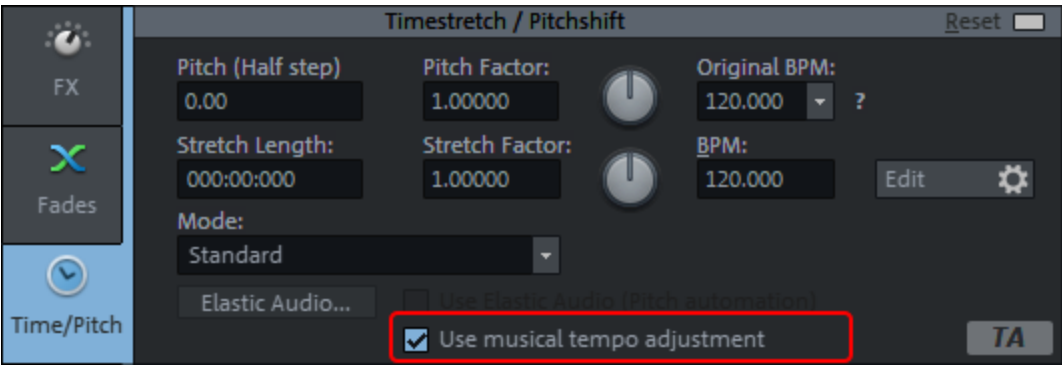
> **ℹ️** **Note**: *For sound pool samples, musical tempo adjustment is enabled by default. If audio files are loaded via the File manager with the* BPM Sync *(↗319) pre-listening option active, tempo adjustment is also activated.*

To determine the time stretching factor for the tempo adjustment, the audio files must have a valid original tempo, which is displayed in the object editor at **Original BPM**. For files recorded in Sequoia, the original tempo of the objects corresponds to the project tempo. You can also add a tempo to any other audio files by opening them as a wave project (↗569) and setting the project tempo (↗354)

 A question mark next to the 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. Options for detecting the BPM value are located in the menu next to the value ▾
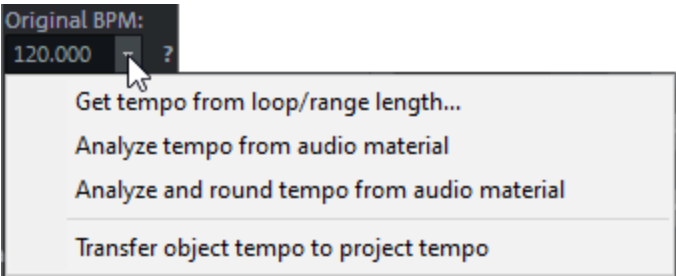
Apply Tempo from Loop/Range...: To determine the tempo, select a range over the object and specify how many quarter notes the selection includes.

Analyze tempo from audio material: **Melodyne** will calculate the tempo. Melodyne must be installed for this. More on this at Melodyne integration (↗222).

Musical tempo adjustment cannot be used simultaneously with Elastic Audio (↗265) (pitch automation), ARA (Melodyne integration) (↗222), or object resampling.

The indicator field in the object editor will show you which of these options is currently being used:

- **PA**: Pitch-Automation (Elastic Audio)

- **TA**: Tempo adjustment

- **SRA**(Sample Rate Adjustment): Adjustment of the sample rate of audio objects by object resampling that was applied when the sample rate of the wave file differs from the project sample rate. 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**: Melodyne-Plug-in

ⓘ *Notes:*

- *The positions of object automations and* audio quantization markers *(↗612) will be taken into account and changed accordingly.*

- *The* Remix Agent *(↗610) cannot be run.*

- *The tempo adjustment is not available for objects that have an active* loop *(↗161) or an object that plays* backward *(↗190).*

AudioWarp

With the help of Audio Warping it is possible to selectively shift audio material within an object in time. This allows temporal inaccuracies in recordings to be corrected.

For this purpose, **Warp Markers** can be set in objects. These Warp Markers can then be moved, whereby the audio material in front of the Warp Marker is temporally compressed and the audio material behind it is stretched or vice versa. This time-stretching is effective until the next Warp Markers in the object.
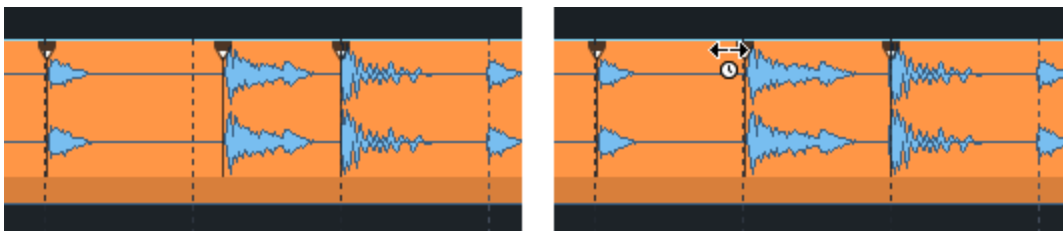
In the picture you can see the typical procedure: To fit the beat in the middle to the beat grid, a Warp Marker was also set on the beats before and after it. By moving the middle Warp Marker, audio material in front of and behind it was compressed and stretched so that the beat at the now exactly desired position in the project.

🅶 To set and edit Warp Markers, switch to **Pitchshift/timestretch mouse mode**. Warp Markers that have already been set are also only visible in this mouse mode.

> 💡 *Tip: To quickly change the mouse mode, right-click anywhere in the project, hold the mouse button, and also left-click.*

To set a Warp Marker, click in the upper half of the object at the desired position. An auxiliary line at the mouse cursor over the entire height of the object makes setting easier.
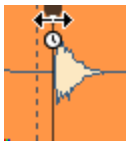
Click and drag the Warp Marker in the upper half of the object to move the point in time in the audio material below the Warp Marker to the desired position. All audio material before and after the Warp Marker until the next Warp Marker (or the object start or object end) will be compressed (played faster) or stretched (played slower) accordingly.

To move the anchor position, i.e. the original position of the Warp Marker in the object to be moved, click and drag the Warp Marker in the lower half of the object.

> ℹ️ *You will notice that even in this case it is not the marker that moves, but the audio material, because the target position of the Warp Marker remains the same, only the time-stretching of the audio material is adjusted to the new anchor position.*

For audio warping, the Musical tempo adjustment (↗364) must be active in the objects. This is automatically activated when you set the first Warp Marker. This also means that the restrictions that apply to Musical Tempo Adjustment (no objects with loop, reverse effect, Melodyne, Elastic Audio or object resampling) also apply to Audio Warping.

Right-clicking on a Warp Marker brings up a context menu where you can use **Warp marker active** to temporarily disable this marker, as well as delete this Warp Marker or all Warp Markers in the object .

Metronome

There is a metronome for acoustic indication of the current tempo.
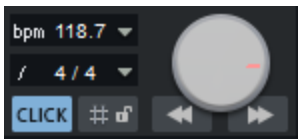
Activate the metronome with the button **CLICK** on the transport console, the menu command **Metronome active** in the menu **Edit** > **Tempo** or the keyboard shortcut **Ctrl + #**.

In the same menu, in the menu **File** > **Program Settings** or by right-clicking on the **CLICK** button on the transport console, open the **Metronome Settings**, where you can configure the metronome.
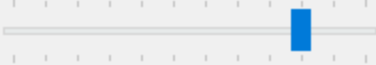
- **Active during playback**/**Active during recording**: Set here whether the metronome is activated during recording, playback or for both.

- **Precount For Recording (Measures)**: Here you can specify the number of beats that the metronome will count in before the recording starts. With **Play VIP during the pre-count** active, the playback starts already during the pre-count.

- **Pre-count clicks only**: With this option enabled, the metronome will stop with the start of the recording.

The metronome can be operated as an audio or MIDI metronome. The audio metronome plays samples, the MIDI metronome outputs MIDI notes via a MIDI hardware device.

Audio Metronome

- **First beat/beats**: This setting allows you to specify individual samples for the metronome for the first beat of each measure (**First beat**) and the other beats of the measure (**Beats**). The two metronome sounds "Metronome1.wav" and "Metronome2.wav" from the "fx-preset" folder of the program folder are preset.

- **Volume**: This slider controls the volume of the audio metronome sounds.

- **Audio device**: Set a hardware output as output device for the metronome here. **Stereo Master** is set as default.

MIDI Metronome

- **MIDI output device**: Here you can set a MIDI output to send MIDI notes for the metronome clicks. A suitable MIDI sound generator (e.g. drum module) must then be connected there, or you use the preset system software synthesizer (Microsoft GS Wavetable Synth).

  > **ⓘ** *It is not possible to use the MIDI metronome with software synthesizers.*

- **MIDI channel**: Set the MIDI channel here. Channel 10, the default channel for drum sounds, is preset.

- **First beat / Beats / Note Value / Velocity**: Set the note values and the velocities for the **first beat** of each bar or the other **beats** of the bar here.

Create Click Track

Use the menu command **Edit** > **Tempo** > **Create Click Track...** to create a new track above the first track in the project. The track (named as "Click") contains an audio object that contains all metronome clicks. In the dialog you can set whether the object should be created for a selected range, from the beginning of the project to the end of the last object or over the entire length of the project.

A Clicktrack has some advantages over using the metronome:

- You can control the click volume in the mixer and turn the metronome on and off by muting the click track.

- For more flexible monitoring, you can route the metronome signal to busses.

- The metronome can be activated during recording only in certain sections.

- The Clicktrack works better than the metronome when you work with the **Varipitch** (↗89) function.

> 💡 *Tip: Enable **Musical Tempo Adjustment** in the Click Track object so that the click track adjusts to subsequent changes in tempo or tempo map.*

VST PLUG-INS

VST is an interface that allows third-party audio software to run as instruments (VSTi) or effects (VST effects) in the form of plug-ins in **Sequoia**. **Sequoia** supports VST effects and instruments according to the VST standard VST2 and VST3.

The plug-ins can be used as 32bit or 64bit plug-in version. Older plugins, which are only available as 32bit program versions, are addressed via a so-called "bridge", i.e. a program mechanism that translates between both formats. However, since this costs additional performance and also stability and compatibility may suffer, we recommend using the 64bit version of the plugins if available.

Like the internal effects, VST effects can be used in various places to shape the sound, and many of the included effects are also designed as VST effects (MAGIX plug-ins).

VST instruments are used for virtual sound generation and are controlled by MIDI objects on the tracks. Their output signal is available as a track input signal and thus also as a channel in the mixer. 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.

Installing 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 **Sequoia** 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. VST3 plug-ins are always installed in the default path for VST3 plug-ins (`C:\Program Files\Common Files\VST3`). For VST2 plug-ins, the plug-in installer usually also suggests a default path (e.g. `C:\Program Files\VSTPlugins`). These default folders (System VST folders) are automatically included in the plug-in search by **Sequoia**. However, this is not really a standard, so 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 (shortcut "Y") and go to **Effects > VST/Rewire**. Under the VST plug-in folder list, click **Add...** and enter the folder path. You can specify and scan any number of plug-in folders.
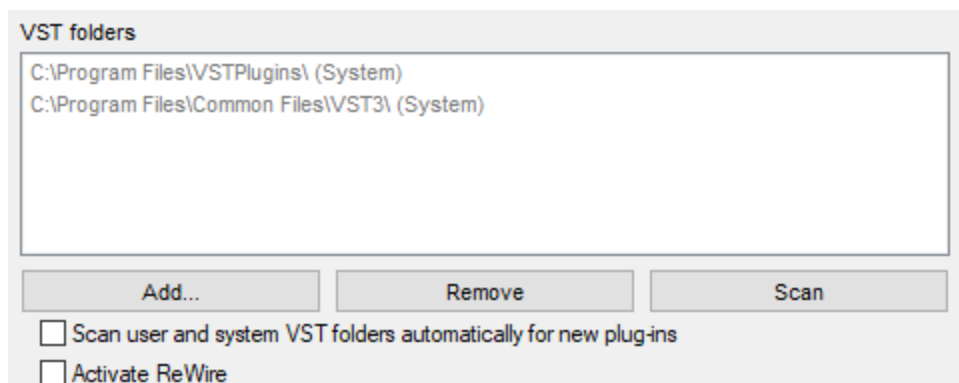
3. Click **Scan** and choose **Scan VST Folders** from the menu to start the plug-in search. This scan may take some time for many installed plug-ins and instruments. Not only are all plug-ins scanned, but they are also checked for their usability in the program. Incompatible or incorrectly installed plug-ins and those that cause the plug-in search to crash are also added to the list as unusable, so that they are skipped the next time you scan and cannot cause problems again. To scan those plug-ins again, use the menu item **Scan failed plug-ins**.

4. Once the scan is complete, you can use the newly found plug-ins immediately.

Tip: 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\MAGIX\Sequoia 17. You must then add all additional search paths again before rescanning.

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 will search for newly installed plug-ins and remove 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.

Tips

   ■ *If you add VST plug-in subfolders into the main VST folder (e.g. "Equalizer", "Filters", "Modulation"), these will also be displayed as subfolders in the Plug-in browser.*

   ■ *A menu command to reload the plug-in folder can also be found at the bottom left in the **Options** menu of the plug-in browser.*

Load Software Instruments

Loading a software instrument can be done at different places in the program:

   ■ Loading via the **Plug-in browser**:

   This is the recommended method, because the Plug-in browser *(↗216) offers a full-text search for instruments and also their presets, and other ways to organize the plug-ins.*

- Select the track.

- Open the Plug-in Browser (key: **B** or click **Add Plug-in...** in an empty plug-in slot in Mixer, Track Head or in the Plug-in section of the Track Editor).

- Filter the browser list with the button **Instruments** and select the instrument in the list.

■ Load an instrument by selecting it as **MIDI output device** for a track:

- In the **MIDI** section of the **Track Editor** under **Out**

- In the track output context menu (right click on the **Mute** button in the **track head** or in the **Mixer** at the very bottom of a channel under **Out**) in the submenu **MIDI**

- In the Keyboard window (↗476)

A rather large menu appears, giving you direct access to all presets of all included software instruments (MAGIX Synths). All other plug-ins can be found in the submenu **New instrument**.
You can also use the same menu to remove an instrument by selecting **No VSTi (audio recording)** from the menu at the top.

■ Alternatively, you can load software instruments directly in VSTi Manager (↗331). To do this, click in the list on **<EMPTY>**, the menu **New instrument** also opens.

> 🛈 *Note that in this case one or more new tracks are always created!*

> ⚠ *A track can only have one **MIDI Out** destination at a time. If you load an instrument in a track that already sends MIDI to another instrument that is also only controlled by this track, this instrument would consequently not be controlled at all. In this case, Sequoia will ask if you want to delete this "orphaned" instrument or keep it loaded so you can re-route it in the VSTi manager.*

Routing of Software Instruments

You can assign a software instrument as a MIDI output device to each track. You can also route the audio output channels of software instruments to any tracks of the project. MIDI (send) and audio (return) of a software instrument can be present together on one track, but they do not have to be.

Dialog Routing for Multichannel Software Instruments

When loading a software instrument, the Routing dialog for multi-channel software instruments opens, in which you can configure the typical routings for a software instrument directly when loading.

Simple: Audio and MIDI together on the same track The software instrument is completely managed in the current track. The track sends MIDI to the instrument and receives the instrument's audio signal. Multiple outputs will be mixed together before the mixer effects.

Extended options: You can specify that for VSTi with multiple outputs, only the first stereo output is routed to the current track.

Multitimbral: Route audio to its own track separate from MIDI: Typically, so-called multitimbral sound generators, which can play back multiple sounds controlled by different MIDI channels at the same time, use multiple MIDI tracks, with each track controlling a specific sound program (part) on a specified MIDI channel. The simplest solution for this is to create the current track as the first (and only) MIDI control track and route all outputs to a new audio-only track. You can then gradually create more MIDI tracks that use the sound generator.

In the **Extended Options** you can also specify that only the first stereo output is routed to the audio track. In addition, you can decide to **don't show the audio return track in the arranger** here. This is useful because these audio tracks do not contain objects or information. In the mixer, these tracks are visible.

ⓘ **Note**: *To display hidden tracks again in the arranger window, use the* Track manager (↗324).

Multi-Channel: Create multiple 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 done automatically by Sequoia (default).

Extended 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 checkbox to route the MIDI output of each single output track to this instrument as well. A different MIDI channel is automatically assigned for each track in ascending order.

- **Don't show the audio return tracks in the arranger**: This is useful because the audio tracks in the arranger do not contain any objects or information. In the mixer, these tracks are visible.

The checkboxes at the very bottom allow you to determine whether this dialog is automatically displayed when inserting a new software instrument into a track and/or when performing actions in the VSTi Manager.

If you have disabled automatic opening, the last selected option will be applied when loading new software instruments. To display the dialog again or to change the options, you can also open this dialog in the VSTi Manager (↗331) at **Routing Settings**.

Editing the Routing of Software Instruments

To route an **audio output** of a software instrument to a specific track:

- Use the VST Instrument Manager

- Select an audio output of a software instrument from the context menu **Track input**. You open it via the Track head of the track (right click on **Rec**), via **In** in the **Audio** section of the Track editor or in the mixer via the **In** button at the very top of the mixer channel.
  From the **Instrument Outputs** submenu, select the output of the software instrument that you want to sound on this track. The menu item is indicated by a check mark. By repeating the process, you can also route multiple outputs of an instrument to one track, or the outputs of other instruments. Deactivation of the instrument outputs works in the same way.

To route the **MIDI output** of a track to the input of a VSTi, open the MIDI output device menu as described at Loading software instruments (↗371):

- In the **MIDI** section of the **Track Editor** under **Out**

- In the track output context menu (right click on the **Mute** button in the **Track head** or in the **Mixer** at the very bottom of the channel under **Out**) in the submenu **MIDI**

- In the Keyboard window (↗476)

At the top of this menu, the loaded VST instruments are listed and you can select one of them.

VST Plug-in Dialog

The plug-in dialog is initially opened when a VST plug-in is loaded. After it was closed, it can be reopened by right-clicking on the plug-in slot in various places in the program (track editor, track head, mixer, object editor).For instrument plug-ins, you can also open it in the VST Instrument Manager (↗331) or via the menu **Track > More > VST Instrument Editor**.

Plug-in

Various settings for the plug-in can be made in the menu **Plug-in**:

Plug-in dialog/Parameter dialog: By default, each plug-in is displayed with its graphical interface. Alternatively, the **Parameter dialog** can be displayed. In this dialog you can select eight parameters of the plug-in, which can then be controlled via sliders. This dialog is also used if the plug-in does not have its own graphical interface.
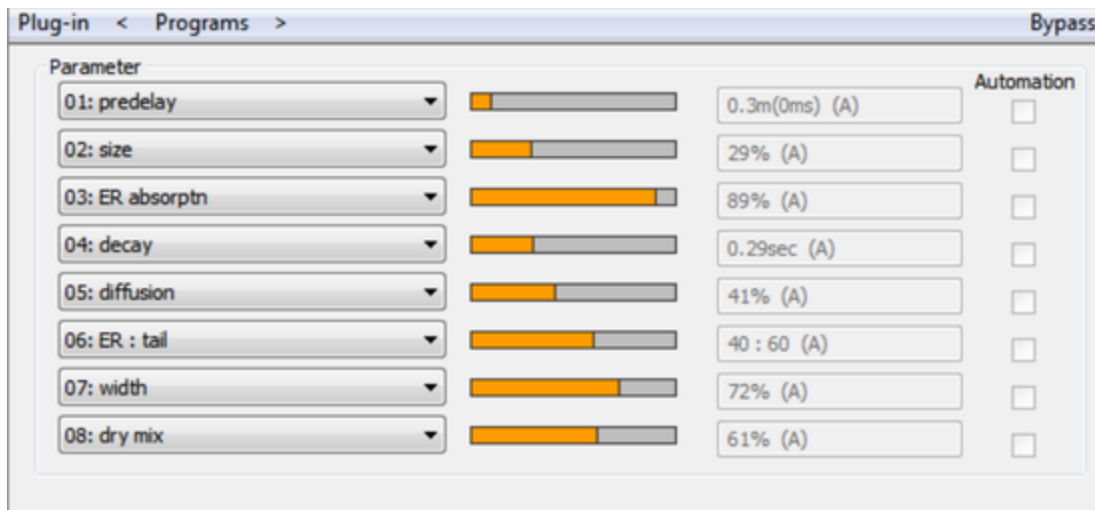
The parameter selection is saved for each plug-in so that you can use the same parameters the next time you use the plug-in. This makes frequently used parameters of the respective plug-in available immediately after loading.

💡 *The keyboard commands **F6** and **Shift + F6** can be used to quickly switch between the two displays.*

FX I/O Matrix (only with maintenance contract): This opens the FX I/O Matrix FX I/O Matrix (↗378). This allows you to change the channel assignment of the inputs and outputs of effect plug-ins. This is particularly useful for correcting incorrect channel assignments for surround effects.

Bypass: Deactivates the instrument and mutes it. Note that some instruments consume processing power even 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 active if the instrument should be heard when playing live and recording. If you want monitoring to be enabled in the track whenever a track is armed for recording, select **Tape monitoring** in the **System options** > **Audio setup** > **Monitoring behavior**.

Receive MIDI: This option is always on for VST instruments. For MIDI controllable VST effects (e.g. vocoder plug-in with integrated synthesizer), you can activate this option so that the effect is available as a routing target for the MIDI output of a track and the audio effect can therefore be controlled via MIDI.

Send MIDI: If you activate this option, MIDI data sent from a VST plug-in can be used as MIDI input of another track.

Process MIDI only: You cannot influence this option. It indicates that the plug-in has been loaded as **MIDI plug-in** (see below).

Permanent options: These additional settings apply to all instances of the respective plug-in.

> ℹ️ **Note:** *We recommend, that you alter the Permanent Options only if compatibility problems occur. If you have made changes here, please reload the project afterwards!*

Restrict to 1 CPU only: All instances of VSTi will be calculated only on one CPU. This way you can avoid a multi CPU conflict between different instances of the same plug-in. If you activate this option for an effect plug-in, all track and object effects on tracks that use this plug-in are calculated on one CPU only.

> ℹ️ **Note:** *When using UAD cards, this option is activated by default.*

Force calculation for silent input: If you have individual tracks calculated as economy tracks via the Playback Engine (↗694), Sequoia normally does not process track effects if there is no audio signal present for a while. This decreases the load on the CPU. However, with this plug-in option you can make the plug-in in the track always calculate audio data, even if no audio signal is currently passing through the effect. This is recommended for reverb or delay plug-ins with long decay times and generally for all plug-ins that produce audio independently of their input.

> ℹ️ **Note**: *When using the Hybrid Engine, also empty or silent tracks are calculated by default instead. You can change this behavior in the* performance options *(↗703) (keyboard shortcut: **Y**) with the options **Deactivate muted tracks for ASIO** and **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: Save an instrument's settings and sounds here. The standard formats for this are: *.fxp for patches or *.fxb for whole banks. Some instruments have their own patch/bank format. Settings are then usually loaded and saved directly via the instrument's interface.

Randomize parameters: 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 interesting results for the sound design thanks to

the random generation of a new sound. However, it should be noted that parameters are only set randomly if they can also be reached from the interface. With some very complex virtual synthesizers or modular systems, some parameters may not be adjustable using this option. Also note that randomly setting arbitrary parameters could generate very extreme sounds. 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 which you change during playback will be recorded via automation. You can also achieve this by holding down the **Ctrl + Alt** key together while changing the parameter with the mouse.

Hardware Controller Easy Learn: If you activate this function, you can learn your hardware controller by moving the plug-in element you want to train and by moving the element you selected at the hardware controller.

Pay attention to the following when doing this:

■ When you are finished assigning the controls be sure to switch "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.

■ Assignment can be cleared again with the **Restore modes** button beside the internal mode in the hardware controller setup

ⓘ *Detailed information about mapping hardware controllers can be found in the included PDF document* `Hardware_ Controller.pdf` *in the section* **Customizing Controllers**.

Programs

Programs features the presets for the corresponding plug-in.

ⓘ *Only the internal presets according to the VST standard are displayed. Many VST plug-ins have their own manufacturer-specific preset management, which is only available through the plug-in's interface.*

For VST instruments: If an instrument has presets, they will also appear on the track that uses this plug-in in the MIDI section of the Track Editor at **Program**. If your MIDI input device can send Program Change messages, you can select programs of the instrument directly in this way.

Sidechain Input

The sidechain is a second stereo input pair that can be used primarily by dynamics effects to control levels using a separate signal from one or more other tracks rather than the program signal itself.

A typical example for using a sidechain send is to control a compressor for the bass using the kick drum signal. This way, every time the kick drum sounds, the bass will be lowered.

If an appropriately equipped VST/ MAGIX plug-in (e.g. AM-Munition) and also the internal Advanced Dynamics effect are loaded in a track or master, you can select one or more tracks of the project as a sidechain source in the lower part of the **Sidechain-Input** menu. When a source is activated, a sidechain bus is created in the corresponding track. This bus is not shown as a mixer channel like normal tracks or busses, it only sends to the sidechain input of the plug-in.

> ℹ️ *In the VST standard, there is no special labeling of VST effect inputs specifically as sidechain inputs. Sequoia assumes for all VST-FX that have more than two inputs that inputs 3/4 can be used as sidechain.*

Side chain in: Activates and deactivates the sidechain function.

Side chain solo: With this option you can listen to the sidechain signal. The effect will be switched to "Bypass", and only the input of the Sidechains will be output. When closing the plug-in dialog, the "Side chain solo" function is automatically reset. "Side chain solo" is also ideal for acoustic control when applying the sidechain filter.

Side chain filter: The sidechain signal can be filtered by a Parametric equalizer (↗262).

Show Sidechains
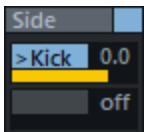
 With a right-click in the **AUX** section of the mixer and Track Editor you can also display the sidechain sends.

An internal sidechain bus is created in the tracks used for the sidechain signal and the sidechain send control is assigned for the destination track. Unlike AUX Sends, the track name is preceded by a **>** character. The signal tap of the sidechain send is preset to "Pre-Fader" (fader display in yellow). This keeps the sidechain signal independent of the channel's volume.

Right-click to switch the selected sidechain to "Direct Out" (fader display in red) or "Post-Fader Out" (fader display in orange), or open the Stereo panorama dialog for panorama definition (↗492) of the respective sidechain send.

> ℹ️ *You can get a good overview of all assigned sidechain sends in the matrix display of the* Routing Manager *(↗332).*

FX I/O Matrix

> ℹ️ *This function is only available with an existing Sequoia maintenance contract.*

The FX I/O Matrix allows you to distribute the channels of the signal that is sent to a plug-in slot to the input channels of the VST plug-in. The output channels of the VST plug-in can in turn be assigned to the output channels of the plug-in slot. This is helpful, for example, to correct incorrect channel assignments for surround effects in surround buses.

You can open the FX I/O Matrix by selecting **FX I/O Matrix** in the **Plug-in** menu of the VST plug-in window (keyboard shortcut: **Ctrl+F6**).
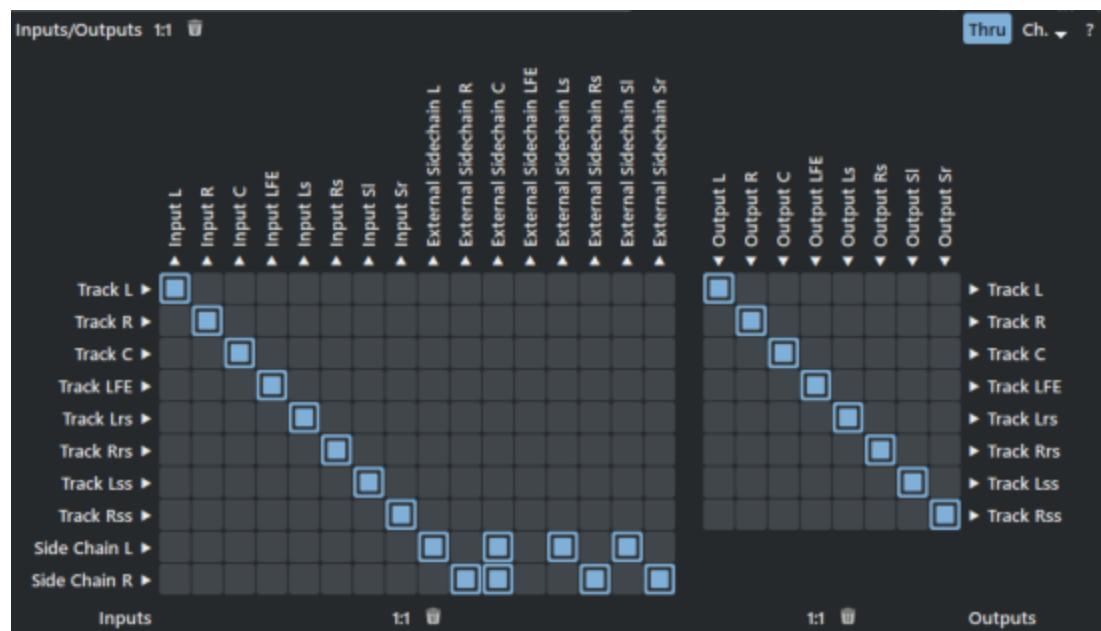
The left-hand matrix allows you to route the track signal (rows) and the sidechain inputsSidechain Input (↗378) to the plug-in inputs (columns). Use the right-hand matrix to route the plug-in outputs (columns) back to the track channels (rows). The sidechain input channels are only displayed if sidechaining is available and activated in the plugin, i.e. a track signal is routed to the sidechain input.

> ⓘ *Sidechain routing is only available with a* Sequoia *maintenance contract.*

There are two buttons for each matrix: **Restore default routing** 1:1 and **Remove all routings** 🗑 . The buttons above apply to both matrices.

Thru: The **Thru** button sends the signal of all track channels that are not routed to plug-in inputs back into the effects chain unchanged. This allows you to use several stereo plug-ins loaded into different plug-in slots of a surround bus or master.
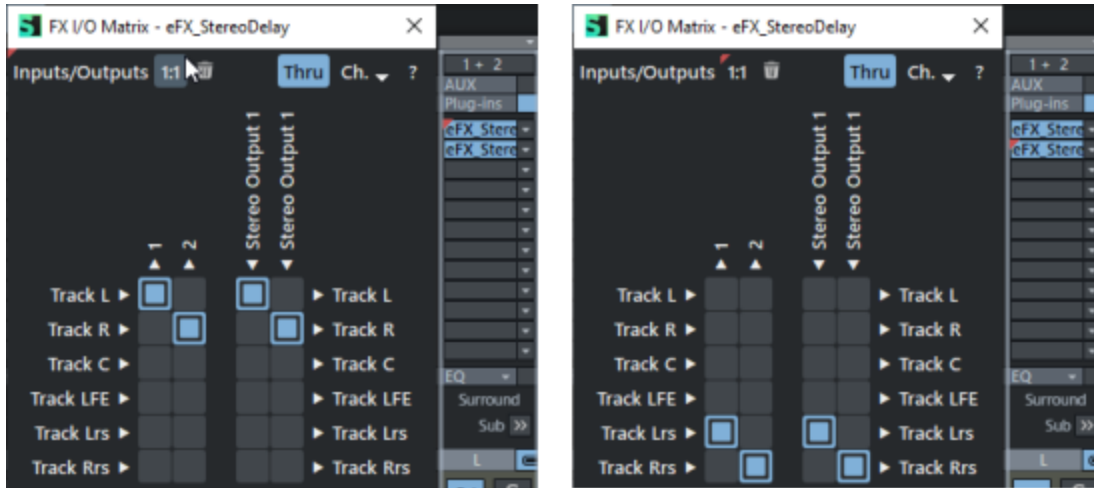
In this example, two stereo delay effects are used in parallel in a 5.1 bus.

Ch.: In the menu **Channels**, the number of channels of a plug-in to be used in the matrix can be limited to mono or stereo. In conjunction with the **Thru** Option, it is possible to use surround-capable plug-ins as stereo plug-ins that are automatically used in a surround bus with the total number of channels of the bus otherwise, and thus use different plug-ins or the same plug-in with different settings on that bus.

> 💡 With the internal effects of *Sequoia*, any groups of channels can be assigned to different instances of the effect, which can be set together or individually. For more information, see Surround effects*Surround Effects (↗530)*

MIDI Plug-ins

The term MIDI plug-in is actually not a "proper" term within the VST standard. Rather, it was coined by users themselves to refer to those VSTi plug-ins that process MIDI input and generate MIDI output from it without generating audio themselves, for example, arpeggiators, chord helpers, or MIDI sequencer plug-ins.

In previous versions of *Sequoia*, the integration of such plug-ins was quite cumbersome: the MIDI plug-in had to be loaded on a separate track, because a track could only contain only one VSTi. Then the MIDI output of the plug-in had to be activated and on another track the actual synthesizer had to be loaded and the MIDI output of the MIDI plug-in had to be selected as MIDI input. The integration of several such plug-ins was not possible at all, since there was only one such "internal" MIDI bus.

As a more comfortable solution there is now the possibility to load VSTis as MIDI plug-ins.They can be loaded in the same track together with a VST instrument, it is also possible to load several MIDI plug-ins (e.g. Chord+Arpeggiator) in a row.

To add a plug-in as a MIDI plug-in, open the track's plug-in browser and select it. Now click on the small triangle next to **Add** and select **Use as MIDI plug-in**. You can also hold down the **Alt**-key while clicking on **Add** (or the **+** icon directly on the plug-in).
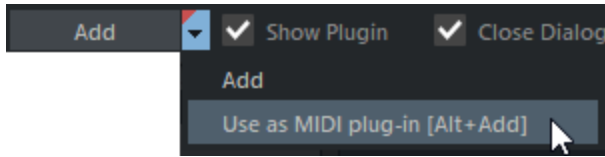
The MIDI plug-in is always inserted before an instrument in the track effect chain. You can now add more MIDI plug-ins, they will be inserted after the first MIDI plug-in before the instrument.

⚠ *The use of MIDI plug-ins is only possible when using the hybrid engine (↗692).*

Recording MIDI Plug-ins

The MIDI plug-in processes the MIDI input signals of the track and the MIDI of the track objects. This means that during a MIDI recording the input of the track is recorded and not the output of the MIDI plug-in. In the example of an arpeggiator, this means the held long notes, not the fast short ones. If you want to record the *output* of a MIDI plug-in, you must activate **Send MIDI** in the **Plug-in** menu of the Plug-in window to be able to select the MIDI output of the plug-in on another track as an additional MIDI input and record it there. (This corresponds to the approach in older versions of Sequoia)

Select a plug-in's MIDI output as an additional track MIDI input by right-clicking the track's Record button and activating it at **MIDI Sender Plug-ins**.
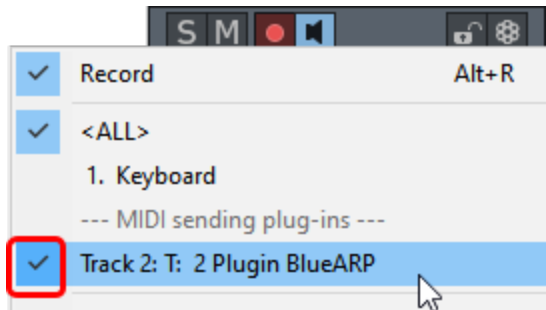
⚠ *Unfortunately, it is currently not possible to select only the MIDI output of a plug-in as the input device on a track . The unprocessed keyboard inputs are therefore also recorded on this track. As a workaround, you can either select an unused MIDI input device or delete the additional notes in the MIDI object after recording.*

ReWire

ReWire is a software protocol that allows you to remotely control another software within Sequoia. This virtually "wires" another DAW software to Sequoia, for example, meaning the ReWire application (client) receives MIDI data from Sequoia and sends your audio output back to and allows direct data exchange. The applications' transport functions, such as play and fast-forward/rewind, are linked by ReWire.

ReWire-compatible client applications (like, for example, Propellerheads Reason) can be integrated into **Sequoia** as software synthesizers.

By default, the ReWire functionality is not activated. You enable it by checking the **Activate ReWire** checkbox in the Program Preferences (key: "Y") at **Effects > VST/Rewire** (↗727). After that, installed ReWire applications can be used as instrument in **Sequoia**.

All ReWire client applications appear as a separate section in the plug-in list of the **Plug-in browser** and are loaded like a software instrument (VSTi). The client application should always be launched after **Sequoia** and should be closed before exiting **Sequoia**.

Some client applications can be opened directly from the host by right-clicking on the name in the MIDI Out slot. Otherwise you have to start your client application manually, it automatically detects the existence of a host application and opens in a special client mode.

The ReWire application can be controlled with MIDI, just like a software instrument. The client application runs, starts, and stops synchronous to the time position of **Sequoia**. Tempo and playback range are taken over.

When using ReWire the MIDI channel of MIDI notes and events is not important and is replaced by the ReWire MIDI bus system. Here, the MIDI object of a track controls a ReWire MIDI bus. This way, all events of a MIDI object in a track refer to this ReWire bus, regardless of which channel number the events originally had. Multi-timbral MIDI objects (like those created in the MIDI file import) cannot be played correctly with ReWire. You can, however, access the ReWire client across multiple tracks on various ReWire MIDI busses.

ReWire supports up to 4096 buses. A ReWire client reports only the actually available MIDI busses to the host (**Sequoia**). You can select the bus (e.g. in Reason: the receiving instrument) within the track in the MIDI Out Channel menu of **Sequoia**.

You can place individual output signals of the ReWire client on different tracks. To do this, directly when loading the Rewire client, use the routing for multichannel software instruments (↗372) or the VSTi Manager (↗331).

> ℹ️ It is not possible to use *Sequoia* as a client in other ReWire applications.

Deactivating VST Plug-ins

Plug-ins can also be completely deactivated so that memory space can be freed up. When reactivating the plug-in, its last state is completely restored.

To deactivate a plug-in, click the corresponding plug-in slot using **Shift + Alt**.

To distinguish the deactivation from the simple bypass, the name is supplemented by asterisk symbols **.

Included Software Instruments

The VST instruments shipped with Sequoia **Independence**, **Revolta 2**, **Vita** and the **Vita Solo Instruments** are referred to as **MAGIX Synths**.

To ensure compatibility with older projects, you can optionally still use some of the old, no longer developed **Object synths** in Sequoia.

MAGIX Synths

Several software synthesizers/samplers are integrated in Sequoia, based on VST plug-in technology:

- **Independence**: a sampler with 70 GB of content.

- **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 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.

ℹ️ *Detailed information about "Independence" can be found in the included PDF document.*

Revolta 2

The Revolta 2 is polyphonic with up to 12 voices. It includes an additional noise generator, step sequencer and is equipped with an extremely flexible modulation matrix. Its effects section of 9 different effects and presets, designed by a famous designer, make its a full-fledged synthesizer for all kinds of lead, sequence, and pad sounds.

Revolta 2 has a number of ready programmed presets. The sounds were created by professional sound designers and already show " out of the box" the great potential of this instrument. However, we would still like to encourage you to try out the various controls to experiment around as much as you like. The sky's the limit to your creativity.

Revolta 2 - Interface

The Revolta 2 interface can 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: Here the volume, panorama position, pitch characteristics ("transpose") and the play modes (POLY, MONO, LEGATO) can be set. GLIDE controls the portamento time.

2. Oscillator section: Two oscillators are available with a smoothly adjustable curve form and a noise generator. The two oscillators can be detuned against each other and frequency modulate with each other.

3. AMP: This is the volume envelope. With this you can control the timing of a sound's volume. A(ttack) stands for the volume increase at the beginning, D(ecay) for the duration of the volume drop to a proportion of the maximum volume specified by S(ustain). R(elease) is the length of time it takes for the sound to fade out. VEL specifies how much the envelope depends on the velocity.

4. FILTER: Here you can switch on a filter which influences the sound. With FILTER TYPE you can select the kind of filter you want to use. "Cutoff" controls the filter frequency and "Resonance" controls the strength of the emphasized filter frequency. "VEL" indicates how much the velocity influences the filter frequency, using "KEY" you can change the filter frequency depending on the note value ("keytracking"). The filter envelope (ADSR slider) influences the filter frequency depending on the time. "env mod" controls the intensity of the filter envelope, "drive" allows you to overdrive the filter.

5. FX1/FX2: Here you can mix in 2 different effects out of a total of 9 available effects.

6. LFO1/LFO2/STEPSEQUENCER: Two LFOs and the step sequencer can be used to modulate single parameters of Revolta 2.

7. Options and modulation matrix: The two buttons open Revolta's options page for general and preset-specific settings and the modulation matrix. In the modulation matrix, modulation sources are connected with modulation targets. Simple modulations like an 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.
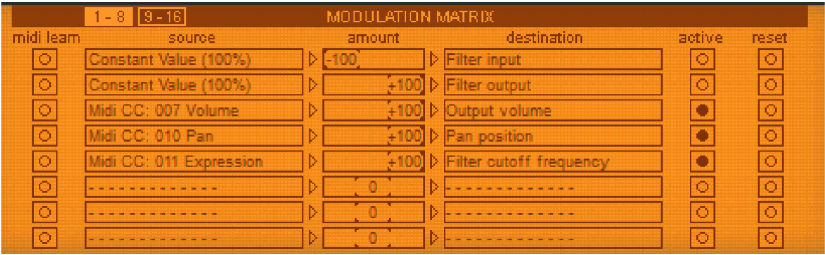
8. VALUE DISPLAY: The value display shows the exact value of the parameter which was just modified. You can also check the usage of the twelve voices.

9. Preset section: Select different Revolta presets here. Each sound can be tested briefly using the loudspeaker symbol; an A-B comparison between two sounds is also possible (e.g. edited and original preset).

Vita Solo Instruments

Vita Solo Instruments are sample players that have an interface specifically tailored to the corresponding instrument.

The basic controls are identical for all these Vita Solo Instruments.

One click on the arrow symbol opens a drop-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.

This controls the general volume of the instrument.

You can turn the instrument keyboard on or off with this switch.

Because these samplers are enhanced for each instrument by tuned effects, the rest of controls function analogously to the already familiar synthesizers such as Vita.

Articulation

Some Vita Solo Instruments have a special feature: In the bass octave (on the keyboard (C0-B0), there are special notes, which let you control the playing style (articulation). An alternative sample set is loaded, which lets the sound even more realistic using various playing styles natural to the instrument, for example note bending and flageolet on the guitar or staccato on the saxophone.

Articulation is switched on and continues until normal articulation is switched on again through the corresponding note (C0).
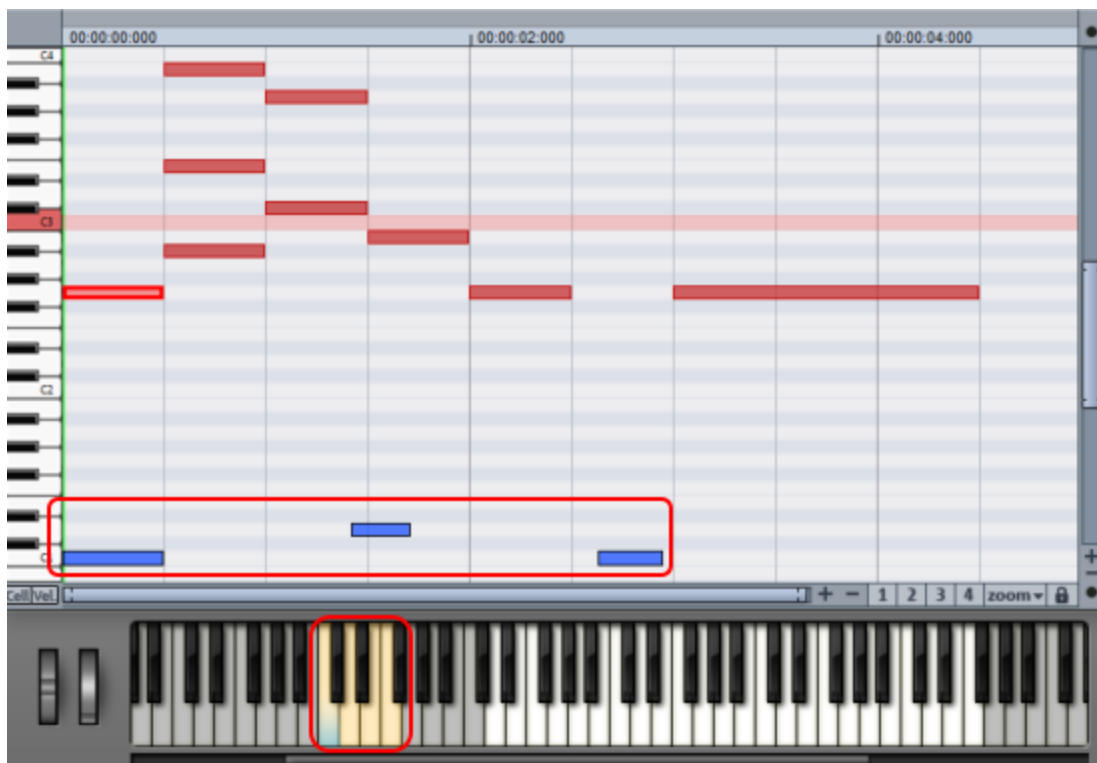
In the keyboard of the instrument surface, the keys for switching the articulation are displayed in a different color. In the MIDI Editor here, you can see a practical application of articulation. The notes from 1:4 are played with a different articulation. Before the last note at 2:2, C1 returns to normal articulation.

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 Interface

1	**Layer selection/Peak meter:** The Vita sounds, also called layers, can be selected here using the arrows. Right-clicking on this display opens the layer menu.
2	**Main parameter**: Set the volume, panorama position, pitch offset ("Transpose"), and the fundamental frequency ("Master tune") here.

③ **AMP**: This is the volume envelope. This allows you to influence the time course of the volume of a sound. A(ttack) stands for the volume increase at the start, D(ecay) for the duration of the volume drop to a proportion of the maximum volume specified by S(ustain). R(elease) is the length of time it takes for the sound to fade out after the note stops.
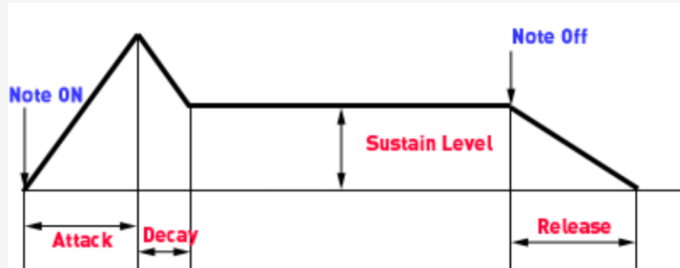
④ **FILTER:** Here you can switch on a filter which influences the sound. With FILTER TYPE you can select the kind of filter you want to use. **Cutoff** controls the filter frequency, **Resonance** the amount of emphasis on the filter frequency. **Velocity** indicates how much the velocity influences the filter frequency, using **Gain** you can balance the volume. The filter envelope (ADSR slider) influences the filter frequency depending on the time.

⑤ **DELAY**: Here you can switch on an echo effect. **Time** controls the delay time and **Level** controls the strength of the echo sound.

⑥ **REVERB**: Here you can switch on a reverb effect. **Time** controls the delay time and **Level** controls the amount of reverb.

⑦ **VALUE DISPLAY**: This always displays the exact values of the parameter that was just adjusted.

⑧ **DYNAMIC SETTINGS**: Usually, the relationship between the created volume and MIDI velocity is proportional. Since some MIDI keyboards need too hard a stroke for loud notes or, conversely, deliver too high a velocity for a light stroke, you can compensate for this behavior with the **MIDI Input Curve**. Using **dynamic** and **dynamic curve** you can manipulate the dynamics of a sound, i.e. the relationship between the loudest and quietest sounds.

⑨ **VOICES**: Here you can control the number of voices played simultaneously. If notes stop playing during fast passages, you can increase the number of voices here at the expense of performance.

⑩ **Keyboard**: This allows you to try out the sounds of the Vita.

Vita - MIDI Specifications

Vita parameter	Control Change #	GeneralMIDI
ModWheel	1	ModWheel
Volume	7	Volume
Pan	10	Pan
Mastertune	14	Mastertune
Sustaining	64	Sustaining
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 are synthesizers loaded as standalone objects that contain both the sound generation and its control in the form of a specially adapted sequencer. Unlike VSTi software synthesizers, which are controlled via MIDI in the same way as MIDI sound generators, synth objects can be moved between tracks at will.

Load object synths via menu **Object** > **Edit MIDI data** >**New Synth Object**. This creates a 4 bar loop object at the current playback marker position. Double-click the object to open the instrument interface of the object synth.

> ℹ️ *The menu item is only available if the object synths are also installed. For installation start the installation manager with menu **Help** > **Download instruments & sounds...***
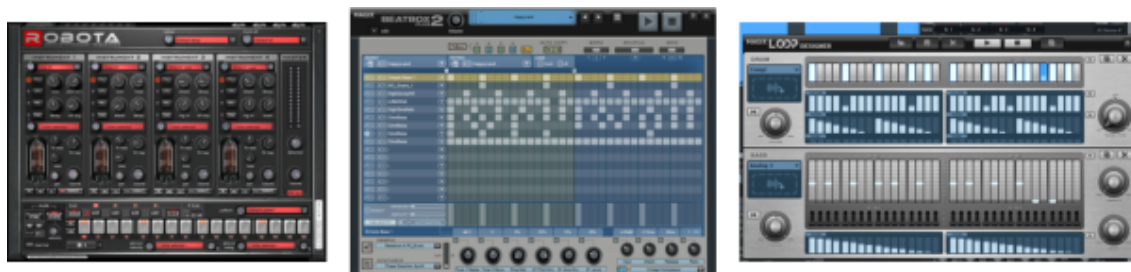
Use the keyboard shortcut **Ctrl + Space** to play a selected object synth solo.

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 machine with hybrid sound generation and step sequencer. The control concept includes sound generation including multi-effect section (one effect per drum instrument), auto-copy and convenient velocity editing.

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.

The BeatBox 2 interface has two states. While closed, you can listen to included sounds or those you've made in BeatBox 2 by yourself without using too much window space.
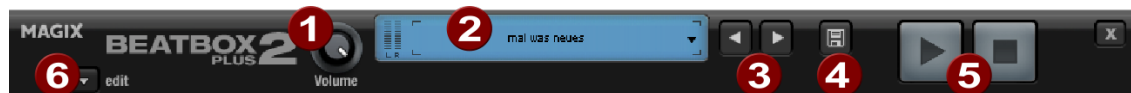
When closed, only the most important controls are displayed:

❶	**Volume controller**: Controls the volume.
❷	**Peak meter and preset name**: The peak meter allows you to visually check the output of the BeatBox 2. Clicking the triangle beside the preset name opens the preset list.
❸	**Previous/next preset**
❹	**Save preset**: The preset includes the drum kit in use, the pattern, and any possible automations.
❺	**Play/Stop**: The playback control in BeatBox 2 starts BeatBox solo, i.e. without playing the arrangement.
❻	**Edit button**: The edit button opens BeatBox 2 for you to program your own beats and sounds.

Maximized BeatBox 2 window

❶	**Drum kit**: This section is where drum kits (collections of different drum instruments) and the individual drum instruments are loaded.
❷	**Selected drum instrument**: The settings in the synthesis section (**5**) and velocity/automation (**4**) always affect the selected drum instrument.
❸	**Pattern editor**: Programs the beat sequence. At the top, various patterns (sequences) can be loaded and saved, and various settings can be made for the view and function of the pattern editor. Beats are programmed in the matrix: A row corresponds to a drum instrument, and a column to the specific time position within 1-4 bars If a box is clicked it triggers the corresponding drum instrument located at that time position.
❹	**Velocity/Automation**: This section has two modes: Velocity and Automation. In the Velocity setting, the velocities for the beats of the selected drum instrument are displayed as bars. In the Automation setting, a sound parameter selected in the Synthesis (**5**) section can be automated.
❺	**Synthesis**: The selected sound parameters and effects settings for the selected drum instrument can be edited here.

In the following, the sections of BeatBox 2 will be discussed individually:

Drum kit

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 can 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. Mute: The **Mute** button mutes a drum instrument.

New drum or effect sounds can by added to the current drum kit by 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. You can drag a complete folder with wave files to the drum kit to create a kit based on those samples.

Context menu

Right clicking a drum instrument always opens 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. It has the standard parameter for all synthesis types and serves as the starting point for your own sounds.

- **Reset automation**: Some 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 for the selected instruments.

Pattern editor buttons
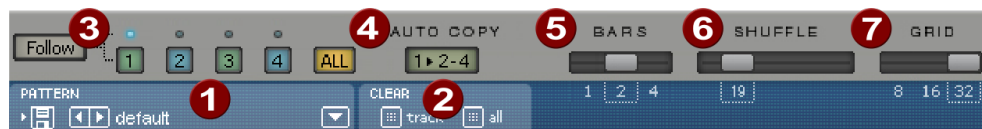
1	**Pattern**: Use the <> buttons to switch through the different patterns. The arrow to the right opens a list of all available patterns, and the save button stores the current pattern.
2	**Clear track/all**: All events for the selected instrument (track) or all events for the pattern (all) are removed by clicking this button.
3	**Bar selection**: The bar you wish to edit can 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** shows 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 makes sure 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 that are set in the following bars are not affected by the Auto Draw function, so that, for example, a variation can be drawn in very easily only in the 4th bar.
5	**Bars**: The maximum length of a drum pattern is four beats. The length can be selected via the small scroll bar above the toolbar list.
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 a bit too abstract – simply try it out, ideally with a pure 1/16 hi hat figure; you'll soon see what the shuffle fader is doing.
7	**Grid**: Set the time resolution for BeatBox here. Choose from 1/8 notes (only for very simple rhythms), 1/16 (default), and 1/32 (for more refined constructions).

Pattern Editor - Matrix
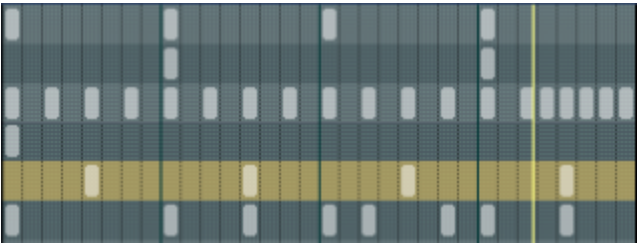
This is the heart of BeatBox. With a mouse click on any position in the drum matrix you can create and delete any drum note. Clicking and dragging draws in a series of notes. Together with the velocity editing options (see Velocity), you can easily create drum rolls.

By holding down the Shift key, a rectangle can be drawn, with which the notes lying in the rectangle are selected (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 clears the selection. The selection is automatically cleared 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 functions

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/coarser

Selected drum instrument

Cursor up/down	previous/next
P	Preview
Enter key	Live input
M	Mute on/off
S	Solo on/off

Velocity

In the Velocity Mode the Velocity/Automation section serves for editing the velocity of the individual drum notes for the selected drum instrument.
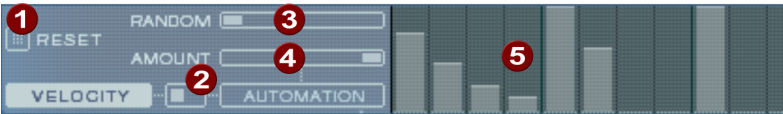
❶	**Reset**: Sets all velocities to 100%
❷	**Mode switch**: Switches the section between Velocity control and Automation (see below).
❸	**Random**: The random parameter adds random variations to the set velocities. This helps make your beats sound more natural.
❹	**Amount**: No function in Velocity mode.
❺	**Velocity levels**: For each set note of the selected drum instrument, the velocity is represented here by the height of the bar and can be edited with the mouse. Multiple columns can be edited at once; see Editing Velocity Values and Automations.

Automation

Every parameter for a drum sound, including effects, can be automated, i.e. can change during a pattern. For example, give your snare drums more liveliness by adding a subtle higher pitch to loud beats, or set accents by adding a reverb effect to only individual beats.

In the Synthesizer section below, select a parameter for automation with the small blue LED above a parameter knob. More about the parameter controls in the Synthesizer section.
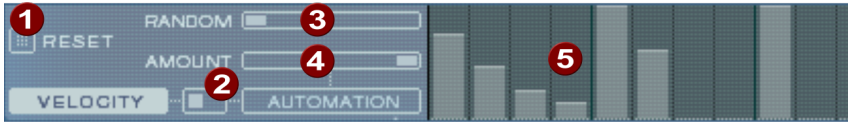
1	**Reset**: Sets all automation values for the selected parameter to 0.	
2	**Mode switch**: Switches the section between Velocity control (see above) and Automation. When a parameter is selected for automation, this happens automatically.	
3	**Random**: The Random parameter adds random variations to 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 controls the total influence of the automation values and the random factor on the selected parameter. With Amount = 0 the drawn automation has no influence at all, with Amount = max the biggest influence. The effects of the Amount controller on the automation values is made visible by a slightly lighter line in the value bars.	
5	**Automation values**: For the selected parameter, automation values can be drawn in here as bars with the mouse. Automation values can also be drawn between the set notes, the sound of the drum instrument will then change during playback.The automation values are added to the original value of the parameter.	

Editing velocities and automation values

Hold down the Shift key and use the mouse to select a number of bars for Velocity or Automation. 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

The **Copy** button copies the selection to the clipboard. If you now drag a selection again within this or any other editor track or automation, you can paste the notes or automation data there using **Paste**.

The three handles allow the velocity or automation values to be edited together.

The middle handle increases or lowers all values.
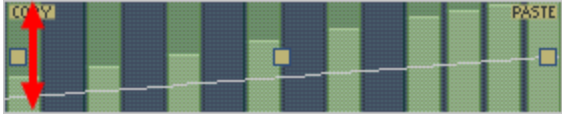
With the right or left handle you can fade the values in or out
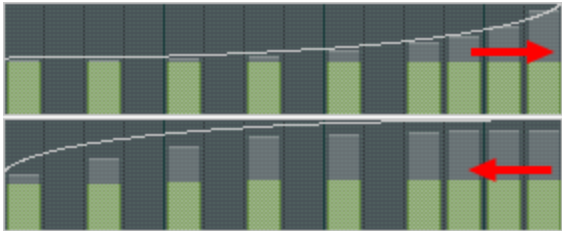
If you move the handles horizontally you can change the curve shape of the transition. A single click in the automation section cancels the selection again.

> **ⓘ** **Note**: *A selection of velocity values is equivalent to selecting the corresponding notes in the Matrix Editor track.*

Synthesizers

In the lower section of the BeatBox you can set the sound of the selected drum instrument.

The synthesizer of BeatBox 2 is composed 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, sampler and synthesizer, is then edited by a multi mode filter. An envelope ("envelope generator) controls time-dependent modulations in all components. The details of the synthesis models can be read in the section Synthesis models.

❶	**Preview drum instrument**
❷	**Save drum instrument**
❸	**Select sample**: Clicking the arrow selects samples from the categories like kick, snare, etc.
❹	**Select synthesis mode**: Switches between the three synthesis models.
❺	**Mix**: Mixing ratio between drum sampler and synthesizer.

6   **Parameter control**: All six sound parameters for a drum sound can be directly adjusted and automated via the parameter controls. The parameters in question depend on the drum sound currently loaded. By clicking the name of a sound parameter, a menu opens for you to add parameter controls to each drum sound's synthesis parameters.

7   **Automation**: This selects the controller's parameter for automation.

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 one of the three synthesis models.
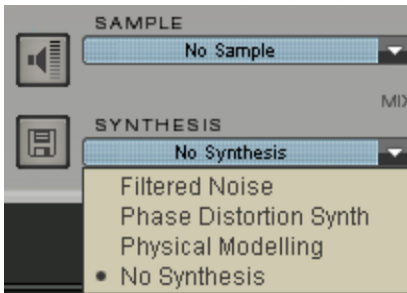
Filtered Noise: White noise is filtered by two band-pass filters with separately adjustable frequency and resonance. This algorithm is suitable for creating synthetic snares and hi-hat sounds.

Phase Distortion Synth: Two oscillators with adjustable phase distortion and variable frequency within wide limits modulate each other (FM/crossmodulation/ringmodulation). 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.

Effect Section

Each of BeatBox 2's drum instruments includes an effects unit which is fed in after the actual sound synthesis and filtering. 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 control**: The effect module has four adjustable parameters, the fourth ("Mix") is permanently available and the others have a function depending on the selected algorithm (see below).
③	**Automation**: This selects the control's parameter for automation.
④	**Effects algorithm**: Click the arrow to select an effects algorithm.

The available effect algorithms:

Mono Delay (Sync / ms)

Simple, monophone delay effect.

Parameters:

- **Time**: Delay time controlled by musical measure (sync) or in milliseconds (ms).
- **Feedback**: Repeats
- **Damping**: Height attenuation of repeats

Stereo delay (Sync/ms)

Stereophonic repetitions, adjustable separately per side.

Parameters:

- **Left / Right**: Delay times, synced or time
- **Feedback**: Unlike the mono delay, no repetitions take place only in the center position of the feedback control. When turned to the left, the delay is of the "ping-pong" type, i.e. the signal is sent alternately to one side so that it bounces between the channels. 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" through modulated detuning of a signal to "thicken" 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 can be understood as the "base" modulation which 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 the 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 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" because air molecules are constantly moving and due to the complexity of reflection processes, 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, simulates 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 can be produced alongside the unavoidable loss of highs which results from "down sampling". Bit resolution is clearly changeable, too.

Parameters:

- **Rate**: Sample rate
- **Crush**: Number of bits

Lowpass: 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". Here a dual-band EQ works on the in and output signals and therefore provides a rich palette of sounds.

Parameters:

- **Drive**: Controls the internal level and thereby the overdrive
- **Low**: amount of bass
- **High**: amount of highs

Analog Filter 12/24dB

An additional filter can be applied to the entire drum sound.

Parameters:

- **Cut-off/Resonance**: Filter frequency and resonance.

- **Type**: Filter type

- **Drive**: Saturation of the filter

Vintage compressor

A compressor with special "analog" control behavior for more pressure on bass drums.

Parameters:

- **Input**: Input level.

- **Attack/Release**: time constants for the 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. It replaces the Drum 'n' Bass machine available in previous versions.

> ⓘ *Tip: Drum'n'bass tracks usually have between 160 and 180 BPMs. The Loop Designer can also be used for other music styles, e.g. for BigBeat (120 BPM) or TripHop (80-90 BPM).*
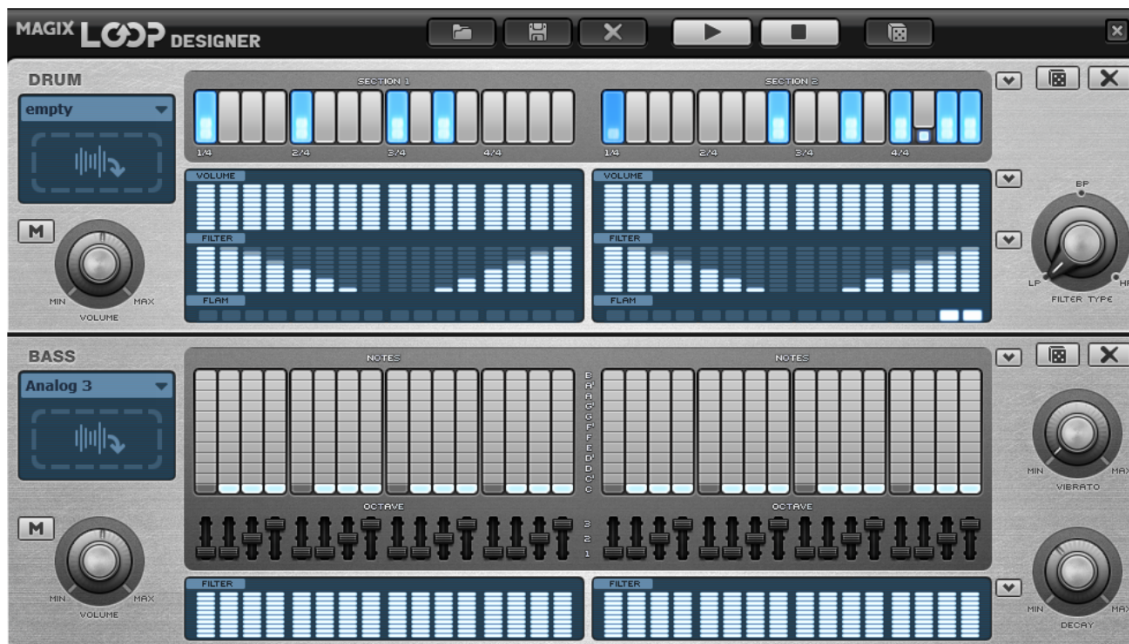
Overview

The upper half of the synthesizer controls the rhythm section, the lower half controls 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

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 gray cells represent the individual sections ("beats") into which the loop is separated.

Pick a different note or playback by left-clicking a gray 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 gray 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 symbol; Play in reverse from this position

Stop symbol: Stop playback

The right mouse button deletes the settings of a step cell; the drum loop is played in its original order.

Clicking the double-arrow button generates a random step sequence. You can change this rhythm as you like.
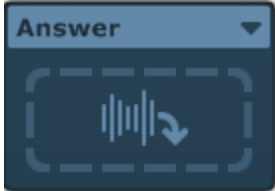
Clicking into the blue field in the left part of the drum section opens a pop-up menu where you can pick the sound of the Drum Loop. When you select another drum loop it will load and be played in the way you programmed it.

> **Tip**: You can also send a loop from a Soundpool or a wave file to Loop Designer. To do so, select the desired loop or file and drag it into the field.

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".

The "Filter type" knob selects the type of the filter sound: "BP" stands for "bandpass", "LP" for "low pass" and "HP" for "high pass".

Bass Section

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.

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.

Furthermore beneath the step rows there are two knobs for sound alterations.

Use the **Vibrato control** to add small variations to the pitch of the bass note.

The **Decay** control determines how fast the sound decays. On the far right, the sound dies out very quickly; on the far left, the sound is permanent.

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. Thereby 1-4 bars are looped in 1/16 or 1/32 steps. Playback positions can be set at each beat position with a click. In Event mode, the beats per instrument are assigned to the beat grid. In Snapshot mode you can automate various sound settings of the instruments.

Structure of Sound Generation

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 complexity somewhat, not all **parameters (1)** can be changed for a sound, but only those for which this makes sense for the selected drum sound (snare, kick, hi hat, ...). There are four selected parameters, each of which is 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, sawtooth or 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 Filter Cut-off, Resonance, Tube, Volume and Panorama are always adjustable.

select	With **Select** you can choose the instrument you want to edit in the step sequencer.
M **S**	Click on **M** to set the instrument to mute, **S** for solo.
	You can preview the instrument by pressing the loudspeaker button.

Master: With **Volume** you control the overall 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 check the output level. If this lights up in the red area, decrease the master volume.

Sound Synthesis - Schematic Illustration

instrument 1 - 4 signal flow

fm / ring modulation

fm/ring decay

freq

envelope generator

FM / Ring oscillator

fm level

ring mod level

waveform generation

osc shape

waveshaper (distortion)

sample playback

oscillator

sin/tri/ saw/ smpl

tune

pitch env

white noise generator

noise

X

"control voltage" (0...f)

envelope

attack

decay

envelope generator

lo-fi section

rectify

rectifier

bit / sample rate reduction

down smple

crush

lo-fi decay

envelope generator

filter

comb filter

multi-mode filter

[delay]

[feed back]

lp / bp / hp

12 / 24dB

cutoff

reso

flt mod +/-

flt mod velocity

envelope generator

flt mod decay

output

compressor

comp

resp

tube

tube circuit

output (+ hp filter + limiter)

output 1 - 4

Detailed Circuit Diagram of a Robota Voice

Description of all control parameters:

- **Pitch Envelope(pitch env)**: controls the amount of pitch drop.

- **Tune**: adjusts the fundamental pitch of the instrument.

- **Oscillator Shape (osc shape)**: The shaper adds additional frequency components to the basic sound of the oscillator by artificially deforming the waveform. A sine wave (shape = 0) can be reshaped up to a square wave (shape = max).

- **Oscillator Waveform**: The fundamental waveform of the oscillators is selected here. You can choose between a sine, triangle, sawtooth or 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**: Attack time of the volume envelope. 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**: Decay time of the volume envelope. The higher the set value, the longer it takes for the sound to fade out.

- **FM/ring modulation frequency (Fm/rng frq)**: Base frequency of the frequency or ring modulation.

- **FM Level (fm lvl)**: FM initially produces a vibrating sound at low frequency, bell-like sound at high frequency and low level, and metallic and finally noisy sound at higher level.

- **Ringmodulation Level (rng lvl)**: Ring modulation generates typical side frequencies.

- **FM/ring modulation decay (Fm/rng dcy)**: Decay time of the FM/ring modulation. With small decay, only the beginning of the drum sound is affected by the modulations.

- **Rectify**: Distorts the audio signal.

- **Crush**: Bitrate reduction. Digital artifacts become audible with higher settings.

- **Downsample (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)**: Decay time of the three LoFi effects for a "dirty" sound. 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)** : Mode of the filter:

  - **Low-pass (LP)** - Sound portions above the cut-off frequency are filtered out.

  - **Band pass (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 in the presets and cannot be changed.

- **Filter Frequency (flt freq)**: Cutoff frequency of the filter.

- **Filter resonance (flt reso)**: Filter resonance. The sound 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 -/+ (flt mod -+)**: Controls how much and in which direction the filter envelope shifts 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)**: controls the dependence of the filter modulation depth on the velocity. If this value is increased, louder beats will generate higher filter curves than quieter ones.

- **24dB**: The filter can operate with a slope of 12 dB or 24 dB. The mode is set in the presets and cannot be changed.

- **Comb Filter (comb filt on)**: A comb filter can be switched on here. This is a feedback delay that produces resonant sounds similar to a plucked string. The delay time and the strength of the feedback are linked to the filter parameters (frequency and resonance). The comb filter is set in the presets 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 time response of the compressor. 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.

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.
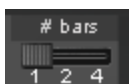
 The maximum length of a drum pattern is four beats. The length can be selected via the small scroll bar above the toolbar list.

The bar you want to edit can be selected with the corresponding **Edit** button. The **Follow** button determines whether the display of the steps of a bar follows the currently played bar.

1>2-4: If more than one bar has been activated as pattern length, the "Auto Draw" mode ensures that the drum notes that are set in the first bar are also automatically set in the following bars. This also makes it easy to create a continuous beat, even with a loop length of four bars. Notes that are set in the back bars are not affected by the Auto Draw function, so it is very easy to draw in a variation in the 4th bar only, for example.

Programming a new drum pattern:

1. Use the slider to select the pattern length.
2. Select **Event** mode.
3. If you are making changes while playing, deselect **Follow** and use the **Edit** buttons to select the bar you want to edit.
4. Use **Select** to select the instrument to be edited.
5. The **Clear bar** button deletes all steps in the current bar for the selected instrument.
6. Turn on the corresponding step buttons. The **velocity** control allows you to set the velocity before you set a beat. To change the velocity of a beat, delete it, adjust the new velocity, and set it again.

7. Repeat the process for the other instruments.

Snapshots

It is also possible to automate the adjustable sound parameters of a drum sound with **Snapshots**. You do this by storing the sound parameters of a drum instrument on the sequencer's step buttons.

Here's how to automate a drum instrument using snapshots:

1. Select **Snap** mode.
2. If you are making changes while playing, deselect **Follow** and use the **Edit** buttons to select the bar you want to edit.
3. Select an instrument and change its sound to your liking. When playback is stopped, you can trigger an instrument with the **speaker** button.
4. Save the sound as a snapshot to one of the step buttons.
5. Now change the sound of the drum instrument and save this setting to other step buttons.

> **ⓘ** **Note:** *Parameter changes are not abrupt, but are internally blended to avoid pops. If two snapshots with extreme parameter differences are too close together, the drum sounds will sound different than intended when the pattern is played.*

6. When playback is stopped, you can use the arrow keys to jump between snapshots. Use **on** to enable snapshot automation.

Groove Control

The secret to "groovy" beats is to play back the individual beats delayed or earlier according to certain patterns. For example, the "shuffle" is used with house beats, whereby each even 1/16 is delayed by a certain amount of time.

Robota features **Groove Velocity** and **Groove Swing** presets. **Groove velocity** presets contain a specific offset for each step of a bar, by which the original velocity is increased or decreased. **Groove Swing** presets contain a time offset for each step by which the step is delayed or advanced. The result is a considerably more lively-sounding drum sequence. The strength of each effect can be adjusted with the **%** knob.
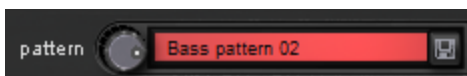
Setups, Drum Kits, Presets and Patterns

An individual drum sound is saved in a "Preset".

The presets of the four voices can be saved together as a "Drum kit"

All sequence information and snapshots are saved in a "Pattern".

Everything together (drum kit + pattern) makes a "Setup".

Load/Save: Select presets, drum kits, patterns, and setups using the endless knobs next to the display panels. To save, click the 🖫 "Save" button. You can enter a new name in the input field.

> **ⓘ 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 applies to the setups, these also contain only the pattern and the drum kit name. So always save (if necessary) in the order: Presets > Drumkit > Pattern > Setup.*

This only applies for creating your own presets. When you save your project, the complete status, synthesizer + sequencer is always saved.

MIDI IN SEQUOIA

MIDI is a standardized command language that helps to control synthesizers.

In addition to the command to play a note (NoteOn, Note Off), there are control commands (Control Change) for internal synthesizer parameters (volume, panorama, the piano pedal, filter settings and others), program change commands (Program Change) to open saved sound programs and for other tasks. These commands are called MIDI events. A MIDI object contains a series of events and the times at which they occur.

Similar to audio objects, MIDI data is represented in Sequoia in the form of **MIDI objects**.

MIDI objects do not contain audio data. They are used to control synthesizers (↗371) that then create corresponding sounds. They are comparable to music notation that is about to be played by a musician.

You can create MIDI objects by loading MIDI files, recording received MIDI data from a MIDI keyboard, or creating an empty MIDI object and drawing in the notes in the MIDI Editor.

Creating, Importing or Recording MIDI Objects

Creating a New MIDI Object

To create a new MIDI object, select **Object** > **Edit MIDI data** > **New MIDI Object** from the menu.

After calling the function a pop up menu opens from which you can select different templates or an empty MIDI object of a preset length (1...8 bars).

The templates are stored in the program data folder `C:\ProgramData\Magix\Sequoia 17\Templates`. You can also quickly reach the program data folder via the link `_ProgramData` in the Sequoia program folder.

You can extend the pop-up menu with your own templates by copying MIDI files into this folder or exporting MIDI objects from the **MIDI editor** via menu **File** > **Export MIDI...** into this folder.
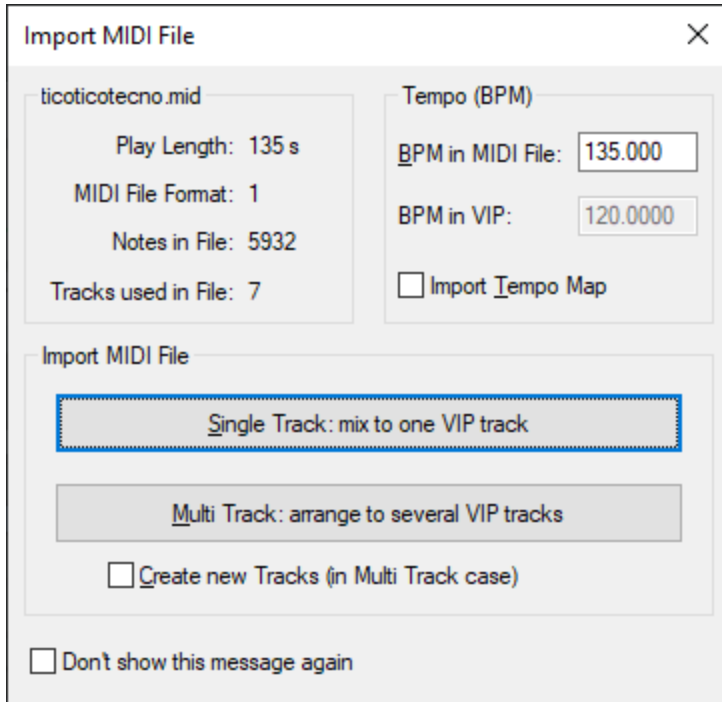
You can also select a range and choose the function Menu **Object** > **Edit MIDI data** > **New MIDI object in range** to create a new, empty MIDI object within the selected range on the current track.

Importing MIDI Files

To import MIDI files (*.mid, Standard MIDI Files (SMF) type 0 and 1) as objects into Sequoia projects:

- Select menu **File** > **Import** > **Load MIDI file...** and choose the file in the selection dialog or

- Drag and drop the files from Windows Explorer or File Manager into the project.

In the following dialog you can set options for the import.

Import MIDI File

ticoticotecno.mid	Tempo (BPM)
Play Length: 135 s	BPM in MIDI File: 135.000
MIDI File Format: 1	
Notes in File: 5932	BPM in VIP: 120.0000
Tracks used in File: 7	☐ Import Tempo Map

Import MIDI File

[Single Track: mix to one VIP track]

[Multi Track: arrange to several VIP tracks]

☐ Create new Tracks (in Multi Track case)

☐ Don't show this message again

Import tempo map: In the Tempo (BPM) section, the first tempo saved in the MIDI file is displayed at **BPM in MIDI file**. (The MIDI file may well contain other tempo information in the case of tempo changes). If the option is active, all tempo information from the MIDI file is transferred to the project in the form of tempo markers. At **BMP in VIP** the resulting tempo of the project is displayed.

Select the option **Single track: mix to one VIP track**, Sequoia inserts the MIDI data into the selected track at the play cursor position as one MIDI object containing all tracks of the MIDI file.

If you select the option **Multitrack: distribute to several VIP tracks**, a MIDI object is created on a separate track for each track of a standard MIDI file type 1. The MIDI objects get the track name from the MIDI file.

If the option **Create new tracks** is active, new tracks will be created for this below the selected track, otherwise existing tracks will be used for the import. Excess tracks of the MIDI file will then not be imported.

> 💡 *Via menu **Object** > **Edit MIDI data** > **Demix MIDI objects by MIDI channels** you can later distribute the imported MIDI data to several tracks separated by channels.*

MIDI files can also be loaded directly into an existing MIDI object via the **File** menu in the MIDI Editor (↗423).

Recording MIDI

> ℹ️ *To find out more about MIDI recording and MIDI recording modes, see MIDI recording (↗99).*

Editing MIDI Objects

> ℹ *This section describes how to edit MIDI objects as a whole. For information on editing the content of MIDI objects, see the section* MIDI Editor *(↗423)*

Arrange MIDI objects

MIDI objects can be positioned exactly like audio objects in the project window, the volume may be modified (middle handle), or fades (in or out) may be added (top right and left handles). They can be split, copied and pasted, and trimmed.

The following differences to audio objects exist:

- While **Split objects** creates two objects for audio objects whose object boundaries are shifted accordingly, the function creates two separate MIDI objects for MIDI objects that do not contain MIDI data outside the object boundaries. Also with the function **Trim MIDI objects**, which you will find in addition to the function **Trim objects** *(↗162)* in the menu **Object** > **Edit** the selected objects will not only be trimmed to the current range limits, but all MIDI data outside these object limits will be deleted.
  This has the advantage that editing several MIDI objects at the same time in the MIDI Editor becomes clearer, since the MIDI events outside the object boundaries, which cannot be heard anyway, are no longer displayed in the MIDI Editor. The disadvantage is that the events outside the object boundaries can no longer be restored by moving the object boundaries.

  > ℹ *You can change the behavior of these two functions by activating the option **Soft Split for MIDI objects** under **System options** > **MIDI**. Then MIDI objects behave during trimming and splitting in the same way as you are used to with audio objects, i.e. the trimmed MIDI data can be restored by moving the object boundaries.*

- Changing the volume of MIDI objects with the middle object handle and fading in and out with the handles on the top right and left of the object is controlled by adjusting the velocity (MIDI note on velocity). Many synthesizers do not change the volume, but rather the sound in relation to the velocity level. If you do not want that, adjust the volume in the mixer instead or by using a controller curve *(↗441)* (usually controller 7).

- MIDI objects always control the synthesizer that is set as the output device in the track where it was placed. If you move or copy a MIDI object to another track, a different synthesizer will be controlled or none at all.

- With MIDI Object Freeze via Menu **Object** > **Freeze Object** > **Freeze Object** *(↗163)*, the audio return signal of a software instrument replaces the MIDI objects with audio objects.

  > ℹ *Note: For this, the audio return signal of your software instrument must be* routed *(↗372) to the MIDI track.*

MIDI Object Editor

The MIDI object editor, similar to the audio object editor (↗183), can be used to edit parameters for MIDI objects and apply MIDI real-time effects such as quantization or transposition.

It is opened with the same menu command (Menu **Object** > **Object Editor**) and keyboard shortcut (**Ctrl + O**) as the object editor for audio objects and is also displayed in the same window or tab in the Dock. Unlike audio objects, however, it is opened with **Shift + double-click** on the object. Double-click opens the MIDI editor (↗423) instead.
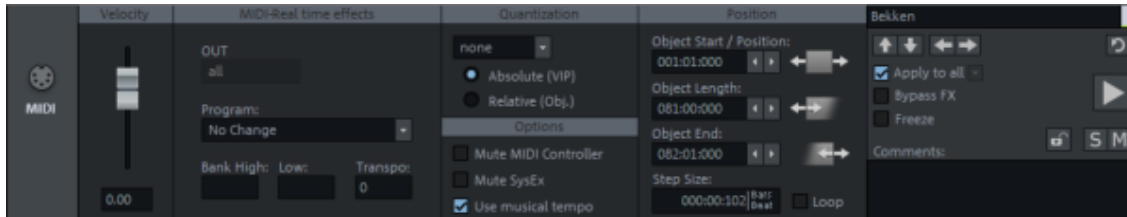
Velocity: Change the velocity value in the object editor to scale the velocity of the notes in the MIDI object. The slider corresponds to the middle upper handle on the MIDI object.

MIDI real-time effects: With these settings the output of the MIDI objects can be changed in realtime, that means settings here do not change the data of the MIDI object:

- **OUT Channel**: If you change the setting from **ALL** to a specific channel, all MIDI data will be output on that channel.

- **Program**: The program change and bank change commands set here are always sent as soon as the object is played.

- **Transp.**: Enter a value here to transpose all notes of the object by the specified amount. This parameter can also be set at the middle object handle in Timestretch/Pitchshift mouse mode, as with audio objects.

Quantize: This allows you to apply a simple, non-destructive quick quantize to the start positions of note events. For a more flexible and detailed adjustable quantization use the quantization function (↗448) in the MIDI editor. However, this changes the notes permanently in the MIDI object, the quantization here can easily be switched off again. With the option **Absolute (VIP)** quantizing is done with respect to the beat grid of the project, **Relative (Obj.)** quantizes relative to the object start.

Options:

- **Mute MIDI Controller**: This option disables all MIDI controller events. It is automatically activated if you have used the command **Extract Controller Curves from MIDI file** in the menu **Object** > **Edit MIDI data** to convert the controllers into automation curves for better editing.

- **Mute SysEx:** Prevents the sending of SysEx data.

- **Preserve musical positions**: This option corresponds to the musical tempo adjustment for audio objects and ensures that when the tempo changes, the time positions follow the changed musical grid. It should only be deactivated in exceptional cases when MIDI events are to be sent at a specific time in the project, independent of the tempo.

Position: The settings here correspond to those in the Audio Object Editor on the tab Fades (↗190). Use the **Loop** option to create a looped object. Unlike audio objects, a MIDI object can only be looped as a whole.

The controls in the right part of the object editor correspond to those in the audio object editor. See Object editor basic functions (↗184).

Extract Controller Curves from MIDI Object

The function Menu **Object** > **Edit MIDI data** > **Extract controller curves from MIDI object** can be used to convert the MIDI controller data (CC) of an object into Automation curves (↗338). This has the advantage that this data can be conveniently edited directly in the Automation Lanes in the project window.
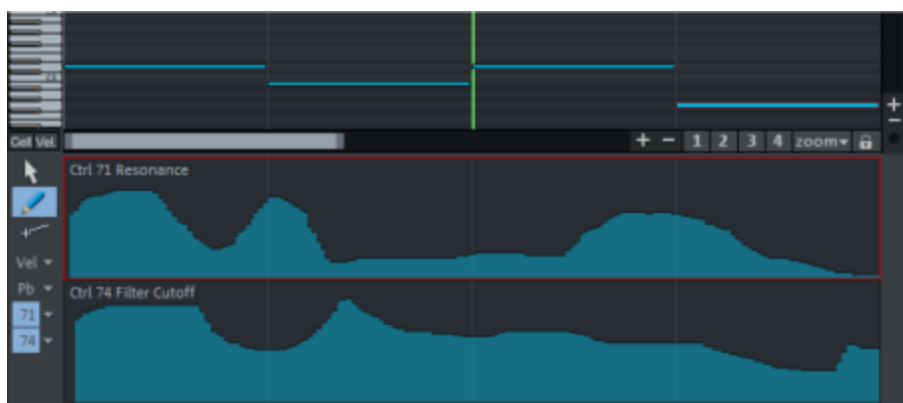
By default, automation curves are displayed in lanes. To view the controller automation curves after executing the function, expand the lanes (keyboard shortcut **Ctrl + Alt + Enter**) and select the curves from the automation menu.
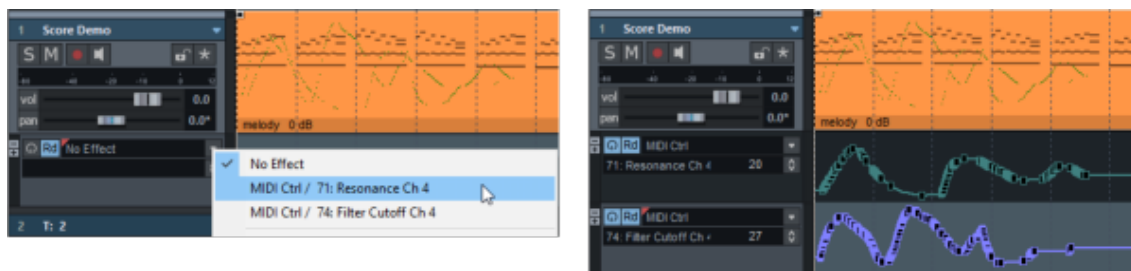
The controller events are not removed from the MIDI object. Instead, the option **Mute MIDI Controllers** is activated in the MIDI object editor so that the controllers are not sent twice.
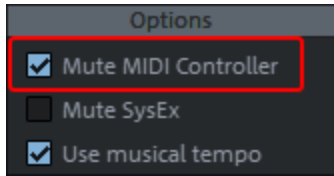
Deactivate this option and the automation curves of the controllers to use the CC controllers from the MIDI object again. However, changes to the automation curves cannot be transferred back to the MIDI object. To finally remove the controllers from the MIDI object, you can use the function **Apply MIDI object and track effects**.

💡 *To update the MIDI controllers in the project during playback in the same way as the other automation curves, activate the option **Recording controller curves during playback** for the tracks concerned in the Track Settings dialog (↗143).*

De-mixing MIDI Objects by MIDI Channels

If a selected MIDI object contains MIDI data on multiple MIDI channels, you can use the function in the menu **Object** > **Edit MIDI data** > **Demix MIDI objects by MIDI channels** to distribute the MIDI data to separate MIDI objects, each containing the data of only one channel. The new MIDI objects are placed at the same time position below the original object.

This function is used to distribute the MIDI data of an imported MIDI file of type 0, in which all instruments are present on one track, separately on tracks according to channels in order to control different instruments per channel.

⚠ *Caution: No new tracks are created in the process. If there are not enough tracks below the original track, the remaining MIDI channels will be missing. Also, existing objects on the tracks below will be overwritten It is therefore recommended to insert the required tracks before executing this function.*

Applying MIDI Object and MIDI Track Effects

Both on the track level (**Track Editor**, section **MIDI** or dialog **Track Properties**) and **in the MIDI object editor** there are various MIDI real-time effects that change the pitch and velocity of MIDI notes during playback.

- **Track**: Transposition, MIDI Velocity Dynamics (↗463)
- **Object**: Velocity, Out Channel, Transpose, Quantize, Mute MIDI Controller

With these effects, you can make MIDI playback flexible, for example, use MIDI Velocity Dynamics to modify the velocity response of your MIDI keyboard as you play it in, or copy a MIDI object and use it transposed in multiple pitches.

If you later want the playback of the MIDI objects to match the notes stored in the objects, you can apply these effects permanently to the MIDI data using the command Menu **Object** > **Edit MIDI data** > **Apply MIDI Object and Track Effects**.

Select the MIDI objects and execute the function. The MIDI data of the objects is adjusted and the effects are reset in the MIDI object editor and in the track.

⚠ *Attention: The function edits only selected MIDI objects, but MIDI track effects affect all MIDI objects of a track. If you have activated MIDI track effects (e.g. transposition) on a track in addition to MIDI object effects, all objects on this track must therefore be selected, since the track effect is reset after the calculation, which means that the unselected objects would now sound different from the selected ones.*

MIDI Editor

The MIDI Editor is used to display and edit the contents of the selected MIDI object(s).

To open the MIDI Editor, **double-click** a MIDI object or select Menu **Object** > **MIDI Editor…**. Once opened it displays the MIDI data from the currently active position over the length of the object or within the selected range.

ℹ *If no MIDI object is selected, it is offered to create one at the play cursor position on the current track. When you select **OK**, the menu opens with templates for new MIDI objects (↗417).*

The MIDI editor can be viewed as floating or docked like the other Sequoia windows.

Due to the abundance of functions, the MIDI editor has its own **menu**. When this section refers to menus (e.g. Edit menu), it always means the **menus of the MIDI editor**, not the main menu of Sequoia!

This menu also has its own keyboard shortcuts for these menu commands, which are only valid when the MIDI editor window is focused. To configure these shortcuts, there is a separate section called **MIDI Editor Shortcuts** in the **System Options** dialog under **Keyboard, Menu & Mouse**. You can quickly access this section of the System Options dialog directly from the MIDI editor, using the **Shortcuts** menu > **Edit Shortcuts…**.

⋮ If the MIDI Editor is placed in the Docker (↗57), you can open the MIDI Editor menu via this button or by right-clicking in a free area of the toolbar.
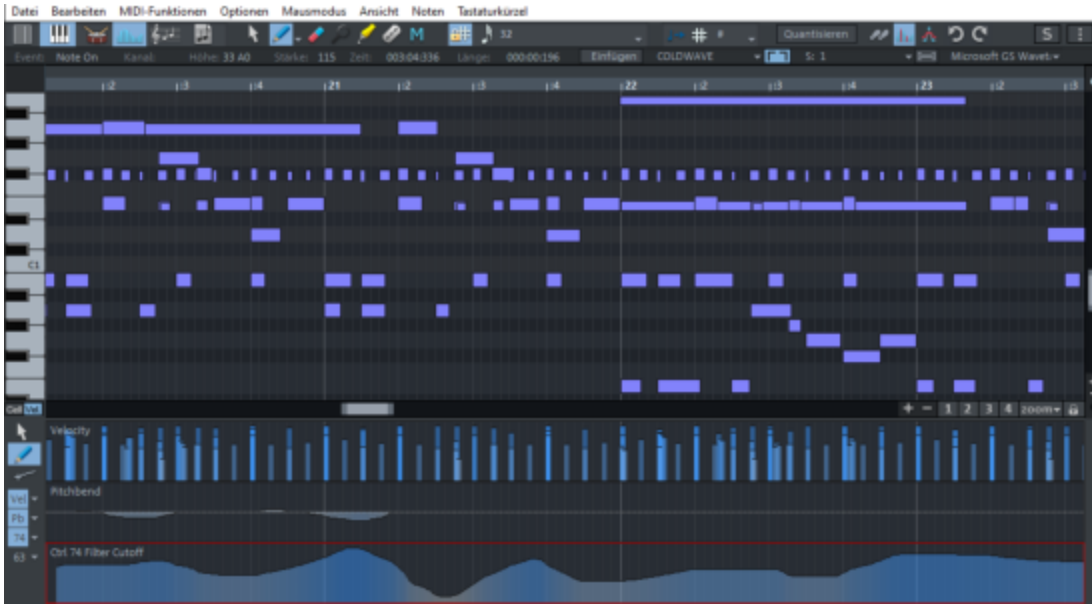
MIDI files can be edited in five main areas of the MIDI Editor:

	Matrix Editor (Piano Roll)
	Drum Editor
	Controller Editor
	Event List
	Score Editor

You can use these buttons at the top of the MIDI Editor or on the **View** menu to show or hide the different panes (Controller Editor/Events List/Score Editor Linear View) or toggle them (Matrix Editor/Drum Editor/Score Editor Side View).

Navigating in the MIDI Editor

Moving the visible part, i.e. zooming (↗105) and scrolling, is done with the mouse wheel and the scroll bars exactly as in the project window. If you change the horizontal section with the scroll bars, the section in the project window is also adjusted accordingly, if you press the **Shift** key while doing so.

Two small circles above and below the vertical scroll bar on the right edge of the MIDI Editor screen indicate by lighting up that MIDI notes are outside the currently displayed vertical screen section.

Selecting Events

Basically, changes like moving or deleting notes apply to all selected MIDI events. Changes to the selection in one editor always apply to all other editors 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.

Select events with mouse and keyboard

Selection	Mouse action
Select the Event	Left-click on event
Select multiple events	Left click on a free spot + drag (only with selection tool).
Add/Remove event to/from selection	Ctrl + left-click on event
Add/Remove several events to/from selection	Ctrl + left click on a free spot + drag (only with selection tool).
Select a range of events	Left-click on first event, Shift + left-click on last event
Select all events	Ctrl + A Depending on the editor, the selection includes notes (piano roll, score), controller events (controller editor) or all events (list editor).
Unselect all	Left click on a free spot
Set or change current event within a multiple selection.	Left-click on selected event

Select current event, deselect all other events	Double-click on event
Selection of all notes of a pitch	Double-click on free space with this pitch or on the corresponding key on the keyboard on the left (piano roll only)
Selection of all notes of a pitch starting from click position	Alt + double-click on free space with this pitch
Selection of all notes behind click position	Shift + Alt + double-click on free space
Select next/preceding note	Arrow left/right

Select MIDI controller events together with notes

With this option, which you can reach in the menu **Options** or via the button, associated CC events are also selected when a MIDI note is selected.

Associated controller events are those events that lie between Note On and Note Off of the note.

Event Display

There are different variants of the color representation of events and different possibilities to filter the events by channel, object and track.

Color Representation

There are several ways in which the event color can be used to display properties of the events. In the menu **Options** you can choose between **Velocity colors** , **MIDI channel colors** or **Track colors**. The default mode is **Use velocity colors**.

General:

- **Unselected notes** have a black border, **selected notes** have a red border.

- Notes that are **outside the object boundaries** in the MIDI object have a white border.

- The **current event** has a bright red, wider border. The properties of the current event are displayed in the edit fields (↗429) above and can be changed there.

- **Muted events** (keyboard shortcut **Ctrl + M**) are displayed crossed out in Matrix Editor.


The fill color of the events depends on the selected option:

- **Use velocity colors**: With this display, **Unselected notes** are drawn in blue, **Selected notes** are drawn in red. The intensity of the red and blue colors indicates the velocity.


Velocity colors: Unselected notes (blue), selected notes (red) and current event (outlined in light red) in the Matrix Editor.

- **Use MIDI channel colors**: In MIDI channel color display, MIDI events are distinguished by color according to their MIDI channel.
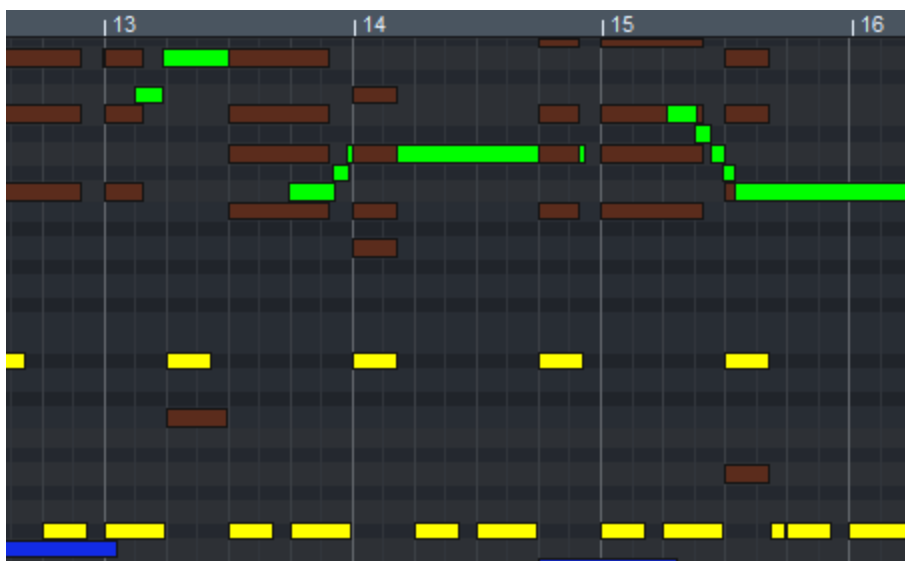
MIDI channel color display

- **Use track colors**: In track color display, the MIDI events are displayed with the color of the track on which the objects are located to which the MIDI events belong. This setting improves clarity when multiple MIDI objects from different tracks are selected.
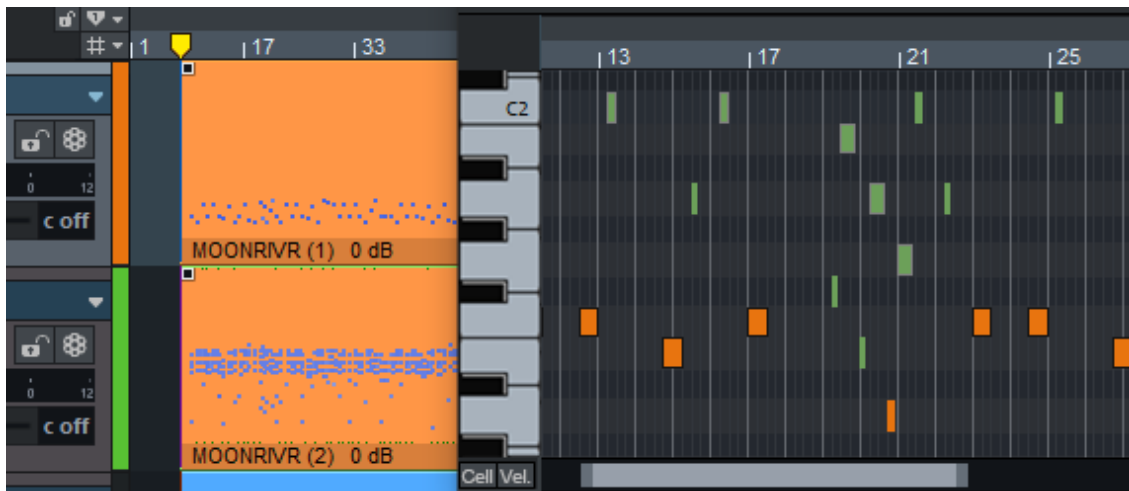

Track color display for MIDI events

If no track colors have been set before, randomly selected colors will be assigned for the track color display in the MIDI editor.

Highlight Current Object and Range

If multiple objects are selected, you can highlight the events of the current object and area. The current object is the last object selected in a multiple selection. It can be changed (as with the current event) by clicking within an existing selection. Also if you select events that do not belong to the current event, the object of these events becomes the current object. If you create new MIDI events, they will always be inserted into the current object.
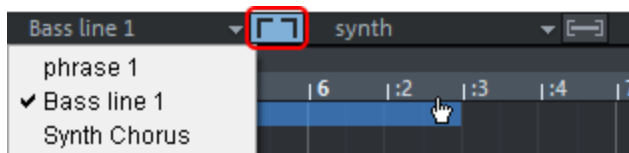

The menu lists all objects selected in the project window. You can select the current object there. If the button next to it is activated, areas outside the activated object are darkened and events of other objects are displayed paler. The current object thus appears highlighted.

Filter Events

By default, the MIDI Editor always displays all events of all selected objects. Therefore, for better clarity and editing, there are filters to lock events against editing. These filtered events will still be displayed, but they will be displayed in a paler color. They can no longer be selected, nor can they be moved, copied or changed.

- **MIDI channel filter**: At **Options** > **MIDI channel filter** you can select specific MIDI channels to be edited in the MIDI editor. In the preset option **All** all channels can be edited. As soon as you select individual channels there, events on all channels that are not selected in the menu will be filtered.

- **Track display filter**:
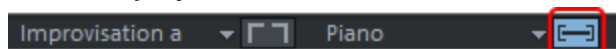
Activate Menu **Options** > **VIP track filter** or this button to restrict editing to the events of the objects that are in the current track.

The current track is always the track on which the current object is located.

If you additionally activate **Hide filtered events** in the **Options** menu, the filtered events will be hidden completely.

> ℹ️ *The* Event list *(↗444) provides additional display filters that only function within the list.*

Editing Selected Events

Menu items and corresponding keyboard shortcuts for editing selected events are located in the menu **Edit**

- **Copy and paste**: You can copy selected events using the common keyboard shortcuts **Ctrl + C, Ctrl + X, Ctrl + V** and paste them to another play cursor position, even between different objects.

- **Delete MIDI events**: Delete selected events with **Del** key or **Backspace**. Press **Ctrl + Del** or **Ctrl + Backspace** to delete all MIDI events.

> ⚠️ *Attention: This affects all events of all selected objects regardless of track and channel filters*

- **Duplicate**: Use the keyboard shortcut **Ctrl + D** to copy selected notes and paste them at the next grid point after the last selected note.
  Another method of duplicating works the same way as duplicating objects: Press the **Ctrl** key, click the selected events and drag a copy of these events to a new position.

> 💡 *Additional commands for editing selected events are located in the menu **MIDI functions**.*

Edit Fields

The properties of the current event can also be changed in the editing fields below the editing tools.
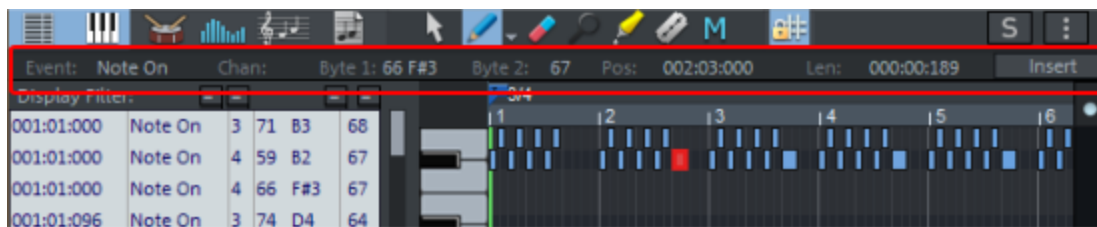
- **Event**: Event type.

  > ⚠️ *Attention: It is not very useful to change the event type when editing existing events. This can lead to unforeseen results. This field is used to enter other MIDI events like Program Changes that cannot be drawn in via the editors. (See below).*

- **Channel**

- **Byte1**, **Byte 2**: The bytes correspond to the two parameter bytes of a MIDI event. For MIDI NoteOn and NoteOff events this corresponds to **pitch** and **velocity**. For byte 1, therefore, the note value corresponding to the numeric value is additionally displayed behind the numeric value. For event types other than notes the parameters have a different meaning, for CC e.g. controller type and value, for some events also the second byte is missing (e.g. Prog Ch).

- **Start time**

- **Length**: This value is only relevant for note events and has no function for other event types. It indirectly controls the position of the NoteOff events associated with the NoteOn events.

  > ℹ️ *The display/entry of times is in bars:beats:ticks. The ticks have a default resolution of 384 PPQ, i.e. 384 ticks correspond to a quarter note.*

- **Insert**: Inserts a MIDI event with the properties set in the edit fields. Normal MIDI events like notes or CC events are easier to create directly in the editor, you only need this button for creating special MIDI events (see below).

Editing MIDI Events

To edit, click and hold down the left mouse button and drag up or down in the fields to increase or decrease the value. If you additionally press the **Ctrl** key, larger value changes will result. You can also enter the value numerically by double-clicking the edit box and entering the desired value.

> ℹ️ *If multiple events are selected, you can change the corresponding values of the events together. If you also hold down the **Shift** key while dragging with the mouse in the edit field for the **velocity** or complete the input process with **Shift + Enter** key, all selected events will be set to the same velocity.*

Defaults for Velocity and MIDI Channel when Drawing in

The input fields can also be used to change the velocity or channel if you want to draw in new MIDI notes with the mouse.

1. To draw MIDI notes on channel 2, draw a note and change the channel in the input field **channel**.
2. If you now draw in further new notes with the mouse, they will all be created on channel 2.
3. In the same way, you can change the velocity with which new notes are drawn.

   ℹ️ *Attention: Only if you change the velocity of a note directly after drawing it, this changed value will be used for the next notes you draw. If you change the velocity of other notes, the velocity of the last note drawn will still be used for new notes.*

4. To clearly set the velocity for new notes, deselect MIDI notes (key **1** for the selection tool and then click on an empty space in the editor). The content of the edit fields can be changed even if no event is selected and the values set then apply to new events.

Entering other MIDI Events Like Program Change Commands (Prog Ch)

You can also use the edit fields to enter MIDI events that cannot be entered via the Piano Roll or the Controller Editor, such as program change commands:

1. Place the playback marker in the MIDI Editor where you want to insert the program change.
2. Make sure that no MIDI event is selected (menu **Edit** > **Deselect** or key **1** for the selection tool and then click on an empty space in the editor).
3. Click on the edit field **Event** and select from the menu **Prog Ch** and set the desired program number at **Byte 1**.
4. Click **Insert** to create the program change.

Step Recording

In step recording, notes are entered using a computer keyboard or MIDI keyboard. However, this is not done in real time, but step by step. You can take as much time as you want for the next note.

The notes are added with a predetermined length and after each entry the play cursor moves one step farther. Step length and note length can be specified by the grid and length quantization values. The note length must not be longer than the step width.

🔲 Use this button or the command **Step Recording** in the menu **Options** to activate step recording.

A red marker indicates the current insert position and the octave in which the notes will be inserted when you use the computer keyboard.

The following keys are used to enter MIDI notes using Step Recording:

Tab	One step forwards (set pause)
Shift + Tab	One step backwards

Ctrl + Up/Down Arrow	Entry octave upwards/downwards
CDEFGAB	Note entry in current octave
Shift	Enter chords: While the Shift key is pressed, you can enter multiple notes without advancing the play cursor. This is how you create chords.

Cell Edit Mode

`Cell` You can switch this mode on and off manually by clicking the **Cell** button next to the horizontal scroll bar.

In Cell Edit mode, you get an alternate view of the MIDI events: the individual time positions of a bar are displayed as a series of cells with on/off states. The actual note lengths of the individual events are no longer taken into account, but a uniform display width is used that corresponds to the set quantization grid. Also the swing and offset settings of the quantization options are made clear by cells of different width and by shifting.

Accordingly, notes are not drawn in this mode by clicking and dragging, but by simple clicking.

The cell edit mode is used for a better overview, especially when displaying MIDI drum instruments, and is particularly suitable for editing MIDI objects in the Drum Editor (↗436), where mostly short, percussive events occur. The display remains limited to the essential information of note start position and velocity.

Velocity mode

`Vel.` If you click the **Vel** button, the events are displayed in velocity mode.

In this mode, the velocities of the events are not represented by their different color intensity, but by the height of the events in the matrix display. The color of the events is always the same in this mode (red/blue or channel/track color).

Click and drag on the top edge of the event to adjust the velocity of the event directly in the Matrix Editor without having to switch to the Controller Editor.

> ℹ **Note**: *For this purpose, the matrix editor must be sufficiently zoomed vertically, since the mouse cursor for setting the velocity is only visible from a certain minimum height of the events.*

In the case of a multiple selection, the velocity change *is applied absolutely*, which means that for all notes the velocity is increased or decreased by the same amount, but the maximum and minimum values (127 or 1) cannot be exceeded.

To apply the velocity change *relatively* to a multiple selection, also press the **Ctrl** key.

To set all selected notes to the same value, additionally press the **Shift** key.

Matrix Editor (Piano Roll)

The Matrix Editor is the default view of the MIDI Editor and is used to edit MIDI notes in a piano roll view.

There are several editing tools (mouse modes) for creation and editing. You can select the tools by clicking the corresponding icons, by the entries in the menu **Mouse mode** and by the keyboard shortcuts **1...9** and **Shift + M**.
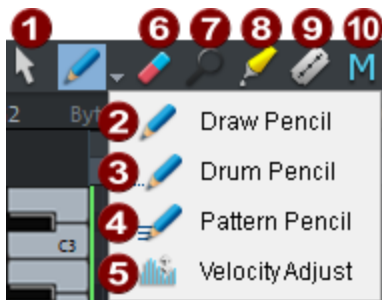
	Tool	Key	Function
❶	**Selection**	1	Click and drag to select multiple notes. For more info, go to Selecting Events (↗425). Clicking on a free area will clear an existing selection.
❷	**Draw Pencil**	2	Left click to draw notes. When the snap is active, the start time and length snap to the current quantization settings. Additionally holding down the **Alt** key overrides an active snap.

③	**Drum Pencil**	**3**	Draw in a sequence of notes. The note length and note spacing correspond to the current quantize settings. + **Alt key**: The pitch of the first note is held for all following drawn notes. Moving the mouse backwards with the mouse button held down removes notes that have just been drawn in.
④	**Pattern Pencil**	**4**	Draw in drum or melody patterns. To create a pattern, draw it in, select the notes, and press **Ctrl + Shift + P** (menu **Edit** > **Create Pattern from Selection**). After that, you can draw in this sequence of notes as a pattern. When drawing in, start at the pitch of the lowest note in the pattern to draw it in at original pitch. However, you can also change the pitch while drawing in. To fix the pitch for single characters corresponding to the first note, additionally press the **Alt key**. Moving the mouse backwards with the mouse button held down removes notes that have just been drawn in.
⑤	**Velocity adjust**	**5**	Click and drag vertically to change velocity values. ℹ *This mode is very similar to the* Velocity mode *below (* Vel. *), but here you don't have to click on the top edge of the notes, but the whole note can be clicked anywhere. Therefore, the notes cannot be moved in this mode.* With multiple selections the value change is absolute, with additional **Ctrl** key relative, with additional **Shift** key all notes are set to the same value.
⑥	**Eraser**	**6**	Clicking on the selected note deletes all selected notes. Dragging the mouse deletes all notes underneath the eraser.
⑦	**Magnifier**	**7**	Zoom mode: left click to zoom in horizontally, right click to zoom out. Additionally press the **Shift** key for vertical zoom. Clicking and dragging over an area zooms into the area.
⑧	**Glue notes**	**8**	Click a note to join that note with the next note of the same pitch. If several notes are selected, all notes of the same pitch are joined together to form a single note.
⑨	**Split notes**	**9**	Selected notes are split into two notes by clicking. If the snap is active, it is split at the next snap position.
⑩	**Mute**	**Shift + M**	Selected notes can be muted or reactivated by clicking on them. The function is also available in the menu **MIDI functions**.

Notes

- Drawing mode (**2**) can be activated from any other mouse mode except Zoom mode by pressing and holding the **Shift** key before clicking.

- The eraser can be activated at any time by clicking and dragging with the right mouse button. You can, for example, add new notes with the pen using a left-click and remove already drawn notes with a right-click, without having to change the tool.

- The Zoom mode can also be temporarily activated by holding the **Z** key. After releasing the key, the selected mouse mode is active again.

- The notes created with the Draw modes get the MIDI channel and velocity from the edit fields (↗429) above the piano roll.

- Keyboard shortcuts for switching between mouse modes (tools) may be freely defined (↗709).

- You can select the next/previous note using the left/right arrows. With arrow up/arrow down the pitch of selected notes can be changed step by step.

Editing Existing Notes

Existing notes can be edited in the same way with the selection tool (**1**), the various drawing tools (**2**...**4**) and the zoom tool (**7**). You can select notes by a single click (see Selecting MIDI events), and click and drag notes to edit their start time, pitch, or note length. Depending on where you click on the note bars several options are available, which are visible by different mouse pointers.

|		
	Change note start time	Click + drag the front edge of the notes, the end of the note is preserved
	Change note length	Click + drag the back edge of the notes.
Shift +	Set note length for all notes of a multiple selection	**Shift** + click + drag back edge of notes sets all notes of the selection to this length.
Ctrl +	Scale note length relatively for multiple selection	**Ctrl** + Click + drag back edge of notes extends the selected notes by the same factor.
	Move note freely	Click + drag the middle of the note. The pitch and start time will be changed.
	Move note start time only	**H** + click + drag the middle of the note. The note is only shifted horizontally and the pitch is maintained.
	Change pitch only	**Shift** + click + drag the middle of the note. Change only the pitch, the position is maintained.
	Delete note	Right-click on note

If you activate the setting **Matrix editor: use different zones for horizontal and vertical movement** in the menu **Options**, there are two shift zones on the note: If you click the notes in the front half, you can only change the position, in the back half only the pitch.

Relative movement in grid (Menu **Options** -> **Relative movement in grid**): If this option is enabled, the distance of an event to the next grid position is preserved during event shifts. This makes it easier, for example, to arrange early entries of instrument groups.

Drum Editor

Clicking this button or using the corresponding command in the **View** menu switches the MIDI Editor to Drum Editor mode.
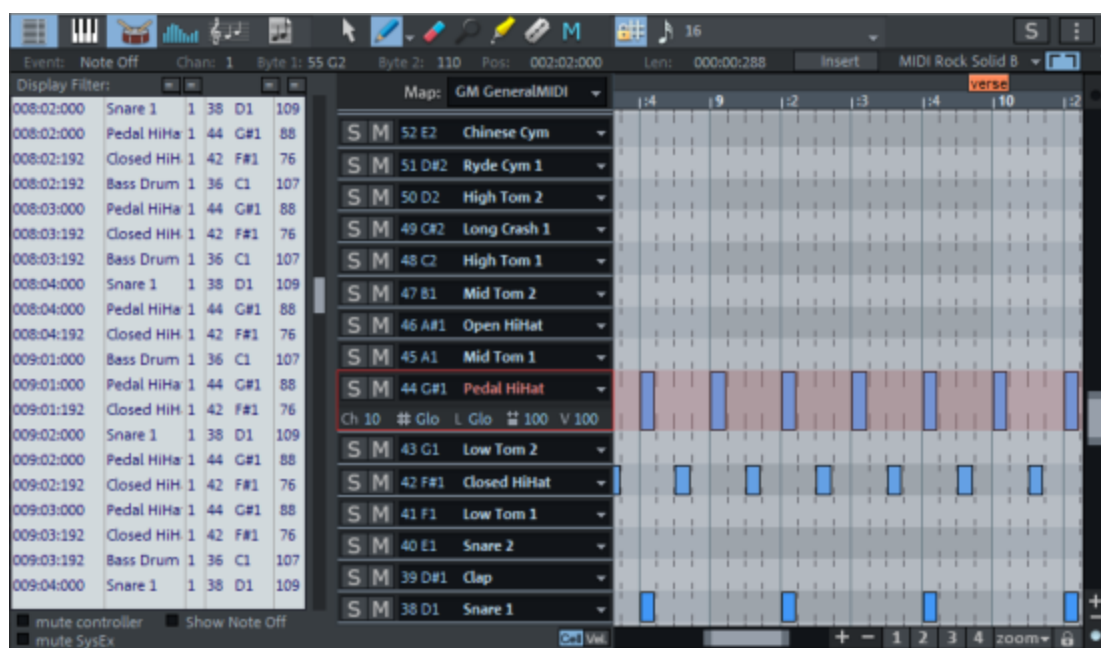
With the Drum Editor you can edit the contents of a MIDI object in exactly the same way as with the Matrix Editor, which also provides the same tools (⤤433) for this purpose. However, its "piano roll" is specifically adapted to the processing of drum sequences:

- For each pitch there is a **drum editor track head** instead of simple piano keys. There you can - individually for each drum instrument - that is, each pitch - assign a name and specify an output note and channel. The grid and quantization settings are individually adjustable for each drum instrument.

- There is also velocity scaling per instrument, so a kind of mixer for the individual drum instruments directly at the events.

- Cell mode (↗432) is used. The display width for each drum event in a cell can also be set individually per instrument in the track head.
- All individual settings for each note can be edited as a whole in a **Drum map** (↗438).

Drum Editor Track Head

Each individual note has its own track head in the drum editor. Here, individual settings can be specified for each instrument. To expand a track head to edit the settings, click on it.

❶	**S/M**	Each individual instrument can be soloed (**S**) and muted (**M**).
❷	**Note number**	The output note for the instrument can be set here. At the time of switching to the drum editor, it corresponds to the current note in the MIDI object. You can change it to use a different drum instrument for the notes of this pitch.

> 💡 *To return the display of the notes to the usual order (low notes at the bottom, high notes at the top), select **Sort Drum Map** from the **Map** menu at the top. This also gathers together the drum instruments used, so that the drum sequence is displayed in the editor in a more compact way.*
>
>
>
> *A drum pattern, before and after sorting*

> ⚠️ *Changes in this field require an application of the mapping when you switch to the matrix editor!*

❸	**Instrument name**	Double-click in this field and enter a name for the drum instrument.
❹	Menu **Grid/quantize settings Colors**	This menu allows you to assign one of eight different colors to the cells of a drum instrument. The dialog for the instrument's quantization options is also opened here. The dialog corresponds to the dialog for the global quantization options (↗449), but the settings for the individual quantization options only apply if an individual grid value is also set for the note.
❺	**K**	Output channel
❻	**#**	**Quantization grid**: **Glo** refers to the global value (↗449).

7 **L**   **Note length**: **Glo** corresponds to the global value

7 🖫   **Display width in cell mode**: With this value you determine the display width of the notes as a percentage of the current grid. The setting has no effect on playback. At "100" the displayed note completely fills the cell.

7 **V**   **Velocity scaling**: The velocity value of each note is multiplied by the % value set here. The scaling is audible, but is not visualized further. The purpose of this setting is to adjust the volume ratios of the individual drum instruments. Software instruments usually include their own mixers.

> ⚠️ *Changes in this field require an application of the mapping when you switch to the matrix editor!*

Drum Maps and Drum Map Editor

Drum synthesizers typically respond to each note of a different pitch with a different sound. Thus, an entire drum kit and a number of additional percussion instruments can be controlled via one MIDI channel. Assigning a MIDI note to a particular drum sound is called "mapping".

Map: **GM GeneralMIDI** ▾   The drum map can be set in the drum editor by clicking in the **Map** menu at the top. You also have the option to set the drum map in the **Track Editor** in the **MIDI** section.

By default, the "GM (General MIDI) Map" is applied to new MIDI objects in the Drum Editor. However, it may happen that your synthesizer uses a different mapping. This means that when you play the drum event, the sound you wish to hear may not be heard (for example, instead of a bass drum, you get a high tom). In this case you have to adjust the mapping.

A Drum map always applies to the entire track and thus always applies to all MIDI objects on a track. You can tell that a Drum map is active on a track when **Map** is visible in the **Transp.** field in the **MIDI** section of the **Track Editor**.

For individual instruments, you can adjust the settings in the Drum Editor track head by changing the note value next to the Solo/Mute buttons.
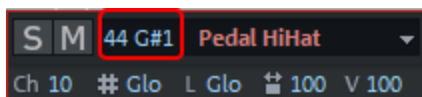
To be able to reuse your changes to a drum map in later projects and also for more extensive changes up to the creation of a completely new drum map, the use of the drum map editor is recommended. You can then also save
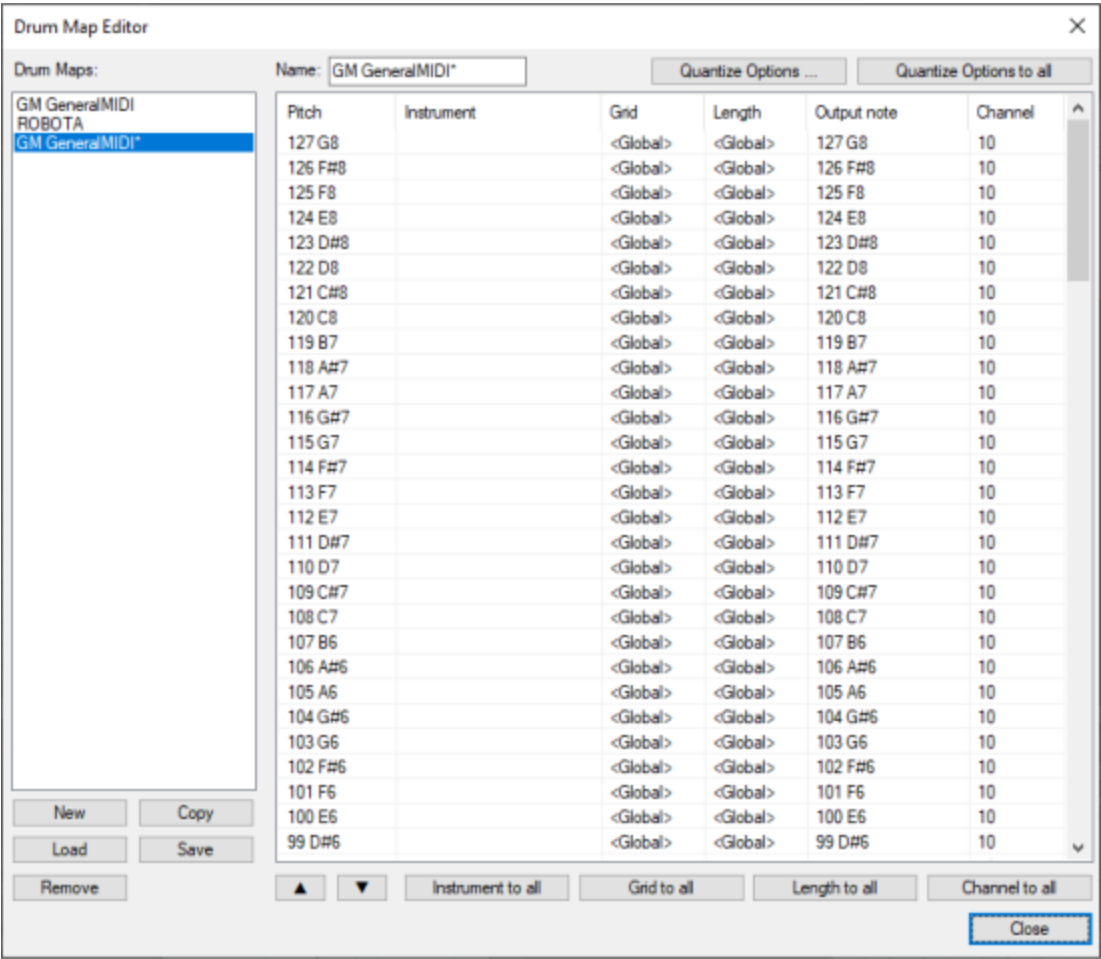
your drum map to a file there. Besides the assignment of the pitch in the editor to the output note, the name of the drum instrument, the MIDI channel and the individual grid and length quantization values are also saved.

In the picture you can see that a new drum map (GM GeneralMIDI*) has been created because changes to the default mapping have already been made in the track heads of the drum instruments.

Via the menu **Map** > **Open Drum Map Editor...** you open the Drum Map Editor with the current drum map. If you want to create a completely new drum map, use the command **Create a new Drum Map**.

ℹ️ A project may contain various different Drum Maps. All Drum Maps saved in the project can be selected via the menu. If you require a Drum Map from a *.map file, you will have to load it into the Drum Map editor first so that it can be shown in the menu.

Pitch	Instrument	Grid	Length	Output note	Channel
127 G8		<Global>	<Global>	127 G8	10
126 F#8		<Global>	<Global>	126 F#8	10
125 F8		<Global>	<Global>	125 F8	10
124 E8		<Global>	<Global>	124 E8	10
123 D#8		<Global>	<Global>	123 D#8	10
122 D8		<Global>	<Global>	122 D8	10
121 C#8		<Global>	<Global>	121 C#8	10
120 C8		<Global>	<Global>	120 C8	10
119 B7		<Global>	<Global>	119 B7	10
118 A#7		<Global>	<Global>	118 A#7	10
117 A7		<Global>	<Global>	117 A7	10
116 G#7		<Global>	<Global>	116 G#7	10
115 G7		<Global>	<Global>	115 G7	10
114 F#7		<Global>	<Global>	114 F#7	10
113 F7		<Global>	<Global>	113 F7	10
112 E7		<Global>	<Global>	112 E7	10
111 D#7		<Global>	<Global>	111 D#7	10
110 D7		<Global>	<Global>	110 D7	10
109 C#7		<Global>	<Global>	109 C#7	10
108 C7		<Global>	<Global>	108 C7	10
107 B6		<Global>	<Global>	107 B6	10
106 A#6		<Global>	<Global>	106 A#6	10
105 A6		<Global>	<Global>	105 A6	10
104 G#6		<Global>	<Global>	104 G#6	10
103 G6		<Global>	<Global>	103 G6	10
102 F#6		<Global>	<Global>	102 F#6	10
101 F6		<Global>	<Global>	101 F6	10
100 E6		<Global>	<Global>	100 E6	10
99 D#6		<Global>	<Global>	99 D#6	10

The **Drum Maps** list on the left lists all the drum maps available in the project. The Drum Map **GM General MIDI** is always available to start off with.
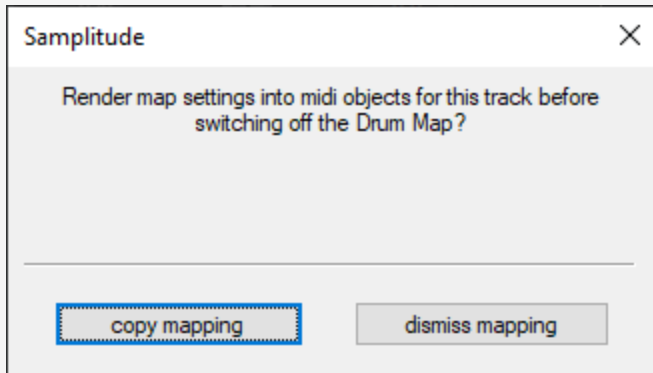
- **New**: This creates a new empty Drum Map.

- **Copy**: A copy of an existing map is created. This way you can quickly create variations of a Drum Map with various note mappings, which can then be switched from within the drum editor.

- **Load/Save**: Use this to save a Drum Map (*.map file). This way you can use a Drum Map you created for a synthesizer in other projects as well. All loaded maps will be displayed in the **Map** menu of the drum editor.

- **Remove**: Removes the selected Drum Map from the project.

Use the **Name** field to rename the selected Drum Map. The settings (mappings) of the individual notes for each Drum Map selected will be displayed below this in table form. Use the arrow buttons below to change the order of the instruments.

- **Pitch**: This is the incoming MIDI note.

- **Instrument**: Displays the name of the drum instrument, for example, "Snare 1".

- **Grid/Length**: Here you can set a grid value for the start time and length quantization of the drum events individually for each instrument. If you leave the value at **Global**, the global values (toolbar in MIDI editor) will be applied.
  In addition, the dialog for the individual instrument's quantization settings (↗449) can be opened via **Quantize options** to specify detailed quantization settings (swing, window...) for the instrument.

- **Output note**: This is the note value to which the drum instrument (the incoming MIDI note in the **Pitch** field) should be mapped.

- **Channel**: You can set a separate MIDI channel for each instrument here. The value set here overrides the channel setting of the MIDI track.

Using the buttons **Instrument, Grid, Length, Channel, Quantize options to all** you can apply the corresponding setting of a selected instrument (except the output note, of course) to all other instruments.

> ℹ️ **Important**: *When you switch from Drum Editor mode back to the normal Piano Roll, you will be asked if you wish to apply the mapping or not.*
>
>
>
> *If you did not make any changes in the mapping (track heads or drum map editor), you can ignore the question and just press Escape or Enter.*
>
> *When **applying the mapping**, the mapping settings are applied to all MIDI objects of the track.*
>
> *An example: You have changed the output value in the track head of the drum editor for the instrument with pitch 35 ("Basedrum 1" in the GM standard) to 36 ("Basedrum 2"). If mapping is applied, these notes will be replaced by corresponding "real" notes with a pitch of 36. Likewise, the velocity scalings are included in the velocity of the notes in the object.*

Controller Editor

The Controller Editor is a graphical editor that allows you to edit the velocity of the notes in the piano roll and other MIDI events such as Pitch Bend, After Touch, Program Change and Continuous Controller events. Continuous Controllers, abbreviated as CC or simply Controller, are used to transfer control values such as filter, volume or pan position.

The Controller Editor is located below the Matrix Editor/Drum Editor and can be shown and hidden with this button or the keyboard shortcut **Alt + V**.

The controller editor is used to display and edit up to four different MIDI controller curves simultaneously.

By default, only one controller lane is shown, in which the velocity is displayed. Pull the dividing line beneath the horizontal scroll bar (**1**) up to enlarge the display of the controller editor. If you now click on one of the controller buttons (**2**), another controller lane is displayed. In the menu next to it ▼ you can select the controller to be edited. Pitch bend, aftertouch or program change can also be displayed.

An asterisk behind the controller name indicates that the MIDI object already contains data for that controller.

The velocity values of the note events (**3**) are displayed as a colored bar in the controller editor. The color scheme corresponds to that of the notes: In velocity colors mode, a darker and higher bar symbolizes larger values; for the other color modes, see Event display (↗426). The bars are located directly below the respective notes.

Also the values of the other controllers are displayed as a bar (**4**), the width of the bar extends to the next event with a different value. Because the controllers typically change more or less continuously in smaller time intervals independently of note events, they appear as ascending or descending ramps. The height of the ramps and their color intensity also represent the last defined value of an event. Selected controller events also appear in red.

The controllers can be edited with the tools (**5**).

> 💡 To make it easier to edit **Pitch Bend** curves, which typically have very small differences in values, you can use **Ctrl**+**Mouse wheel** to zoom vertically into the controller curve.

Combi tool 🔺

(Keyboard shortcut: **Ctrl** + **2**)

The Combi tool lets you select events, edit the values of the selected events, and draw straight or freehand lines.

To select individual events, click the corresponding bar. To modify the controller value, click and drag on the bar end. To select multiple events, click in the controller editor and drag a selection frame. The reference for the selection is the top edge of a bar. Since the selection is two-dimensional, it is thus also possible to select only events in a certain value range.

The selected events can now be edited together with the handles on the selection frame:

❶	**Fade in and out**: Drag the handles to fade controller values in and out.	
❷	**Scale values**: Drag the handle to increase or decrease the values relative to each other.	
❸	**Increase/decrease values**: Drag the handle to increase and decrease all values by the same amount.	
❹	**Increase/equalize value differences**: Starting from the clicked point on the vertical line, the value differences around this point are increased or decreased. Dragging the handle all the way down will result in a straight line at the height of the clicked value, in the other direction it will increase the "amplitude" of an existing curve.	
❺	**Stretch/compress duration**: Drag at the edges of the selection for temporal stretching or compression of the curve.	
❻	**Move selected**: Drag the line at the bottom to move the selected curve.	

> ℹ️ *The display of the selection frame with the editing handles can be deactivated by deselecting the* **Use CC selection controller** *option in the* **Options** *menu.*

The Combi tool is called like this because with the help of additional modifier keys you can quickly switch to the other two tools Because the **Alt** key is used for the drawing tools in controllers for another special function, the modifiers differ depending on whether the velocity or controllers are being edited:

- ■ Velocity:
  - ● **Alt + Click and drag**: Switch to freehand drawing.
  - ● **Shift + click and drag** : Switch to straight line drawing.

- Controllers:
  - **Shift + click and drag**: Switch to freehand drawing.
  - **Ctrl + Shift + Click and drag** : Switch to straight line drawing.
- To delete selected events in the controller editor, press the **Del** key.

Freehand drawing 🖊

(Keyboard shortcut: **Ctrl** + **2**)

With this tool you can draw detailed gradients by clicking and dragging in a controller range. If you drag backwards while drawing, the curve you just drew will be deleted again. Single-click to create a single controller event.

If you edit velocity, no new notes are generated; only existing velocity values are modified. For example, you can easily create crescendi or decrescendi using velocity curves.

Shift + click and drag: This activates the drawing of lines.

Alt + click and drag: Activates snapping to the grid points according to the current quantization settings. Thus, when drawing controllers, only one value per raster unit is generated.

Draw lines ✏

(Keyboard shortcut: **Ctrl** + **3**)

This tool allows you to draw straight lines by clicking and dragging. Click in the controller lane to set the start point, and then drag the end point to the desired location.

If you edit velocity, no new notes are generated; only existing velocity values are modified. For example, you can easily create crescendi or decrescendi using velocity curves.

Alt + click and drag: Activates snapping to the grid points according to the current quantization settings. Thus, when drawing controllers, only one value per raster unit is generated.

Event List

▦ Click on this button, menu **View** > **Show Event list** or keyboard shortcut **Alt + L** to open the event list.

The event list displays all MIDI events as a table. It has two main purposes:

- The list can be used to display and edit also MIDI events that are not accessible via the Matrix, Drum or Controller Editor, such as System Exclusive Messages.

- Using the **display filters** above the table columns, you can filter the displayed MIDI events according to certain criteria in order to select specific MIDI events for editing.

The table lists the events with time position, event type, channel and one or two bytes parameters. For note events, the note name (C1,C#1...) is displayed next to the note number, and for controllers, the type is displayed.

In the Drum Editor, the names of the drum instruments are displayed for note events instead of "Note On".

Time position, channel and the two parameter bytes can be edited directly in the list: Click on the value and drag vertically with the mouse to change the value or double-click on the value and enter the new value with the keyboard, terminated by the Enter key.

> ℹ️ *Note: In order for the keyboard shortcuts for selecting events (e.g. Ctrl + A for Select All or the arrow keys for Next/Previous Event) to refer to the event list, the event list must be focused. You can see this by the red border around the list. To focus, click once in the event list.*

Display filter: For the event type, the channel and the two parameter bytes there are display filters above the columns of the list. This allows you to filter the display of the list according to criteria:

Select an event that has a certain property, for example, a certain pitch, and click the button above the corresponding column. Then only events with the same property (in the example pitch 36 = bass drum) are displayed in the list. Display filters can be combined with one another. For example, you can select all control change events of type 10 (volume) on MIDI channel 6.

Right-click the filter buttons to adjust the comparison condition. The default setting is **=**, which means that all events that do not match the selected criterion will be hidden. With condition **!=** the criterion is applied negatively: All events that meet the criterion are hidden. With **<** and **>** the events are hidden whose properties are less than or equal or greater than or equal to the selected event. For example, you can filter out all note events with a velocity value of less than or equal to 40.

System Exclusive Messages: System Exclusive Messages are device-specific control data for MIDI hardware. You can create such messages using the edit fields (↗429). Double-click on a SysEx event to open a simple text editor where you can edit the data.

System Exclusive blocks consist of a block of data bytes: 0xF0 (System Exclusive Start) followed by two bytes of a unique manufacturer ID, then the manufacturer-specific data and finally the end byte 0xF7. For more detailed information refer to the documentation of the device. With **Mute SysEx** you disable the sending of this data.

Show Note Off: A note actually always consists of two MIDI events, the Note On and the corresponding Note Off event. In the MIDI Editor, Note On and Note Off are always selected together to edit the note length, since this is determined by the time duration between the two events. The Note Off events are hidden in the list by default, with **Show Note Off** they are displayed as well.

Mute Controller corresponds to the option of the same name in the MIDI object editor, see Extract controller curves from MIDI object (↗421).

MIDI Functions

The **MIDI functions** menu contains advanced quantization (↗449) and editing functions for MIDI notes. The commands in this menu are applied to selected events, if no events are selected, to all events.

- **Legato:** If necessary, notes may be lengthened until the next note to be played in legato.
  Keyboard shortcut: **Ctrl + L**

- **Note Quantize (standard):** A standard quantization will be used on all selected notes. If no notes are selected, all notes will be quantized. The default action can be set in the Quantization Settings dialog (↗449). Default Setting is **Start Q**. This function can be accessed via the **Quantization** button in the MIDI Editor. Keyboard shortcut: **Ctrl + Q**

- **Advanced quantize**: In this submenu you can reach the other quantize commands:
  - **Start Q**: The start time of the notes is quantized according to the set grid quantization value. Note lengths remain unaffected.
  - **Start and length Q**: Start time and length of selected notes will be quantized corresponding to the set grid and length quantization values. Quantization is performed with 100% strength ("hard" quantization).
  - **Soft Q**: This command considers the current Soft Q value in the Quantization Options. A Soft Q value less than 100 does not shift the note start time exactly to the grid position, but only by part, according to the strength of the value.
  - **Length Q**: Only the length of selected notes will be quantized according to the set length quantization value. The start time remains unaffected.

- **Quantize note ends to grid:** The end of selected notes will be quantized according to the set grid quantization value. The start time remains unaffected, but the note lengths will change.
- **Undo quantization**:With this command you can undo all quantizations made. This works even after the project has been saved.
- **Quantize Settings...**: Opens the dialog for the Quantization settings (↗449).

■ **Quantize/thin out controller events**: This command replaces selected controller events (↗441) with quantized controllers that replicate the controller course, but only values at the grid points corresponding to the set quantization grid are created.

■ **Humanize**: To make quantized or drawn notes sound more human and less machine-like, you can use the Humanize function to shift the start time of the selected notes by a random value. This corresponds to the button of the same name in the Quantize Settings dialog (↗449), where you can also set the strength of the Humanize function.

■ **Mute notes**: This allows you to mute individual notes and make them audible again.
Keyboard shortcut: **Ctrl + M**

■ **Remove note overlaps (polyphonic)**: Notes are shortened so that notes no longer overlap. However, chords (notes played simultaneously or in very short succession) are recognized and are not corrected.

■ **Remove double notes (monophonic overlaps):** Notes are shortened so that there no longer are any overlaps. This forces monophonic voice leading and removes legato from monophonic synthesizers.

■ **Transform Sustain (CC64) to note lengths**: This function converts sustain pedal controller events (controller 64) into note lengths. All notes which were started after a "Pedal pressed" event (CC64 > 64) will be extended to the "Pedal released" event (CC64 < 64), and the pedal events removed.

■ **Transpose...**: Allows you to transpose the pitch of the selected notes. Enter the shift in semitone steps in the dialog, negative values for a downward shift.

■ **MIDI Timestretching...**: This function allows you to edit the tempo of the selected events. Select one of the options in the dialog:

- **Double/half tempo**: Starting from the start time of the first selected note, the time interval of the following events and the length of the notes are doubled or halved, resulting in a double or half tempo
- **Scale to range length**: The scale factor is determined based on a selected range, so the selected notes are fitted exactly into the range.
- **Stretch manually**: Enter an individual scaling factor.

- **Retrograde (Reverse)**: The selected sequence of notes is played backwards, in other words, mirrored at the vertical center axis.

- **Melody Inversion**: Use this function to mirror the selected notes on a horizontal center axis. The pitch of this axis corresponds to the current note.

  > ℹ️ *The current note is the last selected note in case of multiple selection, it is displayed highlighted. You can set the current note in an existing multiple selection by clicking on it.*

  The current note remains unchanged, but for all other notes the direction of movement of the melody is reversed, i.e. upward movements become downward movements and vice versa.

  > ℹ️ *If no notes are selected, i.e. no current note, the mirroring of all notes of the object is done at the average pitch of all notes $P_{min} + ((P_{max}-P_{min})/2)$.*
  >
  > *Example: Highest note in object P_{max} = 74, lowest note P_{min} = 42*
  > *42 + ((74-42)/2) = 42 + 16 = 58*

- **MIDI Velocity Dynamics…**: Offline version of the MIDI Velocity Dynamics (↗463) effect.

Quantize and Snap

The quantization grid in the MIDI editor serves two functions:

- When drawing in and moving MIDI events, the grid ensures that the notes snap to the correct positions, analogous to the snap (↗79) in the Project window.

- Irregularities when recording MIDI can be evened out with quantization. In doing so, the note starts and/or lengths are moved to corresponding grid positions. In addition, you can use functions such as **Soft Q** or **Humanize** to make the quantization less mechanical and adapt it to the rhythmic properties of the project (**Swing**).

Snap

You make the most important settings for the grid in the two fields of the toolbar in the MIDI Editor.

Use the button in front to activate or deactivate the snap. When snap is active, the positions and lengths of the notes snap only to multiples of the grid values when they are drawn in and changed with the mouse.

> ℹ️ *The note length snaps according to the length quantize value, even if the start position is not on the grid (relative grid).*

If you hold down the **Alt** key while moving or drawing MIDI events, the snap is temporarily overridden.

In the two menus behind it, you set the grid value for the start position and the length. The setting **Beats** at the Start Position causes quarter notes to serve as the grid unit for time signatures based on quarter notes, and eighth notes for time signatures based on eighth notes. The snap therefore follows the time signature and follows any time signature changes. **#** in the field for the length quantization value means that the length quantization value should correspond to the set start quantization value.

The grid display in the MIDI editor adapts to the set start quantize value. If a swing value (↗451) is specified in the quantization options (which are also the basis for the snap), this is also reflected in the grid of the MIDI editor, i.e. the spacing of the grid lines is alternately different.

Quantization

| Quantize | By clicking the button **Quantize** in the upper right corner of the MIDI Editor or the keyboard shortcut **Ctrl + Q** you quantize all selected MIDI notes with the default quantization.

The preset default quantization is **Start Q** and causes only the start time of the notes to be shifted to the set grid values.

For further control over quantization and to select a different default quantization (Start and Length, Length, or Soft Q), open the Quantize Settings (↗449) dialog (see below) by right-clicking this button or using the keyboard shortcut **Alt + Q**.

All quantization functions can always be reached via the menu **MIDI functions** > **Extended quantize**.

If no notes are selected in the MIDI Editor, all notes of all selected MIDI objects will be quantized.

> **i** *Notes:*
>
> - *You can also perform quantization of all selected MIDI objects without opening the MIDI Editor, corresponding menu commands are located in the menu **Object** > **Quantize**.*
>
> - *If the button **Input Q** is active in the section **MIDI** of the **Track Editor**, MIDI recordings are already quantized during recording.*
>
> - *All quantizations are virtual. With **Undo Quantize** (Menu **MIDI Functions** > **Extended Quantize**) you can reset all notes to their original positions. This works even after the project has been saved.*

Quantization Settings

You open the quantize settings in the MIDI Editor menu under **MIDI Functions** > **Extended Quantize** > **Quantize Settings…** or in the main menu of Sequoia under Menu **Object** > **Quantize** > **MIDI Quantize…** or with the keyboard shortcut **Alt + Q**.

The dialog with the settings is not "modal", i.e. you can leave it open if you wish, and test certain passages with different settings.
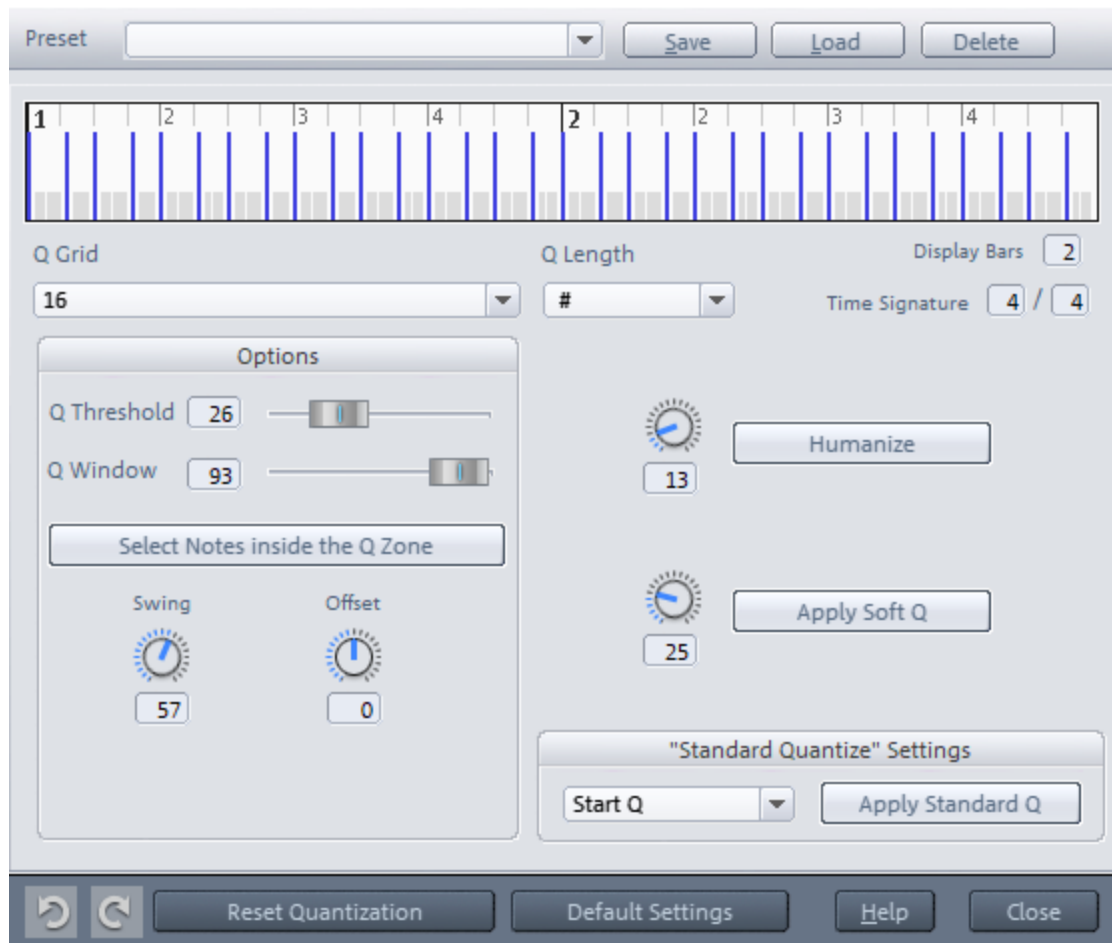
At the top of the dialog, the selected quantization settings are visualized. The blue strokes indicate the target positions for the quantized notes, i.e. the snap points, the length of the strokes corresponds to the velocity for Groove Templates (⤷452). The gray areas around them indicate the quantization window, i.e. the time positions that are affected by the quantization.

At **Display bars/Time signature** you can define how many bars in which time signature the settings window displays. The bar labels of the quantization grid depend on the set time signature.

Q grid/length: Quantization grid and length, i.e. the target points within the bar to which the note start times and/or lengths are moved. Both fields match the fields in the MIDI editor's toolbar. With the value **#** for the length parameter you couple the length quantization value to the (start) quantization grid value.

Q threshold/Q window: The Q Threshold/Q Window parameters let you limit quantization to specific notes to preserve the naturalness of a MIDI recording.

- With the **Q Threshold** parameter, notes that are already close enough to the next snap point are excluded from quantization. Only notes that are far enough from the grid will be quantized.

- Conversely, by reducing the size of **Q Window**, notes that deviate too much from the grid can be excluded from quantization. This makes it possible, for example, to quantize quarter or eighth notes and preserve the in-between sixteenth notes if you reduce the size of the window accordingly.

In summary this means: Events less than the **threshold**-distance or more than the **window**-distance from the grid point will not be quantized. The time range affected by the quantization is indicated in gray in the dialog graphic.

For control purposes, the button **Select notes within the Q zone** can be used to select all notes that would be affected by the quantization.

Swing: With this value you set a swinging style of playing. Here, the distances between every second grid point are of different lengths, corresponding to a stressed/unstressed playing style.

The number is a percentage. A swing of **50** corresponds to a straight play, all grid distances are of equal length (50% to 50%). This is the default setting.

A swing setting of **67** corresponds to the triplet style of playing, typical of blues or swing music. In this case, the double note length (i.e., an 1/8 in a 1/16 grid) is divided into three time sections, the 1/16 notes are alternately given 2/3 (i.e., 67%) and 1/3 (33%) of the length of the eighth note, in other words, a 1/8 triplet in which the first two notes are tied.

A swing setting of **75** corresponds to a dotted note, two 1/8 notes become a dotted 1/8 and a 1/16 note.

💡 *The characteristic swing settings (shuffle) for electronic dance music are between 50 and 66.*

Offset: By changing the value for the offset, you shift the entire quantization grid. Negative values move the quantization to the left, i.e. backward in time; positive values move to the right, i.e. forward in time. The maximum of 100 corresponds with an offset distance of half the grid width.

The display of the blue snap points in the dialog and the grid in the MIDI editor follows the change of this value.

Humanize: The **Humanize** parameter gives you another variation option by allowing you to randomly arrange notes up to a specified distance on positions around the exact quantize value. The setting occurs in % of a 16th note. The specified value thus sets the maximum possible distance of the quantized notes from the exact quantization position.

Soft Q: Soft Q is a quantization where the events are not shifted exactly to the grid point, but only partially, how much is determined by the Soft Q value. 100 shifts the event exactly to the quantization grid point, 50 shifts the event to the mid-point between the current position and the quantization grid point, and 0 means no shift at all.

The **Apply Soft Q** command takes the current strength value in the quantize options into account. The standard quantize command always occurs at 100%. In this manner, you can always select between approximation (soft) and hard quantization without having to adjust the quantization options every time.

Standard quantize settings: From the list, select the quantize action to be performed when the **Quantize** button is clicked. For information on the various quantization commands, see MIDI functions (↗446).

With the two buttons you can undo or redo the last quantization performed.

Reset quantization: All notes will be reset to their original positions.

Default settings: The button resets the options to the default settings: **Q threshold**=0, **Q window**=100, **swing**=50 and **offset**=0.

Presets:

Here you can select from a number of presets:

- **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 events that are within a range of 25% of the grid width left and right of the grid point will be quantized.

- **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", 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 forward a 16th note

- **8th offbeat**: The quantization grid's timing is moved forward a 8th note

- **New groove** and **More life for hi-hat** are provided groove templates

Of course you can create and save your own settings as presets.

Groove Templates

A groove template is a freely defined quantization grid. You can apply a rhythmic pattern on all selected MIDI events or Audio objects with Groove Templates. This allows you to make static MIDI patterns more lively, quantize on existing audio drum loops, or implement special metrics like dotted grid values. For MIDI objects, the velocity and note length can optionally be changed by the groove template in addition to the position. It is also possible to define a rhythmic pattern for drawing in MIDI notes.

A Groove Template enables a freely-definable grid. Usually it has a length of one to four bars and is repeated cyclically. However, the length of the groove templates is basically freely definable. So you could also create a grid based on the bass drum track of an entire song, and then use the MIDI drum pen to draw in a bass that plays exactly to it. The length and start of the groove template are always set to whole bars.

You load the Groove Templates from the Grid Quantize vakue menu on the toolbar. They are located there below the regular grid values.

In the MIDI Editor, you can create new groove templates by yourself by selecting the MIDI events to be used as templates and then choosing the **Edit** menu > **Create Groove Template from Selection** command. The groove template is quantized to whole bar boundaries. So when you create a groove template, make sure it contains enough

notes for a whole measure. Or set a corresponding time signature beforehand via tempo marker, for a template with e.g. the length of 2 eighths set a 2/8 time signature first.

In the file selection dialog you can give your Groove Template a name. Subsequently the Groove Template is immediately active and appears in the grid quantization list.

Additional to each note start position both the note length and velocity are saved in the Groove Template. In the **Quantization Settings dialog**, after selecting a groove template, there are two additional options, **Groove Velocity** and **Groove Note Length**, which determine how much these values from the groove template are included in the quantization. The parameters **Swing** and **Offset** are not available when a groove template is used.

You can also create groove templates from audio objects by selecting the audio object you want to use as a template and opening the Audio Quantization Wizard (menu **Object** > **Quantize** > **Extended Audio Quantization** and in the wizard, run the function **Detect Transients** and click **Create Groove**. If you select a range beforehand, only the AQ markers within the selected area will be considered for the template creation.

Score Editor

The Score Editor is integrated in the MIDI Editor and displays the MIDI data of one or more MIDI objects as sheet music. If you move or lengthen MIDI notes in the Matrix Editor, this change will be visible in the score; if you add new notes in the Score Editor, corresponding MIDI note events will be generated in the object.

Each track can contain staffs consisting of a maximum of 48 staves. When editing objects from multiple tracks, the score is composed of the staffs from each track. The staff of a track can be used as an instrument or instrument group within a score. You can get the full score from this by displaying MIDI objects of all tracks in the Score Editor at the same time. A voice excerpt can be obtained by showing the track of the desired instrument or instrument group only in the score editor.

There are two different views of the Score Editor in the MIDI Editor, the **Linear** mode and the **Page** mode.

Linear View Mode

   This button opens the Score Editor in linear view mode.

The linear view can be displayed in parallel with the matrix (piano roll) and velocity display (controller editor). In this mode it is possible to edit the MIDI data directly in the score.

> **Note:** *With parallel note and velocity display, only the velocity values of the notes of the selected stave are displayed in the Velocity Editor (recognizable by its clef highlighted in red).*
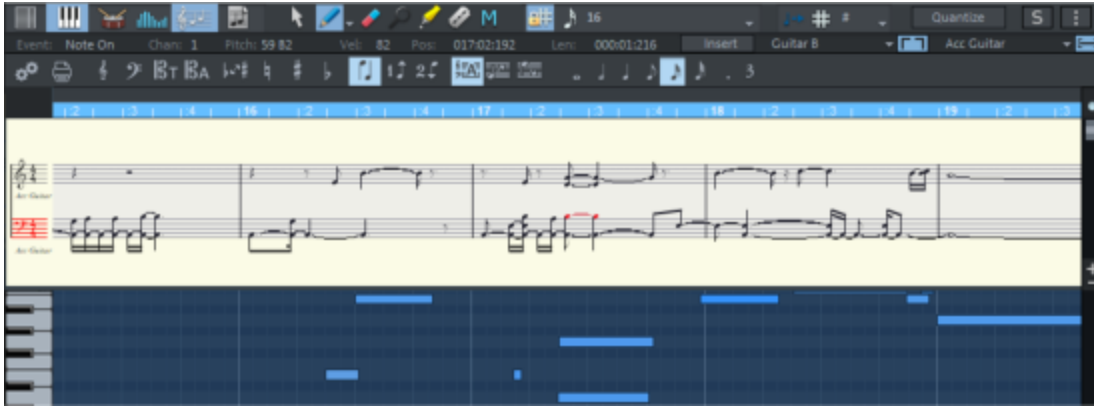
The height of the linear note view can be adjusted by moving the line between the Note and Matrix Editors. The horizontal section can be set above the scroll bar, parallel to the matrix view of the MIDI notes.

Sequoia automatically adjusts the notation symbol size to the selected section. (**Score** menu > **Automatic Zoom**).

Page View Mode

 This button opens the Score Editor in the Page Mode view.

Page mode displays the notes as if they were on a score sheet and also serves as a print preview. In this mode you cannot enter new notes or make changes to existing note events, but you can optimize the graphical representation of the notes on the score sheet. To do this, you can select notes and assign them to the different staves or voices within a stave.

Scrolling in the page view: Use the vertical scroll bar to scroll from page to page. With **Ctrl + mouse wheel** you can zoom in and out.

> 💡 *Tip: Since the page mode also serves as a print preview, you should first set the page format (in the* **Page Settings** *tab of the* **Score Properties** *dialog) to the paper format of your printer before you start designing pages for printing, since the display depends on the page format. You will then see your score exactly as it will be printed.*

Editing MIDI Notes in the Linear View

Select notes: Single notes are selected by clicking, multiple notes by clicking with **Ctrl**- or **Shift**-key. You can select a group of notes with a selection frame.

Change the pitch, velocity or length for one or more selected notes: The data values of the current note are displayed in the info line above the note view. Changing a parameter may have a comparable effect on all selected notes, just like in the matrix editor.

Move and transpose: To move notes, drag them to the desired position. The info line gives you an orientation about the change of pitch and position. The step size during moving is determined by the quantization grid in the MIDI editor.

> ⓘ **Note**: In "Page" mode, notes may not be drawn with the pencil or moved or copied with the mouse. However, you can use the corresponding menu commands and keyboard shortcuts for this purpose.

Copy: Select the desired notes and copy them by holding down the **Ctrl** key and dragging the mouse to the desired position. Or you can use the copy functions of the menu **Edit** in the MIDI Editor.

Insert new notes: Use the mouse mode **Draw Notes** (pencil icon) to draw new notes. Click the desired position, hold down the mouse key and correct the position and pitch as required. When you release the mouse button, a new note is inserted. The length of the note corresponds to the selected length quantize value. To quickly select note lengths, you can use the buttons above the notes.

New notes are always inserted into the active stave. For example, to insert a note on the bottom staff (left hand) in a piano system, first click on the clef at the front of the bottom staff.

 Only notes that correspond to the current key will be inserted. If you want the note to have an accidental, use the buttons for **accidentals** or the natural sign.

> 💡 *You can move the note chromatically in semitone steps using the arrow keys. This way, an inserted **F** in C Major may be transformed into an **F#** by pressing the **Up** key.*

Enharmonic shift: Sequoia sets the sharps and flats according to the selected key. It is often the case, however, that an enharmonic shift can considerably optimize the legibility of certain passages. In this case, you can intervene manually.

 To shift one or more selected notes enharmonically, click the corresponding button. The function transforms flats into sharps and vice versa.

Insert clef: Clef symbols can be inserted at the current play cursor position using the corresponding clef symbol buttons in the active system.

![clef toolbar icons] To insert clefs, first activate the staff where the clef is to be inserted by clicking the staff clef on the far left. Now position the play cursor at the insertion point and click on the desired clef in the toolbar.

Click on a clef symbol with the eraser or the right mouse button to delete it.

Time signatures: The time signature signature symbols are created automatically from the time signature markers of the project. Time signature changes are possible at full bar borders. To do this, create a time signature marker in the **Edit** > **Tempo** menu with the new signature at the desired position. If no time signature changes occur, it is enough to set the time signature of the piece in the transport console.

Delete notes: You can delete notes by selecting them and pressing **Del** or by clicking on them with the eraser or with the right mouse button.

The menu command **Score** > **Selected notes: hide/show in score** allows you to remove selected notes from the score display without affecting MIDI playback. Notes hidden in the score are indicated in the matrix editor by a diagonal line. This feature is handy for making trills easier to read, for example, or for removing "control notes" for switching playing styles from the score.

Adjusting and Optimizing the Score.

The quality of the automatic conversion of MIDI data into a score is usually sufficient for editing MIDI data as efficiency is required here instead of a perfect, ready-to-print display.

The note representation generated from the events of the MIDI objects is always correct in pitch and position. However, this does not mean that the score can be optimally read, since displaying note lengths also plays an important role in this context. In this case, the score permits more flexible interpretation so that 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...

...and finished editing

⚙️ The options for customizing the display can be found in the **Score Settings** dialog. Open the dialog by clicking this button.

💡 *Tip: Use two different versions of a piece, one for MIDI sequencing and one for score printing! This way you can change the MIDI events in the version for score printing as you like, even if it then doesn't sound as it should during 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.*

Score Settings

In this dialog you define all settings for the staves, the interpretation and distribution of the MIDI data to the staves.

At **Current track** you select the track whose settings are being edited and use Number of staves to determine how many staves are created for this track. In the menu **Current track** only the tracks on which the selected objects are located appear. Therefore, if you have selected only one MIDI object, only this track will be displayed in this menu

Presets: These are preprogrammed standard settings for certain instruments or instrumentation such as string quartet, piano or orchestra. By selecting a preset, the note image can be made more readable and clearer in just a few steps.

In the table below, the settings of the staves of all involved tracks are listed one below the other. The list of the staves can be scrolled vertically using the **Up**/**Down** arrow buttons. The active stave is recognizable via the index marked in red.

For each stave there are the following settings:

- **Name**: The name of the stave, on the far left under the clef

- **Clef**: The clef for the stave. Within the stave it is possible to change the clef. With the help of the +/-12 clef option, octave notation is possible.

- **Key**: The key of the stave determines which accidentals apply to the entire stave.

  > ⚠️ **Attention:** *You cannot insert a key change in a staff. As a workaround, you can create an additional track and select a different key for this track in the score settings.*

- **Trp.**: For transposing instruments you can set here an offset by which the notes are transposed (e.g. +2 for trumpet in Bb, -3 or +9 for alto saxophone in Eb).

- **Channel**: A MIDI channel can be specified for the automatic assignment of notes to the different staves. For more on this, see Note Assignment in Multiple Systems below.

- **2. Voice**: A MIDI channel can also be assigned for the automatic assignment of a second voice (direction of the note stem).

- **Quant.**: Using the display quantization you can set the rhythmic resolution of the note display, independently of the actual quantization. For example, you can display a freely recorded and unquantized track in sixteenth notes. The display quantization does not have any influence on note playback, but rather adapts the note display to a grid.
  Set the value to the smallest note value occurring in the sequence. A display quantization that is too fine can result in an unreadable display.

- **Split point**: The split point is also used to automatically assign the notes to the staves: Notes above the split point are assigned to the upper system, notes below the split point to the system below.

In the lower area, the options for interpreting the active stave are displayed.

Changes to the settings for the options, key and display quantization are applied to all staves by default. With the checkboxes you can change this behavior, so you can also set different options for each stave in a system

Interpretation Options

In the lower part of the Score Settings dialog there is a section where you can influence how the score is displayed via a number of options.

> ℹ️ *These options do not affect the playback, but only the note display.*

Simplify note lengths (interpretation): Rests and slurred notes are displayed so that the score is as legible as possible.
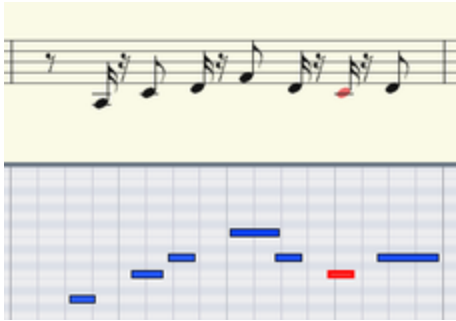
Display without option Simplify note lengths; display quantization 16th.
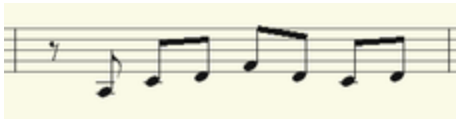
*Display with option **Simplify note lengths (Interpretation)***

Hide note overlaps: Use this option to eliminate slurs over successive notes caused by legato playing, for example.

Original
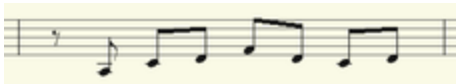
*Display with **Hide Note Overlaps** option*

Automatic staccato articulations: With this option, notes that are shorter than the displayed MIDI note get a staccato symbol.
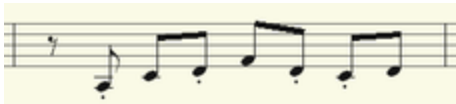
Display with option "Automatic staccato articulations"

Recognize triplets: Activate this option if triplets are present in the MIDI object.

> **ⓘ** **Note**: *Always set the value for the display quantization one step finer than the smallest triplet values to be detected. 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: With this option, no note and rest values occur that are longer than a bar subgroup (pulse, beat). In some cases longer notes are displayed as multiple slurred notes. This can simplify the legibility of the score.

Larger beam 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.
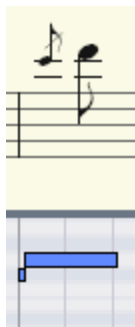
Original   *Display with **Recognize Grace Notes" option***

Force Polyphonic Notation: The display of the notation always appears polyphonic.

Polyphonic Notation

Up to two independent voices can be annotated for each note line. The voices differ in the direction of the note stems: the first voice is always notated with the stem pointing upwards, the second with the stem pointing downwards. Pauses are displayed individually for each voice.

Monophonic representation   *Polyphonic representation*

You can set the voice by selecting the notes and clicking the corresponding buttons:

   Assign 1st voice (stem direction up)

   Assign 2nd voice (stem direction down)

   By selecting Automatic voice assignment, the defined voice assignment can be canceled.

In the case of automatic voice assignment, the MIDI channel of the notes is evaluated for voice assignment: To do this, specify a MIDI channel for the second voice in the **Score properties** dialog. You can set this individually for each stave. All notes on this staff coming from MIDI channels other than the set one will be assigned to the 1st voice. If no MIDI channel is specified for voice assignment and no manual voice assignment has been made, all notes are drawn as 1st voice.

Note Assignment in Scores with Multiple Staves

The assignment of notes to the different staves can be done both automatically and manually.

When automatically assigning notes to a specific stave, either the MIDI channel of the note event, the pitch or a combination of both properties can serve as criteria. The criteria are set in the Score settings (channel, split point).

To manually assign notes to staves:

Click this button to move the selected notes up one stave.

Click the button to move the selected notes one stave down.

This results in the note being connected to the stave (independently of the MIDI channel or pitch).

This manually defined assignment can be undone by pressing the **Automatic Staff Assignment** button.

Page Settings

The page settings determine the arrangement of the entire score on a paper page for printing. You also determine how the notes are displayed in the page view.

To open the page settings, use this button to open **Score Settings** and then click **Page settings**
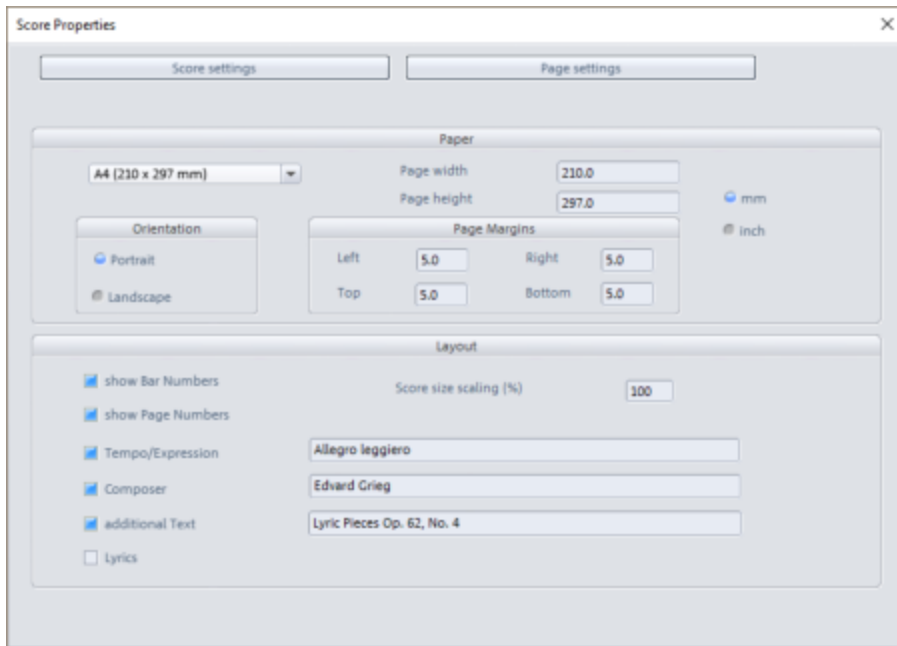
Sequoia automatically creates the layout of the score and caters for optimal division of note lines and systems on the page. You only need to specify the **Page size**, **Orientation**, and **Page margins**. For page size, you can find templates for common page formats in the drop-down menu.

Scale Score Size (%): Use this factor to scale the size of the score symbols in relation to the page. This influences how many bars fit into a note line and thus at which points a line break or page break occurs.

With a smaller value you get more bars/staves on a sheet of paper, the note symbols are displayed smaller accordingly.

Layout: Under Layout you can show/hide various other layout elements such as bar numbers and page numbers by option and add the tempo label, composer and additional text. Also the lyrics (↗472) can be displayed optionally.

The title of the piece (large text at the top center) corresponds to the object name of the first selected MIDI object. To quickly change the object name you can use the keyboard shortcut **Ctrl + N**.

Printing Scores

To print the score sheet, select **Score** > **Print Score** or click the printer button.

The system print dialog will appear, where you can select the printer you want to use and other options, such as selecting the pages and number of copies you want to print. At **Properties...** you can reach further setting options, specific to the selected printer. Be sure to select the same paper size for printing that you set in the Page settings so that the printout is not scaled or printed in the wrong aspect ratio.

The following elements are only visible on the screen and are not printed:

- The lines which mark the page borders on the monitor

- Mouse Pointer

- The color highlighting of selected notes and the playback area.

> ℹ️ *Note: When printing to a PDF file, please make sure that you enable the option to embed the fonts in the document so that the notation symbols are displayed correctly in the document.*

Score MusicXML Export

The score can be exported as a MusicXML file for further processing in music notation applications. Many dedicated music notation programs can import MusicXML files. Converters are also available for transforming MusicXML into other formats. You can find out more at www.musicxml.org.

Converting to MusicXML format has the advantage that, unlike with the standard MIDI file format, many manual settings and optimizations of the score, like the number of staves, key, voice allocation, pitch, etc., remain the same.

Create your composition in **Sequoia** and define score settings here already, so that you can edit or navigate more clearly in the score view. You can then continue editing the score in order to optimize the layout and notation in the familiar environment of your score editing application.

To export to MusicXML, choose **Score** > **Export Score as MusicXML...** from the menu.

When exporting MusicXML, all properties of the score such as staves, notes and rests, bar groups, clefs, polyphonic voices, accidentals, enharmonic changes are included but page format, line breaks and system spacing are not. These are also part of the MusicXML format, but since these parameters can only be edited to a limited extent in **Sequoia**, they are excluded from the MusicXML export.

> ℹ️ *Not every score 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".*

MIDI Velocity Dynamics

MIDI Velocity Dynamics is a MIDI effect that works on the same principle as audio dynamics effects, assigning a specific output velocity to each input velocity value via a characteristic curve to compress or expand the velocity of MIDI notes. It can be used both in real time as a track effect and offline as a MIDI function in the MIDI Editor for editing MIDI data.

Use as a Track Effect

By using it as a real-time track effect, you can influence the velocity response of a MIDI keyboard right at the time of recording.

ℹ️ *The effect influences the playback of MIDI notes. The notes are recorded with the original velocity, but can already be heard with the altered velocity during recording.*

To activate the effect, click the **Velocity Dyn** button in the **MIDI** section of the **Track Editor** or select it from menu **Track** > **More** > **Track Effects** > **MIDI**.

When the effect is activated for the first time, the effect's settings dialog will also open. Changes to the dynamics parameters take effect immediately during playback and MIDI input, and remain so even when you close the dialog.

In the further course you can deactivate and activate the effect with a left click on the **Velocity Dyn** button in the Track Editor. (This corresponds to the **Bypass** option in the dialog). To open the settings dialog again, right click the **Velocity Dyn** button.
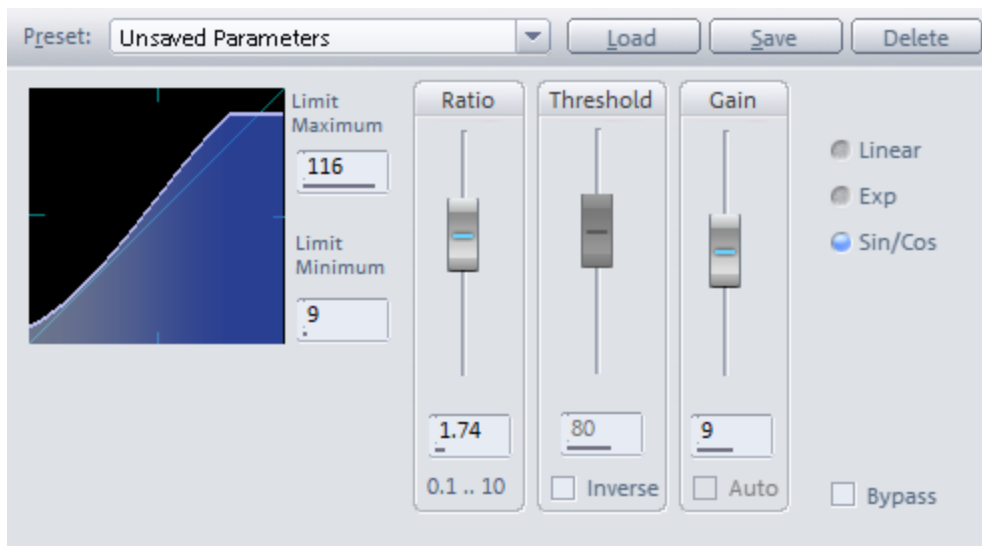
In the **Preset** menu you will find a number of presets for various compressor or expander applications that you can use as a basis for individual adjustments.

The effect is controlled by the following dynamics parameters:

- **Ratio**: Velocity values above the Threshold are affected by this factor, thus controlling the amount of compression (Ratio > 1) and expansion (Ratio < 1).

- **Threshold**: All velocity values greater than the Threshold value are affected by the Ratio factor. If the **Inverse** option is active, only the velocity values below the threshold are processed.

- **Gain**: The gain value is an additional constant value that is added to the velocity, it thus shifts the curve up or down overall. If the **Auto** option is enabled, the gain parameter is automatically adjusted to achieve the maximum output velocity value with the current Ratio/Treshold settings at maximum input velocity.

- With **Limit Maximum** and **Limit Minimum** you can set the minimum and maximum possible velocity value, respectively.

  > 💡 *If you set **Limit Minimum** and **Limit Maximum** to the same value, you can thus set a fixed velocity and simulate a keyboard without velocity, like an organ.*

- **linear, exp, sin/cos**: With the selection of **exp** or **sin/cos** alternative characteristic curves are possible.

Use as Offline Effect

Select a MIDI object and choose **Effects** > **MIDI Velocity Dynamics** > **MIDI Velocity Dynamics...** from the menu to apply the effect to all notes of a MIDI object. To edit only specific notes of a MIDI object, open the MIDI object in the MIDI Editor (double-click), select the desired events, and then choose **Velocity Dynamics...** from the MIDI Editor's **MIDI Functions** menu.

> ℹ️ *If no notes are selected in the MIDI Editor, all notes of the event are edited as well.*
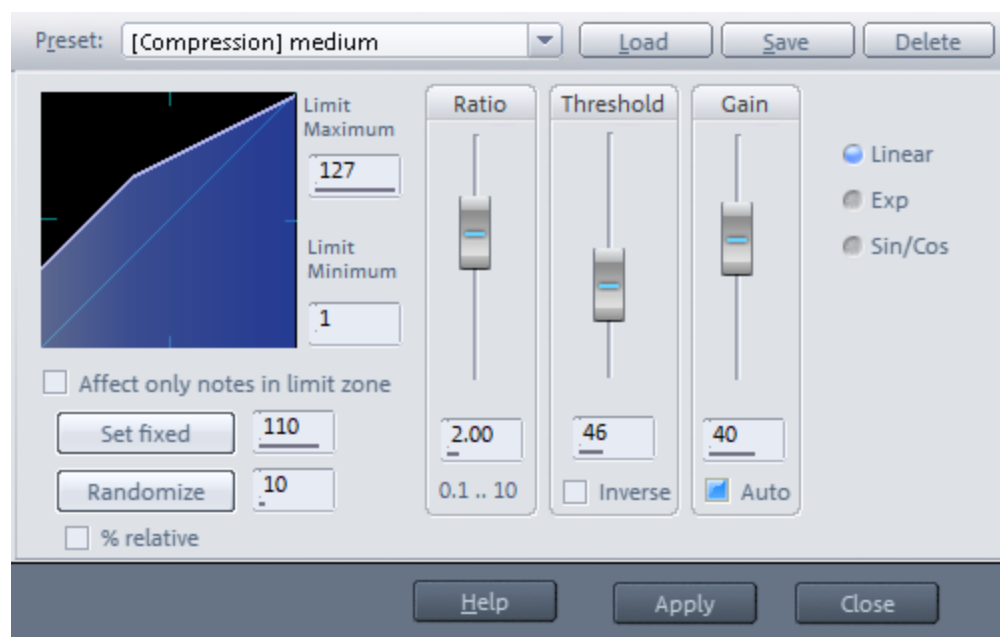
The operation of the dynamics parameters is identical to the real-time variant of the effect. The **Apply button** permanently changes the velocity values in the MIDI object and closes the dialog. **Close** closes the dialog without making any changes.

In addition to the dynamics functions, there are two other functions in the offline version of the effect:

Set fixed: Use this button to set all notes to the specified velocity value.

Randomize: With this button you change all velocity values of the notes individually by a random value. Enter the maximum value of random changes in the adjacent field. This allows you, for example, to make programmed sequences with constant velocity sound more human. With the option **% relative** the change of the values is relative, thus louder passages are subject to greater random fluctuations than quiet ones.

Only for notes in limit zone: If active, only notes whose velocities are within Limit Minimum and Maximum will be modified. This also allows you to set only notes of a certain range of velocity values to a fixed value or edit them with random variations.

MPE

Sequoia supports the new **MPE** (MIDI Polyphonic Expression) extension. MPE is a new control standard for synths which expands the range of possible expressions which can be produced with these instruments. Unlike OSC, for example, this is not a new protocol, but MPE is conventional MIDI, although the MIDI channel is changed 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. So there is no way to assign different pitch or tone changes for individual notes when playing polyphonic. 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, or the Soundplane by Madrona Labs.

On the other hand, the sound generators must also support MPE: They must be able to generate a separate, independent synthesizer voice 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 more channel (Master Channel) is needed for global control commands like program change and global pitch, the maximum polyphony is thus 15 independent 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 www.midi.org.

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 (↗141) or in the MIDI section (↗61) of the Track Editor.
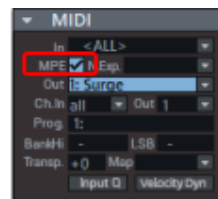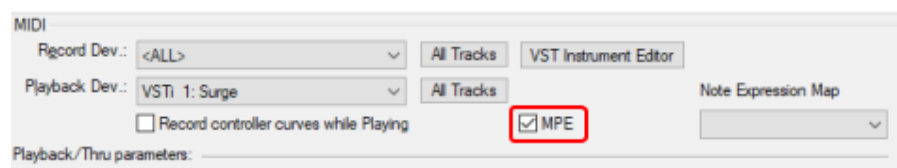
The MIDI editor in MPE mode

The MIDI editor's MPE mode has some special features which make it easier to edit MPE MIDI.

- Due to the very large pitch bend range of 48 semitones, even very small pitch wheel changes produce audible pitch changes. 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.
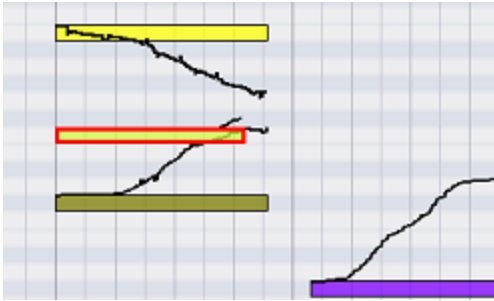
- When you move a note, the controller/pitch wheel data associated with that note (that is, the data located between a specific Note On/Note Off event on a specific channel) are moved as well.

  > ℹ️ *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 whose controller or note was selected last is always in the foreground.
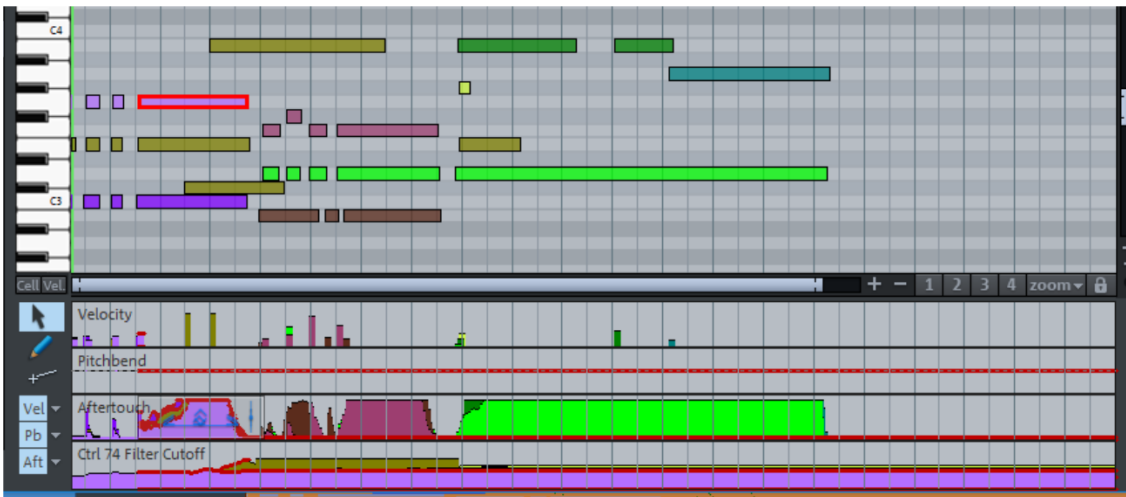
- If this bothers you while dragging a selection frame, additionally hold down the **Ctrl** key while dragging. The selected channel in the foreground is then retained and only events that are also on this channel are additionally selected. If you want to select events from other channels additionally, also press the **Shift** key. To make it easier to edit very small pitch bend curves, you can use **Ctrl + mouse wheel** to zoom vertically into the controller curve.

Master Channel

The MPE standard also requires the definition of a Master Channel. Controllers which are sent over this channel affect all notes, just like normal MIDI. Program Change events 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.
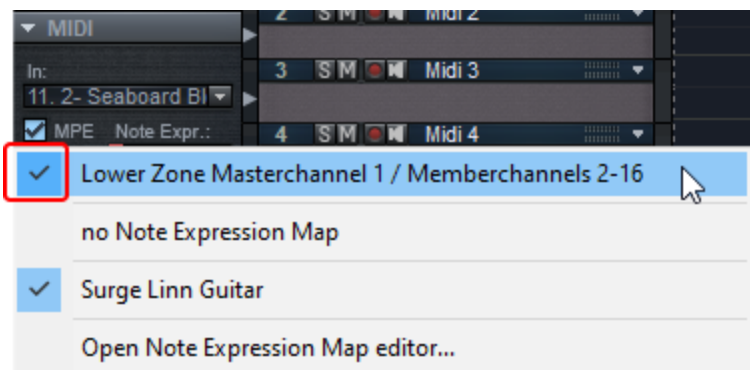
VST3 Note Expressions

Note Expressions is another way to control VSTi synthesizers with per-note sound parameters for more expressive playing. These are an extension to the VST3 standard. Note Expressions are currently only available for certain synthesizers from the manufacturer Steinberg.

A certain number of sound parameters are provided by the plug-in as Note Expressions, which can then be assigned to MIDI controllers in **Sequoia** via the **Note Expression Map**, which then also only apply per note.

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 different channels with corresponding controller values and sends the notes to the sound generator on one channel. Additionally, the expression values assigned to the controllers are transmitted to the sound generator per note via the VST interface.

- Therefore, multi-timbral instruments cannot be controlled with MPE, but they can be controlled with Note Expressions.

Note Expression Map Editor

Open the Note Expression Map Editor in the Note Expression menu: in the Track Editor to the left of the MPE option under **N. Exp.** or in the **Note Expression Map** menu in the Track Options dialog (↗141).
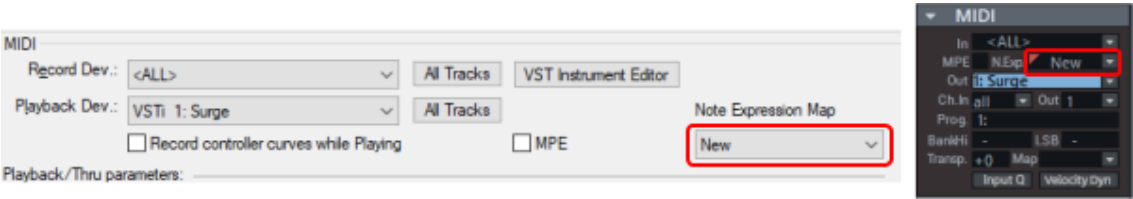
1. Load a VST3 instrument which supports Note Expressions onto the track. Load a preset which has Note Expressions assigned in it as modulation sources, or make these assignments yourself in the VST instrument.

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.
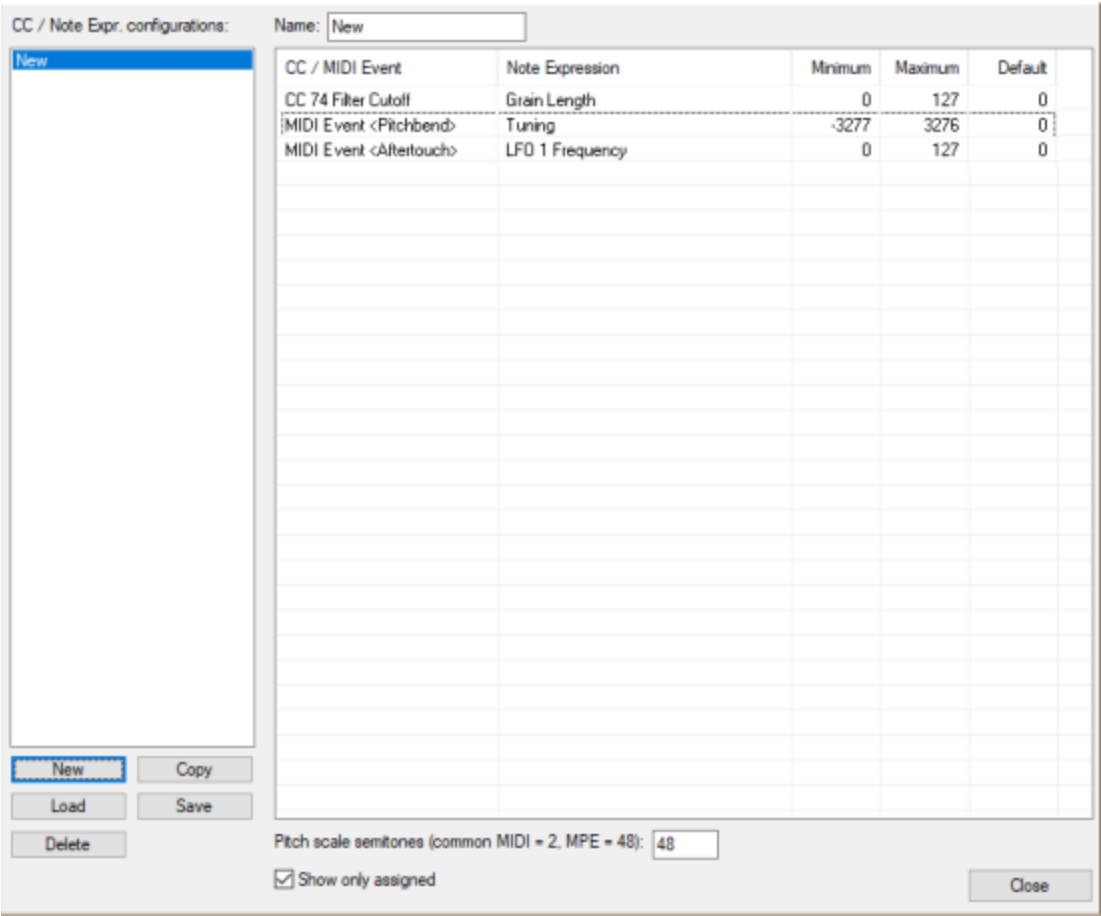
3. Give the map a descriptive name (e.g. Synth_Preset).

4. Assignments are made on the right. The three standard MPE parameters CC74 (Timbre), Pitch bend and Aftertouch are already visible by default. If you want to assign the Note Expressions to other controllers, uncheck the **Show assigned only** option below.

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**. With **Load** and **Save** you can also save the map to a file or load it from a file. **Delete** removes the selected map from the list.

7. With **Pitch Scale Semitones** you define which Pitch Bend Range your keyboard uses, MPE controllers use 48, normal keyboards usually 2

> ℹ **Note**: *The Note Expression parameters depend on the set sound program (patch) of the sound generator and are transmitted by the VSTi when the Note Expression Map Editor is opened (and only then). Therefore it is important to keep the order: First set the patch on the tone generator, 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.
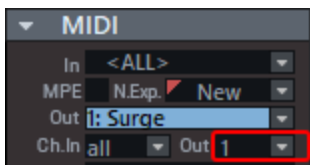
- When using MPE controllers, it is absolutely important to make sure that all channels are selected for use under **MIDI In**.

- If you don't have an MPE-capable keyboard, you can still benefit from the advanced capabilities of MPE or Note Expressions by manually changing notes to other channels in MPE mode in the MIDI Editor Event List (↗444) and drawing in corresponding controller values.

- If you are recording with a normal keyboard which normally sends its notes through Channel 1, be sure to disable the MPE Master Channel in the synth or select a different channel on the keyboard.

Lyrics Markers

Lyrics markers allow you to enter lyrics, comments or stage directions. You can also use the simple markers for this, since they can be named in any way. Using lyrics markers makes certain tasks easier:

- With the standard MIDI file format, it has always been possible to store text in the MIDI file at a precise time. Correspondingly, when MIDI files with text (e.g. karaoke MIDI files *.kar) are imported, lyrics markers are created and then exported along with the MIDI file when it is exported.

    > ℹ *Karaoke MIDI files with the extension `.kar` are not displayed in the file manager, but can be loaded via drag&drop. But since such files are also just ordinary standard MIDI files, you can also simply change the file extension to `.mid`.*

- In the Score editor (↗453), the lyrics markers can be displayed as song lyrics on the score sheet.

- Optionally, lyrics markers can also be displayed on the marker bar, see **Program preferences** > **View options** (↗716).

- The Time display (↗64) can be configured to display the name of the current and following lyrics markers in a field.

The Lyrics markers are the META events "Lyric" and "Text" in a standard MIDI file (SMF). When importing a MIDI file, the standard ASCII characters CR and LF present in the text are replaced by the standard Karoke text character / (see below).

There are other standard karaoke control characters that influence the text output in the time display window. The control characters are always placed at the beginning of a marker text.

Character	Meaning	Note
@	Internal karaoke info like song title and version number	Any lyrics marker that starts with @ will be skipped and treated like a comment, so will not be output in the time display.
/	New Line	The following text should start in a new line.
\	New verse	Clear screen and start at the top. However, in **Sequoia** this is treated the same as New Line.

+	Glue	Sequoia automatically starts a new line in the karaoke window (time display) if the pause to the last note is more than 1/3 bar duration. With the + this automatic is switched off and the marker text is taken over into the current line.

MIDI Editor Keyboard Shortcuts

The keyboard shortcuts can be defined freely (except for a few like spacebar for play/stop), open the editor for keyboard shortcuts, menu and mouse via menu **Shortcuts** > **Edit shortcuts...**

For some basic functions such as scrolling and zooming, all keyboard shortcuts from the VIP are applied. You can explicitly define shortcuts for all commands that are available in the MIDI Editor menu.

File

Import MIDI	Ctrl + I
Export MIDI	Ctrl + E

Edit

Undo	Ctrl + Z
Restore	Ctrl + Y
Cut	Ctrl + X
Copy	Ctrl + C
Paste	Ctrl + V
Duplicate	Ctrl + D
Select All	Ctrl + A
Create pattern from selection	Ctrl + P
Delete selected events	Backspace, Del
Delete all MIDI events	Ctrl + Del Ctrl + Backspace

Select previous event	Right arrow
Select next event	Left arrow

MIDI Functions

Legato	Ctrl + L
Quantize notes	Ctrl + Q
Quantize 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 - Stop running edit notes	Ctrl + F

Mouse mode

Selection	1
Draw Pencil	2
Drum Pencil	3
Pattern Pencil	4
Velocity Adjust	5
Eraser	6
Magnifier	7

Glue	8
Split	9
Mute	Shift + M
Velocity/Controller Selection	Ctrl + 1
Velocity/Controller Freehand drawing	Ctrl + 2
Draw Velocity/Controller Line drawing	Ctrl + 3

View

Show Event list	Alt + L
Show Velocity/Controller Editor	Alt + V

More keyboard shortcuts

Select next event	Right arrow
Select previous event	Left arrow
Pitch up	Arrow up
Pitch down	Arrow down
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 play cursor forward	Page down
Play cursor to next bar	Ctrl + Page down
Move play cursor backward	Page up
Play cursor to previous bar	Ctrl + Page Up

Keyboard

With the keyboard, software synthesizers (↗371) can be played directly via an on-screen keyboard and also recorded.

The keyboard always controls the synthesizer in the track for which MIDI recording has been activated.

You can operate 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 play music properly 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 keyboard.

> ℹ **Attention**: *This only works if the mouse has been clicked in the keyboard once before, otherwise the keyboard entries act as* keyboard shortcuts *(↗709) for various other* **Sequoia** *functions. 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 shift the octave range in which the keyboard can be played using the computer keys.
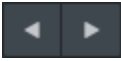
Use the horizontal arrow buttons to select the next/previous sound of the synthesizer, the list box next to it also lets you select the sounds directly.

This button opens the synthesizer editor window for fine-tuning the sound.

Arpeggiator

The arpeggiator is a playing aid that allows you to automatically generate chords by playing and holding individual keys, either as a normal chord or as a broken chord (arpeggio), i.e. a rapid succession of the chord notes.
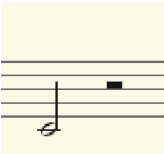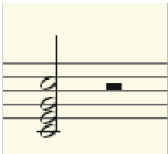
Note C   *C major chord*   *C major arpeggio with 1/16 notes*

1. This button activates the arpeggiator.

2. This switch determines whether the played note will generate no chord, a minor chord or a major chord.

3. This switch determines the type of arpeggio. In the rightmost position 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).

MIDI Panic – All Notes Off

When playing together with hardware and software instruments, "hanging" notes can sometimes occur. They are caused when, after a NoteOn command, for some reason the corresponding NoteOff command does not actually get through to the tone generator, so that the tone sounds permanently.

At the very bottom of the menu **Playback** select **MIDI Panic - All notes off** to end the hanging notes. The command sends a NoteOff message to all MIDI devices that have not been disabled in the MIDI options for all 128 notes on all 16 channels. In addition, the Sustain (controller 64) is switched off, the pitch wheel and the modulation are set to 0. An All Notes Off command is sent to all VST instruments used in the project.

> ⓘ *You can set in the MIDI options (↗697) that individual NoteOff events for all notes on all channels are also sent to the VST instruments.*

The function can also be executed by clicking the stop button in the transport console again after stopping the project!

MIXER

Use the menu **View** > **Mixer** to open the mixer window of Sequoia (keyboard shortcut: **M**).

The mixer can be used docked or undocked like all windows (see Docking (↗56)). By default, the Mixer window shares the docking area with the Project windows.

The Mixer combines the settings of the individual tracks such as volume, panorama, input and output routings, aux sends, plug-ins, equalizers and so on, as they are also found in other forms in the track headers or track editors, on a common interface and thus conveys the familiar look of a mixer as you may still know it from the analog way of working in the studio.

There are also additional control functions that can only be found in the Mixer window, with which you can, for example, link track controls, save the overall state of the mixer in snapshots or reset track settings such as panorama or effect settings for all tracks.

As with the Track Editor, individual areas can be expanded and collapsed, the size of the volume controls changes dynamically depending on the window size and, if necessary, a scroll bar is displayed so that the Mixer can always be displayed optimally according to the available screen space.

> ℹ *In the context of mixing consoles we often talk about channels, in Sequoia a track corresponds 1:1 to a channel, in the following both terms are used synonymously.*

💡 *In the program settings (keyboard shortcut Y) at **Design** > **Skins** or by clicking in the upper left corner of the Mixer window you can select alternative mixer skins.*

Mixer Operation with Mouse and Keyboard

Operation with the Mouse

Mouse wheel: Move the mouse pointer over a control and turn the mouse wheel to change the value. For finer value changes additionally press the **Shift** key, for larger value changes press the **Ctrl** key.

Click and drag: The simplest and most intuitive way to operate, as with any graphical interface, is to simply click to toggle functions and drag to change values. However, there are a few special features:

- The accuracy of setting the **Level Fader** is infinitely variable: Click on the fader and move the mouse pointer to the left or right of the fader while holding down the mouse button, and only then drag the mouse up or down. The farther the mouse cursor is from the selected fader, the smaller the change in value will be.

- **Knobs (pots)** can be adjusted in two different ways: The default behavior is that you click the knob and move the mouse around it to change the value. Alternatively, you can set the pots to behave like sliders, i.e. you can drag the mouse up and down. (at **Program settings** > **Keyboard/Menu/Mouse** > **Mouse**)

Click on value range: Clicking the range below/above sliders and left/right of knobs decreases/increases the value step by step. If the mouse button is held down, the value will continue to change until you release it. The longer you hold down the mouse button, the faster the value changes.

Double-click on numbers: Opens a numeric input field.

Double-click on control elements: Sets the control element to the preset value. Double-clicking again sets the value back to the altered value again.

Right-click on control elements: Accesses either a context menu for the corresponding control element or a dialog with additional settings options.

Keyboard Operation

The Mixer can also be operated completely with the keyboard only:

Arrow keys: Use the arrow keys to move between controls and activate the selection of a control. With **Shift** + **Arrow right/left** you can extend the selection with further similar controls on other tracks if necessary (see below).

> ⓘ *You can also switch between adjacent controls on the same track using the up/down arrow keys. Arrow right/left always changes to the adjacent track.*

Page up/page down: changes the value of the active control. For finer value changes additionally press the **Shift** key, for larger value changes press the **Ctrl** key.

Home: sets the control to its default value. If you press the **Home** key again, the element will be reset to the previous value.

End: This function corresponds to clicking with the right mouse button, i.e. it opens a settings dialog or a menu.

Enter key: opens the numeric input field for a value, switches and opens menus, depending on the selection.

For Plug-in slots

- **Enter** opens the plug-in browser

- **Del** deletes a plug-in from the slot.

- **Ctrl + C** copies a plug-in.

- **Ctrl + V** pastes a copied plug-in. If an occupied slot is selected during insertion, the plug-in is inserted before it.

Select and Group Multiple Controls

You can select multiple controls. First click on a control to select it. Now click on other individual controls with the **Ctrl** key held down to add them to the selection. A click with held **Shift** key adds all controls between the last selected and the clicked one.

Controls selected together are changed together when one of the controls involved is changed.

With **Ctrl + Shift + Click** on a knob or fader you reverse the behavior of the corresponding fader or knob within a multiple selection. It then moves in the opposite direction to the others in the group and when it is moved, the others move in the opposite direction. 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.

To operate a control element that is part of a multiple selection or group individually, click and drag it while holding the **Alt** key.

Shift + **Double click** (Keyboard: **Shift** + **Pos1**) on a control sets all controls of the selection to the value of the clicked control.

As soon as you click on a control element that is not part of a multiple selection, the multiple selection is cleared. If you always want to adjust certain controls together, you can keep the multiple selection by grouping the controls together.
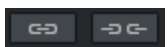
Use the first button to group selected controls in the mixer, and the second to ungroup them.

> ⓘ *Note: There is, independent of the grouping in the mixer, also the possibility to select several tracks and group them (↗132). In such a multiple selection/grouping of tracks, all controls of the tracks are moved together. If this affects controls that are also grouped in the mixer, this grouping is ignored.*

In the mixer, tracks can be selected by clicking on the track name (in the channel below above the output routing) or the track number (below the pan slider).

Drag & Drop

- Drag and drop of the **tracks/channels**: You can rearrange the order of the tracks by clicking and dragging on the track number or track name.

- The **plug-in slots** provide drag-and-drop functionality for copying or moving effects to another track:

    - The sequence of effects for VST plug-ins can be changed within the slots of the track using drag & drop.

    - A plug-in can be moved to another track via drag & drop. It is always inserted at the end of the plug-in chain, regardless of whether you drop it on an occupied slot.

    - Drag & Drop with additional held **Ctrl** key copies a plug-in to another track.

    - Drag & drop of the **Plug-ins** button to another track transfers the contents of all plug-in slots to that track.

    - Plug-ins can be dragged from the dockable Plug-in Browser window directly to a plug-in slot in the mixer.

- The **Equalizer settings** can be transferred by dragging & dropping the **EQ** button.

- Drag & Drop the **FX** button: Dragging an FX button (at the bottom of the track under the peak meter) to another track will transfer the entire FX routing (effects, AUX positions...).

Channel Strips

Each mixer channel corresponds to one track in the project. Accordingly, you will find all the setting options in the channel that you also find in the track editor or in the track header. Each channel strip is divided into different sections, which can be opened and closed individually by clicking on the corresponding arrow symbol.

If the mixer window height is too small, only the section names will be displayed instead. They can then be opened accordingly by clicking on the name.

In (Input)

1 Field for selecting the track. Drag the field to rearrange tracks.

2 **In**: Shows the input(s) for recording on the track. Clicking opens a menu with settings for recording, see Input and output assignment (↗487).

3 **Track name**: Shows the name of the track. Double-click the field to edit the name.

4 **Gain**: Controls input amplification for the channel.

5 **Delay**: Track delay. For more information, see Track settings (↗141). Right-click to select the time display format used.

AUX Sends/Outputs/Side Chains

1 Via right-click menu you can reach the AUX sends dialog (↗498), furthermore the whole section can be switched to display the outputs (↗487) or sidechain sends (↗378).

2 The sliders can be used to set the AUX send values for different AUX buses (↗495). Drag an unnamed control to create a new AUX bus. A left click on the slider deactivates the AUX send. By right-clicking on the fader, you can switch the function of the fader between Pre/Post/Direct-Out (↗498). You can also access advanced settings such as AUX pan editor (↗498).

3 Bypass indicator/switch. Left click on AUX or this button disables all AUX send signals.

4 The **+** icon toggles between displaying 2 or 10 send controls.

Plug-ins

1 The **Plug-ins** button at the top turns all effects in the channel on and off. A visual indicator (*) for plug-ins shows that they were previously active and will be activated again the next time you press the **Plug-ins** button.

2 A mouse click on an empty insert slot opens the plug-in browser (↗216), via which you load a plug-in into the slot. Click on an occupied slot to activate/deactivate the plug-in. Right-clicking on the slot opens the plug-in's interface. VST instruments whose output signal is routed to the track are also displayed in the plug-in slot, indicated by a preceding + symbol

3 The arrow next to the respective insert slot opens a menu with various functions: For example, you can reopen the plug-in browser to exchange the plug-in or remove the plug-in.

4 The **+** icon toggles between displaying 5 or 12 plug-in slots.

Equalizer
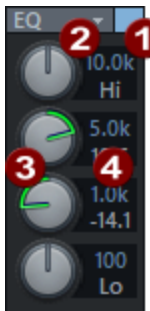
1 This section allows you direct access to the parametric equalizer EQ 116 (↗262). Although the EQ is always visible in the channel, the effect is not loaded in the track until you change the parameters. You can see that the equalizer is active by the blue button. Click this button to deactivate the equalizer.

2 Using the menu with click on **EQ** you can, among other things, copy EQ settings and paste them into the EQ on other tracks.

3 Use the knobs to adjust the gain of the 4 bands. Right-click on one of the knobs to display the EQ dialog where you can make more detailed settings.

4 In the number fields you can edit gain and frequency of the respective band numerically.

Main

Here you will find the most important controls of the track. If the track is a bus, this area is displayed in a slightly different color.

① Panorama: This adjusts the signal placement 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.

**② ** Beside the panorama dial, you'll also find a switch to reverse the phase. Right-clicking on either of these controls takes you to the stereo panorama dialog (↗492), which allows you to make further settings such as panning laws or changing the stereo width.

If the track is routed to a surround bus (↗512) or master, a surround panorama control is displayed at this point. It roughly shows the surround panning of the track, a left click on the control opens a menu with surround panorama presets, a right click opens the surround editor (↗517)

③ Track number

④ Link: This button ⬚ links the channel to the one to the right. All faders, panorama, input, AUX sends, as well as EQ settings and adjustments now affect both channels.

⑤ Automation mode: Left-click to quickly switch the automation mode (↗342) between the last selected mode and **Read**.
Right-click to access all automation modes.

> ⓘ *For detailed information on automation, refer to the chapter "*Automation *(↗338)".*

⑥ Solo: The Solo button mutes all channels except the soloed channels. Keyboard shortcut: **Alt + S**
A more specific Solo button behavior can be set by right-clicking on it:

- **Solo-exclusive**: This setting switches the active channel to "Solo-exclusive" mode. Only this track will still be audible, all other tracks, including the soloed ones, will be muted. Keyboard shortcut: **Shift + Alt + S** or **Double-click** on the button.

- **Solo safe**: This setting ensures that all channels that are switched to solo are automatically monitored with the AUX Return channels that they feed.

> ⓘ *Various global solo modes can also be set in the menu, see* Global Solo Modes *(↗505).*

⑦ Record: This button activates the track for recording. Right-click to set the recording mode and track inputs.

8 **Mute**: The Mute button mutes the selected channel. Keyboard shortcut: **Alt + M**

A 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**: This function mutes the bus inputs routed to this channel.

9 **Volume**: Controls the level of the track. Depending on whether audio or MIDI objects are playing on the track and whether software instruments are being controlled, the slider can control the level in various ways that you specify in the context menu. See the section Control behavior of the volume slider (↗486).

> **ⓘ** **Note**: *The volume control has a maximum value of +12dB. You can set higher values by entering them numerically. To do this, double-click on the volume value display below the peak meter and enter the desired value.*

10 **Monitoring:** By clicking this button you activate the monitoring function. The signal fed to the input is passed on to the output.

> **ⓘ** *For more details on monitoring, see* Monitoring Settings *(↗692).*

11 **VCA:** This button opens the VCA menu for the channel. There you set the fader as VCA master or assign the fader to a VCA group, see VCA-Groups (↗490).

12 **FX**: This button opens the effect routing dialog (↗220) for the respective track, which you use, among other things, to specify the effect order in which the effects are applied.

Right-click the button or click the arrow to open a context menu where you can load preset effect settings. You can also copy, paste, reset, save or load track effect settings (↗220) here.

13 **Track name**: Shows the name of the track. Double-click the field to edit the name.

14 **Out**: Sets the output for the channel. See Input and output assignment (↗487)

Control Behavior of the Volume Fader

Depending on whether a track is recording MIDI or audio, and whether the audio signal of a software synthesizer is being processed in that track, the volume fader can perform different control functions.

The audio output of a software synthesizer is in the simplest case (↗372) generated, processed and mixed on the same track as the MIDI data that this instrument receives. This gives several options how the volume fader of this track controls the volume of an instrument: the level of the generated audio signal, the velocity of the MIDI notes or the MIDI volume (MIDI controller CC7). These are not identical parameters: It is possible, for example, to place a MIDI instrument played loudly at high velocity quietly in the mix on the audio side and vice versa. For this reason, you can select the mode for volume faders via the fader's context menu:

- **Audio track volume**: The fader controls the volume of the track's audio output. This is the default setting for new tracks. (This may differ in project templates)

- **MIDI velocity (scaling)**: The slider scales the MIDI velocity of MIDI notes on the track.

- **Audio volume and MIDI velocity**: The two previous options can also be combined.

- **MIDI Volume (CC7)**: Controller 7 (MIDI Volume) is the standard controller assignment used to control the volume in MIDI sound generators themselves.

- **No MIDI Volume (CC7) Change**: This option is only available when MIDI Volume is selected and is enabled by default when this option is selected. This has the following reason: The standard behavior of a volume control is that it does not influence the track volume in the 0dB position. There is no such "neutral" position for the MIDI controller CC7. Therefore, when switching to CC7, sending the volume control is first prevented with this option. It is automatically deactivated as soon as you pull the volume control.

Input and output assignment

In (Input)

At the very top of the channel strip, you specify the input and recording mode for the track.

> ℹ️ *Note: Since tracks in **Sequoia** process MIDI and audio, it is actually not quite correct to speak of audio tracks or MIDI tracks. We use these terms here as shorthand for "tracks that record audio" and "tracks that record MIDI".*
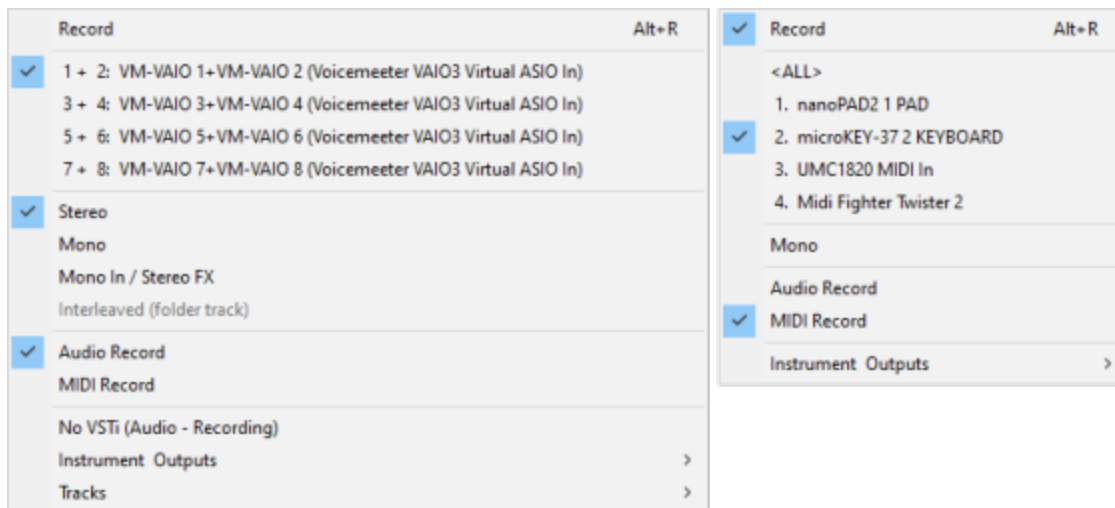
- **Record** arms the track for recording and corresponds to the Record Arm button.

- Below this you can see the available audio and MIDI inputs. Select here the input at which the desired signal is present. For MIDI tracks, <ALL> is preset, which means that all available MIDI input devices are recorded.

- **Stereo**: You can select a pair of inputs and a two-channel stereo input signal will be recorded.

- **Mono**: You can select a single input and a single channel mono input signal will be recorded.

- **Mono In / StereoFX** You can select a single input and a single-channel mono input signal will be recorded, but the track itself will remain a stereo track where the effect calculation will be performed in stereo.

- **Audio recording / MIDI recording** determines whether you want to record audio or MIDI input on the track. It is automatically switched to **MIDI** if you load a software instrument into the track (↗371).

- With **MIDI tracks**, you can select the output of the VSTi whose audio is to be played back on the track at **Instrument outputs**. This also works for **audio tracks**: If the track is set to audio, you can select the VSTi output from all VST instruments in the project as audio input there and thus record the audio output of the instruments. The selected hardware audio input of this track will then be ignored.

   > ℹ️ *See also* Routing software instruments *(↗372) and* VST Instrument Manager *(↗331)*

   To record the actual audio input on an audio track again, select **No VSTi (Audio Recording)**.

- **Tracks**: You can select and record audio output from other tracks, buses, or the master as the track's input.

Output
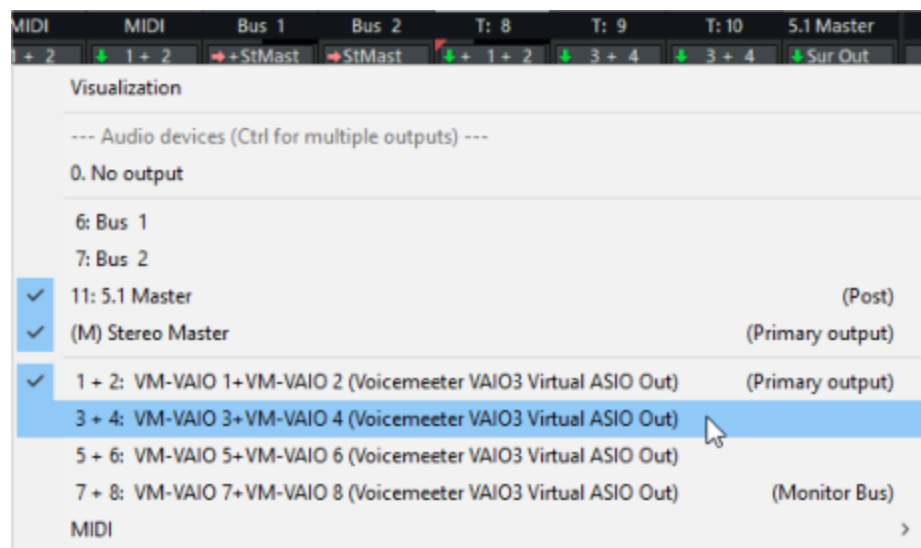
 The stereo master, another stereo output, a Submix bus and/or a surround bus can be selected as the output of the track. In the MIDI section of the menu, you can set the MIDI (VST) output device here. With **No output** you do not assign an output to the channel strip.

The stereo master output is labeled with and (M) and is the default output.

You can route the output signal of a channel to multiple hardware outputs, busses, or masters simultaneously, for example, to create a stereo mix in the same project alongside a surround mix, or for monitoring a mix on different speaker systems.

Select additional stereo audio outputs by clicking on them while holding the **Ctrl** key.

If you set the AUX section of the channel to **Show outputs** by right-clicking on it, you have an overview of the used outputs.
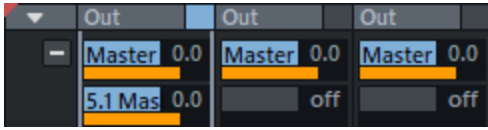
By right-clicking, you can set the selected output as "Direct Out" (knob display red), "Pre-Fader Out" (knob display yellow) or as "Post-Fader Out" (knob display orange) or call up the Stereo panning dialog (↗492) for panorama determination for the respective output.

> 💡 *If you have routed a track output to a bus or master with a different number of channels, e.g. a surround bus to an additional stereo master, you can access the UpMix or DownMix matrix with Ctrl + right-click on the fader of the output or via the menu entry at the bottom of the output menu of the bus.*

To unassign the output to a single output, click the **Out** button again, hold the **Ctrl** key and select the output you want to disable.

> ℹ️ *You can also get a good overview of all assigned outputs in the matrix display of the* routing manager *(↗332).*

Applying Effects to Multiple Channels Simultaneously.

To place multiple instances of the same effect on multiple mixer channels:

1. Select the channels to which you want to apply the effect. To do this, first click on the channel number of the first channel to select it. Then hold down the **Shift** key and select the number of the last channel. This also selects all intermediate channels. If you press the **Ctrl** key instead of the **Shift** key, you can add any other channels to the selection.

2. Now select an insert effect for one of the selected channels and adjust it to your liking. As soon as you close the effect window, the effect with these settings will be loaded into all selected channels.

   > ⚠️ *This only works if you close the effect dialog immediately afterwards without opening other effect dialogs first.*

3. If you later select multiple channels and make changes to one instance of the effect, those changes will be applied to all instances of the effect in the selected channels.

   > 💡 *This behavior can be deactivated in* **Program Preferences** > **Effects** > **General** *with the* **Copy effect settings to selected tracks option***, separately for effects that you have newly inserted or that you had switched off and reactivated, and for effects that already exist.*

4. To remove the effect from all selected channels, it is enough to remove it from one insert slot. All other instances of the effect will be removed as well.

VCA Groups

VCA is the abbreviation for Voltage Controlled Amplifier. These are used in analog mixing consoles. A fader no longer directly controls the level of a channel, but only a control voltage, which in turn adjusts a mixer channel via a voltage regulator (VCA). Since one control voltage can also be used to control several VCAs, the level of several channels can be adjusted together via a VCA group.

Why do you need VCA groups? To adjust the level of several channels together, you could route the channels to a Submix bus and adjust the level of the channels together at the channel fader of the Submix bus. However, if these channels each feed AUX sends that are routed post FX, i.e. are dependent on the level of the channel, these AUX levels (which determine the dry/wet balance of an effect, for example) cannot be controlled via the Submix bus. When using VCA groups, the individual channel signals and the AUX send feeds that depend on them are still controlled.

You can access the VCA group functions of **Sequoia** via the **VCA** buttons of the channels. When you click a VCA button, a context menu shows you whether and via which VCA master (VCA group) the corresponding fader is controlled.
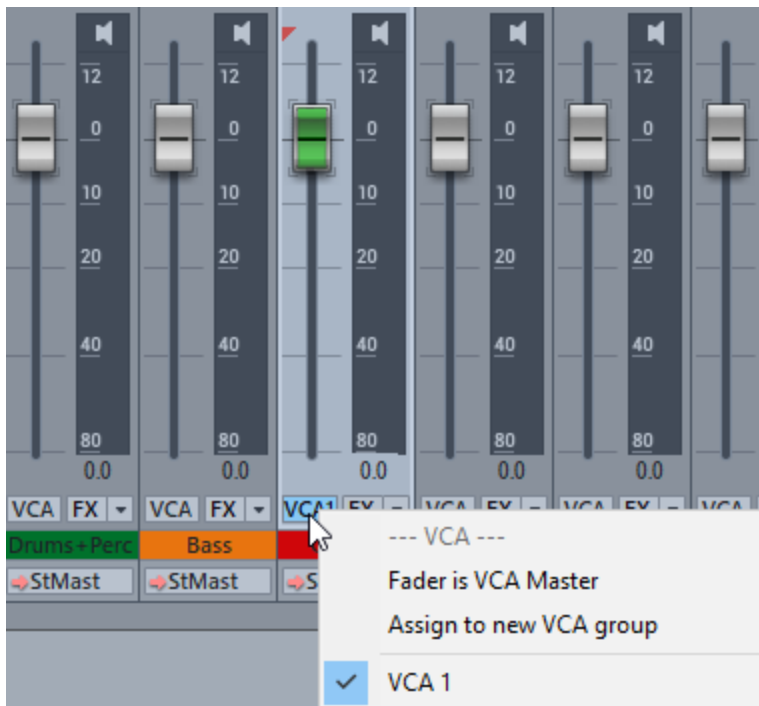
To assign channels to a new VCA group:

1. Select the channels that you want to control via a VCA group and click the **VCA** button of one of the channels.
2. Select the option **Assign to new VCA group**.

3. A new VCA master channel is created on the right edge of the mixer in front of the master, to which the selected channels are assigned.

4. The VCA button lights up on the channels and the VCA group number is displayed on the VCA buttons

Ghost Fader

Even if a channel has been assigned to a VCA group, you can still control the volume of a channel individually. The values are summed and a "ghost fader" appears in the channel showing the resulting level of the channel. When you change the volume in the channel, the ghost fader moves accordingly along with the channel fader.
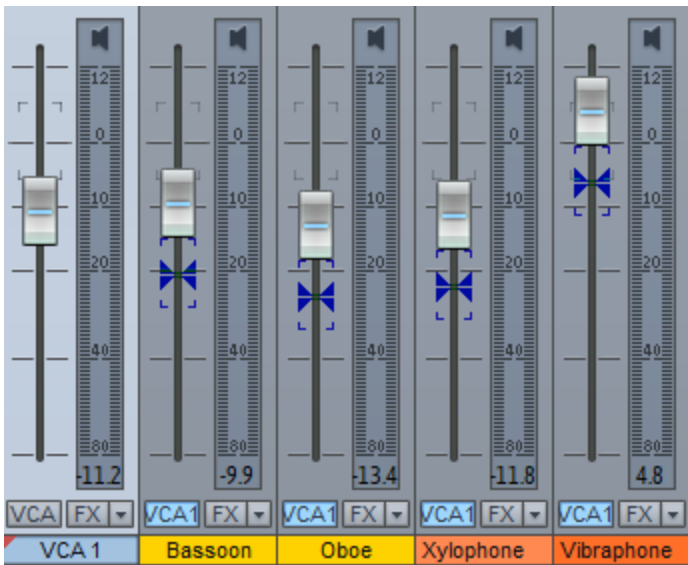
You can create as many VCA groups as you like. However, each channel can only be controlled by one VCA group at a time. If you assign a channel to another VCA group, the previous assignment is canceled. VCA groups can be controlled by other VCA groups, i.e. you can assign the master fader of a VCA group to a group in turn ("nested" VCAs). This works up to a maximum of four levels.

Changing a Channel Fader into a VCA Master

Each fader can serve as VCA master for other channels. If you select the option **Fader is VCA Master** in the VCA menu of a channel, this channel appears as a VCA group in the menu.

ⓘ *The channel fader still controls the volume of this channel as well.*

So you can also do the VCA grouping like this, by creating a new track, activating the **Fader is VCA Master** option there and then selecting this VCA group in the tracks that are to be controlled. The track name is used as the group name in the menu and on the buttons for VCA control.

💡 *Since there is little space on the buttons, name your VCA master as briefly as possible.*

Remove VCA Group Assignments
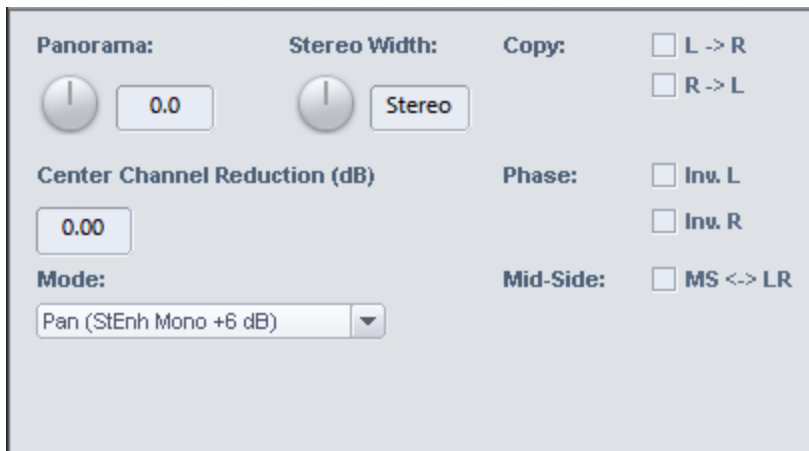
Remove the VCA group assignment for a channel by clicking the group again in the channel's **VCA** button menu. When deactivating the **Fader is VCA Master** option in a VCA Master all assignments for this VCA group will be canceled.

> 💡 *Assigning and deleting VCA assignments can be done in the* Routing Manager*VCA (↗334).*

Panorama Editor

Right-click on the Panorama slider in the Track Head, Mixer, Track Editor or Object Editor to open the Panorama Editor.

In addition to panning for objects and tracks, which you can also set with the normal panorama slider, you will find further options for influencing the stereo image and phase settings in this dialog.

With mono tracks, the signal is converted from mono to stereo in the signal flow at the position of the panorama controller. From this point on, effects or plug-ins can be included in stereo. You can change the routing position in the Effect Routing dialog (↗220).

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 width. If the slider is set to the leftmost position, you will only hear the mono signal. To the right, the side signal (difference of L-R and R-L respectively) is mixed in, thus increasing the stereo base width.

> ℹ️ *With the* Multiband Stereo Enhancer *(↗289) you can perform a frequency-selective processing of the stereo base width in three bands.*

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. If you activate both options, the left and right channels will be swapped.

Center Channel Reduction (panning law): To compensate for volume fluctuations during panning, it is common practice to additionally lower the track volume in the pan center position. The setting depends on the audio signal. Usual settings are:

- **0 dB**: This setting is recommended for stereo material. There is no volume reduction in the center position, i.e. when panning to the right, the right channel remains at unchanged volume, when panning to the left, the left channel remains unchanged. The audio material is not changed if set to the center position. With mono signals on a stereo track, this setting may increase the 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.

- For stereo signals **-3dB** or **-4.5dB** are also common settings.

Phase: Here you can invert the phase for the stereo channels individually. By default, the Phase button in the Mixer and Track Editor inverts the phases of both stereo channels. You can change this behavior via the option **Use mixer phase switch only on left channel** at **System options** > **Effects** > **General**. Then the phase button only works on the left channel and you can use it to correct the classic "phase shifts" that occur with incorrect wiring and that often affect only one stereo channel.

Mid-Side: With the option **MS->LR** you can convert a signal present as Mid/Side into a stereo signal. It is assumed that the mid (mono) signal source is on the left and the side signal is on the right.

Mode: Here you can set 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

In the Panorama editor for tracks there are some more options

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. To remove the output routing, select **Remove output routing** from the menu ▼ on the tab. Click on one of the tabs to open the corresponding Panning editor.If the track is routed to a surround bus or master, the Surround Editor (↗517) dialog appears accordingly.

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.

This option can also be set directly in the Effect Routing dialog. To do this, deactivate the Stereo Pan entry in this dialog.

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 "mid/side" and describes a method in which the stereo information is recorded in a center channel ("M") and a side channel ("S") rather than in two channels left and right.

MS recordings

With MS microphoning, you get recordings in which there is no left and right channel, but the mid and side parts are in separate channels. You can process MS recordings with the included presets of the dialog **Track Panning**. These are assumed to be stereo files with M on the left and S on the right.

Stereo processing of MS original material

To extract the correct stereo picture from an MS file, duplicate the source material and place it on 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, you can use the preset **Side signal (stereo) (from MS source)** to play back only the S signal in stereo, i.e. +S on the left and -S on the right. Then mix both tracks together with 0 dB each.

🔆 *Tip: For this use case you can also use the **MS-> LR** option in the dialog.*

MS processing of stereo original material

Another application is the independent processing of center and side content, even if the source material is in stereo format. Again, duplicate the source material and place it in 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 Mid/Side processing

In addition, presets are available for using stereo signals in mono tracks:

- **Mono (get mid signal from stereo source)**: With this, only the mid component of the stereo signal is used in a mono track.

- **Left Channel only**: This allows you to use only the left channel of the stereo signal in a mono track.

- **Right Channel only**: This allows you to use only the right channel of the stereo signal in a mono track.

- **Side signal (mono) from stereo source**: This allows you to use only the mono side part of the stereo signal in a mono track. This way, you can apply all subsequent object and track effects to the side part as mono effect.

- **Side signal (stereo) from stereo source**: This allows you to use only the stereo side part of the stereo signal in a stereo track. This allows you to apply all subsequent object and track effects to the side part as stereo effects.

- **Convert side signal (mono) to stereo**: This allows you to convert the mono side part of the signal to stereo.

Buses

Busses are used to "collect" audio signals from other tracks to treat them together in the mix. Since in Sequoia each track corresponds to a mixer channel, a bus is also initially a normal track with all the properties of a normal track, i.e. it can accommodate track effects and be automated, has an output and an input. But beyond that, there are differences.

- **AUX buses**: An AUX bus differs from a normal track in that you can send the audio signal from any track to this AUX bus. To do this, use the **AUX-Send** slider in the mixer.

AUX busses are mostly used for effects like reverb or echo, when they are to be used by multiple tracks at the same time. For this purpose, a signal portion from the corresponding track is sent to the AUX bus ("AUX Send") and effects are applied in the AUX bus. The volume fader of the AUX bus represents "AUX Return".

- **Submix busses**: A Submix bus differs from a simple track in that any track can be routed to this bus instead to the master or an output device.
  A Submix bus thus combines several tracks. Volume, panning and effects processing can thus be done for all tracks routed to the Submix bus together in this bus. For example, you can combine all drum tracks into one Submix bus, so that the entire drum kit can then be controlled via the volume control of the Submix bus.

Via Menu **Track** > **Track type** or the Track Options Dialog (↗141) each track can also be configured as AUX and/or Submix bus.

Since a bus represents a normal track in the project window, this track can also contain objects. Normally you will not place objects in a bus yourself, but they will be created, for example, if you freeze the bus (↗139)

> ℹ️ *To prevent you from accidentally placing objects on a bus track, buses are created as locked by default.* 🔒

So, since you usually don't need these tracks in the arranger, to save space you can hide them in the arranger with the command Menu **View** > **Hide Submix/AUX busses**.

> ℹ️ **Attention**: *This function hides the bus tracks even if they are set to visible in the Track Manager in the column* **Arrangement**.

For the surround mix there are other bus types:

- A surround bus corresponds to a normal **Submix bus** with surround functionality. All tracks routed to a surround bus get 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.
  If there is no Surround master 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 hardware outputs that are specified for this purpose in the Surround Setup dialog (↗513).

  > ℹ️ *Technically, the Surround master is also such a surround bus, which has the additional property of assigning the surround channels to the audio outputs and cannot itself be routed to a Surround bus.*
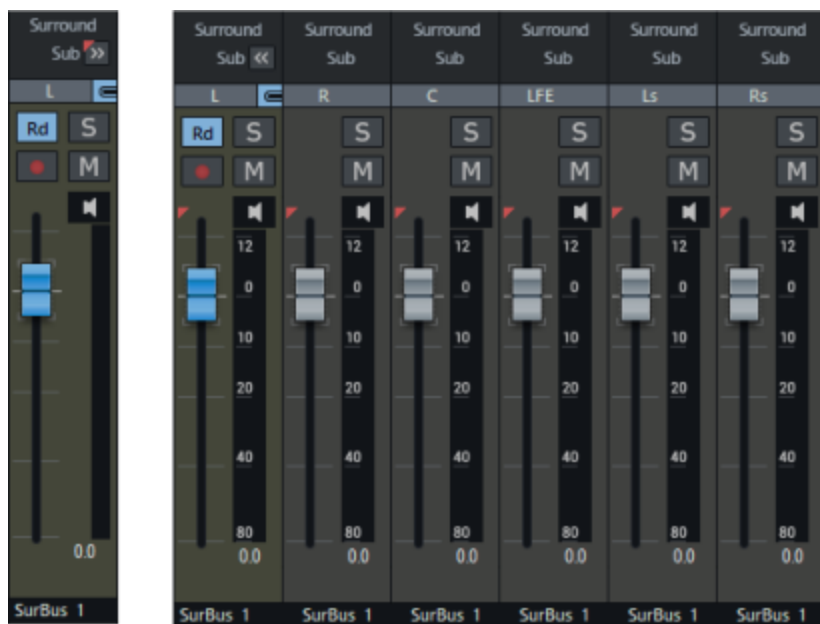
The individual channels of a Surround bus are normally hidden, by clicking on **Sub >>** they can be displayed. If you disable the link button ⊑ , you can also set the channels separately.

- **Surround AUX bus**: If a surround master is available in the project, an AUX bus can also be created as a surround AUX bus. A surround AUX bus allows integration of surround effects. To adjust the surround panning of a surround AUX send of a track, open the **(Track/AUX) Surround editor** in the context menu of the send control.

Creating AUX busses and Submix busses

To create an AUX bus, select **Insert New Track** > **New AUX Bus** from the **Track** menu. You can also create a new AUX bus by dragging an AUX send control of a previously unused slot in the AUX section of the Track Editor or Mixer. This creates a new AUX bus automatically. New AUX busses are always inserted at the far right of the mixer before the masters, after existing AUX busses.

To create a Submix bus, select **Insert New Track** > **New Submix Bus** from the **Track** menu. The Submix bus is generated directly after the selected channel.

💡 *The Mixer Setup dialog (⏴646) lets you quickly add multiple buses to the project at once.*

AUX routing

To set the signal tap of the AUX Send control as **Direct Out**, **Pre-Fader** or **Post-Fader** use the context menu of the control:
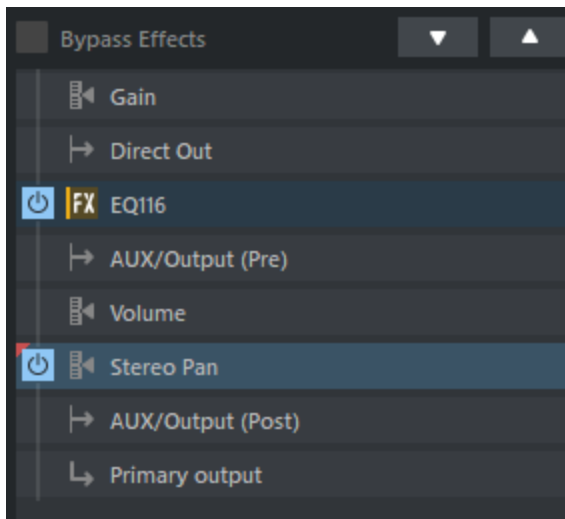
You can see the positions of these signal taps within the processing chain of a track in the FX routing dialog. Here in the picture they correspond to the default setting. Use the arrow buttons in the upper left corner to move them in the signal flow.

Direct: means that the AUX send signal is tapped directly after the **Gain** control, i.e. the input gain. The AUX slider in the mixer channel and track editor is displayed in red.

Pre-fader means that the AUX send signal passes through the input gain, equalization and effects processing to be tapped before the volume control. The AUX signal is thus independent of the channel's volume setting. One main use for this is to set a separate monitor mix that then goes to a separate hardware monitor output via the AUX bus. The AUX slider in the mixer channel and track editor is displayed in yellow.

Post-fader means that the AUX send signal is tapped after the volume control, after it has passed input gain, equalization and effects. AUX sliders that control newly created AUX sends are preset Post. The AUX slider in Mixer Channel and Track Editor is displayed in orange.

AUX sends routing dialog

In this dialog you have access to all AUX sends used by a track, even if the corresponding section in the Mixer/Track Editor is minimized. Here you can also create new AUX sends or determine the send value and pan of existing AUX sends.

Open the AUX Send Routing dialog in the context menu of the AUX button of a mixer channel or track editor with **Track AUX Sends...**.
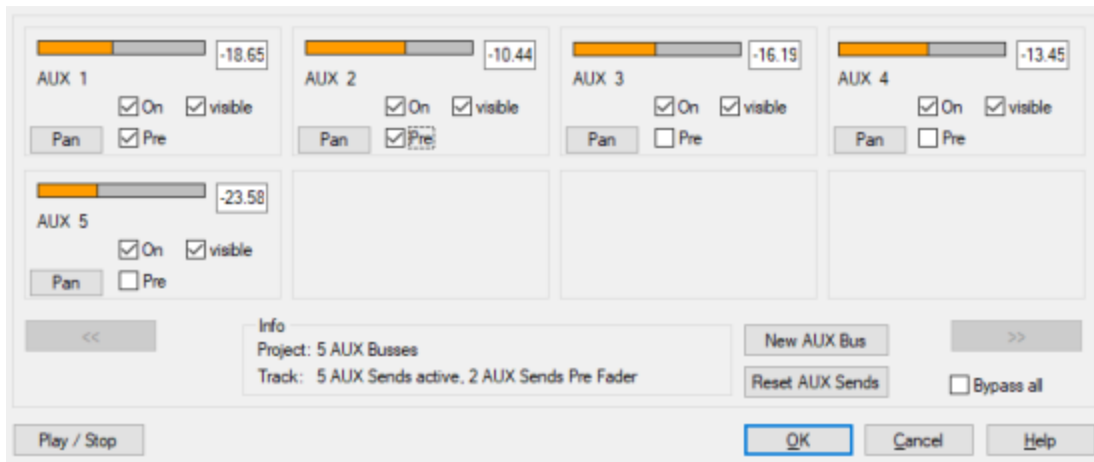
You can enter the Send level of the respective AUX bus numerically or drag the orange bar.

On: Activates and deactivates an AUX send path.

Pre: By default, all AUX buses are routed "post-fader" in Sequoia. To switch them to "Pre-Fader", place a checkmark the "Pre" box.

Visible: Shows and hides individual send controls in the AUX sections in Track Editor and Mixer.

Pan: Opens the Stereo Panorama dialog (↗492) for the AUX sends. 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 settings are reset.

Master section

A master is similar in structure to a track. In the following, only the differences to the controls of the track are described and the controls that are additionally available in the master.

The **In** section is not present, the input signal is composed of all tracks and buses routed to the master. Therefore, there is also no section **AUX**.

The Plug-Ins section, here called **Master Plug-Ins** works exactly the same as the one in Tracks. In addition, you can select an export preview function for AAC or MP3 (↗501) in the plug-in browser here.

The **master equalizer** is identical to the one in the tracks, but there are additional knobs for adjusting the frequencies of the bands.

Stereo enhancer: The panorama control is replaced by a stereo base width control. Clicking on **StEn** opens the dialog of the multiband stereo enhancer (↗289). With the Stereo Enhancer you change the stereo image of the overall signal separately for three frequency ranges. As with EQ, as long as no adjustments are made, the effect is not active.

Mono: This button causes the overall signal to be played back in mono. This can be used to test mono compatibility.

Normalize (N): This is the master normalization. When you click this button, the master level is adjusted so that the loudest part played reaches 0 dB. The basis for this is the maximum level reached during the last playback process, which is displayed above the peak meters.

> **i** **Note**: *If the level display is clicked in the stopped state, the play cursor will jump to the position indicated by the level.*

Link: This button links the left and right channels of the master signal.

Master automation: The master can also be automated. However, the master track is hidden by default in the arranger. To edit the automation curves for the master, select **Show master track in arranger** in the context menu of this button.

> **i** *For detailed information on automation, refer to the chapter "Automation (↗338)".*

Master Out: Only hardware outputs are available in the Master Out menu. Specify the playback device for the master here. Select **Master inactive** if you do not want to use a master output in the project (for example, to use multiple buses routed directly to output channels instead).

Mix to File

You can perform a mixdown in real time, in which the output of the master is written to a file during playback. You can change your mix during playback, just as mixing was done in the analog studio: The mix is played back and the controls on the mixing console are moved live.

1. By clicking the **Mix to File** button you can specify the name and location of the wave file. Once you click **On**, the master output will be written to a wave file during playback.

2. Now start the playback of the project. During playback, any parameter may be changed to record any live sound adjustments that are made.

3. Stop the playback.

4. When you are satisfied with the recording, uncheck the button **On**. Otherwise, each time you start play back, the wave file will be written again, overwriting the previous mix.

5. Alternatively, you can click **Mix to File** again. Then a new recording file with a consecutive number is created.

> ⓘ *You do not necessarily have to use the **Mix to File** function to record mixer movements during playback. Use the* Automations *(↗338) function instead!*

MP3/AAC preview
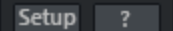
In the Master Plug-ins section there is a preview function that allows you to prepare your project for MP3 or AAC export. This plug-in lets you preview in the mixer in real time how the exported file will sound with the selected encoder settings.

To do this, you can change the export settings such as bitrate or encoding quality in the plug-in dialog. With **Get settings from MP3 export** you copy the MP3 export settings from the export dialog. **Apply settings to MP3 export** transfers the settings back to the export dialog.

Global Mixer Functions

At the edges of the mixer window there are a number of buttons for cross-track control of the mixer window and mixer behavior.

Mixer Setup and Help

`Setup  ?`   **Setup**: Click on the **Setup** button to open the Mixer Setup dialog (↗646). **?** opens the help for the mixer, i.e. the text you are just reading.

Sends on Fader

With this function you place the small sliders for one AUX path each on the large channel faders. This makes it more convenient to set and compare the AUX send level for the tracks. (This corresponds to the AUX page on digital mixers, as these usually do not have separate AUX controls)
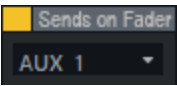
Select an AUX bus from the list box to activate the function. The function can be switched on and off by clicking on the yellow button.

Reset Mixer

The buttons under Resets allow you to reset various properties of the mixer.
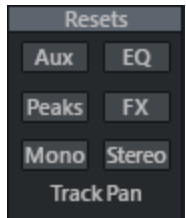
Reset AUX: This resets all AUX sends.
Reset EQ: This resets all equalizer settings.
Reset Peaks: This resets the LED peak meters (peak hold display).
Reset FX: This removes the effects in all channels.

Reset (Mono): This resets the entire mixer to its default settings for mono tracks. The tracks are alternately set to 100% Right and Left and all special settings of the Track Panorama settings (↗492) are reset.

Reset (Stereo): This resets the entire mixer to its default settings for stereo tracks. The pan controls are set to center position and all special track pan settings are reset.

Loading and Saving Mixer Settings

You can save the state of the entire mixer into Snapshots. (For hardware mixers this is called "Total Recall"). This includes the levels of all volume and AUX send controls as well as all track effect settings and routings.

Click one of the buttons to save. To load the snapshot, click this button again. To overwrite a saved snapshot, click the button while holding down the **Shift** key. Double-click in the name window to name the snapshots. The last loaded snapshot is highlighted.

Load mixer settings / Save mixer settings: With these buttons you can save the current mixer settings to a file (*.mix) to use them in other projects and load saved mixer settings.

In the context menu of a snapshot button you can reach further functions:

- **Load/Save Snapshots**: This corresponds to the function of the button, **Delete Snapshot** deletes the snapshot from the button

- **Next Bank/Previous Bank**: A project can contain a total of 32 snapshots in 4 banks, use these commands to change the displayed bank.

- **Save mixer to file**: This corresponds to the function of the Save mixer settings buttons (without snapshots).

- **Save mixer with snapshots**: Use this option to save the current mixer settings and all snapshots. When loading mixer settings that contain snapshots, you are asked whether the snapshots should also be loaded. If so, the snapshots of the project are overwritten by all snapshots of the file.

- **Load mixer from file / Load mixer from file (including track names)**: You can load mixer settings both with and without track names.

- **Load last mixer:** When loading a snapshot, the current mixer settings are temporarily cached and can be retrieved with the command **Load last mixer**. This also allows an A-B comparison between the snapshot's and the current settings.

- **Include record/monitoring states:** If this option is activated, the record arm and monitoring states are also adjusted when loading snapshots and mixer files.

-💡- *With Menu **View** > **More** > **Load Mixer Snapshot** you open a menu from which you can select the desired entry with the arrow keys and confirm it with the Enter key. If you also assign a* keyboard shortcut *to this menu item, you can switch snapshots in this way completely keyboard controlled.*

Playback Functions
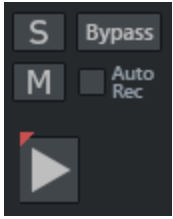
Solo/Mute: Use the **S** and **M** buttons to toggle all Solo/Mute buttons on and off.
Bypass: This button allows you to turn off all effects active in the project except for the volume and pan settings of the channels for comparison purposes.
AutoRec: This display will light up if the level automation is written in the master.

Play/Stop: This button can be used to play and stop the project, a right click opens the transport console.

View
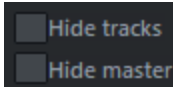
Hide Tracks/Hide Master: These buttons can be used to hide the tracks or the master section in the mixer.

Mixer layout snapshots: If the arrangement has many tracks, usually only a part of all mixer channels is displayed in the mixer window. The scroll bars at the bottom and right edge of the mixer can be used to move the visible section.

In the Track Manager, you can also hide individual tracks in the mixer (↗134). The four snapshot buttons allow you to save this visibility state of the tracks together with the current section, i.e. the first track displayed in the mixer, and then restore a specific view with one click. To save, click one of the four buttons on the left edge of the mixer. To load the snapshot, click this button again. To overwrite a saved snapshot, Shift-click the button.

-💡- *With Menu **View** > **More** > **Load Mixer Layout Snapshot** you open a menu from which you can select the desired entry with the arrow keys and confirm it with the Enter key. If you also assign a keyboard shortcut to this menu item, you can switch the layouts in this way completely keyboard controlled.*

Solo and Monitor Control

At the bottom right, there are basic setting options for the monitor control.

ℹ️ *For more options and settings for monitoring, use the* Monitoring Section *(↗507).*

The Solo knob determines the monitoring volume in solo mode, while the Monitor knob additionally influences the level at the monitor output. The value corresponds to the Volume setting in the Monitoring section (↗507).

The monitor output can be selected at the very bottom. This corresponds to the first output device of the monitoring section.

> ℹ️ *Monitoring via separate monitor buses is only available when using the* hybrid engine *(↗692).*

Via the switch **AFL/PFL** you can tap the signal for solo playback in two different places:

- **PFL**: With PFL (Pre Fader Listen), the signal is tapped before the volume fader and effects, but after the input gain.

- **AFL**: With AFL (After Fader Listen), tracking effects, pan and volume settings are taken into account and affect the monitoring signal.

This switch corresponds to the Global Solo mode **Global: Solo PFL**

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 tracks in solo status are output via the monitor bus.

Global Solo Modes

The button for selecting the monitor bus allows you to define global default settings for solo playback. This can also be accessed by right-clicking on the solo buttons.

Global Solo Modes	>	✓	Global: Solo In Place
Monitor bus inactive (---)		✓	Global: Solo Safe
			Global: Mute/Solo Safe Pre Sends
1 + 2: VM-VAIO 1+VM-VAIO 2 (Voicemeeter VAIO3 Virtual ASIO Out)			Global: Solo PFL
3 + 4: VM-VAIO 3+VM-VAIO 4 (Voicemeeter VAIO3 Virtual ASIO Out)			Global: Solo Exclusive
5 + 6: VM-VAIO 5+VM-VAIO 6 (Voicemeeter VAIO3 Virtual ASIO Out)			
✓ 7 + 8: VM-VAIO 7+VM-VAIO 8 (Voicemeeter VAIO3 Virtual ASIO Out) (Monitor Bus)			

Global: Solo In Place Solo in Place lets you hear the channels set to Solo in the mixer as they are placed in the stereo image of your 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. This option is active by default.

Global: Solo Safe. In Solo In Place mode, this option causes any channel switched to Solo to automatically be listened to along with the AUX Return channels it is feeding. This option is active by default.

Global: Mute/Solo Safe Pre Sends (only available in Hybrid mode). This is an additional "Solo Safe" function for AUX sends/outs configured as pre or direct sends.

Global: Solo PFL (only available in Hybrid mode) You can also access this option via the **AFL/PFL** button. It sets the behavior of the solo tracks to PFL. In this case, the signal is tapped before the fader and effects, but after the input gain. In the deactivated state, the solo tracks are tapped after the fader.

> **ℹ Note**: *Economy Tracks cannot be monitored with PFL.*

Global: Solo Exclusive: In this mode, the Solo button switches this channel exclusively to Solo. For all other channels the solo status is cleared. If you also want to listen to the AUX return channels of the "Solo" switched channels, **Global: Solo Safe** must also be active.

> **ℹ Note**: *If the mode **Global: Solo Exclusive** is not active, click with **Shift + Alt** on the "Solo" button of a channel for **Solo exclusive**. Conversely, if "Solo Exclusive" is active, you can listen to multiple channels at the same time in "Solo" state using the same key combination when you click Solo.*

Global Solo modes without monitor bus

If you cannot or do not want to use the Hybrid Engine or do not want to use a Monitor Bus for other reasons, you can also select **Monitor Bus inactive (---)**. The following differences then exist:

- Solo In Place is always active and cannot be deactivated.

- Also in "Solo in Place" mode it is possible to switch between Pre Fader Listen and After Fader Listen. With a monitor bus, PFL is no longer In Place.

> **⚠ Attention**: *In this mode it is possible to influence the master level with the monitor control. This means you can set up the listening level in the mixer without using any external volume controllers. The audible result in this case differs from the peak meter display. Therefore, the numerical level values in the mixer are displayed in red. Additionally, the Solo knob influences the volume of the soloed tracks that are played through the master section. However, the internal level for the export remains unchanged.*

Monitoring Section

The Monitoring section provides a separate workflow for monitoring inputs, busses or masters. It is possible to switch between four 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.

> ℹ️ *Note: The* Monitoring *(↗692) of the tracks is a different, independent function. This means monitoring the input signals of tracks before and during a recording. The monitoring section is generally used to control the audio output of the program.*

The settings in the Monitoring Section apply across all projects, i.e. they are independent of the loaded project.

To display the Monitoring Section, go to the menu **View** > **Monitoring Section** or use the keyboard shortcut **Ctrl + Shift + S**.

> ℹ️ Monitoring via separate monitor buses is only available when using the hybrid engine *(↗692)*.

Input
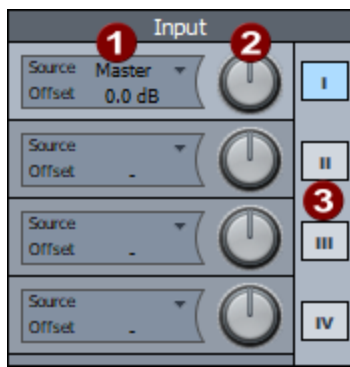
❶ Select Source: Default source is the Master, but you can also select any input pair or (stereo) bus as input source.When you select a surround master, you can open the downmix matrix in the menu to create a stereo downmix if a stereo output is set under Output.

When using a separate monitor bus, you can use **Solo on Monitoring section** to limit soloing to monitor output.

Sources selected here that are not available in the current project are shown in red.

❷ Input Offset: Here you can adjust the level of the input signal independent of the selected output.

❸ Switch Source: You can switch between 4 different sources.

> ℹ️ *If the master or a bus has been selected as the input, and their outputs are routed to the same output device as a monitoring section output, the signal will not be played back via both paths.*

Volume/Output/Plugins

1 **Output selection**: A separate output device can be set for each output. Settings in the ranges **Volume** (except the large knob for the global volume) and **Plug-ins** apply separately to the respective output. To display the settings in these areas for the respective output, activate the corresponding field below **Output**. Use the buttons **1...4** to activate the according output.

2 **Output level offset**: The controls can be used to trim the monitor output level per output by setting an offset to the global monitor level.

3 **Device selection**: Click the arrow ▼ to select the device for the output. The device for output 1 corresponds to the device that can be set under Monitoring (↗501) in the mixer.
If you deactivate the **Exclusive** option in the menu, other outputs can be activated independently of this output. You can also mark an output as **Headphone**. For headphone outputs, volume control is only through the offset control, independent of the global monitor level. This state is indicated by **!!** behind the dB display in the large knob.

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 others 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 global monitoring volume. The value corresponds to the setting **Monitor** (↗504) in the mixer at the bottom right.

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**: Mute switch.

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.
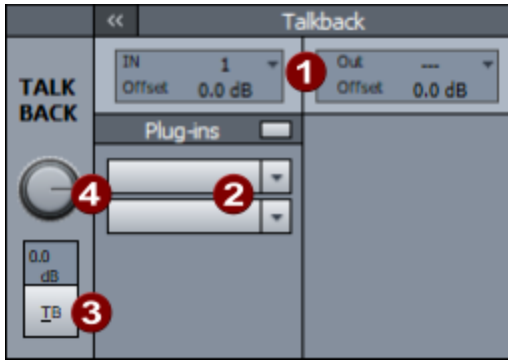
1 **Input/Output**: Here you define the input and output device for the talkback.

2 **Plug-ins**: There are also plug-in slots for the talkback.

3 **Activate talkback**

4 **Talkback Volume**

Keyboard shortcuts for the monitoring window

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. So you must have clicked in the window at least once after opening it.

> 💡 *For talkback, you can define a global keyboard shortcut in the* **Program Preferences** *(↗709) in the section* **Keyboard/Menu/Mouse** *under* **Special Keys** *that also works independently of the focus of the monitoring window.*

Alt + 1..4	Select Input 1-4	**H**	Invert phase
	ℹ *These keyboard shortcuts are global.*		
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)
N	Mono (Input L+R to Output LR)	**Shift + T**	Talkback on/off

Monitoring section signal flow
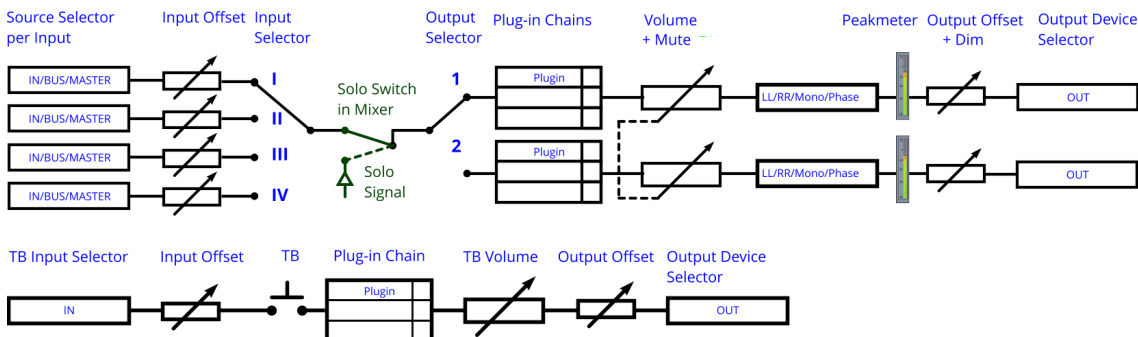
Mixer Signal Flow

The following illustration shows the audio signal flow of a track during playback.

Hardware In
Primary output of other tracks
Record Arm
Object Effects Chain*

Track Effects Chain*
Solo PFL
Wave on HD
Gain
Stereo Pan
Gain
Objects
EQ
AUX send Direct
Reversed
Plug-In
EQ
Resampling / Timestretching
Plug-In
Plug-In
ARA2
...
Plug-In
Object Effects Chain
Volume + Fade in/out
...
AUX send Pre
Volume
Solo AFL
Monitoring
Object AUX Sends
Stereo Pan
AUX Bus In
AUX send Post
Submix Bus In
Direct/Pre/Post
Sidechain Sends (to other tracks)
Track Effects Chain
AUX Sends
Solo to Monitoring
PFL/AFL
Busses Additional Masters Hardware Outs
* Default order, is changeable
Primary master

Notes:

■ 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 effects chain (direct, pre, post) can be specified.

■ One output must always be routed Post, this is the Primary Master. It is used for the peak meters in the mixer and is preset as input in the monitoring section.

■ The switches **Record Arm** and **Monitoring** are interdependent depending on Monitoring switching behavior (↗695) and playback state.

■ Monitoring cannot be enabled for tracks that use the output of other tracks as input.

- Record Arm is not available for buses.

- The bus inputs of a track are only active if the track is an AUX or Submix bus and always receive the sum of all aux sends or out routings of the other tracks, regardless of the playback state.

SURROUND SOUND

Sequoia offers extensive options for mixing audio material in various multichannel formats - from LCR (Left-Center Right) to various 5- and 6-channel formats 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. Target formats are e.g. 5.0 ITU 5.1 ITU or 7.1 SDDS.

In the **Surround settings**, you can assign existing playback devices to the channels of a surround master. Use the **Surround Editor** to distribute the audio from the tracks of an arrangement that are routed to a surround master across the available surround outputs. In the mixer, the panorama knob in these tracks is replaced by a Surround panorama display.
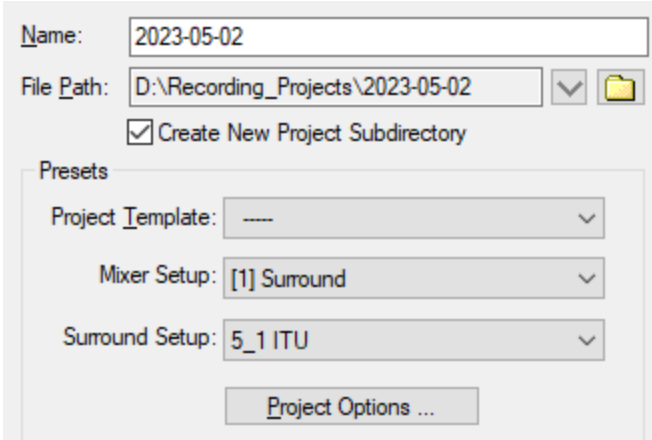
You can also route individual objects directly to the surround master and distribute them in the surround panorama, regardless of whether the track on which they are placed is routed to a stereo or surround master.

The Surround Editor offers different modes for arranging signals in 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, with the necessary upmix or downmix being performed.

Create New Surround Project

In the **Setup for new Project (VIP)** dialog, select the option **Surround** under **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 a Surround Master in the selected format (e.g. 5.1 ITU L, R, C, LFE, Ls, Rs). All mixer channels of 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

Click the **Setup** button in the project's mixer. The **Project Options** > **Mixer Settings** dialog opens.
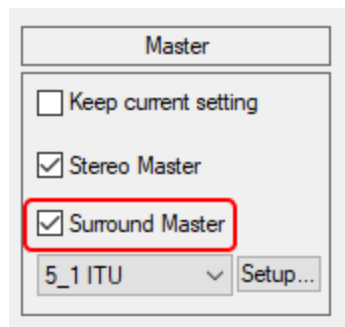
In the section **Master**, activate the option **Surround Master** .

In the list box below you can select your preferred surround format (e.g. 5.1 ITU). For further adjustments and the assignment of the surround channels to the output devices, you can open the dialog Surround Settings (↗513) with the button **Setup**.

Surround Settings

The setting of the physical outputs of the surround buses is done in the dialog **Surround settings**.
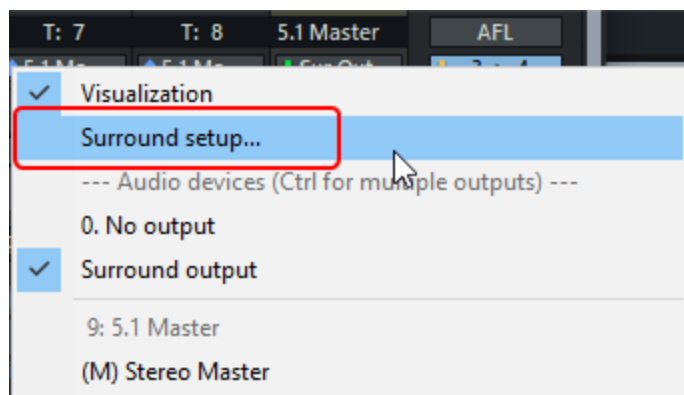
To open the dialog, use the **Setup...** button in the Surround Editor (↗517) or in the Master section of the Mixer Settings (↗646).

This window is used to specify the surround format in which the mix will be made. Here you can load various 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

Here you can load already created surround formats or save newly created formats. A preset features the Surround buses, their name as an abbreviation, the sequence, and the position coordinates of the speakers. When loading a preset, the assignment of the playback devices is also taken over.

Audio device settings:

- **Load/save in preset**: If the option is active, when saving presets, the current playback device configuration is saved in the preset regardless of the default configuration.

- **Get default for preset**: This applies the default configuration for the playback devices that was assigned to the active preset.

- **Set default for current preset**: This assigns the current assignment of the surround busses to the playback devices to the active preset as default configuration. This configuration is applied when the preset is loaded.

Channel Configuration

Name: Setting of the name of the individual surround channels. The abbreviation in the following column **Short** is generated from this name. For the abbreviation, the first letter of each word used is used as consecutive letters.

- 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 export, the abbreviation is automatically appended to the file names.

> ℹ **Example**: *Select the name `SurroundMix` when Surround bouncing a 5.1 Surround mix as 6 mono files. Then, wave files are created with the names: `SurroundMix_L.wav`, `SurroundMix_R.wav`, `SurroundMix_C.wav`, `SurroundMix_Ls.wav`, `SurroundMix_Rs.wav` und `SurroundMix_LFE.wav` generiert.*

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 of the sound card to the respective surround channel. You can save a default device configuration for each surround preset (see above).

Lock channel position: If the option is active, the position of the speakers can no longer be changed in the Surround Panorama Editor in sound field mode. It is always active by default and should only be switched off for special tasks (e.g. variable speaker arrangements).

Arrow up/arrow down button: In the configuration table, one line corresponds to one surround channel. Use the arrow buttons to move a selected row down or up in the table.

> ℹ *The order of the surround channels in this table also determines the order of the channels when they are displayed in the mixer, in the Surround Editor (e.g. Peak Meter), and in the surround effects dialogs.*

LFE Filter

For the LFE (Low Frequency Effect) channel, you can activate a filter in the surround settings. You can choose between a 6-band EQ116 (default setting) and an FFT filter.

Cut off frequency: Here you can set the cut-off frequency for the low-pass filter.

High frequency reduction: The reduction of the signal above the cut-off frequency is set to 12 dB/octave for the 6-band EQ.

Edit EQ: This opens the dialog for adjusting the corresponding filter.

Initialize all tracks to stereo position: Activate this option to obtain a correct downmix for a stereo export of a surround mix, i.e. you can then perform a stereo export from a surround project with the downmix set correctly (⤢527).

Track and Object based Surround Panning

The arrangement of audio in the surround space can be done in two different ways, track based or object based.

Track based Surround Panning

By default, the audio signal of all objects passes through the channel strip of the corresponding mixer track. This track can be assigned a position in the surround panorama via the Surround Editor (⤢517) in the associated channel strip of the mixer. All objects contained in this track are thus also positioned at this point in the surround panorama.

Object-Based Surround Panning

To assign individual objects their own surround position independently of track-based panning, each object can be routed directly to a surround master or another surround submix bus.

To do this, open the object editor of this object by double-clicking on it, switch to the tab **FX** and open the Surround Routing menu in the section **Pan** by clicking on the **Surround** button.
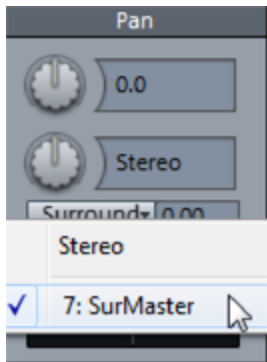
Select a surround bus or master from the menu to route the object to the appropriate bus or master. Hold down the "Ctrl" key while selecting in the object editor to route the object to multiple busses at the same time and thus combine different surround pannings. An additional slider appears in the object editor which can be used to adjust the object's send level to the Surround Bus or Master.
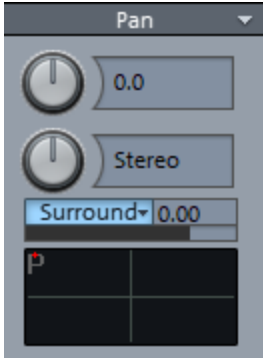
Right-clicking in the surround panorama field below opens the corresponding Surround panorama editor (↗517). Here you can set the positioning in the Surround panorama. Object-based Surround panning is displayed in the object view in the arranger with the appendix **-> Sur**.
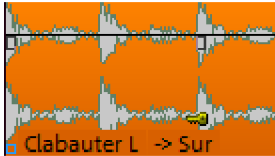
> ⓘ **Note**: *With object-related surround panning, the audio signal no longer passes through the channel strip of the associated mixer track. All settings made there (AUX send, EQ, etc.) have no effect on this object.*

Surround Editor

In the Surround Editor the panning of a track or object on the Surround bus can be set and edited.

Open the Surround Editor for a track by:

- ■ Right-clicking on the surround pan display of the track in the Mixer/Track Editor or
- ■ Right-click on the "pan" button or the pan slider in the track head or
- ■ Choosing **Track** > **More** > **Pan/Surround Editor...**.

For object-based surround panning, open the object editor for the respective object by double-clicking and then right-click on the surround pan display.

Surround Panner VST Plug-ins

You can also perform surround panning with a third-party VST plug-in. To load the plug-in into the surround editor, select the plug-in in the menu on the output tab at the top of the surround editor or in the left-click menu of the surround pan control in the mixer at **Surround Panner (VST)** or open the plug-in browser via the menu item there.

Controls of the Surround Editor

At **Mode** you can choose from seven different display modes: Sound field mode, Panning law mode, Angle mode, Matrix mode, VBAP mode, and the 3D modes VBAP (3D) and Sound field (3D).

> ℹ️ **Note**: *The following is an initial description of the controls that exist in all modes except Matrix mode. Other parameters that are specific to certain modes are explained in the section* Panorama modes in the Surround Editor *(⤢521).*

The speaker channels of the surround setup are shown as blue circular areas. 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 circle. The sound source to be positioned is displayed as a red circular area, with a corresponding setting at **Pan L/R** also as two circular areas.
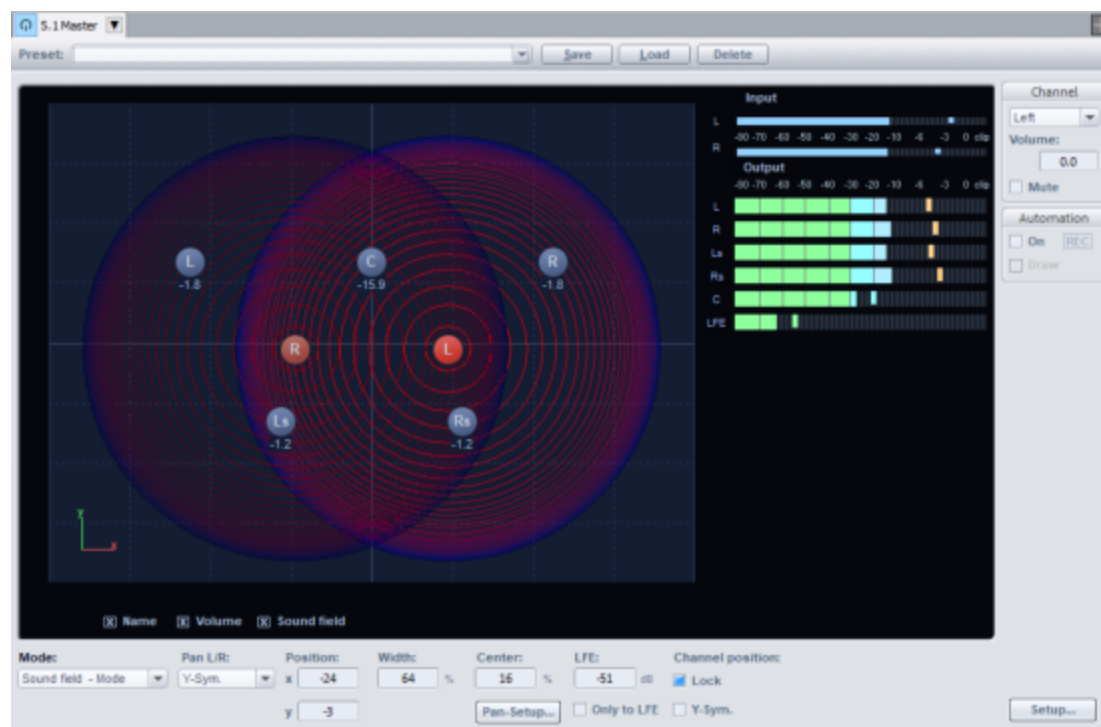
The following additional information is displayed:

Name: The loudspeakers are labeled with the abbreviations specified in the "Surround Setup" window.

Volume: The level readout below the speakers indicates the value of the signal portion that is given from the sound source to the assigned surround channel. If a stereo mode (X-Sym., Y-Sym., XY-Sym., Parallel) is set under "Pan L/R", the sum of the levels of both stereo sources is displayed. Click on one of the two sound sources with the Shift key held down to display only the level value of this source.

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.

ℹ️ *You can hide the display elements **Name**, **Volume** and **Sound field**" by deactivating the corresponding options.*

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 Stereo and mono signal processing in Surround projects (↗526).

Position: Normally you drag the sound source with the mouse to the desired position. For precise position information, enter the position directly using the number fields **x/y**. The values can also be changed by dragging the mouse vertically over the number field, or you can use the mouse wheel to make incremental changes.

When dragging the sound source with the mouse on the graphical interface, you can restrict the movement by pressing the following keys simultaneously:

■ **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 of the sound source to the coordinate origin (position x=0, y=0) remains the same. The result is a circle.
  Result: Circular movement

■ **a** or **r** + **Mouse movement**: The sound source can only be moved diagonally. A straight line starting from the initial position of the sound source and passing through the coordinate origin of the panoramic field defines the course of this movement.
  Result: Diagonal movement with constant angle.

Pan Setup: see Pan Setup (↗520)

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.

Input: Peak meter for the input signal. Mono input signals are displayed in both scales of the input meter.

Output: Level meters of the individual channels of the surround bus/master. Only the level components from the audio currently being edited in the Surround Editor are displayed.

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 circles in the panorama field.

Automation: see Automation in the Surround Editor (↗528)

Presets: Save your own frequently used surround editor settings as a preset. In addition to the position of the sound source, the mode and the settings of Pan L/R are also saved.

> 💡 *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 Setup

In the **Pan Setup** dialog box, you can define even more advanced settings for the Surround Editor. The dialog window can be opened by clicking on **Pan Setup** in the Surround Editor or by right-clicking on the sound source.
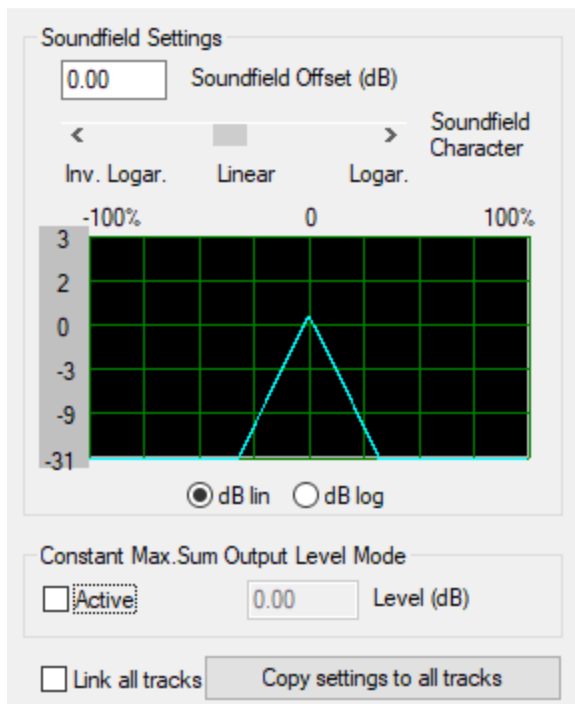
Soundfield offset (dB): If sound sources are positioned directly on a single surround channel, the offset set here is taken into account. Thus, for example, individual compensation can be made if such signals are too prominent.

Soundfield Character (only in Sound field or Angle mode): The level drop characteristic of the sound field can be adjusted continuously. **Inv. 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.

Constant max. sum output level mode: If this option is activated, the maximum level of all channel outputs of the Surround Panorama Module will not exceed the set value. Unintended fluctuations of the overall level during position changes can thus be compensated. The function is especially useful when you want to move specific sources around the room via automation.

Link all tracks: The following settings within the Surround Editor are applied to all tracks of the arranger and changes are always executed in sync on all tracks: Sound Field Offset, Level Drop Characteristic, Sum Output Volume, Center, Pan Law/Width, LFE Level and Settings, Channel Settings like Level Change 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.
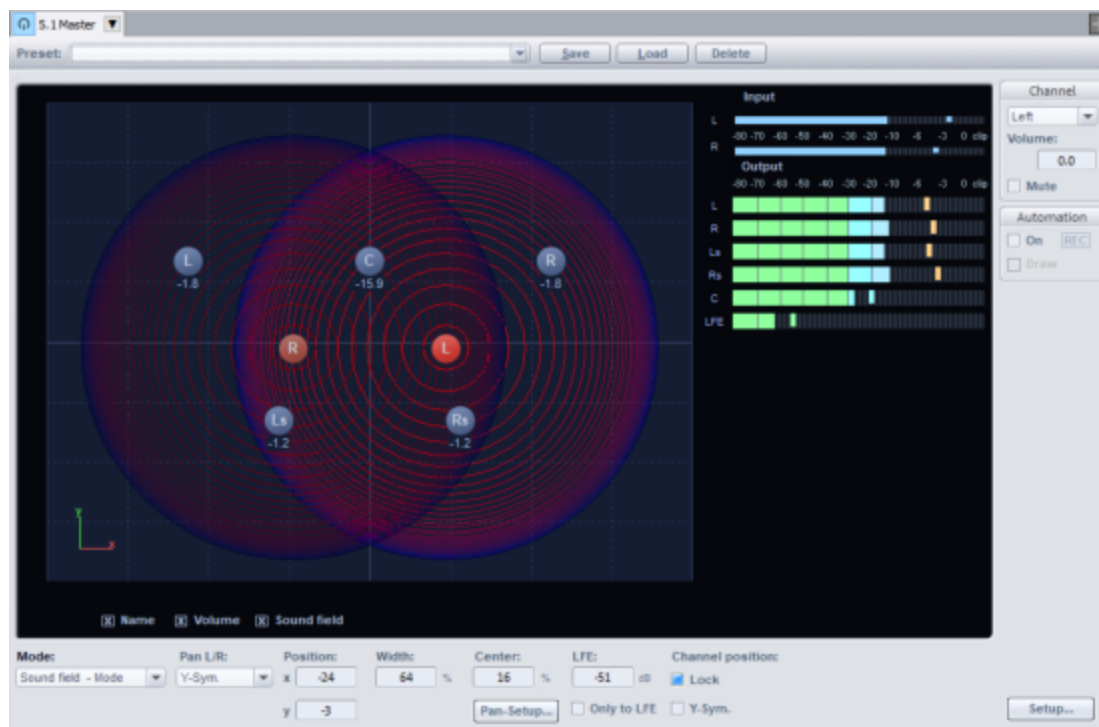
Example of use: Very accurate localization, particularly for movements.

For movements in constant direction (e.g. overflight of a jet) level conflicts may occur - for this the Angle mode is more suitable.

Width: This parameter influences the sound field width of the loudspeakers.

Pan Setup...: 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.

Channel position: By default, the positions of the speaker channels are fixed in the control panel against accidental shifting. If you deactivate the option **Lock**, you can move the speakers freely in the sound field. You can then additionally activate the option **Y-Sym.**, then when moving one channel, the associated other channel will be moved symmetrically to the Y-axis.

Panning-Law Mode

This mode uses a surround panorama display familiar from many digital mixing consoles.
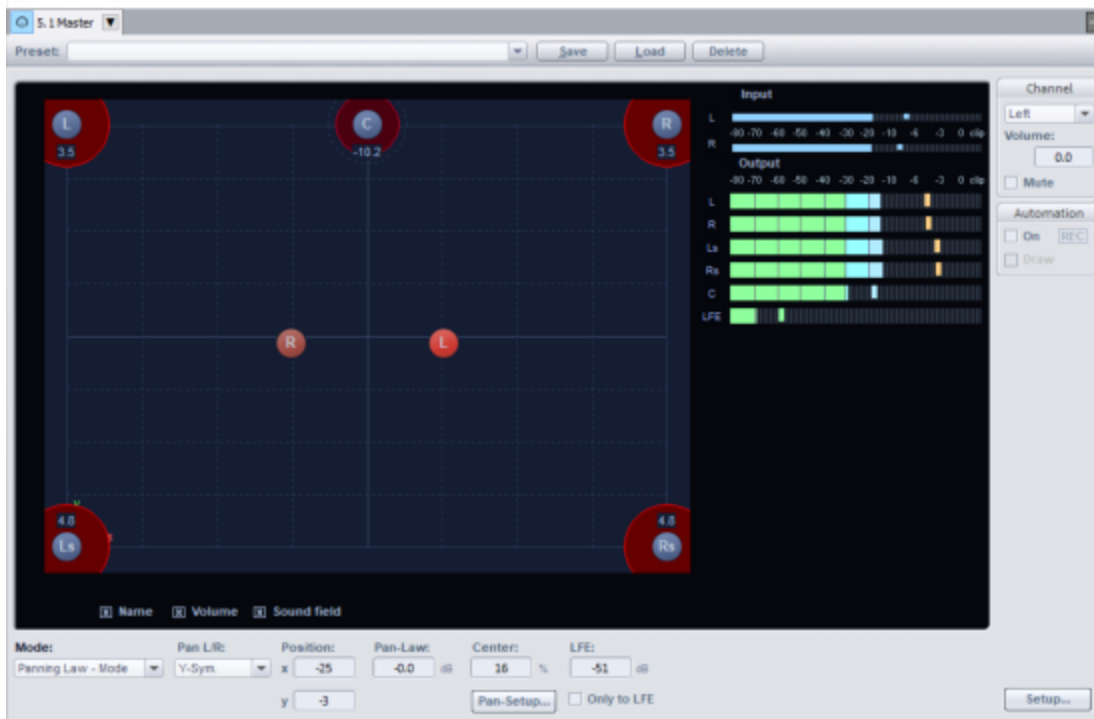
The positions of the speakers are shown at the outer edges of the usable panorama. By clicking the "Sound field" option, you can visually display the graphical levels displayed across the Surround channels. The level distribution between two neighboring loudspeakers follows the -3 dB law, which means 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 when panning, you can additionally lower the volume in the pan center 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 correspond with each other (i.e. the middle axis of the sound field is pointing directly at the loudspeaker: angle difference = 0), the level of this channel is 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 less suitable for distance panning.
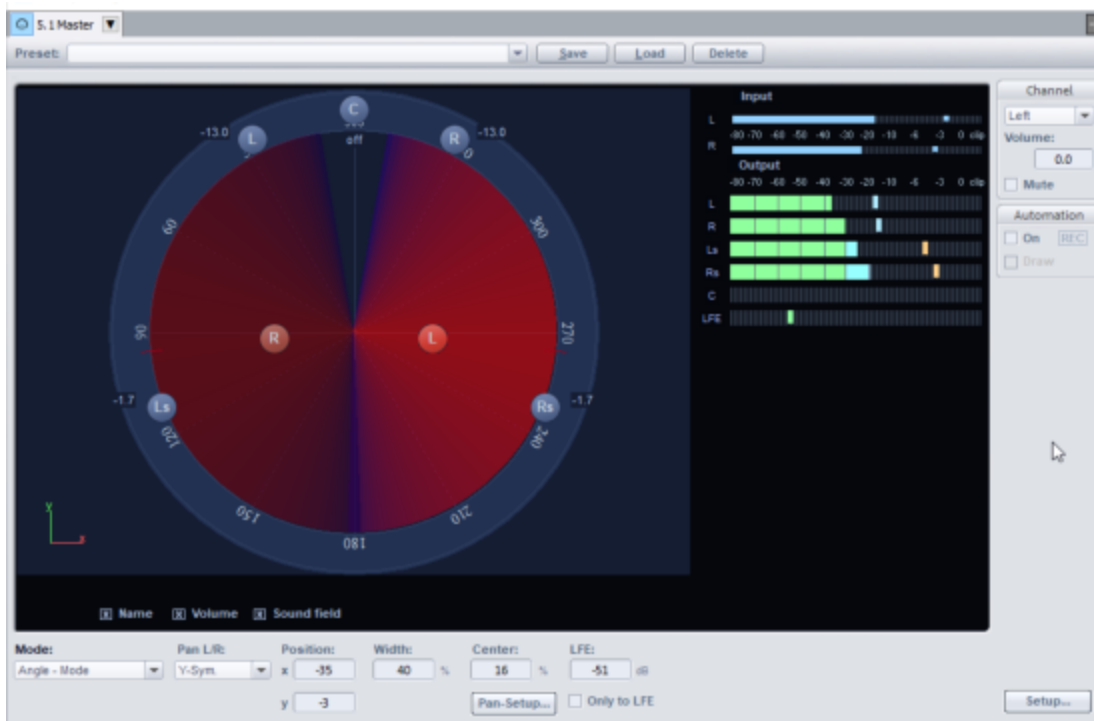
Width: This parameter determines the size of the sound field opening angle.

Pan Setup…: 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 – Maximum sum output level constant: If this function is selected, the maximum level of all channel outputs of the Surround Panorama Module will not exceed the set value. In angle mode, this avoids level drops of the sound field during movements in connection with large opening angles.

Matrix Mode

In Matrix Mode you can directly enter the level that will go from the input signal to the individual Surround channels.

Double-click on the number field to enter values manually. You can also adjust the levels by dragging them with the mouse. 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.
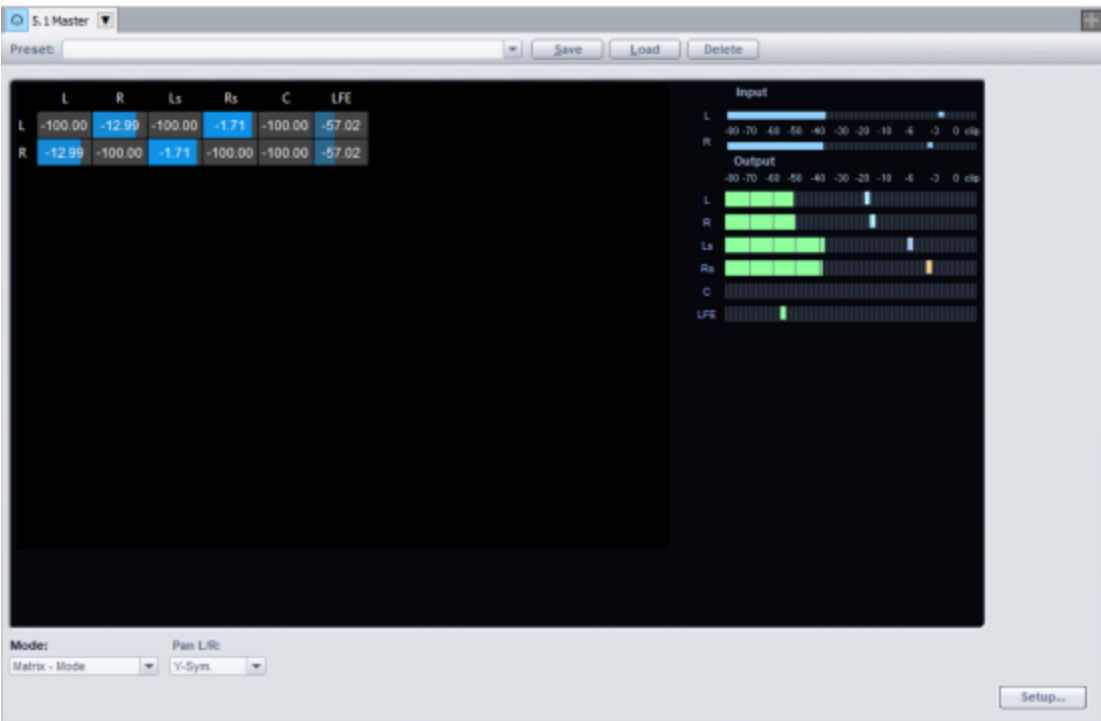
Differing from the other modes, the settings of **Pan L/R** in Matrix mode have the following meaning.

Pan L/R Mono: The levels of the left and right input signals are adjusted together.

Pan L/R Stereo: The levels of the left and right input signals can be adjusted individually.

The remaining "Pan L/R" settings do not apply. No mirror sound sources will be formed.

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.
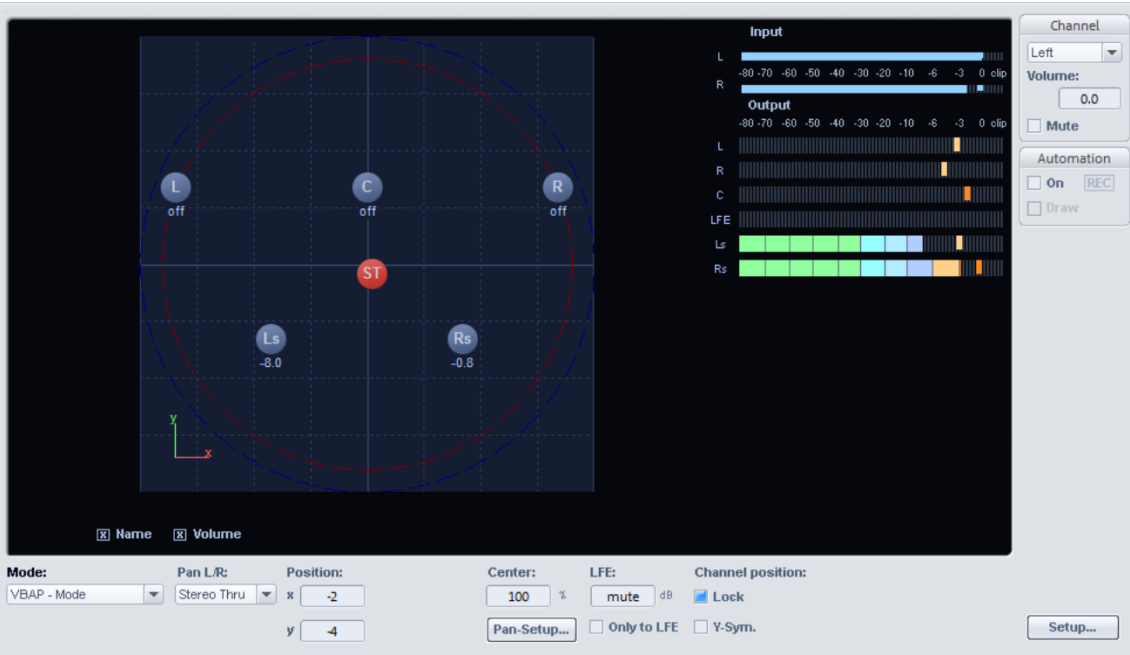
The operation differs from the sound field mode only in the point that no specification of the width of a loudspeaker sound field is required.

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.
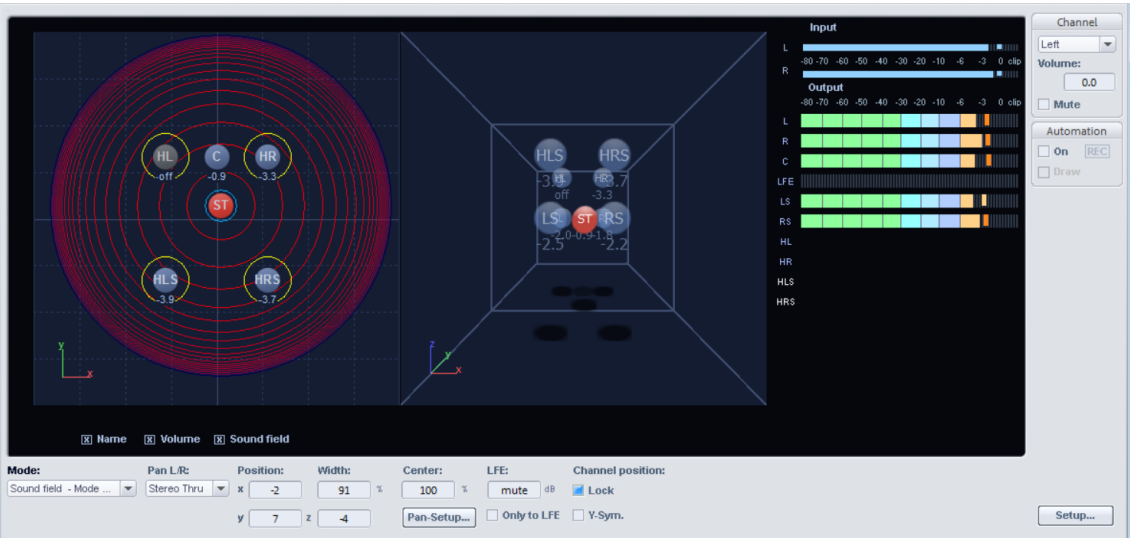
Pan L/R: Stereo and Mono Signal Processing in Surround Projects
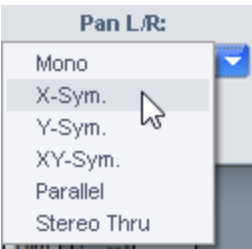
Mode	Stereo signal	Mono signal
Mono	Mono sum is positioned as a single mono sound source in the panorama.	The mono signal is positioned as a single mono sound source in the panorama.
X-Sym:	The left and right channels are symmetrically aligned to the x-axis. For example, this enables you to pan a stereo signal to front L (Left) / LS (Left Surround).	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 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).	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 left and right channels are symmetrically aligned to the x- and y axis. For example, this enables you to pan a stereo signal to front L (Left) / RS (Right Surround).	In addition to the original mono source, a second mirrored sound source is positioned with this signal, mirror-symmetrically with respect to the X and Y axes.
Parallel:	The left and right channels maintain a constant distance from each other when they are moved, they are moved parallel. While dragging a sound source, hold the Ctrl key to change the distance between the sound sources.	The original and mirrored sound sources maintain a constant distance from each other when they are moved, they are moved parallel. While dragging a sound source, hold the Ctrl key to change the distance between the sound sources.
Stereo Thru:	The same level proportions as for a mono source are applied to the different channels, depending on the position of the sound source. However, only the left signal is used for all left channels, and only the right signal for all right channels, and the mono sum for the center and LFE channels.	Identical to the mono mode.

Working with Multiple Surround Masters.

In **Sequoia** 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 (↗487) additionally to the surround master and 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, right-click the pan slider to bring up the Panorama Editor (↗492) of the track.

At the top of the editor you can now switch to the tab with the surround editor for the track.

Surround Upmix or Downmix

A master bus can also be routed to any other surround or stereo master, with the necessary upmix or downmix being performed. For example, you can create a stereo mix on an additional 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.
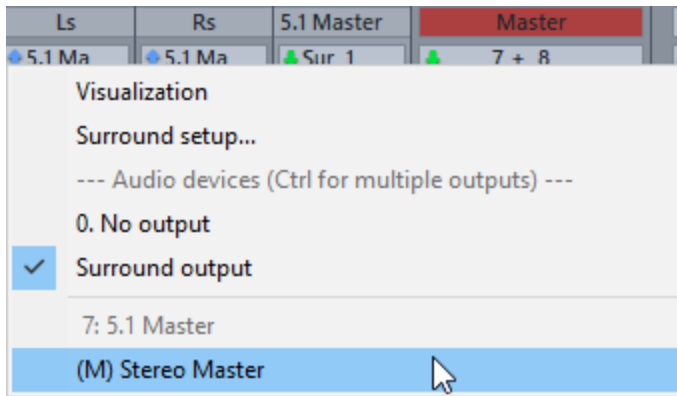
To open the downmix dialog in which you can set the downmix coefficients, select the corresponding entry at the bottom of the output menu.
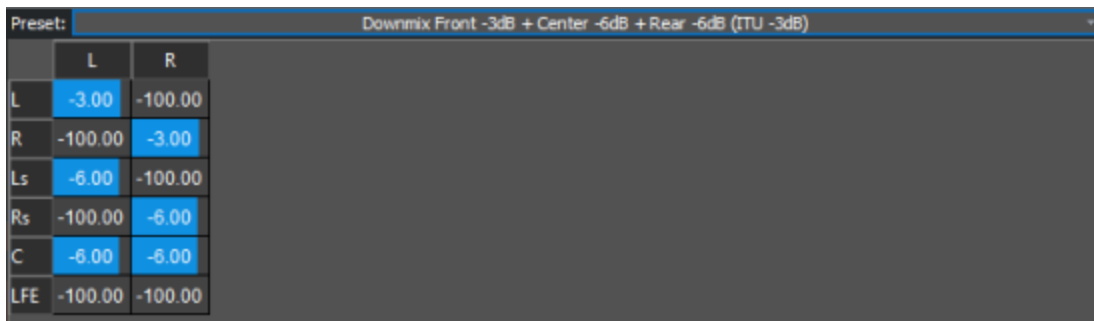
In the preset list above there are common presets for downmixes of 5.1 into stereo. Additionally, you can also set and save your own downmix coefficients. The downmix is taken into account during the stereo master export of a project.

Automation in the Surround Editor

Recording Surround Automation

To automate panning movements, first activate the Automation option in the Surround Editor.
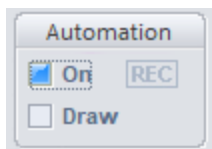
Start playback. If you now move the sound source with the mouse during playback, this movement will be recorded.
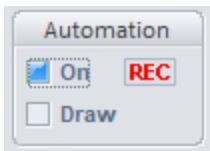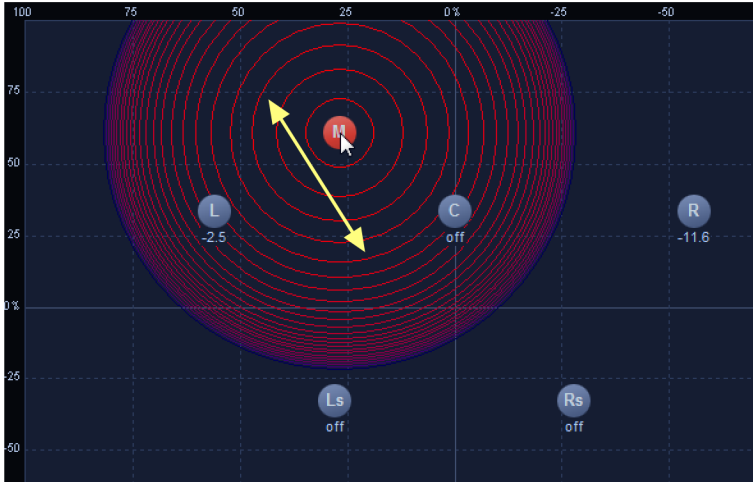
It is displayed in the project window as a curve.
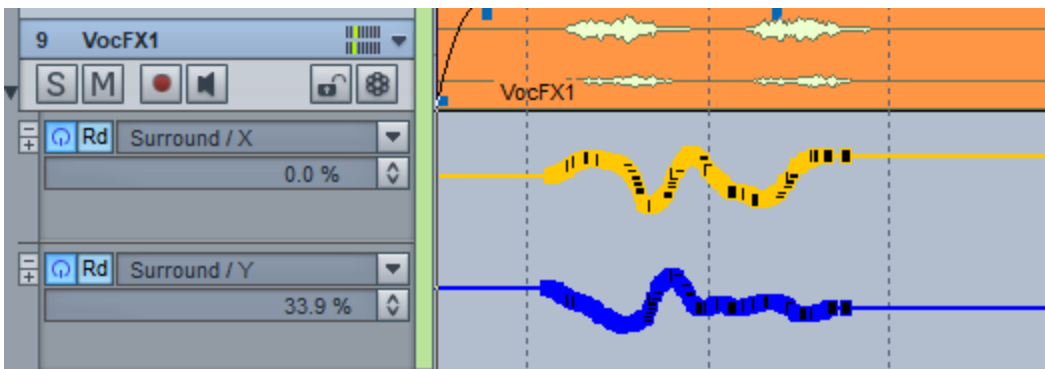
You can edit the automation by using the Automation Draw Mode in the project window.

> ℹ️ *With object-based panning, i.e. when a single object is separately routed to a surround bus and the Surround Editor has been opened from the Object Editor, the automation is recorded and edited accordingly as an object automation curve.*

Drawing Surround Automation Curves

You can also draw in a motion path directly in the Surround Editor. To do this, you must first define the range over whose length the simulated movement extends.

> ℹ️ *With object-based panning, this time range always corresponds to the entire object. If the automation should only take place over a shorter section, you must split the object.*

Now additionally activate the **Draw** option.
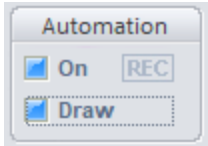
Now draw in the desired motion of the sound source in the Surround Editor.
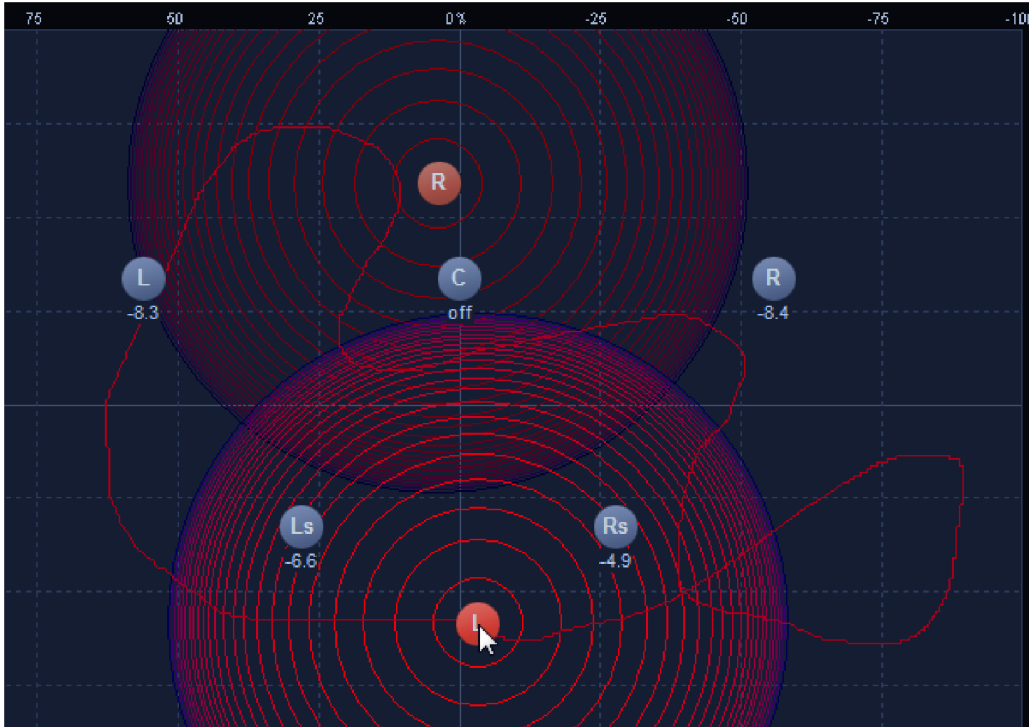
The drawn in motion path is converted into automation curves. The sound source is moved over the selected time period along the complete path at a uniform speed. Here, too, subsequent editing of the curve is possible in the project window.

Surround Effects

All effects can also be used in surround busses and masters. There are some differences here:

- Normal (stereo) plug-in effects are loaded as multiple individual instances into the respective surround channels, and the parameters of the plug-in interface apply equally to all instances.

- Internal effects and surround-capable VST plug-ins such as the 3D Reverb or third-party plug-ins are automatically detected and used as one instance.

- With **Sequoia** 's internal effects, there is also the option to make separate plug-in settings for each surround channel. In addition, the settings for the different channels can be combined into groups (e.g. one EQ setting for R/L and one for Rs/Ls).

Surround Control Groups

When internal effects are loaded as a surround effect, a new section with controls for the surround channels appears at the top of the effects dialog.

The buttons in the top row, named according to the surround channels, allow you to view the effect settings for the respective channel.

Use the **S** buttons to solo the individual surround channels.

The surround channels can be grouped together. This causes:

- All channels of the group are calculated together in one instance of the effect you loaded in the mixer. (But of course still in separate channels).

- The parameters set in the effect dialog affect all channels of the 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 with stereo.

Form or disband a group: To form a group, click the **+ -** button in the Surround Effect dialog. Now click on the buttons of the Surround channels that you want to group or ungroup. The corresponding Surround channels will now be marked with points of the same color.

Click the **+ -** button again to end group editing. You can now view the effect settings of each group by clicking on one of the associated channels.

Surround Export
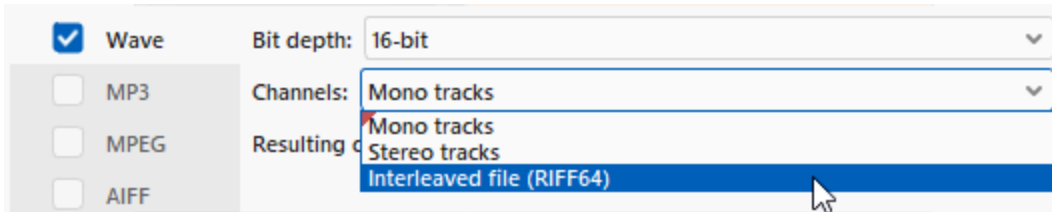
Surround Export allows you to render a surround mix. The outputs of the individual surround channels are used as the output signal.

Open the window **Export…** with menu **File** > **Export…**.

In the **Source** section at **Routing**, select the surround master whose surround mix you want to export. It is also possible to export a stereo downmix at the same time if you select Surround Master (e.g. 5_1 ITU) + Stereo Master in the list box.

Format: With Surround Bouncing you have the choice between the following output formats (Wave or FLAC):

- ■ Mono tracks
- ■ Stereo
- ■ Interleaved (RIFF64)

If mono tracks are selected, one mono file is saved per surround channel, if stereo tracks are selected, 2 surround channels are saved in one stereo file. RIFF64 (also called RF64) is a format that can be used to combine individual channels of a multitrack production (e.g. for 5.1) into one interleaved file. A single file containing all surround channels is created.

The abbreviations of the individual surround buses defined in the surround settings (↗513) are appended to the selected names for the files to be created.

Example: You have chosen the file name "5_1_Surround" when exporting a 5.1 surround mix into 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".

Select the other settings as for the regular export (↗661).

ADM Editor

The ADM editor allows the creation of next generation audio according to the ADM standard (Audio Definition Model). ADM files can be created in accordance with the general specification of the EBU (European Broadcasting Union), as well as those that conform to the requirements of the **Dolby Atmos** distribution format. These are commonly used in smartphones, cinemas, and home entertainment systems.

> ⓘ *A helpful introduction to the concept of ADM can be found at* https://adm.ebu.io/.

Introduction - General Information about ADM

The ADM standard (Audio Definition Model) describes the technical properties of audio using metadata. This provides a high degree of abstraction and makes it possible to depict complex audio data streams. In addition to the actual audio data, additional properties can be encoded within the audio file:

- **3D Audio**: Sound sources that are called *ADM objects* in ADM terminology can be statically placed in a room or moved. Their positions are specified directly and converted by the renderer into volume components for the respective loudspeaker channels during playback (in accordance with the ITU-R BS.2051 standard).

  > ⚠ *ADM objects should not be confused with* **Sequoia** *objects. These are different concepts with different meanings. In this help chapter, the term "objects" always refers to ADM objects.*

- 3D audio as a multi-channel surround mix can be integrated as a direct speaker object (Dolby Atmos: Bed) (background audio). Depending on the speaker configuration used, a downmix or upmix takes place in the target system during playback by the renderer. Other audio types such as ambisonics, matrix or binaural audio (headphone output in which the spatial information is encoded into the stereo signal using filters and phase shifts) are also possible.

- Content (contents or programs) can be provided with a language setting and the finished file can contain several alternative language versions. Objects can also be marked as interactive, allowing the audio data to be moved in 3D space during playback, the volume to be adjusted separately, or even muted completely.

- Data streams can be marked as having different levels of importance so that a decision can be made as to which data streams are omitted when transmitting content with limited capacity.

The metadata is defined independently of the code (codec agnostic). This means that ADM audio data is described using the model. The renderer is free to decide which of these properties are evaluated and mapped and how. Since not all renderers currently support every feature of the ADM specification, a target renderer must be specified for export and preview (monitoring). Currently, **Sequoia** supports the renderers EAR (EBU ADM Renderer), and Dolby Atmos.

The exported ADM file is a broadcast waveBroadcast Wave Manager (↗650) file in which all audio data is stored in an unstructured format. It also contains a section with the ADM metadata in XML format. This metadata defines both the logical structure of the file, such as the grouping and movement of the sound sources, as well as the technical specifications including sample format, number of channels, and the layout of audio streams.

> ℹ *Example Playback of a sports broadcast of an international match:*
>
> - *The background (bed) is the live atmosphere from the stadium, which is available as channel-based 5.1 surround audio. The volume can be controlled separately during playback.*
> - *The commentary by the presenters of the two teams can be selected alternatively and the volume can also be controlled.*
> - *There are also jingles that are moved around the scene as dynamic 3D audio objects.*

Structure of the ADM metadata

The ADM metadata has a hierarchical structure and can contain the following elements and sub-elements:

- **Program**: A complete audio production with all content played during a certain period of time. (The term "programs" comes from the ADM standard and refers to a complete audio unit, not a TV or radio program). An ADM file can contain several programs, for example for different languages. A program contains one or more contents.

    > ℹ️ **Note**: *The **Dolby Atmos** renderer only supports one program.*

- **Content**: Part of a program, e.g. a dialog or the background music. Typical metadata assigned to the content includes the language of the dialog and the types of content (e.g. dialog or music). A content contains one or more objects.

- **Object**: The actual audible components of the content. An object is either

    - An **Object**: Any audio content with a specific position in 3D space. The surround mix according to the perceived position in the room is left to the ADM renderer, which realizes this according to the loudspeaker configuration used.

    - A **Direct Speaker Object**: Channel-based surround audio with a specific speaker configuration, for example the audio output of a surround bus.

    Objects can be marked as interactive. This makes it possible to mark objects as mutually exclusive (complementary objects) - for instance dialogs in different languages - or to specify objects that can be adjusted separately in position and volume during playback by the listener. In addition to the ADM objects, there are other specific object types that are specialized for particular applications.

A further section of the ADM metadata defines how the audio data within the wave file is grouped into the individual objects (tracks, streams, channels, packs). These specifications correspond to the conventional format parameters for audio data, including the number of channels and speaker assignment (for channel-based objects) as well as sample rate, bit depth, and codec. In Sequoia, this information is derived from the format of the source material and the channels of the Sequoia objects, tracks and surround busses used in the project and therefore does not need to be edited directly.

Overview of the ADM workflow

When creating a 3D audio mix with the ADM editor, the basic procedure is as follows:

1. Create a new project with 48 or 96kHz sample rate.

2. Assign the audio to the ADM objects on the tracks in the project. Import all the required audio files onto separate tracks. When importing an audio file that is already available in a surround multi-channel format, e.g. for use as a bed in a Dolby Atmos mix, a corresponding surround submix folder is created.

3. Open the ADM Editor and create programs and content according to the planned structure of the ADM and create the corresponding objects for all audio content: For the surround folders as a direct speaker object, for

the other audio tracks as simple objects.

> 💡 *For a simple standard layout with a single program and one content, as the standard used in the Dolby Atmos format, you can use the **AutoFill** button. An object is automatically assigned to each simple track, while surround submix busses or folders are converted into the corresponding direct speaker objects.*

4. Place the objects in 3D space using the surround panner of the corresponding tracks.
   If necessary, automate the object movements. In all tracks in which objects have been linked, the standard panorama slider is replaced by a special version of the surround editor.
   For acoustic monitoring of the panning: Activate **ADM monitoring**. This creates a surround bus into which the render plug-in for the corresponding output format is inserted. In the Monitoring settings dialog, set the output format that matches your speaker system.

5. Add the required information for interactivity or language variants (e.g. attributes, complementary objects, interactivity) to the objects (only for ADM).

6. Export the ADM or Dolby Atmos audio file. If Dolby Atmos was selected as the target format, the file can subsequently be converted to the desired distribution format using the appropriate encoder.

The ADM editor window

The ADM editor window enables the creation and management of ADM (Audio Definition Model) files for immersive audio formats such as Dolby Atmos. It provides precise control over audio objects, metadata and export options.
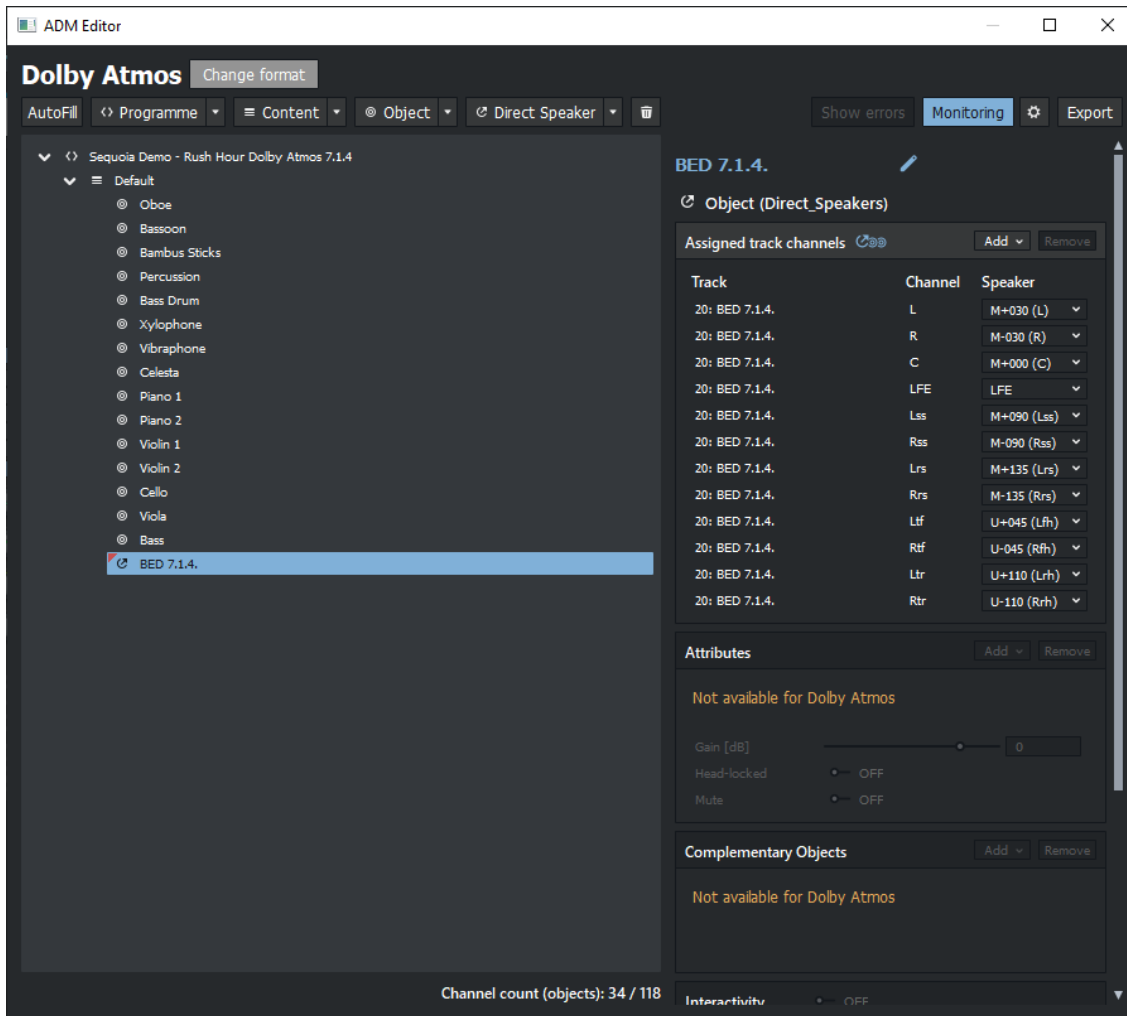
Selection of the target format

The ADM specification is flexible and serves as a basis for future audio applications. Distribution formats such as MPEG-H Audio or Dolby Atmos are based on ADM, but require additional specifications (e.g. speaker arrangements, number of channels, sample rates). To ensure that the ADM file is compatible with these formats, the corresponding render profile must be defined when creating the ADM layout.

Sequoia supports the following render formats:

- ADM (EBU ADM renderer)
- Dolby Atmos

After selecting the target format, the button changes to **Change Format**. You can also use it to change the target format later.

> ⚠️ *Attention: Changing the target format can lead to an **invalid project structure**. If the new target format does not support certain properties, **data may be lost**!*

ADM

This option can be used to generate generic ADM audio. There are no restrictions regarding the number of programs, channels or objects. However, an ADM audio file created in this way may not be compatible with Dolby Atmos . That means this option is primarily suitable for your own customized immersive software solutions, for archiving or for further processing in other software.

Dolby Atmos

Dolby Atmos is a widely used format for film sound and found in home theater systems, professional cinemas, and streaming services. It is supported by a wide range of hardware and software solutions.

The following restrictions exist compared to the general ADM specification:

- Sample rate: The sample rate is limited to 48 kHz or 96 kHz for Dolby Atmos projects only.

- Mono objects: Objects only ever have one audio channel. However, two audio channels of a stereo track can be assigned to two connected objects.

- Maximum number of objects: 128 objects at 48 kHz (64 at 96 kHz). The first 10 objects function as direct speaker objects for a "bed" surround mix in 7.1.2 format.

- Program restriction: Only one program is permitted in the hierarchy Program → Content → Object.

Terminology variants

Although the Dolby Atmos format is based on ADM, it uses its own terms for some of the concepts from the ADM specification in its actual implementation. In the ADM Editor, we use the more general ADM terminology for both format variants. The following table therefore compares the terms of the different terminologies:

ADM	Atmos
Program	N/A
Content	Content
Object	Object

ADM	Atmos
Complementary objects	N/A
Direct speaker objects	Bed

Set up ADM structure

AutoFill

With the **AutoFill** function, you can make all the necessary track-object assignments with a single click. ADM content is created for each folder with tracks. Tracks that are not in a folder are assigned to the "Default" content as ADM objects.

The following operations are carried out with AutoFill:

- A program is created that contains one item of content.

- One object is created for every track.

    > **ℹ** *With Dolby Atmos, in which only mono objects are permitted, two objects are created internally for stereo tracks. The two objects are displayed as one unit and can only be edited together.*

- A direct channel object is created for each surround bus or surround folder.
  The surround setup of the busses or folders is transferred to the channel/speaker assignments.

> **ℹ** *Note: For a Dolby Atmos project where no more complex structures are permitted, you can use AutoFill to create the complete structure by creating tracks for all objects and surround folders for all beds and subsequently performing* **AutoFill***.*

Manual creation of contents, objects and direct speaker objects

For more complex projects with several programs and contents as well as additional metadata for the individual objects, as is possible with the ADM output format, all elements of the ADM hierarchy can also be created manually.

To do this, use the buttons **Program**, **Content**, **Object** and **Direct speaker**. Select an existing element in the tree structure to add the new element below the selected one. Click on one of the buttons to create an empty element. There are further options for creating elements in the menus at the buttons:

- **Programs > From project name**: Creates a new, empty program that is named after the project.

- **Content > Empty**: Creates new, empty content - corresponds to clicking the button directly. This option is also available for **Object** and **Direct Speaker** buttons.

- **Content > Reference existing content**: Existing content can be referenced from the submenu. The referenced content contains all objects of this content, but may also contain other objects or attributes. This function is also available for **Object** and **Direct Speaker** buttons.

- **Object / Direct Speaker > From track selection (single object)**: Creates an ADM object or Direct Speaker object and assigns all track channels to this object.

- **Object/Direct speaker > From track selection (per track)**: An object is created for each selected track, bus or folder.

> **ⓘ** *After creating an empty object, you must manually define the channels of the associated tracks. In the "Assigned track channels" area, click on **Add** and select the track and track channels from the menu.*

Assigned track channels
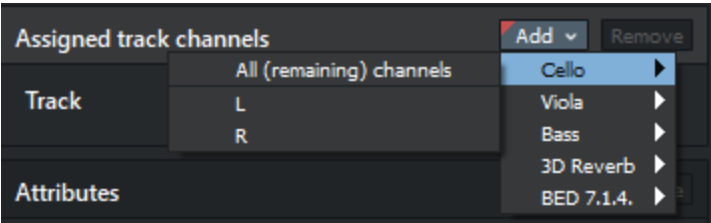
With Direct speaker objects, you must also specify the associated speakersSpeaker configurations (↗549).
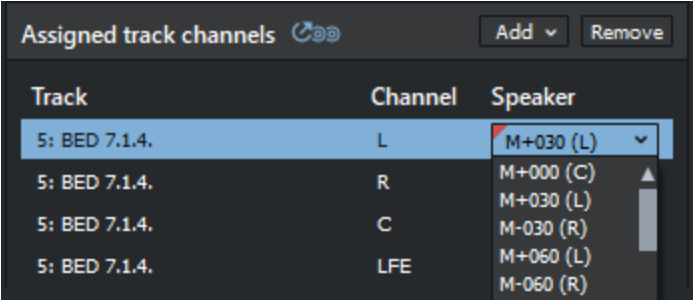
Bed formats that are not natively supported by Dolby Atmos are composed of a standard bed and additional object channels as follows:

Target format	Composite
5.1.2	5.1 bed + 2 objects (2 top speakers)
5.1.4	5.1 bed + 4 objects (4 top speakers)
7.1.4	7.1 bed + 4 objects (4 top speakers)
7.1.6	7.1 bed + 6 objects (6 top speakers)
9.1.4	7.1 Bed + 6 Objects (4 Top Speaker + Lw + Rw)
9.1.6	7.1.2 Bed + 6 Objects (4 Top Speaker + Lw + Rw)

Attributes

In addition to the mandatory track channel assignments and, in the case of direct speaker objects, speaker channel assignment, the elements can also be assigned additional attributes.

Attribute	Item	Target format	Description	Value range
Name	Programs, contents, objects	ADM, Dolby Atmos	Designation of the element	
Language	Programs, contents	ADM	Language of the element.	Selection of a 3 letter code from the menu or any own entry
MaxDuckingDepth	Program	ADM	Maximum strength of automatic ducking for each object within the program	-62... 0 dB
Content kind	Contents	ADM, Dolby Atmos	Type of content	Selection from a range of predefined values in the menu
Gain	Objects	ADM	Additional gain for all audio data belonging to the object	-100...12 dB

Head-locked	Objects	ADM	The perceived location of an audio element is linked to the head position.	on/off
Mute	Objects	ADM	Additional switch to switch an object on or off	on/off
DisableDucking	Objects	ADM	Allow/prohibit ducking for individual objects	on/off
Importance	Objects	ADM	Allows the renderer to discard an object below a certain level of importance	0...10 (0 is least important, 10 is most important)
Position Offset (X,Y,Z)	Objects	ADM	Apply an offset to all positions	No limitation in all directions

Complementary objects

Complementary objects (only available for ADM) are mutually exclusive. They can exist in different contents. Only one object can be active at a time. Select an object within a content, click on **Add** and select another object in the content. You can use the checkbox to decide which ADM object is set as the default ADM object and is therefore played if no selection has been made.

Interactivity

Objects that are marked as interactive can be changed live during playback within certain limits if the target format supports this.

ℹ️ *Note: Interactivity is only available for the ADM target format.*

Name	Description	Parameter
On/off	Switch object on or off	on/off
Gain	Change in the object level	Min/max -100...+12 dB
Position	The position of the object can be changed.	Azimuth: -180°...180° Elevation: -90°...90° Distance: 0...1

Surround Panning of Objects
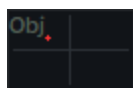
 As soon as a track is assigned to an object, the mixer panorama control replaces the surround control. Right-click on this element (or on the panorama slider in the track header or track editor) to open the surround editor. This differs from the standard Surround editor and is therefore labeled "Obj".
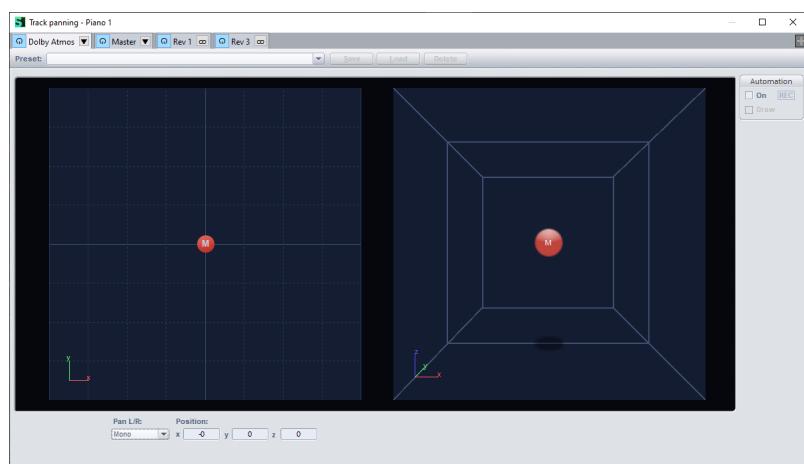
Object surround editor for a mono signal

> **ⓘ** Note: The Object surround panner does not allow the LFE channel to be used. Objects are placed in the room, while LFE signals should ideally not be localized. The LFE channel can only be controlled via a direct speaker object (bed).

Dolby Atmos projects only supports mono signals in an object. When an object is created from a stereo track, the system automatically creates two objects in the background, which are assigned the right and left channels respectively. These objects are moved as a single unit in the Surround editor. Additionally, press the Alt button to move a channel individually. There are additional key combinations to restrict the movement of sound sources. You can find more information on this at Surround editor controlsControls of the Surround Editor (↗518).
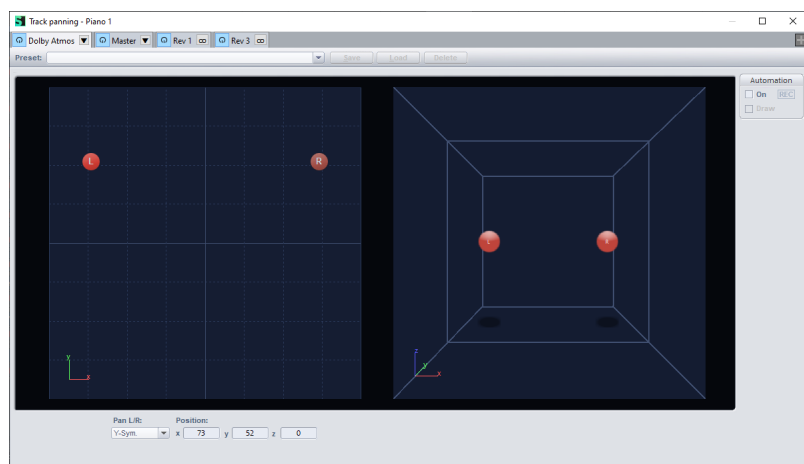
Object surround editor for a stereo signal

Dynamic metadata - Automation

In addition to static attributes, objects can also be positioned in space using dynamic metadata. This metadata changes during project playback and enables the movement of objects. The movement is controlled using Automation curves Automation (↗338). In addition to the position parameters (X, Y, Z), Dolby Atmos has other parameters that can be automated:

Name	Description	Parameter
Size	Controls the size of the object. A value of 0 means that the object is localized exactly at the specified X, Y and Z coordinates as a point-shaped sound source without spatial expansion, which means that the sound is positioned precisely and reproduced directly. A value of 1 means that the object reaches its maximum spatial extent and acoustically covers the entire listening room, whereby it is evenly distributed in the room and reproduced by all relevant loudspeakers so that its exact position is no longer perceptible.	0...1
Speaker zones	This allows the objects to be assigned to specific speaker groups in order to restrict playback to certain levels of the speaker system.	The following zones can be assigned: **0 - all** **1 - no back** **2 - no sides** **3 - center back** **4 - screen only** **5 - surround only**
Object snap	When active, the object position is moved to the nearest speaker and then "snaps" to this speaker.	on/off
Elevation	With Elevation off, the output of the object is limited to the listener level; the top speakers are no longer used.	on/off

Monitoring with the Render plug-in

Monitoring To create a surround mix, the surround output must be rendered for the selected output format must be rendered so that it can be played back via the speakers.

By clicking the **Monitoring** button, a surround bus is created and the renderer for the output format is loaded into the bus.

> ℹ️ *The render plug-in is an internal effect plug-in that is loaded in the first plug-in slot of the bus. However, the bus with the renderer does not behave like a normal bus. The audio signal goes directly from the objects into the render plug-in. From there, it is routed directly to the hardware outputs (speakers). The monitoring section can optionally be used for headphone monitoring. Other bus functions (gain, volume, EQ, other plug-ins...) are not available.*

Click on the cogwheel ⚙ to open the render plug-in settings.

Settings of the render plug-in

The settings dialog of the render plug-in offers various monitoring options:

- Peak meter:
  On the right and left side of the dialog are peak meters for the input signals (track channels, sorted by ADM objects) and for the output signals (loudspeakers).

- Mute & solo mode:
  The "M" and "S" buttons can be used to mute or solo the objects individually.
  In addition, the volume control allows you to adjust the volume of all objects displayed on the left-hand side of the renderer. This allows the volume of the ADM mix to be changed, for example to achieve a certain loudness.
  The volume control works independently of the mixer and only influences the input signals of the renderer. Technically speaking, it offers an offset of -24 dB to +24 dB on the audio signals.
  Comparison: While the mixer faders have a range of -100 dB to +12 dB, the volume control is used exclusively for global adjustment within the renderer.

- Master Trim: Controls the overall volume of the input levels for all signals in the renderer. Enables amplification by up to +24 dB or attenuation by -24 dB.

- Loudness measurement (loudness metering):
  The measuring instruments for loudness measurement are located at the bottom right of the dialog.
  Settings:

  - **On** → Activates the loudness measurement.
  - **Reset** → Resets the loudness measurement.
  - **Reset on Start** → Resets the measurement each time playback is started.
  - **Hold on Stop** → Pauses the measurement when playback is stopped.

ⓘ *You can find more information on loudness measurement in the section* Loudness meter*Loudness meter (↗559).*

Further settings for the renderer can be made in the middle:

- **Active program**: Only one program can be played at a time. This setting has no function for Dolby Atmos, as there is only one program in this case.

- **Target speaker layout**: Select the speaker configuration of your playback system on which the surround mix will be played. Open the surround settings of the render bus via **Surround Setup**. Select a preset there that connects the channel assignments of your audio hardware with the corresponding speakers.

- **Headphone output** (only for Dolby Atmos): In addition to output to loudspeakers, the render output can also

be provided as a stereo or binaural 3D mix for the monitoring section
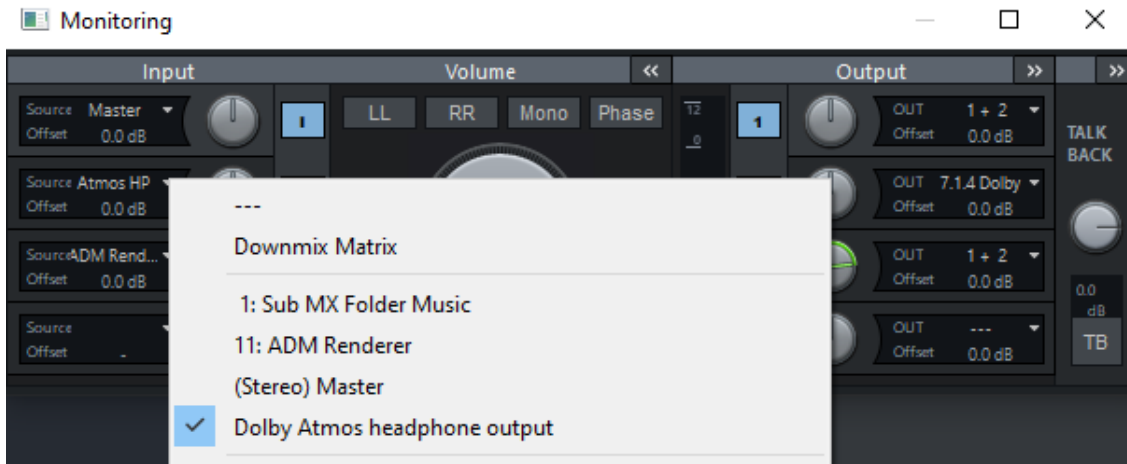
The **Dolby Atmos renderer** offers a number of other settings that not only affect the renderer, but are also transferred to the exported Dolby Atmos BWF wave file:

- **5.1 to 2.0 downmix/5.1 or 5.1.x downmix**: If the target speaker format is set to stereo, a downmix method according to Dolby standards can be selected. It is also possible to downmix more complex channel formats to 5.1 or 5.1.x (with raised speakers). When downmixing complex channel formats to stereo (not with "Stereo direct"), conversion to 5.1 takes place before the final downmix to stereo is performed. More information on the individual settings is available at Dolby Support.

- **Trims**: The Trims function allows you to adjust the volume for the middle and upper levels. The balance between these two levels can also be adjusted. If **automatic** is activated, the system automatically adopts the preset values.

- **Binaural headphone modes**: For binaural playback on stereo headphones, a special render mode can be defined for each object and each bed channel special render mode can be defined for each object and each bed channel. Binaural processing improves spatial sound perception by converting the entire 3D sound information into a stereo signal. The positioning of the objects and bed channels can be influenced using the following modes:

  - **Off** → No binaural processing
  - **Near** → Close to the listener
  - **Mid** → Average distance
  - **Far** → Far away

These settings enable optimized spatial perception, depending on the playback environment Playback environment and the desired sound image

Loudness calculation in the Dolby Atmos renderer

The Dolby Atmos renderer contains an integrated loudness measurement based on the settings on the "Loudness" page.

The calculation takes into account the basic settings and the program loudness and ensures that these parameters are used for the measurement. The exact maximum peak value (true peak) and storage options are not taken into account, as these are not included in the automatic calculation.

Parallel to the render signal, the system internally generates a stereo or 5.1 mix, which is used for the loudness calculation. The "Loudness" switch activates the measurement and display of the rendered audio data.

The measurement is controlled via the following options:

- **Reset** resets the measured values.

- **Reset at start** performs a complete reset of the loudness analysis at the start of playback.

- **Hold at stop** ensures that the measured values are retained after the stop and are not automatically reset.

Export in ADM broadcast wave files

You can use the export function to save the ADM structure of a project as an ADM broadcast wave file. This file can then be converted into the desired distribution format using a suitable encoder. Depending on requirements, complete projects, specific sections or defined areas can be exported.

Item	Description
Time selection	Defines the export area. Selectable options: *Complete project* - Exports the entire project. *From project start to last object* - Exports the project from the first to the last object. *Selected range only* - Exports only a previously defined area. ⓘ *If no area is active, the entire project is exported.*
Length	Displays the length of the selected export range.
Bit depth	The bit depth is permanently set to **24 bit** and cannot be changed.
Sample rate	The sampling rate always corresponds to the sampling rate set for the target format.

Item	Description
Add FFOA	(First Frame of Action, not for ADM target format) Specifies a **time offset** for the export. The time format can be adjusted directly in the input field.
Loudness and level analysis	Activates a **loudness and level analysis** of the exported audio data. The results are saved in a **text file** with the file name postfix "_atmos" next to the exported file. The displayed values help to comply with the loudness requirements for different distribution platforms.

```
Results of loudness calculation (Dolby Atmos)          ✕

Created with: Sequoia 17.3.0.25135 64 Bit
-----------------------------------------------------

Stream: 0
Norm:
Target Level: -18.0 LUFS

Loudness: -1.8 LU (-19.8 LUFS)
Max. Momentary: 7.4 LU (-10.6 LUFS)
Max. Short-Term: 5.7 LU (-12.3 LUFS)
LRA: 17.4 LU

Max. True Peak: 0.3 dBTP

PPMmax: 0.0 dB

File Name:
Channel 0: Le Tigre.wav
-----------------------------------------------------

                                              OK
```
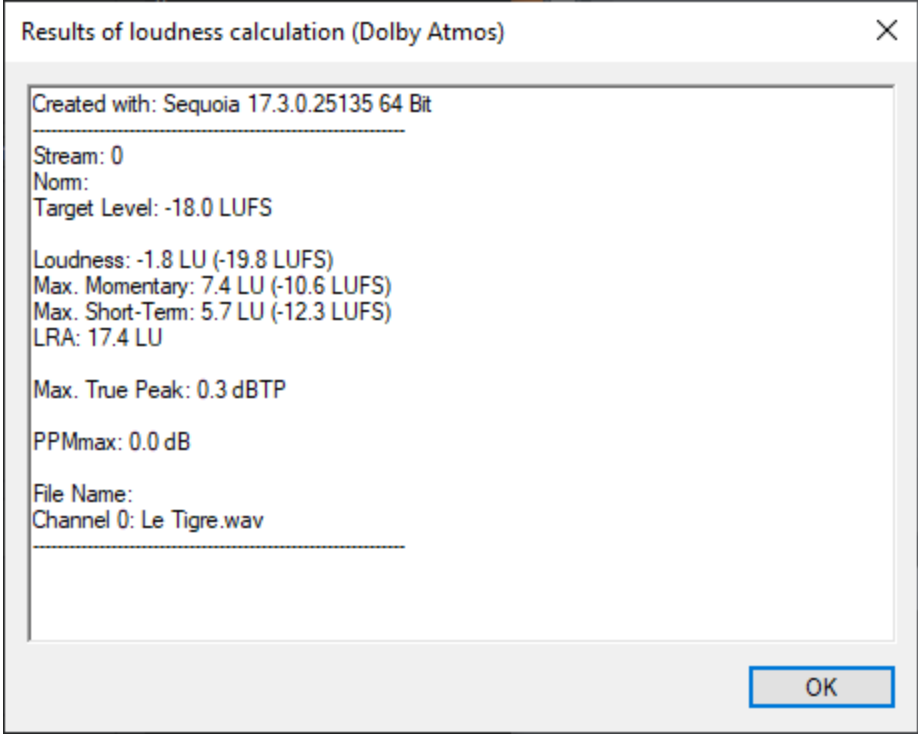
Speaker configurations

The loudspeakers are arranged on three levels.

- **Middle level (M)**: This is where the stereo and surround speakers are located, and where the listener is located.

- **Upper level (U)**: This is where additional ceiling speakers are located.

- **LFE**: Because LFE signals ideally should be perceived evenly throughout the room, a fixed position is not specified.

In the ADM standard, there is also a bottom level (B) (speakers positioned below the listener). Additionally, a second LFE is possible.

Other special speaker positions are:

- "Voice of God" (T+00): Speakers directly above the listener.
- Screen edge speakers (M+SC/M-SC): Speakers at the edges of a screen in front of the listener.

Names of speaker configurations

With Dolby Atmos, the designation is in the form `M.L.U`:

- `M` = Number of speakers in the middle level
- `L` = Number of LFE channels
- `U` = Number of speakers in the upper level

> ℹ️ **Example**: *Dolby Atmos 7.1.4 = 7 speakers in the middle level (M), 1 LFE (L), 4 speakers in the upper level (U).*

With ADM, the naming is in the form `U+M+B`:

- `U` = Number of speakers in the upper level
- `M` = Number of speakers in the middle level
- `B` = Number of speakers in the bottom level

Additional speakers such as LFE or T+00 ("Voice of God") must be mentioned separately.

> ℹ️ *Dolby 5.1 Surround corresponds to this nomenclature: 0+5+0 + LFE.*

With ADM, the speaker positions are specified in angles (polar coordinates), starting from the central position of the listener.

- Elevation: Determines the height of the speaker and is predefined for the three speaker levels: +30 (upper), 0 (middle), -30 (bottom). As a result, only the level designation is used for the elevation.
- Azimuth: Defines the direction within a plane and is appended to the plane name.
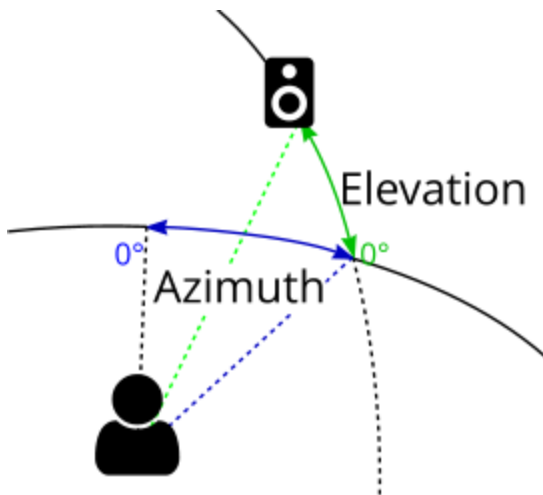
The angle is measured counterclockwise, so the left-hand side has positive azimuth values and the right-hand side has negative azimuth values.

> ℹ️ **Example**: **R** (right front) = middle level, azimuth -30° = **M-30**, L_{TR} (Left top rear surround) = Upper level, azimuth +135° = **U+135**.

Track	Channel	Speaker
2: SurBus 1	L	M+030 (L) ⌄
2: SurBus 1	R	M-030 (R) ⌄
2: SurBus 1	Lrs	M+110 (Ls) ⌄
2: SurBus 1	Rrs	M-110 (Rs) ⌄
2: SurBus 1	C	M+000 (C) ⌄
2: SurBus 1	Ltf	U+045 (Lfh) ⌄
2: SurBus 1	Rtf	U-030 (Rfh) ⌄
2: SurBus 1	Ltr	U+135 (Lrh) ⌄
2: SurBus 1	Rtr	U-110 (Rrh) ⌄
2: SurBus 1	LFE	LFE ⌄

The ADM speaker positions are used for assigning the channels of direct channel objects.In Dolby Atmos mode, the corresponding Dolby speaker names are also displayed in brackets.

In the Surround Setup dialogSurround Settings (↗513) (see "Settings" menu), you will find the corresponding presets for the Dolby Atmos speaker configurations. Detailed information on speaker placement and other configurations can be found on the Dolby website: https://www.dolby.com/about/support/guide/speaker-setup-guides.

Detailed information on other possible loudspeaker arrangements according to the ADM standard can be found in the ITU Recommendation ITU-R BS.2051-3.

Template

To make it easier for you to get started with the ADM 3D surround mix, we provide you with a template for a Dolby Atmos mix. The template `Dolby Atmos 7.1.4` is a template with the minimum features for a Dolby Atmos 3D project. It is ideal for testing and experimenting and can be quickly adapted and expanded to suit your requirements.

The template contains the following components:

- A 7.1.4 surround bus that is linked as a direct speaker object in the ADM editor and serves as a bed. The 7.1.4 bed serves as a central mix area for channel-based content. This can be used to create a conventional surround mix in which the audio signal is distributed to the fixed bed channels using the surround panners.

Track 1 (mono) and track 2 (stereo) are assigned to the 7.1.4 bed. You can quickly add all the tracks you need for your project by selecting the relevant track and selecting **Track menu** > **Duplicate tracks**.

■ Dolby Atmos objects
Track 3 (mono) and track 4 (stereo) are intended for object-based content and are routed directly to the surround master. These tracks are assigned in the ADM Editor and are processed as "Objects" in the Dolby Atmos renderer. Objects are not distributed to fixed channels in the surround mix, but the position in the room is determined. The Dolby Atmos renderer takes over the mix according to the selected speaker configuration. Here too, tracks for additional objects can be added quickly by duplicating the corresponding tracks.

⚠ *Please note: You must then link these objects in the ADM Editor by selecting the new tracks and selecting from the menu of the button* **Object** > **From track selection (per track)**.

■ LFE sends
Since Dolby Atmos objects may not contain LFE content and the LFE channel can only be controlled using the 7.1.4 bed, there is an additional aux send. This allows audio from object tracks to be sent to the LFE channel of the 7.1.4 bed, if necessary.

■ Dolby Atmos renderer and monitoring
The Dolby Atmos renderer is inserted on the surround master. This makes it possible to monitor the Dolby Atmos mix via a 7.1.4 loudspeaker system. The preconfigured headphone output also allows for binaural monitoring of the mix via the monitoring section.

TOOLS AND ASSISTANTS

This chapter describes other various assistants and tools that support you when working with Sequoia.

Undo and Redo

In Sequoia, when working on virtual projects and on wave projects, you can undo the last work steps by selecting Menu **Edit** > **Undo**, using the keyboard shortcut **Ctrl + Z**, or the button on the upper toolbar .

You can restore undone operations by choosing Menu **Edit** > **Redo**, keyboard shortcut **Ctrl + Y** or button on the top toolbar .

The preset maximum number of recorded work steps that can be undone is 99 for virtual projects and 10 for wave projects. The number can be changed in the Undo settings (↗708).

> ℹ️ *Notes*:
>
> - *When editing audio material with offline effects (↗227), the "Undo" function can only be executed if the option* **Create copy** *is active in the respective effect dialog. The option is active by default. If you are sure that you do not need an "Undo", you can disable this option to save disk space and speed up the application of effects.*
>
> - *You can undo work steps even after saving a project. However, if you close the project and reopen it later, you will no longer be able to return to a state before saving.*

Undo History

Via menu **Edit** > **Undo history...** a list with short descriptions of all work steps that have been recorded for the undo function can be displayed, together with the time at which they were executed. When you click on a work state, the project is set to the state it was in after this work step was performed. This allows you to quickly return to an older state without having to undo all the steps one by one. You can also use the **arrow up/arrow down** keys to move through the individual undo steps.
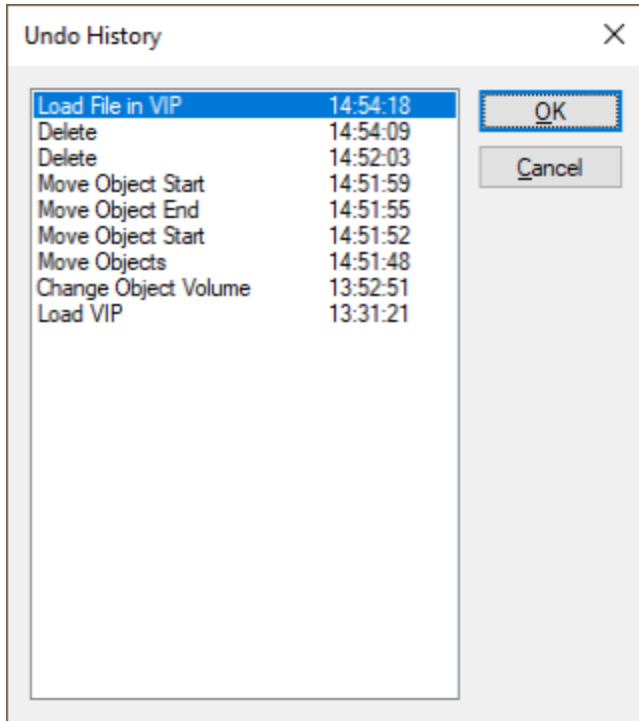
The history can be cleared with menu **Edit** > **Delete undo history**, after that undo is no longer possible.

Using the small arrows on the Undo/Redo buttons of the toolbar, you can reach menus through which you can also access the work states from the undo history.
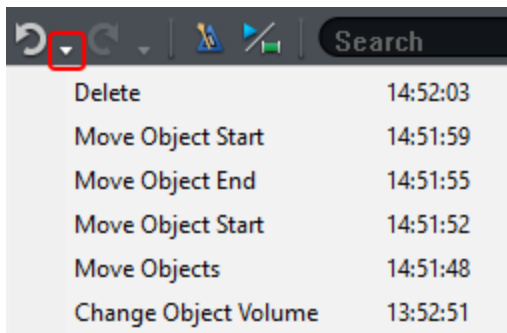
Visualization

The visualization window provides a set of measurement tools for the graphical representation of played or recorded audio. The visualization window can display several of these measuring instruments simultaneously.

To display the visualization window, select Menu **View** > **Visualization** or the keyboard shortcut **Ctrl + Alt + Shift + V**. By default, the visualization opens in the Docker (↗56) but can then be docked or floated anywhere.

You can open as many visualization windows as you want. Different inputs and outputs (maximum 8 tracks, buses or masters) can be selected for the individual visualization windows.
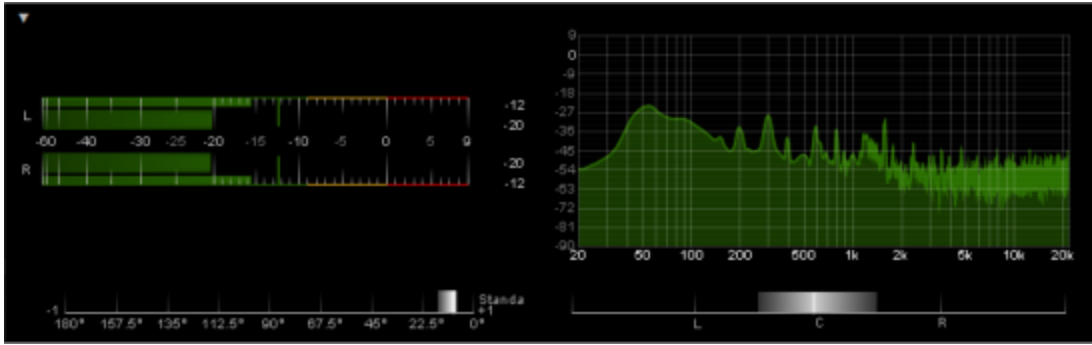
Customizing Visualizations

The arrangement and selection of measuring instruments is specified in various **visualization layouts**. To select a layout, right-click in the visualization window and select it in the submenu **Visualization layout**.

First load a layout in the desired format (1x1..., 2x2...) and then right-click on the individual sections to replace individual visualizations with others. You can resize the displays by clicking and dragging the dividers.

The following display instruments are available:

- **Peak meter** (level meter),
- **Loudness meter**
- **Surround Meter**
- **Phase oscilloscope (vectorscope)**
- **Correlation meter**
- **Direction meter**
- **Spectroscope**
- **Spectrogram**
- **Bit meter**
- **Oscilloscope**
- **Tuner**.

At **Presets** you will find presets for the selected display instrument.

Via the menu item **Settings...** you can open the dialog **Visualization settings** to completely adapt a measuring instrument to your requirements. In the first tab you can also select the type of the measuring instrument. In the following tabs there are further settings, depending on the display instrument set. You will find notes on this in the following sections. Custom presets that you save in this dialog are also listed in the **Preset** menu of the context menu.

When you have set all your displays, you can save the whole customized layout under a new name with **Visualization Layout** > **Save Layout...** to load it again later with **Load Layout...**. Custom layouts that are saved in the default layouts folder are also listed in the **visualization layout** menu.

Select output or input
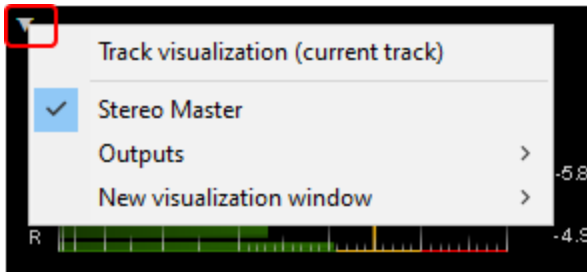
To select the source for visualization, click on this small triangle in the visualization window or open the submenu **Output** in the context menu. Preset is the **Stereo Master**. At **Outputs** you can choose from the available hardware outputs. An input signal for a track or bus can be selected at **Track input**. **Track visualization (current track)** selects the signal of the currently selected track.

> ℹ️ *The selection **Track visualization** can also be found in the menu **Track** > **More...***

If **Monitoring** is active for a track, a track visualization shows the input signal of the track.

In the submenu **New visualization window** you will find again the same menu for output selection, a new visualization window for the selected output will be opened. Up to 8 different sources can be displayed in visualization windows.

Reseting Visualization

Double-click on a measuring instrument to reset it or select **Reset this visualization (double-click)** in the context menu. With **Reset all visualizations** all displays in the visualization window are reset.

Peakmeter
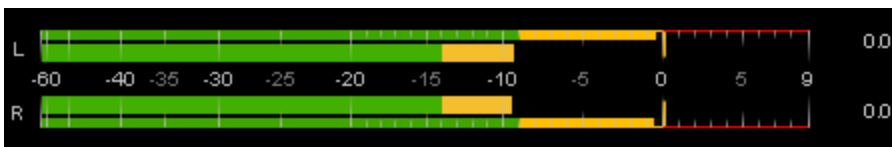
The peak meter displays the level in dB. By default, the peak meter is a combined instrument (**Multimeter**) and displays with the outer, thinner bars a **Peak Program Meter (PPM-Meter)**, and with the inner, thicker bars a **VU Meter**.

If you want to see only the PPM display or only the VU display, select the corresponding display instrument PPM meter or VU meter at **Instrument selection**. When selecting **Multimeter**, you set in the tab **Multimeter** under **Display configuration**, which of the two displays should be shown outside and which inside.

Both meter displays are based on standardized level meters with precisely defined display characteristics. While the PPM meter is a measuring device for peak levels, the VU meter averages the level values over a certain measuring period.

PPM-Meter

The PPM meter displays the instantaneous level of the signal. The level peaks are displayed as a vertical line at the top of the level meter, which moves with a delay for better readability. To the far right of the bars, the peak value is displayed numerically.

Headroom: This allows you to set a level offset that is added to the measurement. This is due to the fact that different systems are calibrated differently. For example, a value of 9.0 (IRT) causes the level to be displayed with + 9 dB more.

Integration time (Q-PPM): The response time of the peak meter is delayed by the set value, so that the display does not react quite so quickly at individual level peaks. The inertia generated is modeled on the behavior of analog peak meter displays.

Return time: This determines the speed at which the level peaks run back after a maximum.

Peak hold (ms): With this value you specify how long the level peaks should remain at a maximum. With **Manual** the level peaks remain permanently and are only reset when you reset the visualization (double-click).

DC filter cut-off frequency: With this high-pass filter you can filter out a DC voltage component so that it is not included in the peak meter measurement.

Clipping threshold: This value specifies the dB value above which a level is displayed as clipping. The range of the scale above this value will be marked in red and also the peak level indicator and the numerical value if you exceed this value.

Minimum number of clipped samples: This value specifies how many consecutive samples may be above the clipping threshold before the signal is considered clipped.

Activate True Peak measurement: In True Peak measurement, the measurement is performed with fourfold oversampling.

PPM-Meter Scale

In this tab you can set different display options for the scale of the peak meter.

Scale: Choose here from different presets for scale display with different scale divisions and display characteristics as used in different European countries.

Reference level (scale offset): Set the reference level as an additional scale offset here. The scale offset is added to the value on the scale.

Minimum/Maximum: Here you set the minimum and maximum value of the scale.

Color settings: Enter the colors and thresholds for overmodulation(clipping), critical, optimal and undermodulated ranges.

> ℹ **Note**: *The color defined for the undermodulated range is only used if you define a threshold value for the optimal range that is above the level minimum.*

VU Meter

For the VU meter there are settings in this tab analog to the PPM for **Headroom**, **integration time** and the **cutoff frequency** for an optional **DC filter**.

The integration time of a VU meter is typically much longer than that of a PPM meter, since an averaged level is to be displayed.

The checkbox **+3dB IEC** ensures that the VU level headroom is raised by an additional 3dB in accordance with the standard (DIN IEC 60268) relative to the headroom specified at **PPM meter**.

> ℹ *If the peak meter is operated as a pure VU meter, you can also display the current peaks with **Display current peak hold**.*

Furthermore, the display of the **RMS** value can be additionally activated. The RMS value is displayed numerically to the right of the VU scale.

The settings for the **VU meter scale** are similar to those for the PPM meter scale.

Presets for the K-Metering System

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.

"K-System" refers to a metering system developed by Bob Katz that has become a standard in mastering audio monitoring. 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. With K-System Metering it's not so much the loudness, but rather the musical dynamics that take center stage. Thus, a reference volume is set for the 0 dB level, which no longer corresponds to the maximum level, as was usually the case in the past.

According to the program and audio material, three different meter scales can be used:

- **K-20**: 0dB reference at -20dBFS
  K-20 is recommended for audio with large dynamics such as classical or film sound.
- **K-14** : 0dB reference at -14dBFS

K-14 is recommended for rock and pop productions or for surround sound.

■ **K-12**: 0dB reference at -12dBFS

K-12 is recommended for radio and television stations.

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.

The corresponding presets for this can be found as VU-Meter Presets **K-12 Broadcast**, **K-14 Music** and **K-20 Cinema** in the Visualization Settings dialog and in the context menu of the peak meter at **Presets**.

Loudness meter

Loudness is the subjective perception of volume when listening to audio material. This sense varies among listeners, depending on several listening conditions such as level, music style, age and even psychological state.

The **EBU Recommendation R128** is a standard for the standardized measurement of loudness with the aim of harmonizing audio productions and level measurements. R128 is based on the ITU-R BS.1770 standard which was established by the International Telecommunications Union.

The core of the measurement of the average loudness impression is the integrated loudness, averaged over the entire duration of the playback. For standardized loudness, the audio material is normalized to a target value of -23 LUFS according to EBU R128.

> ℹ️ *If compression is applied in the master, files may be created whose true peak level is slightly or even significantly below the maximum level. In this case, to take full advantage of the dynamic range of the files, the compression must be adjusted to reach the target loudness value. For this you can use the* **loudness adjustment** *(↗279) in the menu* **Effects** > **Amplitude***.*
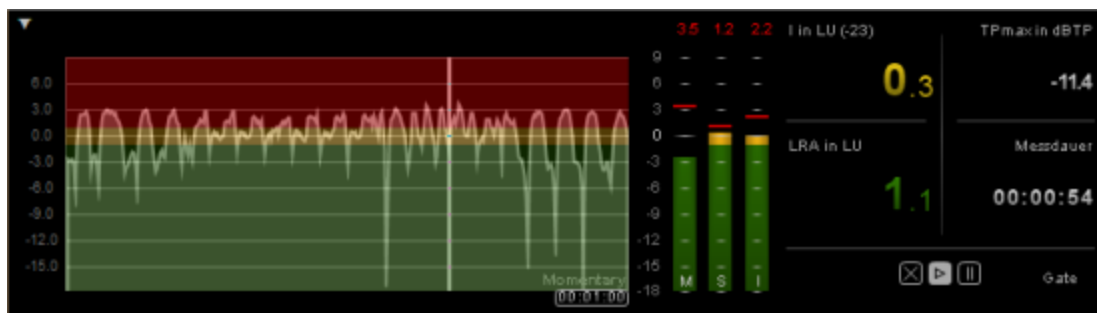
By switching to leveling and normalizing by loudness, a consistent loudness level can be achieved that can be maintained throughout the production chain. The end result is an even loudness impression for the listener and audio material that contains fewer artifacts caused by excessive dynamic compression.

EBU Recommendation R128 pays special attention to three parameters that are essential for the characteristics of an audio signal:

- **Program Loudness - Program Loudness**: Long-term integrated loudness over the duration of the audio material - measured in LUFS (Loudness Units, referenced to Full Scale) or in LU (referenced to a target value, e.g. -23 LUFS according to EBU R128). This parameter indicates the average program loudness. It calculates the mean value over the length of the piece.

  The LU indicator is the level difference to the target value. For example, a measured value of -20 LUFS corresponds to 3LU difference to a target value of -23 LUFS. The graphic display for the relative program loudness in LU would show +3 in this case.

- **Maximum True Peak**: The maximum value of the audio signal when measured continuously on the time axis in dBTP. 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). The R 128 standard permits a maximum True Peak Value of -1dBTP.

- **Loudness Range (LRA)**: Value changes in the loudness measurements of the audio material - measured in LU (Loudness Units). This value shows the difference between loudest and quietest measured value (silence excluded).

The combination of these parameters allow more dynamic mixes without affecting the loudness.

Momentary and **Short Term** are other measures of loudness with shorter integration times (3s for Momentary, 400ms for Short Term) that can be used to assess the loudness of individual sound events.

> 💡 *Further information about EBU Recommendation R128 can be found* in the Wikipedia*!*

Loudness meter - Parameters

I in LUFS/LU: Program loudness value for the "Integrated Loudness" in measured loudness units (LU).

LRA in LU: Loudness Range, measured in LU as a relative value.
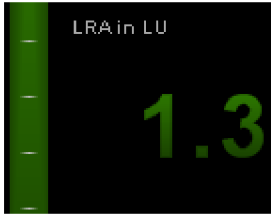
True Peak TPmax in dBTP: Peak level measured with oversampling.

Using Oversampling-True Peak Metering according to ITU-R BS.1770 true peaks can be ascertained (dBTP - referenced on Digital Full Scale). The measurement accuracy 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 Peak Value of -1dBTP.

Measurement Duration: Time indication for the duration of the entire loudness measurement so far - i.e. of program loudness and loudness range.

With the three buttons below you can interrupt the measurement (Pause button), continue (Play button) or reset (Reset button **X**). Pressing the reset button resets all display values.

If the **Gate** indicator in the lower right corner is lit, the level is too low for a certain time. The measuring algorithm then excludes these values from the calculation of the integrated value assuming these parts are silent and therefore can be omitted.

Loudness Meter - Settings

Basic settings

With **Loudness standard** you set the standard the loudness meter should follow. If you select **User** as the loudness standard, you can freely adjust all parameters. If, on the other hand, you select **EBU R128** or **ITU-R BS. 1771**, the parameters are preset according to the selected standard. Parameters prescribed by the standard are grayed out and cannot be changed.
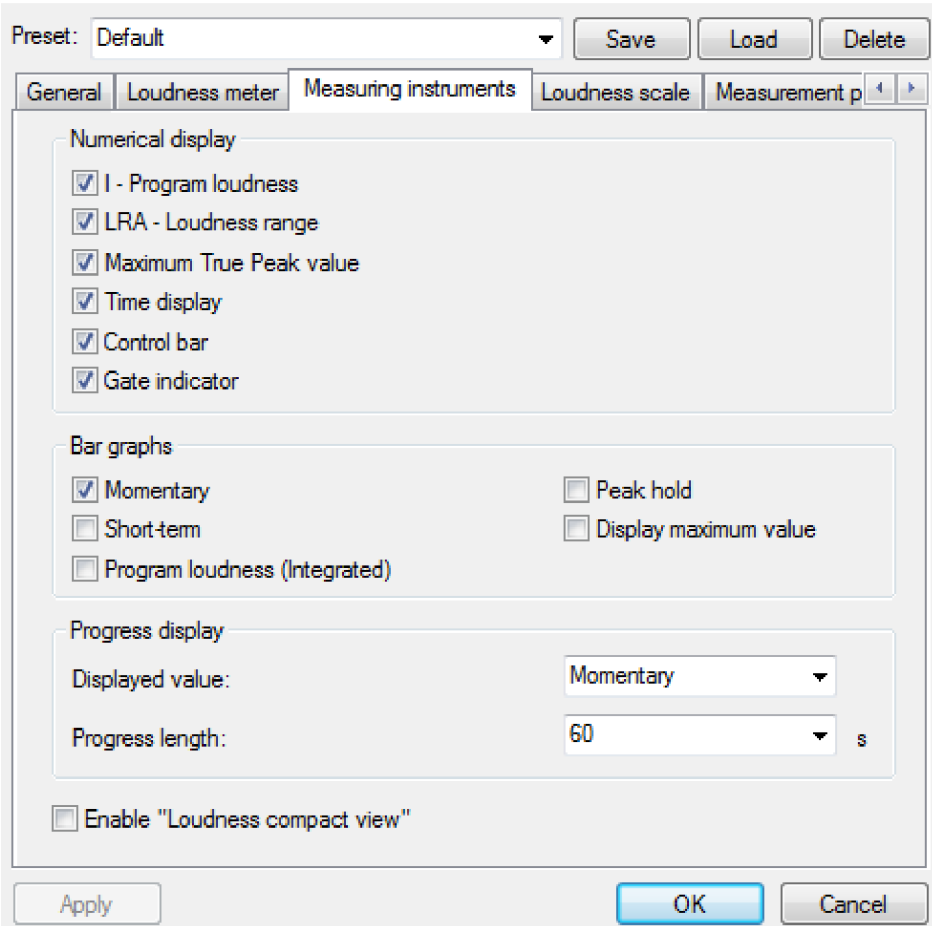
Specified for EBU R128 is the **target loudness** -23 LUFS. This level represents the default value for content production in broadcasting.

Clipping Threshold: Set the clipping level, i.e. from what point clipping should be indicated. This value is prescribed to -1dB for EBU R128. This leads to the fact that, for example, a measured True Peak peak level of -0.8 dBTP is already displayed as clipped.

Automatic reset after play start: With this option you reset the measurement every time you start playback. This option corresponds to the "Reset" button ☒ in the display.

Measuring instruments

In the **Measuring Instruments** tab you can configure which numeric displays, instruments and other controls should be displayed in the Loudness Meter.

Numerical display: Select the different measurement values for the numerical display, the display of the control bar for starting and pausing the measurement and the gate indicator, which are all displayed on the right of the dialog.

Bar graphs: Select which of the three measured variables Momentary , Short and Integrated Loudness should be displayed as bar graphs in the center and whether the maximum values should be displayed numerically or as a peak hold indicator.

Progress display: For the Progress graph, select which measurement - **Momentary**, **Short Term**, **Integrated**, **LRA** (Loudness Range) or **M + S** (Momentary + Short) - should be displayed. **Off** hides the Progress display altogether. With the **Progress length** you define which time duration the Progress display represents.

Enable "Loudness compact view" : This switches the loudness meter to a simplified view without progress and bar graphs.
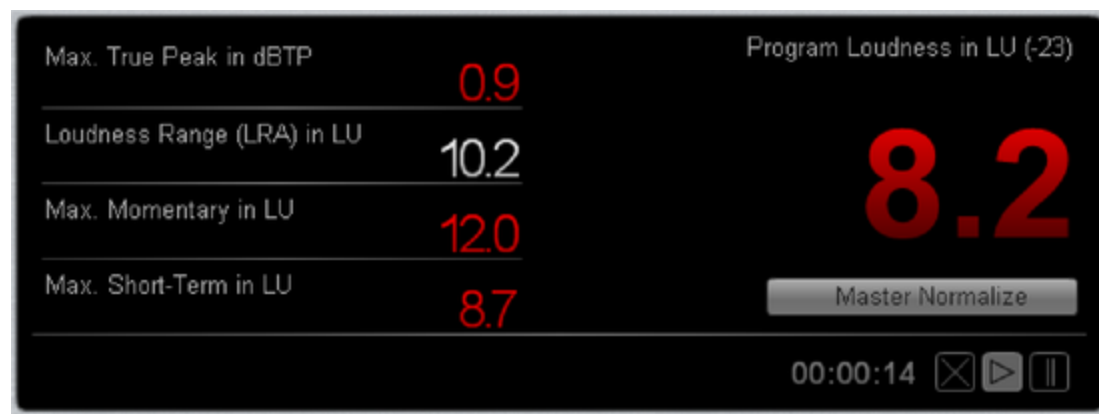
Loudness Scale

In the tab **Loudness scale** you can select different standardized scales. This also defines the value range of the display. For example, the entry **EBU R128 +9** means that the scaling display extends to +9 LU. At **Scale range** you can set the limits of the LU scales individually.

At **Color Settings** you can define color codes as well as threshold values for the overdriven range, the tolerance range around the target value, the normal range and the undermodulated range.

Measurement Parameters

In the **Measurement Parameters tab** you can set the time parameters for the **Momentary** measurement and **Short** measurement, the gate thresholds for the **Integrated** measurement, and the lower and upper limits of the **Loudness Range (LRA)** measurement. The specified default values for EBU R128 are especially marked.

> ℹ️ *These standardized values are initially locked to prevent them from being changed accidentally. They can only be changed if you first select **default** at **Preset**.*

In case of a future change of the standards and additional requirements of the broadcasters, you can adjust the affected values here 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.

Surround meter

The surround meter displays the levels based on the surround presets or angles set in the surround settings (↗513). It shows each speaker's level in a polygon on a sound representative area.
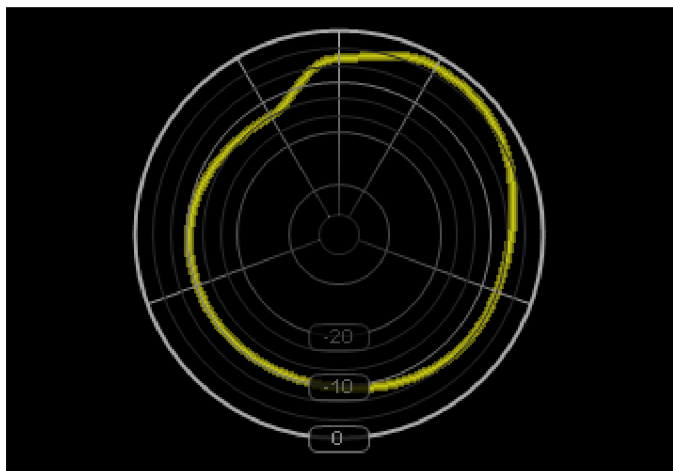
Phase Oscilloscope (Vectorscope)
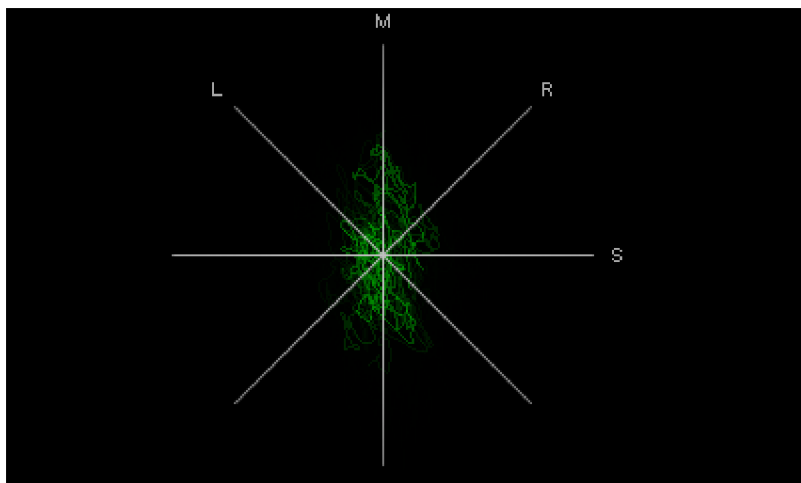
The Phase Oscilloscope gives you information about the distribution of the stereo image in your recording. A mono recording would be displayed as a vertical bar in this view. A stereo signal is by contrast shown as "diffuse ball," because multiple instruments were assigned to the mix in different panoramic positions.

The wider the display, the wider the stereo field of the recording. Note that this means that components of the signal are in phase opposition. The signal is less mono-compatible, as these signal components cancel each other out during mono playback.

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

Settings dialog

You can select different **grid lines**: **LR** (left-right), **MS** (mid-side), **LRMS** (LR+MS) and **Calibrate** to calibrate signals.

Due to the **Pre-amplification mode** the image is always automatically enlarged in such a way that the readability of the distribution of the signal in the stereo image remains guaranteed even at different levels. You can also switch off this mode and set a constant amplification factor instead.

The curve is drawn on top of each other, only after a certain time older traces fade away, resulting in a graph that "averages" over a certain time. **Speed** determines how fast the curve fades. The slower, the denser and more diffuse the representation becomes. Finally, you can also set the **Color** of the curve.

Correlation Meter
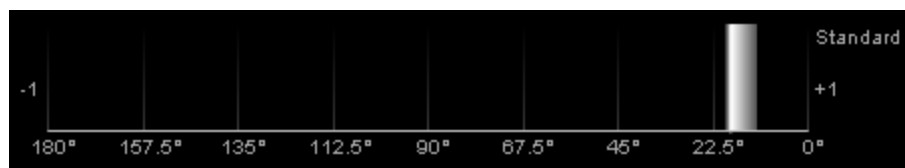
With the correlation meter you can read phase shifts between the two stereo channels and thus the degree of mutual cancellation of signal components with 180° phase difference. If the signal indicator is in the left, red area between 90° and 180°, the signal is no longer reproduced properly in mono.

In the settings dialog, you can specify the colors for the mono range, stereo range and mono-incompatible range in the **color settings** and also the threshold values from when a signal is considered stereo and from when it is no longer mono-compatible.

Direction meter
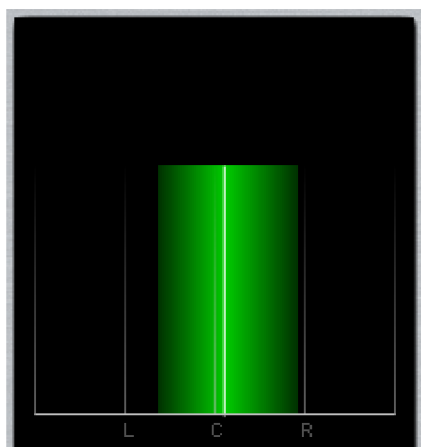
The Direction meter displays the signal's detected direction. The width of the plot corresponds to the degree of correlation.

In the settings dialog, you can specify the colors for the mono range, stereo range and mono-incompatible range in the **color settings** and also the threshold values from when a signal is considered stereo and from when it is no longer mono-compatible.

Spectroscope
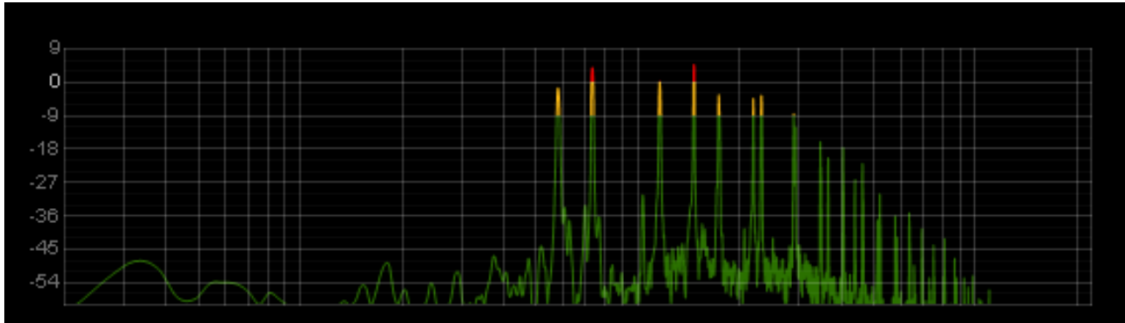
In the spectroscope, the signal is split into individual frequency ranges (frequency bands). The height of each frequency band displays the volume of the relevant frequency range. This way you can tell if certain frequency bands are over-represented.

Spectroscope Settings

Line/bar representation: Select the desired display type from the list box. Default is **Filled (smoothed)**. If you choose **Line (smoothed)** the area under the line is not filled, if you choose **Line** the points are connected by straight lines instead of interpolated curves. If you select one of the numbers, the corresponding number of **bars** will be displayed instead of a line.

Ballistics (only for bars): There are three different modes of bar display.

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 the right signal. The side signal can also be represented by selecting Left-Right.

FFT parameters: Here you can set the parameters for determining the spectral values by changing **FFT size,overlap** and **window function**.

Time constants: The time constants correspond to those of the peak meter (PPM).

> ℹ️ *The settings of the **Spectroscope Scales** correspond to those of the* Peak meter *(↗556).*

Spectrogram
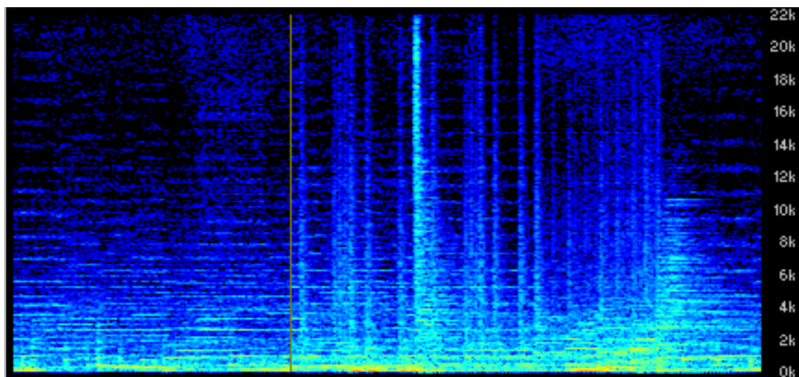
In the spectrogram, the signal is represented as frequency components over time. The volume of frequencies is visualized by its brightness. The spectrogram is suitable for identifying noise in recordings that cannot be easily identified acoustically. They are highlighted with colors in the spectrogram.

In the settings you can define the **FFT parameters** and the analyzed **Stereo channels** as in the Spectroscope. For mapping the levels of the individual bands of the spectrum to the brightness values, various **Color schemes** can be selected in the view options. The option **Logarithmic** selects a different mapping formula and produces a slightly brighter graph.

Bit Meter
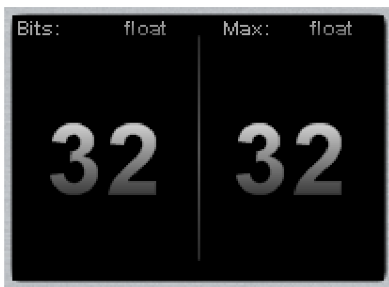
The bit meter shows you with which bitrate the played signal is calculated and which maximum processing rate is possible.

Oscilloscope
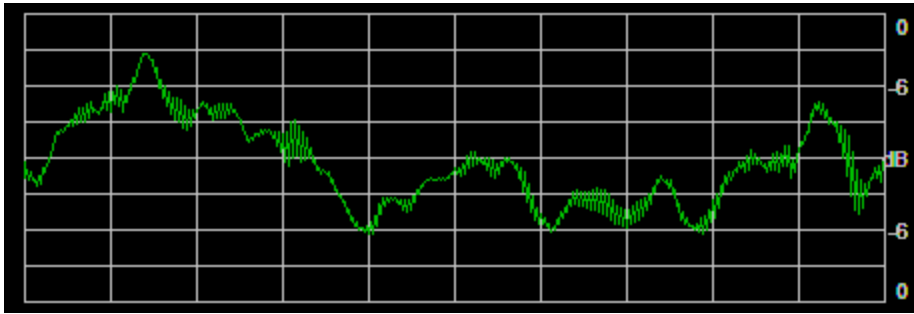
The oscilloscope displays the signal amplitude over time, i.e. the waveform. This has no practical use, but looks quite pretty :)

Tuner

The tuner displays the pitch of the analyzed signal. Use the visualization to tune your guitar or other instrument.

The pitch of each string is shown large in the center of the display. The calibration marks show the deviation of the played to the exact pitch. Red triangles on the right and left show whether you need to tune the string higher or lower. With exact tuning the triangles turn green.

In the preset operating mode **Chromatic** the tuner displays the pitch of the notes played in semitone steps. From the list box at the top you can select common tunings for guitar, bass and ukulele. Then only the relevant pitches of the strings for the respective instrument in the corresponding tuning are displayed.

This is how you tune your guitar:

1. Select the track to whose input the guitar is connected.
2. Open the visualization window (menu **View** or keyboard shortcut **Ctrl + Alt + Shift + V**) and select the visualization **Tuner** in the context menu (right click).
3. In the visualization window, click the arrow in the upper left corner and select **Track visualization (current track)**.
4. Make sure that monitoring (⬈692) is active for the track.
5. The pitch of each string played is shown large in the center of the display. The calibration marks show the deviation of the played to the exact pitch. Red triangles to the right and left show the direction the strings should be tuned. If the tuning is exact, the triangles turn green.

Wave Editing

With *Wave editing*, you no longer edit the objects in the virtual project, but the audio files themselves stored on your hard disk. This type of audio editing is also called "destructive" or "offline" editing.

Audio objects refer to audio files that are open in the background. They refer to the audio data and contain working instructions on how to process this data, i.e. which audio data from the file to use and how to process it with effects.

In Sequoia, therefore, all simple editing of individual, finished audio files, such as shortening, cutting, mastering, in other words: the classic applications of a wave editor, can generally also be done in a virtual project. Load your audio file into a project, make the necessary edits to the object, and export the project back to an audio file.

For certain applications, however, it can still be faster and less complicated to edit the audio material directly: For example, if ready-mixed audio files are only to be shortened or recorded raw material is to be cut in advance, i.e. the audio material is to be deliberately changed on the hard disk in order to gain storage space and work more performantly with smaller files. The additional work steps: Creating a project, importing the file and final export are not required.

Also, for audio files used in a virtual project, it can sometimes be helpful to access the audio data directly. An example: you have made a recording, edited different takes together, and in the result you have a large number of objects accessing the same audio file. Now you discover a noise that affects the entire recording. You would now have to make edits to each object separately to eliminate the noise, if you were working purely virtual. At this point it is easier to directly process the audio file, e.g. with the FFT filter. After that all objects use the corrected material.

Loading Audio Files for Audio Editing

To edit audio files directly in Sequoia, open them as a wave project:

- Select Menu **File** > **Import** > **Load audio file...** (Keyboard shortcut: **W**) If a virtual project is already selected, you must activate the option **Load file as wave project** in the file selection dialog, otherwise the file will be imported as an object into this project instead.

- Drag audio files from the Sequoia file manager or Windows Explorer to the title bar of the program window or project docker.

To open the **audio file of an object in a virtual project**, double-click it while holding the **Shift** key. Alternatively, you can also select the option **Edit audio data** > **Wave Editing...** in the **Object** menu.
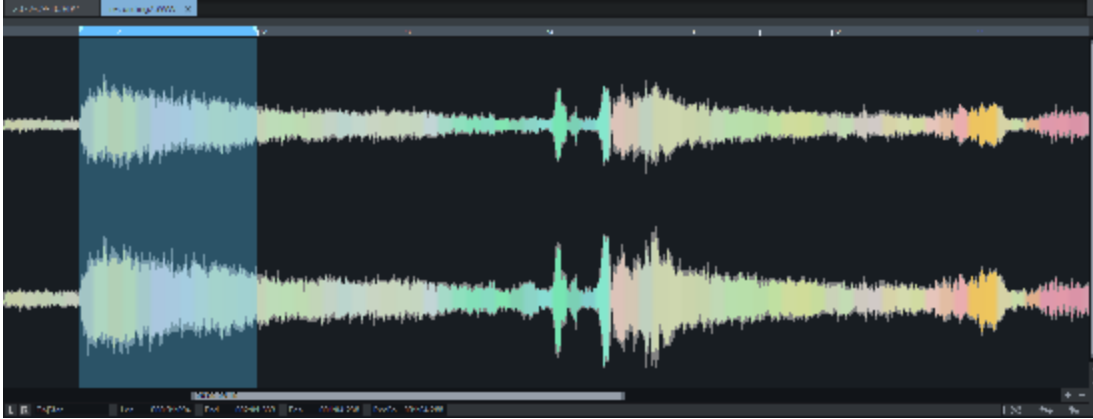
The range in the audio file used by the object is selected as range in the wave project.

If you edit an audio file that is used in several objects with **Wave editing...**, all objects in the VIP will then access the edited audio file and the edits will affect each of these objects.

If you want to edit the audio file only for a specific object, select **Object** > **Edit audio data** > **Edit copy of wave project...**. 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. After editing, the edited copy and the original file are available as Take (↗328) of the object.

Destructive and Real-time Wave Editing

Audio editing can be done in two different modes: Destructive wave editing and Real-time wave editing.

■ The **destructive wave editing** (also called offline editing) is the default mode when using the menu command **Object** > **Edit audio data** > **Wave editing...** or **Shift + double-click** to open the audio file associated with an object.

> ⓘ *You can change this default behavior by activating the option **Open audio files in destructive editing mode** in the program settings at **Program** > **General**.*

■ Audio files (e.g. wave files) that are opened directly via the file menu or the File manager in a separate project window (menu **File** > **Import** > **Load audio file...**) open by default in **Real-time wave editing mode**.

Regardless of how the audio file was loaded, you can switch between the two modes using Menu **File** > **Project Properties** > **Destructive Wave Editing Mode**. In destructive mode the wave project is marked in the title bar with the addition **-destructive**.

You can also recognize the mode by the changed background color. In the picture you can see the same wave project in destructive mode on the left, and in real-time mode on the right.

Destructive Editing

In destructive mode audio files are edited directly on the hard drive. Changes in the project are immediately written to the audio file. To undo edits in this mode with **Ctrl + Z**, the function **Undo enabled** must be on for audio files in the **Program preferences** under **Program** > **Undo**.

⚠ *It is not possible to delete audio material from the audio file that is currently used by audio objects in the project.*

Real-time Wave Editing

Real-time wave editing is a compromise between working in the virtual project and destructive editing, combining the advantages of editing directly on the audio material with the advantages of non-destructive virtual working you are used to with objects in virtual projects.

The audio file is opened in its own wave project window, just as in destructive mode, and can be edited there in exactly the same way. In contrast to destructive editing you will notice the following differences:

- All **editing functions** (cut, copy, delete and paste) are non-destructive. Operations are performed much faster because no data is written to disk and no data needs to be copied for the undo function. All changes are made only when the audio file is saved and then included in the audio file as a whole.

  ℹ *The positions where these operations are performed are marked with dotted lines.*

  ⌕ *The auto-crossfade option (↗169) can be enabled for any cut or paste operation during real-time audio editing. The crossfades are marked by lines in the waveform. However, the crossfades cannot be edited.*

- **Real-time effects**: For the wave project there is a Mixer window reduced to the master channel (keyboard shortcut **M**). The effects of the master channel (master plug-ins, EQ, stereo enhancer) can be used in real time as in virtual projects, can be combined and adjusted during playback. Applying effects to the audio file also happens only when saving the wave project.

> **ⓘ** *Note: The application of the effects of the menu **Effects** happens immediately and destructively also in the real-time editing mode. Read the section* Destructive effect calculation *(↗227) in the Effects chapter.*

- **Volume curve**: If you select the mouse mode Draw volume 🖌 or Draw automation 🖌 , you can draw a volume automation curve in the project window.

Working in Wave Projects

The audio file will be opened in a separate window (wave project). In the title bar of the wave project window, you will see the name of the file, the bit resolution, the length of the sample and the resulting required memory. Navigation and playback controls with zoom, scroll, range and marker functions are the same as for virtual projects.

Wave projects are played back using the standard audio output device. You can set it at **Program Preferences** > **System Options** > **Playback**. (Keyboard shortcut: P)

Mouse modes: All mouse modes are available that are useful in the context of editing wave projects: The range mode 🔳 replaces the preset universal mode and is used for range selection. The Draw volume and the Draw automation modes 🖌 🖌 can be used to draw a volume curve in the real-time wave editing mode (↗571). In Destructive wave editing mode, the volume drawing mode lets you edit the volume directly. Scrubbing mode 🔊 , Zoom mode 🔍 and Spectral mode 🌈 work as in virtual projects. The Draw wave mode 🖌 (see below) is only available in wave projects.

Editing: The same range-related editing functions (↗123) as in the VIP can be executed via the **Edit** menu and the corresponding keyboard shortcuts.

Effects: The effects of the Effects menu are applied "offline". Please refer to the notes at Destructive effect calculation (↗229).

Creating a new wave project by drag & drop: To create a new wave project as a copy of a part or the whole audio file, select a part or the whole audio data (keyboard shortcut: **Ctrl + A**) and drag it from the project window to a free area of the program background.

Import an open wave project into a virtual project: A wave project or a range from a wave project can be imported into a simultaneously opened virtual project. Activate the desired wave project window, select a range in it if necessary and activate the VIP immediately afterwards. Menu **Object** > **More** > **New Object** inserts the wave project or the range selected in it as a new object in the VIP in the selected track at the play cursor position.

Waveform Drawing

In wave projects, you can use the mouse mode **Draw wave** 🖌 to draw in the waveform of a file. The waveform display shows a suitable zoom level, and the mouse pointer becomes a pencil. Changing the waveform in the wave project window is suitable for manually correcting individual very short glitches or drop-outs.

Converting Mono/Stereo

Under Menu **File** > **Advanced export** > **Convert Mono/Stereo** there are two commands for wave projects only, with which you can split stereo wave projects into two mono files and vice versa.

- **Stereo Wave > 2 Mono**: This saves the left and right channel of a stereo wave project (File.wav) each separately in a mono file (File_Left.wav, File_Right.wav).

  The function can also be applied to objects in the project. In this case, the objects with stereo files are replaced by two superimposed objects with mono files each, whose panorama positions are set to left and right.

- **2 Mono > Stereo Wave**: This allows two mono audio files to be combined into one stereo audio file. The command opens a dialog that displays all open mono files. With **Load file** you can load additional files. Select the right and left files from the opened audio files using the ˆ buttons. With the button **<->** you can swap the channels. Click **Merge** and specify the name of the new stereo file that will be created in the project folder.

  > ℹ️ *Note that only files with the same bit depth and sample rate can be connected. The files can be of different length, the resulting length is equal to the longer of the two files.*

Build Loop (offline)

This function is only available for wave projects and helps to create loops that are as seamless as possible. To do this, select an area that should correspond approximately to the finished loop. To find suitable loop areas, start playback in Loop mode and move the area limits to optimize loop playback.

Now select the menu **Effects** > **Sample manipulation** > **Build physical loop (offline)**. The function sets the limits of the range to the zero crossings (↗121) in order to avoid value jumps at the range limits, which would otherwise be audible as clicks. In addition, the audio material at the end of the loop is cross-faded with audio material from before the start of the loop to ensure a smooth transition from the end of the loop to the start of the loop.

To define the length of the crossfade, set a marker in front of the selected range; the audio material between the marker and the start of the range is then used for the crossfade. Please note that the distance from the marker to the start of the range must not be greater than the range itself so that a crossfade is possible.

CDs and DVDs

You can import the content of an audio CD or audio DVD to Sequoia. For audio CDs the metadata of the tracks is taken over or can be determined via the gnudb online CD database.

You can burn any virtual or wave project directly from Sequoia onto audio CDs or DVDs.

In addition to labeling the individual CD tracks when burning a project, the CD track indices are also used to add metadata for export into MP3 or other compressed formats.

To burn the project in other software, projects can be exported as a wave file with an additional .cue or TOC file containing the position and metadata of the CD tracks.

For digital transfer to a pressing plant in the form of DDP files, you can start the supplied external software **DDP Export** from Sequoia.

Importing Audio CDs and DVDs

Importing audio data from an audio CD or DVD is done completely in the digital domain without any sound loss. The titles are saved as wave files during import and inserted into the project as objects.

Importing Audio CDs

To import audio CD tracks into a virtual project, choose Menu **File** > **Import** > **Import Audio CD Track(s)…**. The CD Import dialog opens:
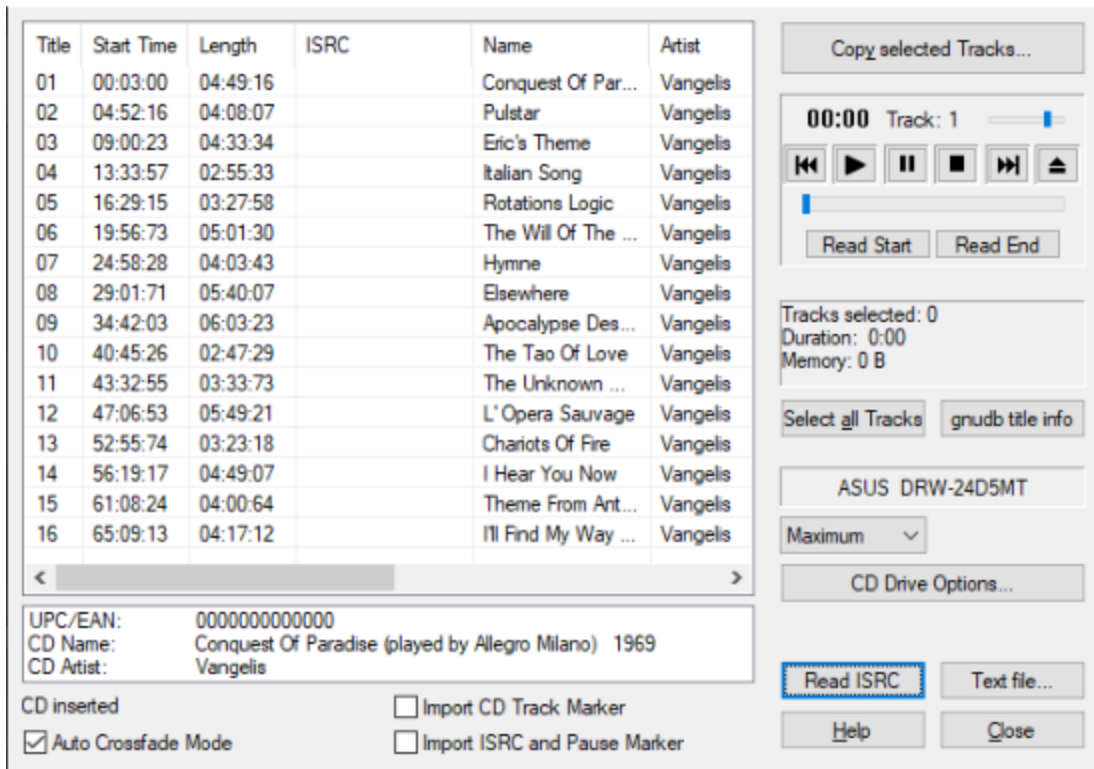
1. If you have more than one drive installed, click the **CD Drive Options** button and select the desired CD drive. Above the button you can see the name of the selected drive.

2. The content of the CD is displayed in the track list. Click **gnudb title info** to retrieve CD title information from the online CD database gnudb.With **Read ISRC** you can take over the ISRC codes from the CD, if available. With **Text file...** you open the system text editor with a text file containing this title information.

3. If you select a track in the list, you can use the playback control to preview the CD track. You can use the slider to access a specific position. To import only a part of a CD track, click at the beginning of the section **Read start** and at the end **Read end**.

4. To select multiple titles use the Shift and Ctrl keys, for all titles click **Select all tracks**.

5. Some options can be set for the import:

   - **Auto crossfade mode**: Auto crossfade mode is activated in the project and crossfades are created between the imported titles.

   - **Import CD track indices**: If this option is enabled, CD track indices are set in the project. The markers contain the metadata from the track list so that when you subsequently export to other formats that support metadata (MP3, CD Text when burning to a CD...) you preserve this information.

   - **Import ISRC and Pause indices**: The ISRC codes are also transferred to the CD track indices and the CD pause indices are also read from the CD.

6. Click **Copy selected tracks…**.

7. The **Import Audio** dialog opens. Choose a file name for the audio file. With **File type** and **Format settings** you can also select a file type other than .wav and thus convert the CD tracks to MP3 already during import, for example.
   By default, all CD tracks are saved into one file and only objects per each track are created. If you choose the option **Each track in its own file with names**, you can select different naming schemes in the menu below that derive the file name from the metadata.

8. Clicking **OK** will copy the audio from the CD drive to your hard disk. You can now copy more tracks from the CD or insert another CD or close the dialog.

Importing Audio DVDs

To import audio data from audio DVDs, select **CD/DVD** or via menu **File** > **Import** > **Import Audio DVD…**.

In the dialog **Import DVD-Audio** you proceed exactly as in the dialog **CD import** (see above). However, it is not possible to read in only parts of an audio DVD title, the controls for metadata and title indexes (gnudb, ISRC) do not exist for Audio DVDs.

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

Search Title Information Online

With the online CD title database **gnudb** the title information for an imported CD can be retrieved from the Internet. This query is based on the exact combination of the exact track lengths and order of the tracks on a CD. This also works if the tracks are loaded into the project individually (e.g. as MP3 files) and are in the right order. The track lengths can also deviate by a few seconds from the exact track length, but in general the correct CD is still recognized.

The gnudb title query is available via:

- the button **gnudb title info** in the dialog **Import CD**. Here, the database query is based on the track lengths of the inserted CD,

- the button **gnudb title info** in the dialog **Make CD**,

- Menu **CD/DVD** > **Get gnudb title info…**. In this case, the search is performed using the CD track indices available in the project.

gnudb Options

In case the query does not work, various settings can be made in the menu **CD/DVD** > **gnudb options**:

- **gnudb user data**: This sets the user data that Sequoia will use when requesting the gnudb. The use of the gnudb is anonymous and the user data is actually arbitrary, but the gnudb may need to handle many requests at once and therefore must be able to identify users. If you have problems accessing because someone happens to access gnudb at the same time using the same preset data, you can change it here.

- **gnudb proxy options**: If you have problems connecting to the gnudb server, you can choose another server from the list here if necessary or, as a first workaround, set the **timeout** value higher. An increased work load causes the server to react slowly and a connection cannot be made properly.

- **clear gnudb cache**: The gnudb online database creates a cache on the local hard disk. The cache contains all files that have already been imported via gnudb online. If necessary, this data can be accessed again at any time without always having to be online. However, you may wish to delete this cache at some point, for example, if it contains erroneous files or if new files are made available online.

Search CD Title Online

The gnudb title search can also be used if the exact title lengths are not known, because, for example, after a recording from cassette or record, there is only one large file in which all titles are available one after the other without title indices. Although you can use the function Menu **CD/DVD** > **Indices** > **Indices after silence...** to determine the track lengths based on the pauses, this does not work very well when the tracks segue into each other.

But if you know the album the tracks are from, you can also find out the track names and exact lengths by querying the gnudb CD database online. To do this, proceed as follows:

1. Place the recording in the first track.
2. From the menu **CD/DVD** > **Search CD online and set track indices...**. The Search CD online dialog opens.

Search for CD online   ×

Search for your CD manually using the CD name or the artist.
The trackmarkers will then be opened from the online database.

    Open Internet search page...

Was your search successful?
Once you click on the "Disc-ID" link for your desired CD,
you will receive a text site with the title data.

Copy the Internet link from your browser into the following text field:

    [   ]

(e.g. https://gnudb.org/gnudb/rock/940c260b)

☑ Set CD track marker again

    Apply CD title data   Cancel   Help

3. Start up your Internet browser and go to the gnudb search page by pressing the **Open Internet search page** button.

4. Enter the name of the album or band into the search field. One or more albums matching the search query will be listed. If you know that a certain album fits your recording, then click on **Details**. It will display the CD track list you were looking for alongside other details.

5. Click the link at Disc ID (an 8-digit number/letter combination, e.g. bf102c0e). The gnudb record for this CD is displayed.

6. Copy the URL (Internet link) from the address bar of your browser onto the clipboard.

7. Go back to Sequoia and paste the Internet link into the text box at the bottom of the dialog. Then click on **Apply CD title data**. The CD title indices with the corresponding names are set in the project.

⚠ *Attention: It may happen that the length of the first track does not match the track length provided by the database if the first track starts very quietly (e.g. intro or applause in live recordings) and thus the start of the recording was too late. As a result, all track markers will be a little bit too far back. In this case, move the second track marker forward while holding down the Ctrl-key; all subsequent track markers will then be moved by the same amount and should be positioned appropriately at the start of each track.*

Editing Track Indices and Metadata

Use the Track Indices (also called Track Markers) to divide a project into individual tracks for burning to CD or exporting to multiple individual files.

They can be edited in detail in the CD-Index-Manager (↗581), CD Title Indexes can store metadata other than the marker name.

Setting Individual Track Indices

Place the play cursor at the desired position and set track indices using the commands in the menu **CD/DVD** > **Indices** or these buttons. The numbering of already set, subsequent indices is automatically adjusted.
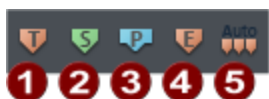
To display the track index buttons, select the workspace (↗58) **Mastering**.

① **Set CD Track Index** Track indices split a project into individual sections that correspond to individual CD tracks when burning to CD, and split up individual files when exporting with **Split at markers**.
Shortcut: **Ctrl + Alt + I**

② **Set CD Sub Index** Sub indices are used to mark individual sections within a track and can be accessed by some CD players.

③ **Set CD Pause Index** The CD pause index is a special sub index (index 0) used to mark the end of a CD track. At this point, some CD players switch to absolute silence during playback until the next track index, and the time in the display runs as a countdown to the start of the next track.

④ **Set CD End Index** This index marks the end of the CD. When burning and by default when exporting, the project is used from the project start to the end of the last object. If the last object still has a reverb tail after it ends, place a CD end marker at an appropriate distance from the end of the last object to ensure that this audio is exported with it.
You can also use the end index to specify the end of an export to keep additional audio material behind it, such as unused recordings and other material in the project that should not included in the export.

> 💡 *You can also specify the starting point for burning and exporting, because it always starts at the first CD track index, which does not have to be at the project start.*

❺ Set Track indices at object edges This function inserts track indices at each object in the project. In the submenu **Set indices at object edges - Options** some additional options for this function can be set by selecting the corresponding menu item:

- **Also set pause indices at object ends**: This option creates pause indexes at the object ends in addition to the track indexes.

- **Time offset for indices at object edges**: Select here a time interval with which the indices are set before the object edges.

- **No indexes at object crossfades**: With this option, no indexes are created at objects that are blended to the previous object via a crossfade.

If the audio files of the objects contain metadata like artist, title, album... (ID3 tags of MP3 files) or ISRC codes from Broadcast Wave files (↗653), this function transfers them to the track indices.
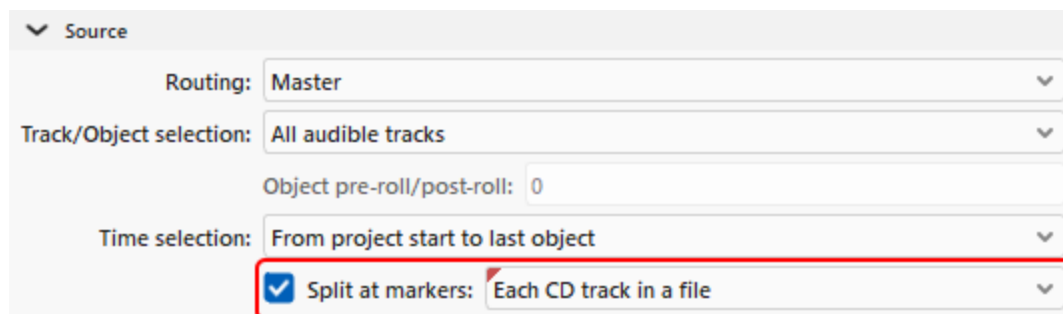
Use the menu option **Remove Index** to delete an index from the project. To select it, place the play cursor on the index position. **Remove all indexes** (keyboard shortcut:**Ctrl + Alt + Shift + I**) removes all indexes.

The menu option **Set track indices after silence...** opens a special version of the dialog Set markers after silence (↗115), which lets you set CD track indices automatically based on detected pauses in the audio material.

Managing Metadata via Track Indices

CD Track Indices were originally intended only for dividing a project into individual titles before burning to CD. With the ability to burn CD text, the indices gained the ability to store the title of a CD track and other metadata. Even though CD burning has become less important than exporting to compressed formats, Sequoia still uses the CD track indices to store such metadata.

Use the function **Set indexes at object edges** to transfer metadata from the objects' audio files to CD track indexes. Afterwards, this data can be edited via the **CD-Text/ID3-Editor**. To export the project to individual files and, in the process, enter the CD track index data in the metadata (ID3 tags) of the exported files, select the option **Each CD track to a file** in the Export dialog (↗665) at **Split at markers**.

Editing Metadata with the CD Text/ID3 Editor

Via menu **CD/DVD** or button **CD Text** in the **CD Index Manager** you open the dialog **CD Text/ID3 Editor**, where you can edit the metadata of the CD track indices.

The track names correspond to the name of the CD track indices in the project. All other information can only be displayed in this dialog.
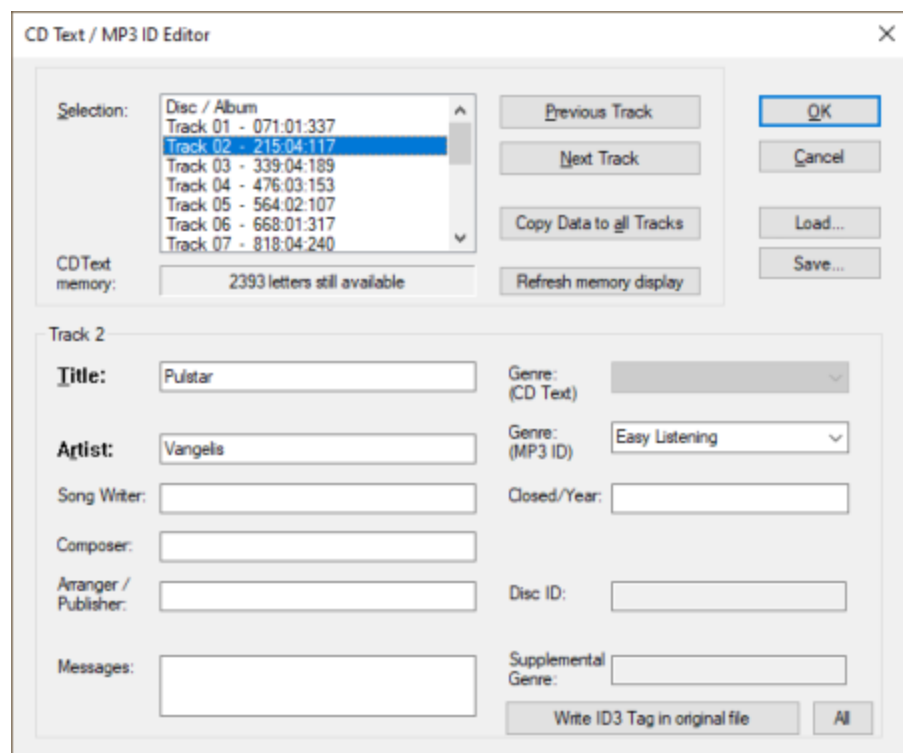
> **i** **Note**: *The maximum number of usable characters for a CD is 2824. At* **CD Text Memory**, *the available remaining number of characters is displayed. After making changes, click* **Recalculate Display** *to update the display.*

Write ID3 tags to original files: If the metadata comes from MP3 files loaded into the project, this button allows you to write changes made in this dialog back to the MP3 files in question without having to export them separately. With the button **All** you cause this for all such MP3 files in the project.

> **i** *To do this, the objects must be on the first track and exactly at the marker positions.*

CD Index Manager

Keyboard shortcut: **Ctrl + Shift + Alt + D**
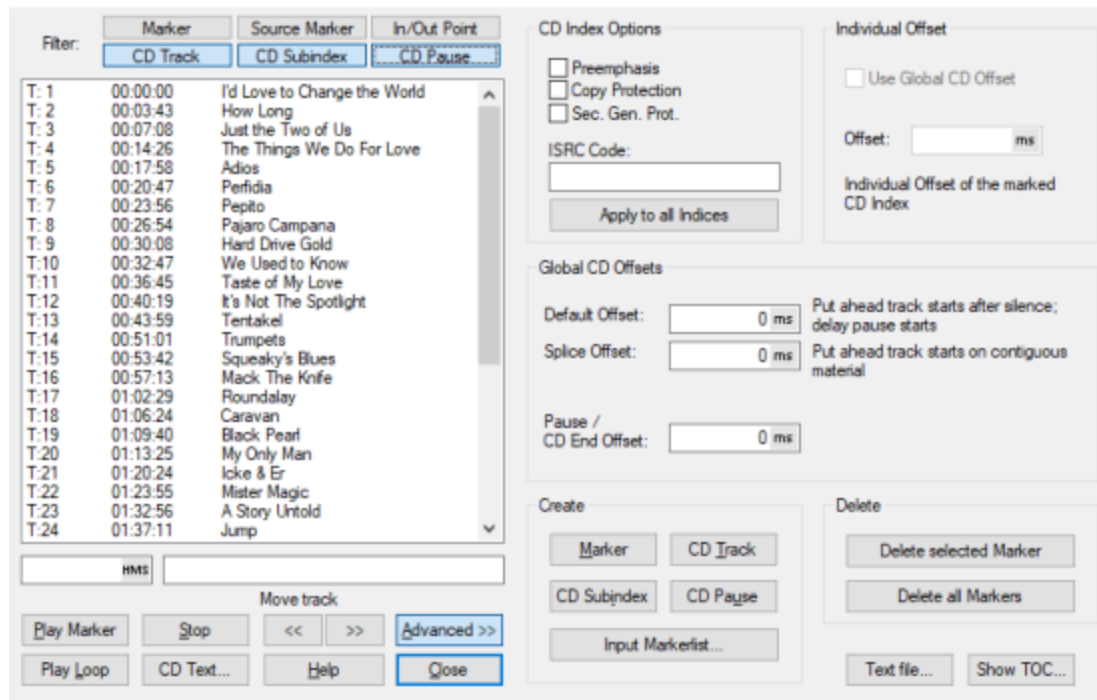
In this dialog all markers and CD indices of the project are listed. The **Filter** buttons at the very top of the dialog allow you to show only certain marker/index types. Select an index to change its position and name in the input fields below.

Play marker starts playback from the position of the selected marker. **Play Loop** plays a range (pre-roll + post-roll time) looped at the marker position. **Stop** stops playback.

With **CD Text** you open the **CD Text/ID3 Editor** (↗581), which allows you to edit additional metadata of a CD track.

Use the arrow buttons at **Move track** to change the order of the CD tracks. This also moves the audio material at the title index if it is on the first track.

If you click the button **Advanced**, more options will be displayed:

CD Index Settings: Here you can set different flags for the CD tracks: **Preemphasis**, **Copy Protection (SCMS)** and **Second Generation Protection**.

Preemphasis means that when the CD is created, the high frequencies are boosted before A/D conversion. During the A/D conversion itself, high-frequency quantization noise is always generated. The audio signal is now written to the CD with the treble boost. During playback, the CD player detects the preemphasis flag and corrects this frequency boost again, resulting in a reduction of quantization noise. In practice, preemphasis is very rarely used because the signal-to-noise ratio during CD creation is already so large that quantization noise is negligible.

You can enter **ISRC code** for each title. With **Apply to all indices** you can transfer the ISRC code to all indices and increase the numerical value by 1, 10, 100 or 1000 in each case. Furthermore, you can transfer the flags individually or collectively to the indices.

Preemphasis, **copy protection (SCMS)**, and **second generation protection** may be transferred and copied together for all indexes.

CD offsets shift indices relative to their position in the project during the CD burning process to ensure that a track start is played back correctly.

Due to technical limitations, there are delays when playing a track in a CD player compared to latency-free playback on a computer. A CD player needs a certain amount of time between triggering playback and outputting the audio signal at the output. On the CD, the title, pause and end indices should therefore be set somewhat earlier or later than in the corresponding project.

> **ⓘ Note**: *Title indices are shifted to the left by the selected time span, end and pause indices to the right.*

Individual Offset: If you disable **Use Global CD Offset**, you can apply an individual offset value for each track or pause index.

Global CD Offsets: The same offset values are applied to all indices. Title indices are written earlier, pause and end indices later.

- **Default Offset**: Offset value for track indices with silence between tracks. A common value here is about 160...200 ms (12...15 CD frames).

- **Splice Offset**: For direct transitions, without silence between titles, a lower offset value is applied. A common value here is about 67 ms (5 CD frames).

- **Pause/CD end offset**: Offset value for pause and end indices.

Create: Using the corresponding buttons you can create **markers**, **CD tracks**, **CD sub indices** or **CD pause indices**. The index is created at the current play cursor position, its position can then be changed in the time input field.**Marker list input...** (↗117) opens the dialog for quick input of markers from a list.

At **Delete** selected or all markers can be deleted. **Text file...** opens the system text editor with a text file listing all markers and positions (Marker List Export (↗327)). Only the marker and CD index types selected at **Filter** above will be exported. **Show TOC...** opens a text window where you can format the TOC as a text file with various options and then pass it to an external text editor in *.rtf format.

CD Arrangement Mode

This mode activates two automatic mechanisms that make it easier to create projects for burning CDs . You can activate the CD Arrangement mode via the menu **CD/DVD** > **CD Arrangement mode** or in the general project options (↗643)

- A default pause of two seconds is inserted between audio files loaded consecutively into the first track. The length of the inserted pauses can be changed via Menu CD/DVD > **Set pause time...**.

With the function Menu CD/DVD > **Set start-pause time…** you set the length of the pause before the first track. The standardized value is 2 seconds and should only be changed in exceptional cases. The start pause time is not visible in the project and is only inserted during burning.

■ You can change the order of CD tracks by dragging and dropping the track indices of the project. As soon as you drag a title index to another position in front of or behind another title index, an object on the first track located at this title index is moved to this position and the other objects and title indexes are adapted accordingly.
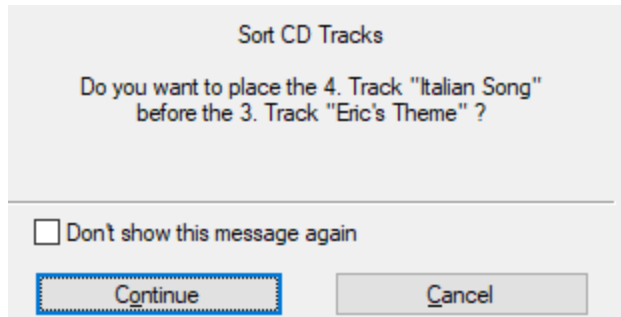
Create Audio CD

To create an audio CD from the active project, choose Menu **CD/DVD** > **Create Audio CD…** or click this button on the top toolbar.

The **Make CD** dialog opens.
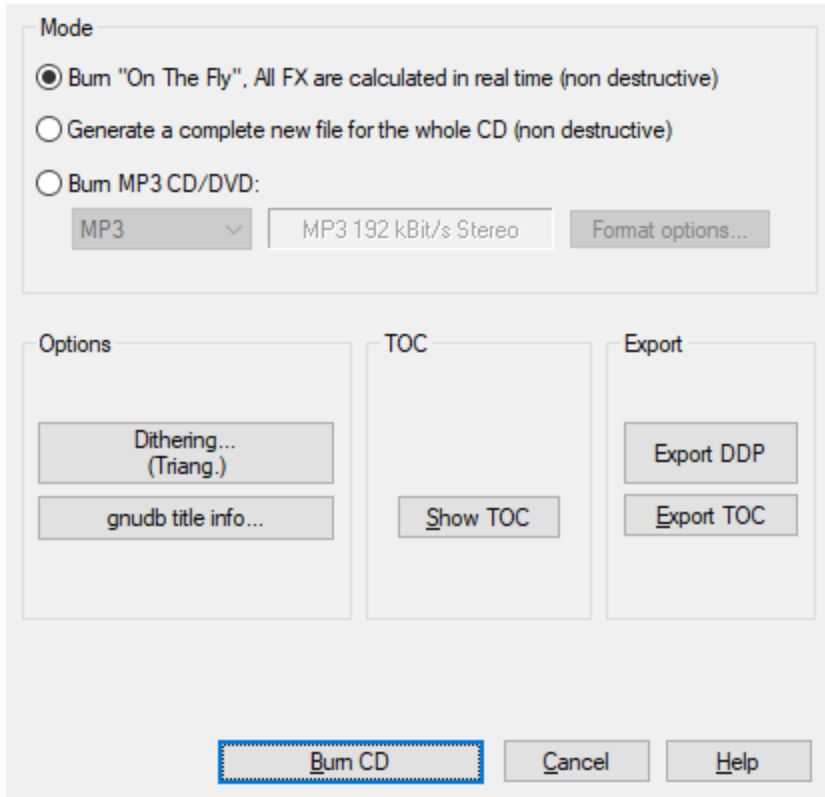
Mode: There are three different modes how to create the CD.

- ■ **"on-the-fly"**: writing the CD in real time. The project is simultaneously played in the background and the output is written to CD. Sufficient processing power must be available for this, since the burning speed is typically many times the playback speed. For example, if you burn at 24x speed, the computer load during playback must be just under 4.2%. All CD Recorders have safety measures in place to prevent burning failure if the data is not delivered fast enough, but the advantage of high burning speed is then lost due to repeated aborting and restarting. Use this function to burn simple projects, e.g. copies of finished titles.

    🔅 *To test whether writing in real time is possible, you can also select the option **Simulate recording** in the following dialog.*

- ■ **Generate a complete new file**: Use this function to create your CD if the project is too complex and computationally intensive to burn on-the-fly. The project is first exported to a new audio file. You can also try reducing the burning speed first. However, only blank CDs optimized for very high burning speeds are now available, and they no longer work at low burning speeds (<10). We therefore recommend this method for burning more complex projects such as complete mixes.

- ■ **Burn MP3 CD/DVD**: This exports all tracks into individual MP3 files according to the selected format option and passes them to the separate burning program **MAGIX Speed BurnR** to burn them onto a data CD.

Options:

- **Dithering settings**: The bit depth of the audio output must be reduced from 32bit float to 16bit for burning to CD. The **Dithering options** button displays the dithering mode used for this purpose. You can change it by clicking the button. See Dithering settings (↗723).

- **gnudb title info**: In case you burned an imported CD and missed the transfer of the CD track information from the gnudb.org database, you can still add this information to the CD track indices with this button.

TOC: Sequoia creates a TOC (Table of Contents) file in the project folder with the name of the current project and the extension *.TOC before writing the CD. This file contains all data about the CD.

- **Show TOC**: This opens a text window where you can format the TOC as a text file with various options and then pass it to an external text editor in *.rtf format.

Export: Besides burning to audio or data CD, it is also possible to export the CD only in the form of audio data and a description file.

- **Export TOC**: The project is exported to a single wave file. In addition, a .TOC and a .cue file are generated.

- **Export DDP**: Creates a DDP image for transfer to a pressing plant with the separate program **DDP Export**.

With **Burn CD...** you open the Burn CD dialog where you start the burning process.

ℹ️ *The dialog can only be opened if there is a recordable CD in the CD burner.*

Select the **Write speed** in the menu, below it you can set some more options:

- ■ If in doubt, you can simulate writing with **Simulate recording**: This allows you to test the burning without actually writing data to the blank.

- ■ **Prepare CD-R for additional data session(s)**: The audio CD is not closed and can be supplemented with additional content with a data session.

- ■ **Activate Burn Proof**: In case on-the-fly burning does not deliver data fast enough, burners provide this safety measure to not destroy the blank. The option should be disabled only in case of problems.

- ■ **Burn with CD Text**: Metadata can be burned onto the CD as CD Text information.

- ■ **CD Text settings**: This opens the CD Text/MP3 ID Editor (↗581).

- ■ **Number of copies**: You can also burn several copies of the same CD in a row. After the burning process is finished, the CD will be ejected, then insert the next blank and the next CD will be burned.

With **Write** you start the burning process.

CD Disc Options

Via menu **CD/DVD** > **CD Disc Options...** you can edit further settings for the current CD.

CD Title: Title of the CD which will be written as CD-Text on the CD.

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 can be set. This is not important in "disc at once" mode, since the CD will always start with title no. 1.

Allow bonus track before 1st track: This makes it possible to burn a "hidden track" on the CD that is located before the first regular track (track 2) . The "hidden track" may only be accessed on standalone CD players via the "STEP BACKWARD" button.

Viewing CD Recorder and Blank CD Information

- To view information about inserted blank CD such as type, number of tracks, number of sessions, total memory, free memory, and status, select **Show CDR Disc Information...**.

- To view information about the active CD recorder drive, select **Show CDR Drive Information**. These include: Manufacturer, product name, product revision, cache size and various properties supported by the drive (disc-at-once, indices, second generation copies, catalog numbers, ISRC, CD-Text, CD-RW).

DDP Export

An additional program DDP Export will be opened.

This additional program enables the export of a CD project that is compatible with DDP-CD mastering standards. It is opened using the **DDP Export** button in the **Create CD** dialog or menu **CD/DVD** > **DDP Export...**.

Set the name for the audio file to which the audio data for the CD will be written. Sequoia exports the project and then displays the loudness. Then the program **DDP Export** will be opened.
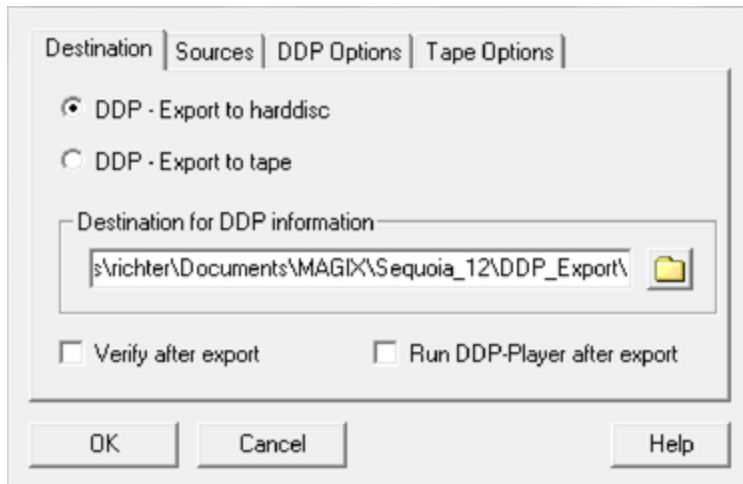
The project can be exported either to hard disk or onto tape (for example, Exabytes EXB-8505, EXB-8500). You can then deliver a project exported to disk to the pressing plant via file transfer.

Destination for DDP information: When exporting to hard disc, a destination folder can be specified here. If exporting to tape, enter the path of the tape drive here. With the function **Verify after export** you start an import of the image for verification after the export.

The **Run DDP player after export** option opens the "MAGIX Sequoia DDP Player by Sonoris" directly after export.

Sources: The **CD-Info file** (*.tcd) belonging to the project is created automatically by Sequoia. If the DDP export was not started from Sequoia, but directly from the Sequoia program folder, you can select the corresponding files (CD Info *.tcd + audio file/data file) here. **CD info details** shows the content of the CD info file.

DDP options: The formats of the **1.01** and **2.00** standard for DDP export can be used and also specify a **Master ID**. This ID is used by the manufacturer for testing purposes.

Tape options: Enter a **Volume ID** and an **Owner ID** for archiving.

With **OK** the CD project is exported, creating the following files: `DDPID` (DDP-Id-Stream), `DDPMS` (DDP-Map-Stream), `Image.DAT` (audio data), `CDTEXT.BIN` (CD text information), `PQDESC` (PQ description), `CHECKSUM.TXT` (CRC32) and `CHECKSUM.MD5`. For electronic transmission to the pressing plant, it is best to pack all files together in an archive (zip, rar...).

> ⓘ *Note: Sequoia also creates a 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.*

Create DVD-Audio

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.*
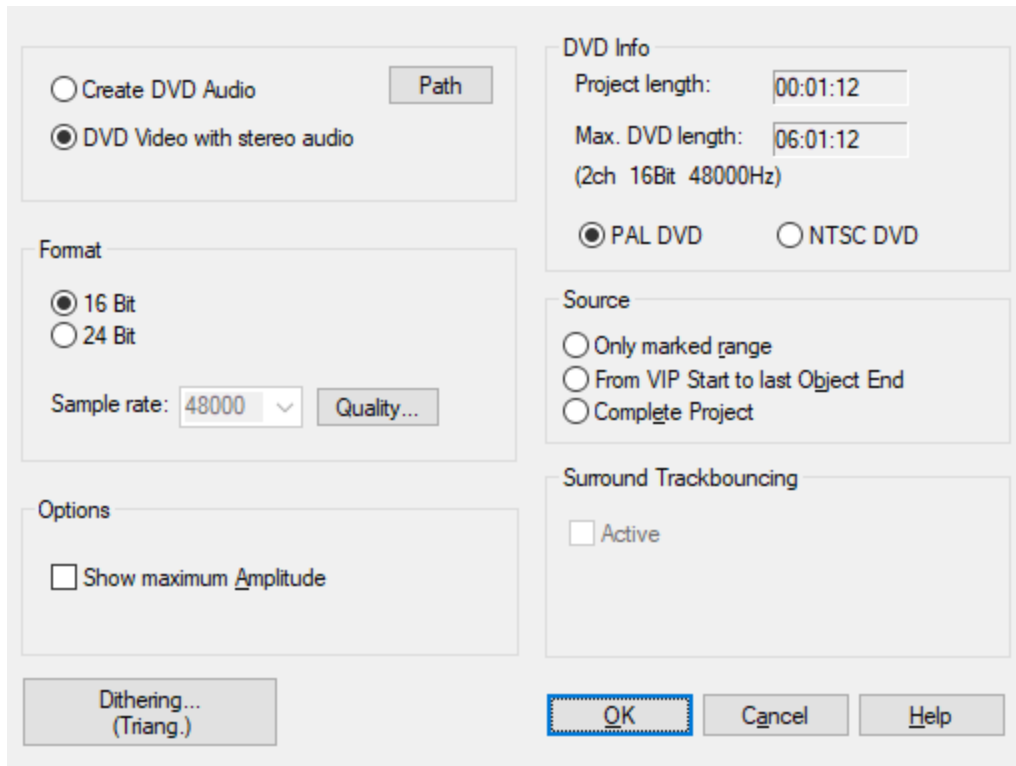
Sequoia burns so-called "black discs", i.e. DVD audio without graphic menus.

> -ⓥ- *You can create video DVDs with menus, videos and slideshows using the video editing and DVD authoring program "MAGIX Movie Studio".*

As an alternative to DVD Audio, you can also create an ordinary DVD Video that contains an uncompressed stereo audio track without video.

Like audio CDs, audio DVDs can be burned directly from the program, all you need to do is set track markers. You can burn both an audio CD and an audio DVD from the same project.

Choose Menu **CD/DVD > Create DVD Audio...**. The dialog **Trackbouncing - DVD** opens.

Select either **Create DVD audio** or **DVD video with stereo audio**.

> ℹ️ **Note**: *Please note that DVD menu authoring is not available for **DVD-Video with Stereo PCM**. For extensive authoring options we recommend "MAGIX Movie Studio".*

The actual creation of the DVD image and burning is done with the external program DVDAudio.exe. The path to this program is preset. If the program has been moved, you can set the current installation path with the **Path** button.

Format: You can choose between **16** and **24** bit audio.

Sample rate: For **DVD- Video** 48Khz must be used. Therefore, if the project has a different sample rate, resampling must be performed. At **Quality...** you can reach the corresponding settings.

With **DVD-Audio** you can freely choose the sample rate, 44.1 kHz and 48 kHz as well as double and quadruple sample rates of it are possible. Up to 6 discrete audio channels are supported, so configurations like 5.1 surround, stereo or 4.0 are possible.

The only limitation is that the maximum guaranteed data rate of around 10 Mbit/s for hardware players, may not be exceeded. Therefore, for example, with 5.1 Surround 24 bit is only possible up to 48 kHz sample rate, at 96 kHz the 10 Mbit 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

Options:

- **Show maximum amplitude**: Here you can display the maximum level in dB after the export process, in order to be able to set outboard equipment correctly for further processing or to correct the master level. After the export is finished, a corresponding information window appears.

- **Dithering**: Every export 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). You can also dither according to the system options or call the dithering options of the system options. The button value in brackets (e.g. **Triang.** or **POW-r 1**) indicates the currently set dithering algorithm.

  > ℹ️ *For detailed information, see* Dithering *(↗723).*

- **DVD Info**: Here you can find out details about the **project length**, **maximum DVD length** and choose between **PAL DVD** and **NTSC DVD**.

- **Source**: Specify whether the export should be performed only over the length of the range selected in the project, from the beginning of the project to the last object (plus the reverberation time) or the complete project

- **Surround track bouncing**: Surround audio can only be used with DVD Audio, the project must have a surround master for this. If this option is selected, you will be asked before the export in which surround format the export should take place.

Clicking **OK** first exports the project to one or more audio files according to the settings made, then opens the **Create Disc** dialog.

Select burner and burning speed. The red button creates a DVD image first, and the writing process starts after this is completed.

> ℹ️ **Note**: *You need twice the space of a DVD on the hard disk, because first the project is exported and then the DVD image is created from this audio data, which is then burned.*

Batch Processing

Batch processing allows you to automate work processes. You can specify certain edits for an audio file, which are then applied to all files in a list (the batch) of any size. The definition of the edits, including format conversions and save naming scheme, is called a **job**. You can also define several such batch jobs, which can be collected in a job list and executed independently one after the other.

Possible editing options are:

- Loudness or peak normalization
- Linear fading (in and out)
- all real-time effects (including restoration and VST effects)
- Time Stretching / Pitch Shifting
- Remove clipping
- Remove DC offset
- File format conversion: bit resolution (8/16/24/32Float), sample rate, stereo/mono settings, data compression, dithering.
- Save in all available export formats.

Settings can be made for the location and naming of the created files as well as how the original is to be kept.

In the **Error Handling** tab, you can also specify how to handle files with errors.

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.

The process for batch conversion is as follows:

1. In the **Source files** tab, select the files for editing.
2. In the **Effects** tab, select the required editing processes.
3. In the **Target format** tab, select the output format.
4. In the **Target files** tab, specify where and under what names the files should be saved.
5. On the **Error Handling** tab you can configure how to handle corrupted files.

Jobs

You can run multiple batch jobs in succession. The job list located on the right edge of the batch window is used for this.

With **Save** you can save a job. This allows you to reuse jobs once configured in subsequent sessions by loading them back into the job list with **Load**.

New jobs are created by clicking the **Add job** button. When you click a job, the settings of this job (source files, effects, destination format, destination files) are displayed in the dialog. **Run Jobs** starts batch processing and runs all loaded jobs. **Delete job** removes the selected job.

ⓘ *All jobs remain loaded in the list until you quit Sequoia, even if you close the batch window in the meantime.*

Source files

Create the list of files you want to edit.
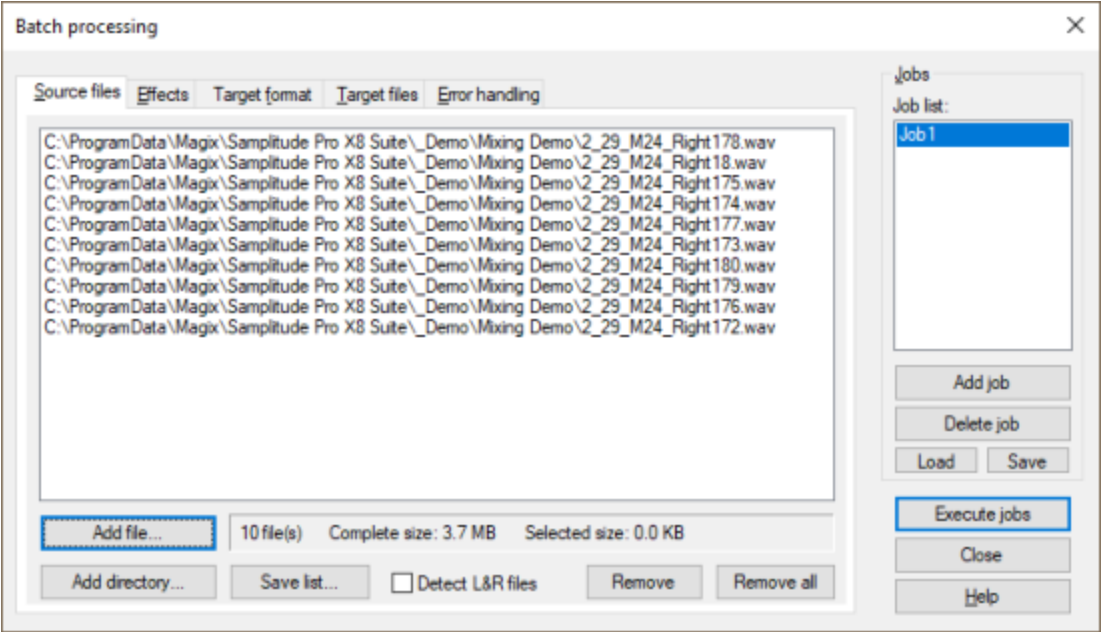
Click **Add File** to open a file browser to select the files you want to include in the list. Multiple selection of files is also possible here. Select **Add directory** to add all of the audio files in a folder (including all subfolders). You can add files in all file formats that Sequoia can import. With **Save list** you can save a list of your file selection in *.m3u format. 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

Here you specify the effect processing for the files:
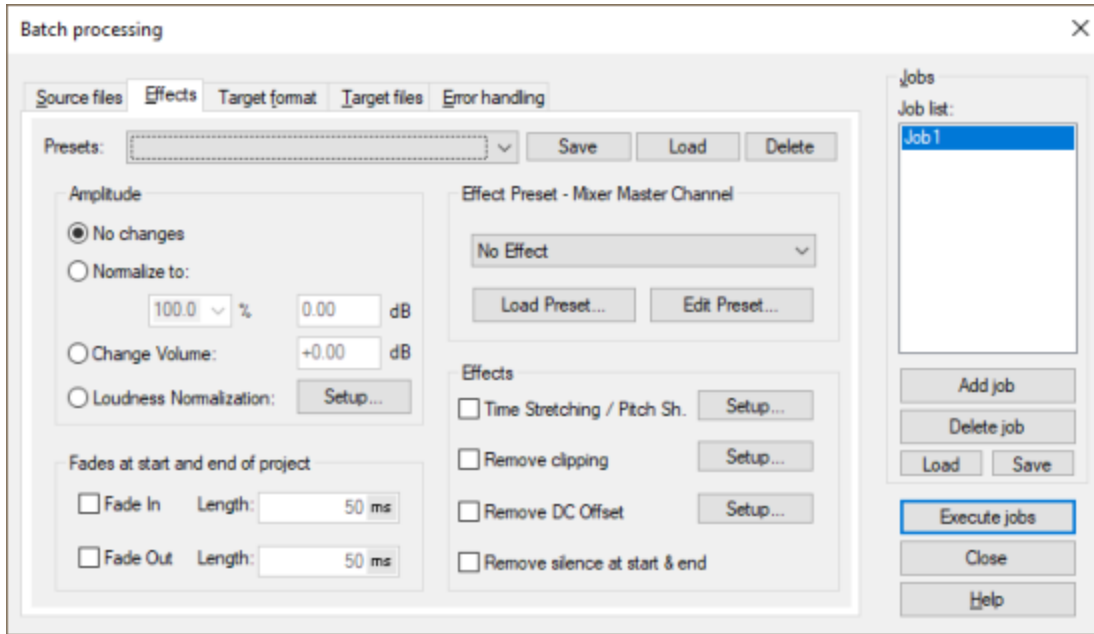
Normalize to: You can specify a target maximum amplitude value in % or in dB. For example, a value of 75% corresponds to -2.5dB. You can also enter **volume changes** in dB.

> ℹ️ *For detailed information on normalizing, see the section* Normalizing *(↗291).*

Loudness Normalization: Here you have the possibility to apply a loudness adjustment to bring the selected files to a certain loudness value. For details, please refer to Loudness Adjustment (↗279).

Fades at start and end of project: Linear fades can be added to the start and end of files.

Effect Presets - Mixer Master Channel: In this section you can use any combination of all real-time effects from Sequoia in batch processing. The presets used for this purpose correspond to the mixer snapshots (↗503). Saved mixer snapshots can be selected from the menu or loaded into batch processing as `*.mix` file with **preset**.

To create mixer snapshots for batch processing:

1. Load one of the files you want to edit into a new project.
2. Load the effects you want to use into the mixer's master channel (at **Master Plug-ins**) and make the necessary settings.
3. Save the mixer settings to a mixer preset. To do this, click the **Save Mixer to File** button 🖫 below the mixer snapshot locations.
4. Load the saved mixer snapshot in the **Effects** tab of the batch dialog from the menu.

> ℹ️ *You can also use **Edit Preset** to open the* Effect Browser *(↗216) with associated FX Routing dialog directly from the batch processing to edit the effect chain for batch processing. However, the "detour" via the mixer makes it possible to try out the effects on the audio material before using them.*

Additional effects: Beside the Mixer effects the non-real-time effects **Timestretching/Pitchshift**, **Remove clipping** and **Remove DC offset** can be used. The **Setup** buttons open the corresponding effects dialogs.

Remove silence at start and end removes silence at the beginning and end of the file.

Target format

Here you specify a destination format for the files. This allows you to convert files to other file formats.
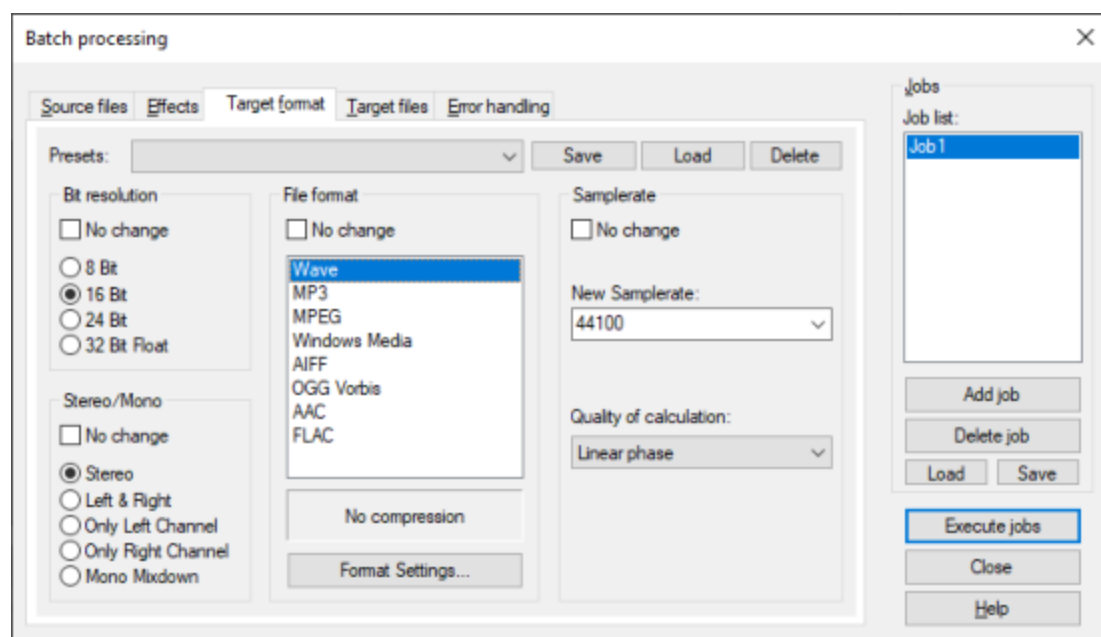
You can choose the bit resolution (8/16/24/32 bit float), the sample rate, stereo/mono/left/right as well as the save format with the corresponding **format settings**.

> ℹ️ *To learn more on the subject of sample rate change, see* Adjusting the sample rate *(↗298). To learn more about file formats, see* Export*Exporting Audio (↗661).*

> 💡 *All settings of this tab can be saved as preset for later use.*

Target Files

Here you define where and under which name the processed files are saved and what should happen to the source files.
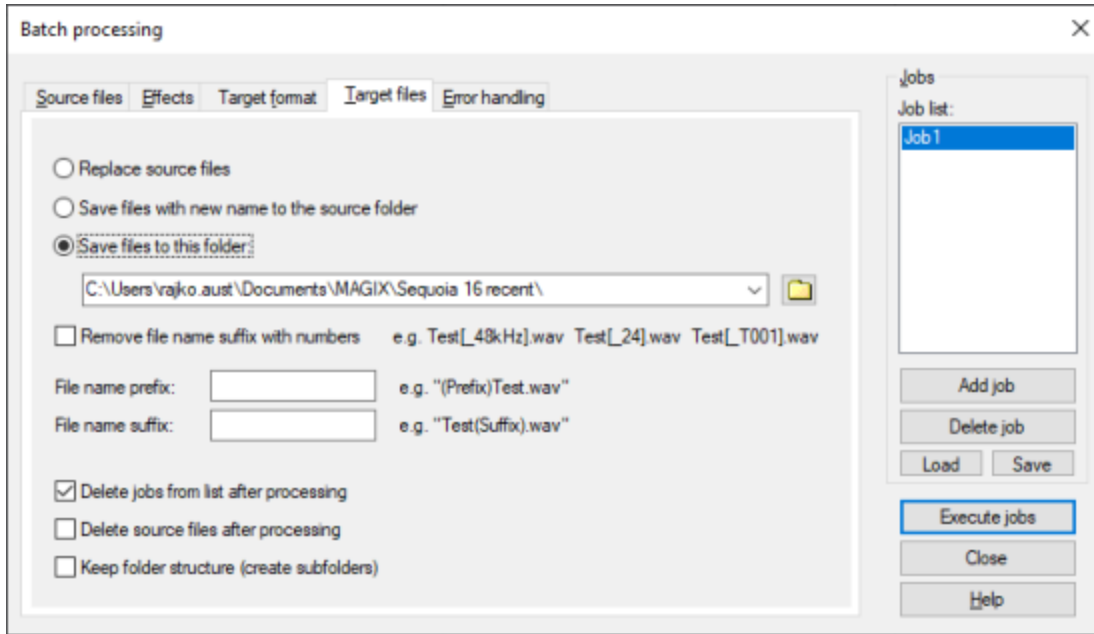
Replace source files: The original file is replaced with the edited one. If the file is used in a virtual project, the VIP is closed first.

Save files to the source folder with changed names: The edited file will be saved in the source folder next to the source file - adding the additional characters to the file name that you can enter in the fields **File name prefix** and **File name suffix**.

Save file to this folder: The edited file will be saved in the folder specified here. If you have entered characters in the **file name prefix** and **file name suffix** fields, the file names will also be expanded accordingly.

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: This option deletes the source files after the job is executed. A confirmation prompt will appear.

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

Error handling detects files that deviate from the loudness norm with respect to certain criteria and should therefore be handled separately. These files are stored in a special error folder.

To enable error handling, select the **Activate error folders** option and select a folder where files will be stored. All other files are processed normally and saved in the defined target folder.
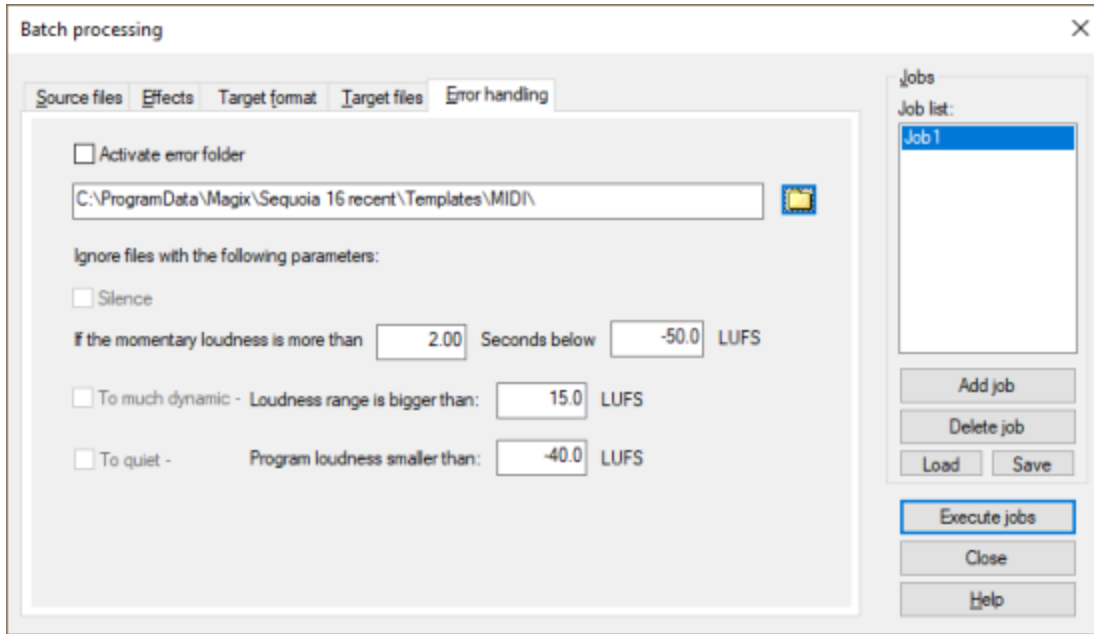
Silence: Sort out files that are below the specified momentary loudness (LUFS) for longer than the specified duration.

Too much dynamic: Sort out all of the files that exceed a certain loudness range (LUFS).

Too quiet: Sort out all of the files that fall below a certain loudness (LUFS).

Opening Batch Processing using the Command Line

By calling Sequoia.exe with the command line parameter `-batch <job>.xml` the batch processing can be executed from the command line. For example, you can use it to perform regular tasks such as converting and archiving completely automated by using batch scripts.

```
C:\Sequoia\Sequoia.exe -batch "C:\Batchfolder\job.xml" | Out-Null
```

> ⚠ *The normal Windows command line interpreter, when executing a .bat script, does not wait for a command to complete before executing the next command. If you want to invoke batch processing from batch files and use the results of batch processing later in the script, you must run such scripts in Windows Powershell.*

Watch Folder

The **Watch Folder** feature allows you to automate all audio file edits and conversions that are possible via batch processing as soon as the files are saved to a specific folder. You can do this in the following way:

1. From the **menu, select Edit** > **Watch Folder...**. A dialog opens that is mostly identical to the **batch processing** dialog.
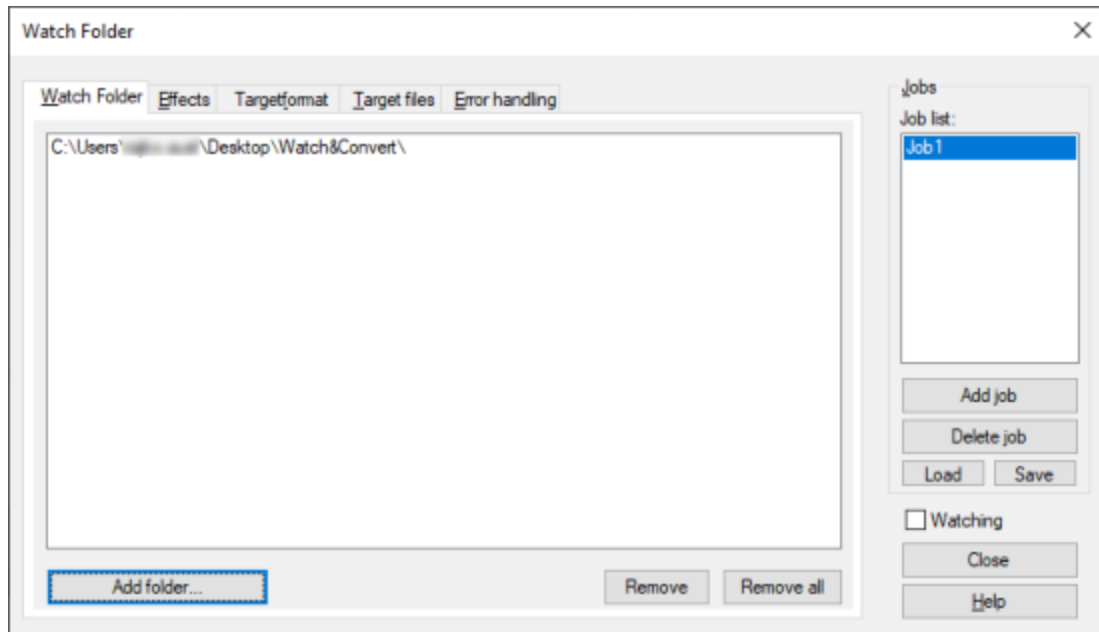
2. In the **Watch Folder** tab, add folders to the list of watched folders using **Add Folder…**.

3. Define the edits and conversions you need. For details, see Batch Processing (↗591). As with batch processing, you can also define multiple jobs, for example saving in several different formats. These jobs can also be saved for later reuse.

> ℹ️ *The storage format for Watch Jobs is different from that for Batch Jobs. However, the presets from the **Effects** and **Target format** tabs of the batch processing can also be used for the Watch jobs.*

4. Once you enable the **Watching** option, all compatible audio files in the watch folders will be processed and saved according to the defined jobs. As soon as new files are encountered in the watch folders, they are processed as well.

5. The monitoring function is only active as long as the dialog is open. However, the settings of the dialog are retained until you exit Sequoia, so you can continue monitoring with the same settings when you open the dialog again.

To use the Watch Folder feature all the time in the background, you could simply open a second instance of Sequoia and keep it open in the background. However, it is better to open Sequoia with the special **command line parameter** –watch to avoid hardware conflicts and to save your computer's resources. In this case, Sequoia will only open the Watch Folder dialog and can remain open in the background.

```
Sequoia.exe -watch [path]\Watchfolder_Job_Name.watch.xml
```

> 💡 *Batch job processing can also be done from the command line, start Sequoia with* `Sequoia.exe –batch [path]\Batch_Job_Name.watch.xml`

Synchronization

Synchronization allows you to couple other audio programs or studio hardware such as tape machines, drum machines, video recorders, or sequencers to Sequoia so that all devices and programs run in time or clock synchronization, i.e. musical tempo, playback or recording speed, and position at the devices involved match.

Sequoia can be configured for synchronization as a master (leader) or as a slave (follower). As a master, it generates the synchronization data for other components in the system. As a slave, Sequoia receives the synchronization data and follows it during playback and recording.

To switch synchronization on and off, use this button on the transport console or menu **File** > **Project properties** > **Synchronization active** (keyboard shortcut: **G**).
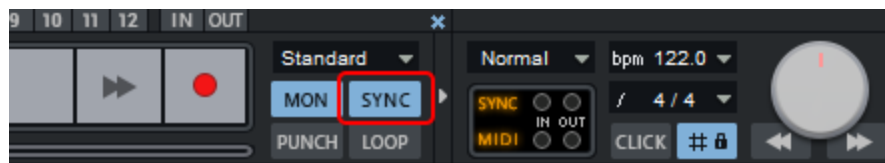
All settings for the different synchronization options are located in the dialog **Synchronization**, which you open with a right click on this button or via menu **File** > **Properties of the project** > **Synchronization settings...** (keyboard shortcut: **Shift + G**).
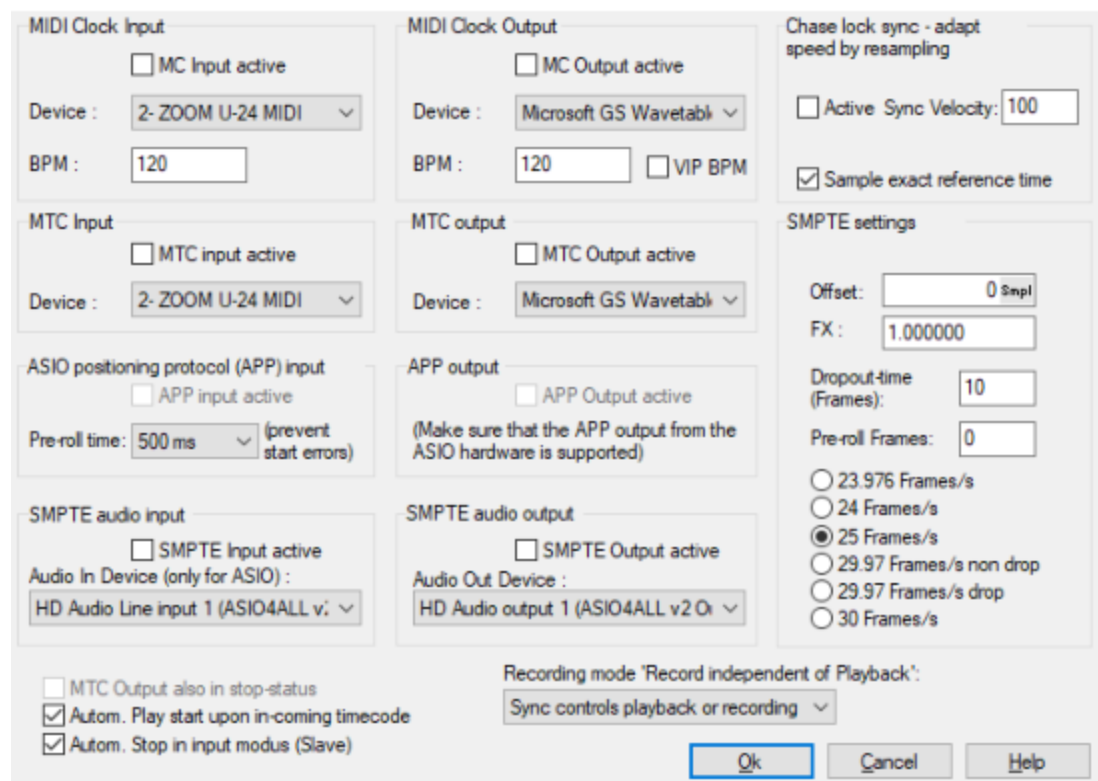
Synchronisation Protocols

The protocols (data formats) used for synchronization contain information about the start position, start and stop commands and, in some formats, position information that is constantly transmitted.

Sequoia receives and transmits the synchronization protocols **MIDI Clock** (MC), **MIDI Time Code** (MTC), MIDI Machine Control (MMC), ASIO Positioning Protocol (APP) and SMPTE Timecode Audio.

MIDI Clock

The MIDI Clock is pure tempo information, the time reference is the musical tempo. For this purpose 24 **Timing Clocks** (MIDI System Realtime Message 0xF8) are transmitted per quarter note (24 ppqn - *pulses 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 receiving device determines the current tempo from the number of clocks received per elapsed time. But beyond that MIDI Clock does not transport any current information about the time position in the project.

With the messages **Start** 0xFA (start at project start), **Stop** 0xFC and **Continue** 0xFB (start at position) the playback of the receiving device can be started and stopped synchronously. Before each **Start** or **Continue** message, the **Song Position Pointer** (SPP) 0xF2 is also sent. It indicates the current bar position in 1/16 notes and thus allows synchronization of the project positions at least at playback start.

MIDI clock synchronization is suitable for synchronizing devices with tempo and clocks as time reference like hardware sequencers or the LFOs or arpeggiators of external synthesizers.

At **MIDI Clock Input/MIDI Clock Output** in the dialog **Synchronization** you activate the synchronization via MIDI Clock:

- With **MC Input active** you activate the MIDI clock synchronization of Sequoia as slave (follower). At **Device**, select the MIDI device from which the MIDI Clock signal is to be received for synchronization.

  > ⓘ *Sequoia is not able to adjust its project tempo to an incoming MIDI clock. Instead you have to set the tempo of the received MIDI clock at **BPM**. The project tempo should also correspond to this value, so that in the project the time display in bars matches the MIDI clock. For synchronization with MIDI Clock we recommend to use Sequoia as master or, if possible, to use MTC (see below).*

- With **MC Output active** you activate the MIDI clock synchronization of Sequoia as master (leader). At **Device**, select the MIDI device through which the MIDI Clock signal is to be sent.
  The option **VIP BPM** ensures that the current project tempo is used for the MIDI clock. If you disable it, you can also send the tempo set at **BPM**.

MIDI Time Code (MTC)

With the MIDI Time Code (MTC) the current time position is transmitted continuously, via a MIDI port in the form of a SMPTE time code, which is transmitted as **MTC Quarter Frame** (MIDI System Realtime Message `0xF1`) four times per frame. Each message contains a part of the absolute time position as data byte. Eight such MTC Quarter Frames result in the transmitted time code as an absolute time position in the project in hours:minutes:seconds:frames. This means that the frame rate in the components involved must be the same for MTC synchronization to work. Therefore, select the same frame rate in the SMPTE settings (see below) and on the device involved.

At **MTC Input/MTC Output** in the dialog **Synchronization** you activate the synchronization via MTC:

- With **MTC Input active** you activate the MTC synchronization of Sequoia as slave (follower). At **Device** select the MIDI device from which the MTC signal is to be received for synchronization.

- With **MTC Output active** you activate the MTC synchronization of Sequoia as master (leader). At **Device**, select the MIDI device through which the MTC is to be sent.

There is no transmission of the playback state as with MIDI Clock, since it is clear from the change (or non-change) of the time code whether playback is running or not. For this, there are some MTC specific options that affect the playback and recording behavior:

- **MTC output also in stop state**: Eight MTC messages must be transmitted within two frames to transmit a complete timecode. If with Sequoia as master (leader) the time code is transmitted just with the start of the playback, the slave (follower) component synchronizes to the current position only after a delay of one frame, so it is in a wrong position for a short time after a manual position change while in stopped state (moving the play cursor).
If the option is active, the current time position is sent continuously even when playback is stopped to avoid this. The option is also important for devices such as tape recorders or video recorders, so that they can detect these position changes in the stop state and transport the tape to this time position.

- **Automatic play start on incoming timecode**: If Sequoia receives timecode as a slave (follower), Sequoia starts playback as soon as a timecode is received.

- **Automatic stop in input mode (slave)**: If Sequoia receives MTC as slave (follower) and the corresponding track is ready for recording, Sequoia starts recording as soon as MTC is received. If there is no more MTC, Sequoia stops recording and automatically goes into playback mode. When timecode is received again, Sequoia starts playback. If you want Sequoia to continue recording when you receive MTC again, uncheck this option.

APP (ASIO Positioning Protocol)

The ASIO Positioning Protocol allows the transmission of synchronization information via the digital inputs or outputs of the sound card (SPDIF, ADAT) or special interfaces (LTC, Video Burst In). This allows you to synchronize with timecode as with SMPTE Audio or MTC without the need for an additional audio or MIDI connection. This protocol is only supported by very few, special audio cards.

Activate the operation as slave (follower) or master (leader) with the corresponding options **APP Input active** or **APP Output** active. With active input, you can set a **Pre-roll time** to ensure stable synchronization.

SMPTE settings

SMPTE Timecode is a conventional synchronization standard for linking different audio and imaging devices via absolute time reference. With each SMPTE message, the exact amount of time that has elapsed since the start of the project is sent.

The SMPTE timecode uses 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 frame rate corresponds to the number of frames per second common in film and video.

Go to **SMPTE Settings** and select the appropriate project frame rate here. This frame rate is used for synchronization with MTC and SMPTE Audio. If you use video files in the project, you should also match their frame rate.

- **SMPTE offset**: Here you can specify an offset that is subtracted from the received 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 (↗643).

- **FX**: This legacy option can be used to compensate for synchronization inaccuracies in old projects. It should not be changed for working with current projects.

Dropout time (frames): Specify here after how many consecutive unrecognized frames the synchronization should be considered failed.

Pre-roll Frames: Here you can enter a number of frames that Sequoia should ignore before synchronization starts. This can be used to respond to the fact that analog devices first need a certain amount of time to reach the correct speed. To prevent Sequoia from synchronizing to an invalid time, the preroll frames are skipped before the synchronization takes effect.

SMPTE Audio

For synchronization with external video or audio devices, you can use the SMPTE audio signal of these devices (slave) or have these devices controlled by a pre-recorded SMPTE signal or one generated in Sequoia (master).

■ With **SMPT Input active** you activate the SMPTE synchronization of Sequoia as slave (follower). At **Audio In Device**, select the audio device at which the timecode is received. Playback and recording now follows the received timecode.

In a track in the Track options (↗141) (or menu **Track** > **Track type**), activate the **Timecode** option to set this track as **timecode track**. Then you can record and playback the timecode on this track and thereby synchronize the external device with Sequoia.
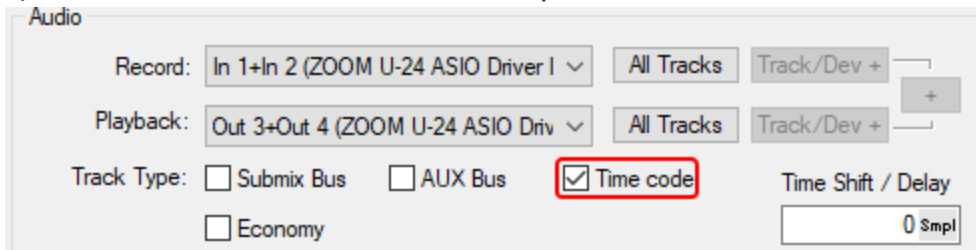
■ With **SMPTE Output active** you activate the SMPTE synchronization of Sequoia as master (leader). At **Audio Out Device**, select the audio device that will send the timecode. If no timecode from a previous recording is to be used, you can use the SMPTE generator (↗604) to generate timecode yourself, which then controls the external devices.

💡 *In the* time display *display fields can be configured to show the incoming SMPTE timecode (**Sync timecode input**) or the content of the **Timecode track***.

Chase Lock Sync

With chase lock synchronization, audio playback from Sequoia in slave mode exactly follows a received timecode signal by changing the playback speed through real-time resampling. Thus, Sequoia is able to synchronously follow a timecode source with variations such as analog tape machines or VCRs over a longer period of time if the timecode is located on a track of a multitrack tape machine, for example.

■ **Active**: Here you activate Chase Lock synchronization.

   ℹ️ **Note**: *Note that if Chase Lock is enabled, real-time resampling will also occur during recording according to a received timecode with all possible fluctuations. This causes increased CPU load and may result in unwanted changes to the audio material if this is played later with other timecode references.*

■ **Sync Velocity**: The sync velocity influences the speed of the tempo adjustment. The larger the value, the faster Sequoia follows a tempo change of the master device, but the greater the pitch fluctuations in the audio material. The default value for the sync velocity is 100. Increase it if the synchronization is not exact.

■ **Hardware Pitching**: When using hardware pitching, the chase lock synchronization changes the sample rate of the sound card by direct access. This achieves particularly precise synchronization without additional CPU

load. However, this feature has to be supported by the sound card.

- **Sample exact reference time**: If you select this option, Sequoia takes the time position of the sound card as clock reference (timer) and not its own internal timer. This ensures sample-exact synchronization between the recorded audio material and the sync signal.

ℹ️ *If you use Wordclock for digital sample rate synchronization of your devices with Sequoia, you should not use Chase Lock.*

Wordclock Sync

When digital data streams are exchanged between multiple devices using protocols such as ADAT, SPDIF or MADI, a clock reference must be established throughout the interconnected system. The clock signal is transmitted via the digital input and can come from involved computers, but also external devices like converters or mixers can serve as clock reference. For these systems, there can be only one master but several slaves. This digital word clock is not a time code, no position information is transmitted. The clock signal is just a digital pulse and is used to synchronize the timing between the connected devices and to keep the sample rate constant everywhere to prevent transmission errors. Professional digital audio devices normally feature a word clock input and are able to generate and receive a clock signal.

To set the clock source, use the **Clock Source** menu in the Audio Settings (↗691). There the ASIO driver of the sound card may provide options to synchronize the sound card to an external clock source.

SMPTE Generator

This function allows you to generate an SMPTE audio signal for synchronization with analog equipment.

1. Create a new track and open the SMPTE generator with menu **Effects** > **SMPTE generator...**.

2. Set the parameters of the timecode signal to be generated:

   - **SMPTE Frames**: Select the frame rate

   - **SMPTE Start**-time in hours, minutes, seconds and frames, default is 00:00:00:00. if the timecode should correspond to the project time.

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

- **Sample rate** in 44.1 kHz, 48 kHz or 96 kHz, preset is the project sample rate.

3. Confirm this by clicking on **OK**. In the following dialog you can specify the name and location of the audio file with the timecode (by default `SMPTEGENERATOR.wav` in the project directory).

4. With **OK** you close the dialog, the audio file will be generated and loaded on the selected track.

5. Route the output of the track to a free hardware output connected to the SMPTE sync input of the analog device.
   You can also mark the track on which the SMPTE object is located as **Timecode track** (dialog Track Options or menu **Track** > **Track type** > **Time code**). Then the SMPTE signal of this track is always output via the output specified in the dialog **Synchronization** (↗599) at **SMPTE Audio Output**.

MIDI Machine Control (MMC)

 Sequoia supports the synchronization of external devices via MMC. In the project options (menu **Project properties** > **MMC Settings...**) you can configure MMC remote control.

Choose between the working modes **Receive MMC** and **Send MMC**:

- **Receive MMC (slave)**: If Sequoia shall work as slave, activate this option. Sequoia follows JOG shuttle operations, play marker position, track mute, track solo, track input monitoring, track record ready, play, stop, fast forward, rewind, and record commands from external devices.

- **Send MMC (master)**: Sequoia works as master. Whenever you execute commands such as Play Marker Position, Track Mute, Track Solo, Track Input Monitoring, Track Record Ready, Play, Stop, Fast Forward, Rewind, and Record, the external device follows. If you additionally set "Receive MMC", Sequoia receives the current playback position of the device and displays it as an additional blue play cursor in the timeline.

When playing a range in **Sequoia**, the MMC device stops when the end of the range is reached.

- **Transport console only controls external device**: If this option is active, the transport console no longer controls the playback and recording of **Sequoia**, but controls the external MMC device. Play, Stop, Fast Forward, Fast Backward and Rewind no longer have any direct effect on **Sequoia** playback. The **Record** button starts the recording in **Sequoia** synchronously to the playback of the MMC device.

- **Use space bar for remote control of external devices**: If this option is active, start and stop via space bar control the MMC device.

- **Input MIDI device**: Select here the MIDI device from which MMC is received.

- **Output MIDI device**: Select here the MIDI device to which MMC is sent.

- **MMC Input-ID**: Enter here the Input MMC ID from **Sequoia**.

- **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	

Backup Recording with Two Sequoia Systems

To ensure the safety of the recording during important recording sessions, you can work with parallel systems. Instead of using a hardware device to make an additional recording of the stereo sum or a complete multitrack recording, you can very easily make such backup recordings with a second Sequoia system, synchronizing the recording control with MMC in the process.

1. You need two DAW systems with Sequoia or Samplitude ProX and a MIDI connection between the two.

   ℹ️ *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.*

2. Select in the **MMC settings** the appropriate preset **Record Backup (Master)** or **Record Backup (Slave)** at the master and slave system.

3. Now create an identical project for master and slave, activate the necessary tracks and deactivate the display of the confirmation window at the end of the recording (**Program preferences** > **System options** > **Recording**) by selecting **No confirmation dialog** at **Extended options**. In addition, the recording option **Set cursor to record end position - Next recording starts at this point** should be activated for the slave system. This will

add each new take to the end of the project even when an insert punch recording is applied in the master.

4. Alternatively, parallel recording is also possible with the presets **Parallel Systems (Master)** and **Parallel Systems (Slave)**: In addition to the transport functions, Cursor Position, Mute, Solo, Input Monitoring and Record Ready are also transmitted to the other program. This creates an almost identical project on the slave system, with Punch In recordings also inserted in the right place. Only take names and markers are not taken into account. Thus, even after a failure of one of the systems, you get a complete project including take structure.

Calculating Loudness Values

This command calculates the loudness (↗559) of the master output for the entire project or selected range and displays the loudness values in an information window.

To calculate loudness, select an object or range and choose Menu **Edit** > **Loudness calculation (master output)…**. You can also find the function in the context menu of files in the file manager.

In wave files the loudness values are included in the metadata of the file (according to EBU 3344). For all other files they are only shown.

Click the button **Correct program loudness with master volume** to correct the master level so that the program loudness matches the loudness target value of the selected loudness standard (**Program Settings** > **Effects** > **Loudness**).

Waveform Generator

The waveform generator provides you with a generator of test signals.

1. You open it via the menu **Effects** > **Waveform Generator…**

2. Set the parameters of the test signal:

   - Mono or stereo, in a **resolution** of 16 bit or 32 bit float.

   - **Sampling rates** in the range between 22050 -384000 Hz.

   - **Waveform** Rectangle, Triangle, Sine, Cosine, Sawtooth (up), Sawtooth (down), White Noise, Pink Noise or Brownian Noise.

3. Set the **length** of the generated signal in samples, milliseconds or as SMPTE time (hh:mm:ss:ff).

4. Set the frequency (Hz) and level of the signal at **Waveform Parameters**.

5. Click **OK**. A save dialog will appear where you can specify a file name and location. The default is the project folder. With **OK** the test signal is generated and an object is created on the selected track at the cursor position.

WaveColor Audio Search

This function allows you to find identical or similar sounding sections within audio files. For this purpose, the analysis data of the WaveColor algorithm are used, which are also used for the color-graded display (↗716) of the sound properties of audio data.

1. Audio search works on wave projects. Double-click an audio object while holding down the Shift key to open it as a wave project.

2. Select the audio material from which you want to find similar audio material in the wave project and copy it to the clipboard with Ctrl + C.

3. Start the audio search with **Menu Object** > **WaveColor Audio Search...**

4. Set the **Sensitivity** of the detection with the fader in the dialog. Accordingly, more or less markers labeled **Match** are placed in the project at the points where the algorithm detects similar audio material.

5. After closing the search dialog, the found positions can be jumped to with a mouse click on the respective "match" markers.

Remix Agent - Tempo and Beat Detection

> ℹ️ *This is a deprecated program function. The tempo of an audio object can be determined faster and more accurately in the* **object editor** *(↗191) in the section* **Time/Pitch***!*

Remix Agent allows you to perform an analysis of the tempo of longer audio files, typically complete pieces of music from which a remix is to be created. There is an automatic detection of tempo and measure beginnings, which you can then edit manually. At the end you can split the object into individual objects of the length of a bar, match project tempo and object tempo, and write the tempo information to the audio file.

The audio material can have a length between 15 seconds and 10 minutes and should be rhythmic.

Start the Remix Agent via menu **Object** > **Remix Agent...** or from the context menu of objects.

Now follow the instructions in the dialog!

On the last page of the wizard, you decide what to do with the result of the beat analysis:

■ **Create remix objects**: This option cuts the song bar by bar into individual objects that you can then reuse in the virtual project.
Audio quantization: Objects are adjusted to a fixed project tempo, slight tempo fluctuations are corrected by object timestretching. With the option **Use resampling for small corrections** resampling is used instead for small corrections. Optionally, the loop mode can be activated for the objects.

> ⓘ **Note**: *If you subsequently change the tempo of the project, clearly audible pitch changes occur in the remix objects.*

Apply object tempo to arrangement: The project takes over the found BPM value.
With the option **Use crossfades** the remix objects will get default crossfades. If **Grouping** is active, the remix objects are grouped. Optionally, you can save the **tempo and bar information to the audio file** and assign a new **background color**to the objects.

■ **Adapt tempo**: You can either apply the tempo of the analyzed audio to the project tempo or apply the project tempo to the tempo of the analyzed audio.

- ● **Apply arrangement tempo to object tempo**: The object gets the tempo of the project. There are three options for this:

  — **Use timestretching**: With timestretching, the pitch remains constant, but sound changes occur due to timestretching.

  — **Use resampling**: Resampling changes the pitch but preserves the sound quality.

  > ⓘ **Note**: *If you subsequently change the tempo of the project, clearly audible pitch changes occur.*

  — **Use audio quantization**: Audio quantization applies tempo adjustments to the audio file.

  Save tempo & bar information: Also here you can additionally write the tempo and beat information into the audio file

- ● **Apply object tempo to arrangement tempo**: As with the option **Create remix objects** the project takes over the determined tempo.

— **Change global settings**: Use this option to set the project tempo to the determined value.

— **Adjust tempo map**: Tempo markers are placed at the beat boundaries, with **Use quarters** at each beat. With **Use bar pos. instead of tempo markers** grid position markers are set instead of tempo markers.

- **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 remain unchanged. Optionally, bar position markers can be set in the project at the bar beginnings (**Create bar markers within current range** or on each bar beat (**Create quarter markers within current range**).

Audio Quantize

Audio quantization is used to match audio material to a beat grid. It is particularly suitable for adjusting drum recordings and other percussive audio material with clearly distinct transients (attack sounds). The objects can be split at the individual drum beats and the objects with the individual beats can be moved to the respective grid positions. If this results in audible gaps, these can be closed by moving the object boundaries or by time-stretching.

Typical application include:

- Correction of minor errors in a drum recording.

- Separation of individual drum beats for targeted application of object effects and creative editing of drum beats.

- Subsequent tempo changes can be made without time-stretching only by adjusting the object positions.

The workflow of audio quantizing consists of a typical sequence of individual steps. In the window **Audio quantization** all steps are combined in one dialog. (Menu **Object** > **Quantize** > **Audio Quantization Wizard...**. The individual steps and some additional functions can also be reached via the menu **Object** > **Quantize** > **Extended Audio Quantize**.

1 **Detect transients**: All level peaks of the selected objects are marked with beat markers ("AQ").

> ℹ️ *A beat marker is a special audio marker that is stored directly in the audio file. To make these markers visible, in the **View options** of the program preferences (keyboard shortcut: **Shift** + **Tab**) under **Objects** activate the option **transients (AQ)**. For detailed information on audio markers, see the section* Audio markers *(↗116)*

In the menu you can choose between an algorithm optimized for percussive audio and a universal one. The transient detection is started as soon as you move the slider **Sensitivity** and the found transients are updated accordingly. The higher the sensitivity, the more markers are set. By using **Recalculate** and **Delete**, you can recalculate or delete an already created audio material analysis.

2 **Consolidate transients**: If you edit objects on different tracks that contain different instruments, the positions of the transients will differ slightly, also due to delay differences caused by different microphone distances. To avoid too many cuts and very short objects when splitting the objects later, this command moves beat markers that are on different tracks together within a certain time range (**time window**) (default: 20 ms) to the same position (the foremost one).

3 **Split at Transients**: This command splits all selected objects across all tracks at the beat marker positions of all objects. After that, each beat should be present on all tracks as a single object, this is a prerequisite for the subsequent quantization of these objects.

> ℹ️ *You don't necessarily have to use the transients of all tracks for the cut: For example, if you want to align all drum tracks to the bass drum and snare tracks, first select only the bass drum and snare track objects and perform transient detection. Then select the objects of all drum tracks and execute the **Split at Transients** command. This splits all objects on each bass drum beat and on each snare beat, but the bass drum and snare objects are not additionally (and unnecessarily) split on the transients of the other drum tracks.*

4 **Quantize Object Positions (AQ)/Soft AQ**: These commands are used to perform the actual audio quantization. All selected objects will be moved to the corresponding grid positions according to the **Quantization settings**. The quantization settings correspond mostly to those of the MIDI quantization, but unlike MIDI events, no quantization of the length is provided. Also the **presets**, which can be loaded and saved at the top of the dialog, correspond to those used in the MIDI quantize dialog. **Default** resets the options to the default settings: **Q threshold**=0, **Q window**=100, **Swing**=50 and **Offset**=0.

> ℹ️ *For more information about the* Quantization settings *(↗449) please refer to the MIDI section.*

Soft AQ: While with **Quantize Object Positions (AQ)** the objects are moved exactly to the grid positions, **Soft AQ** moves them only a bit toward the grid position. The slider **Soft Q** determines how far: if it is set to 50, the new object position is exactly halfway between the starting position and the targeted grid position.

❺ Remove object gaps: After an object quantization has been performed, gaps may occur between objects, which can cause audible dropouts in the audio signal for certain signals with a long decay phase. The **Remove Object Gaps** command lets you choose between two ways of closing the gaps between audio objects:

- ■ **Insert audio from next slice's front**: This moves the object start position of the following object to the left until the gap is closed.

- ■ **Timestretching**: The object located in front of the gap is extended by timestretching. You select the respective method in the Audio Quantization Wizard.

The objects that now directly follow each other are crossfaded, according to the set **Overlap(Crossfade) length**.

❻ The **ONE CLICK BUTTON** automatically executes the **Detect transients**, **Consolidate transients**, **Split at transients**, **Quantize slice positions (AQ)**, and **Remove Object Gaps** commands in one go.

The dialog contains some more controls that support the work with audio quantization:

❼ Reset quantization: All objects will be reset to their original positions.

       With the two buttons you can undo or redo the last quantization performed.

❽ Create Groove: This command creates a groove template from the AQ markers of the selected objects in the selected area, which you can use for audio and MIDI quantization. Make sure that the selected range is full bars long (2 bars is best).

> 🛈 *For detailed information on groove templates, see* MIDI in Sequoia > MIDI Editor > Quantize and Grid > Groove Templates *(⁊452)*

> 💡 *Tips:*
>
> - ■ *Before using audio quantization, make sure that the transient display is enabled in the program preferences: In the **View options** (keyboard shortcut: **Shift** + **Tab**) at **Objects**, the option **Transients (AQ)** has to be on.*
>
> - ■ *If you are using a longer recording, first split the objects once so that you have a short representative section of about 8-16 bars as a separate object. The analysis of the transients for a complete song can take a very long time and when adjusting the sensitivity it is constantly repeated. When you have found the appropriate sensitivity for the smaller part, you can select the rest of the objects and enter the found value numerically in the **Sensitivity** field.*

Generate MIDI Triggers from Transients

This function allows you to copy the detected transients of an object to a separate MIDI object.

Select an analyzed object with AQ Markers and choose Menu **Object** > **Quantize** > **Extended Audio Quantize** > **Create MIDI Triggers from Transients**. A new track is created below the object. In it a MIDI object is created which

contains the found transients as MIDI events with corresponding velocity. You can now load a VST instrument in the track to duplicate or replace the audio track with additional sounds from drum synthesizers.
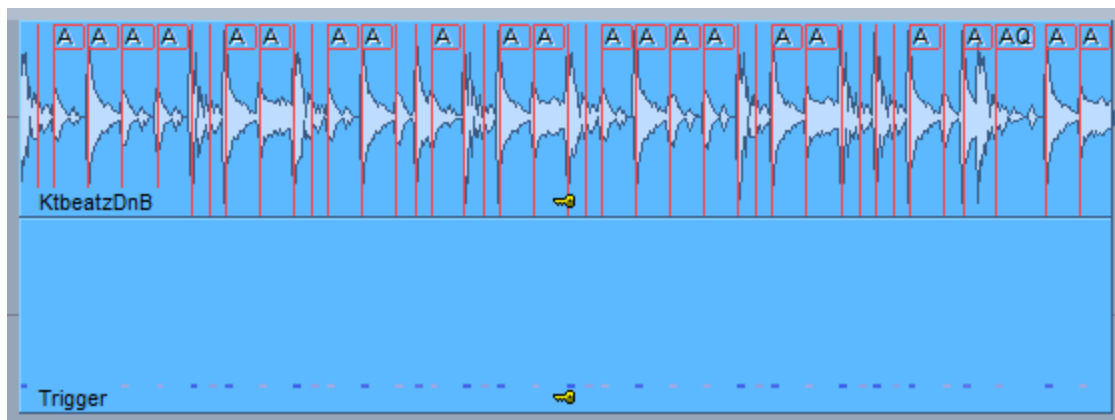
Video

Although **Sequoia** is not intended for video editing, you can also load video files into **Sequoia** to create, edit, or surround mix a movie soundtrack. These are then played along with the project and displayed in a separate video window.

Loading Video Files

This can be done by drag&drop from Windows Explorer or File Manager or via menu **File** > **Import** > **Load video file...**. Supported file formats are MPEG2 (mpg, mpeg, m2v, mp2, vob...), MPEG4 (mp4, mov), AVI, DV, MXV (MAGIX own full frame format) as well as image files in JPG or BMP format.

For the video file, a video object is created in a separate video track above the first audio track. The video object shows its content as thumbnails.

Optionally, the original video sound can also be imported and placed on its own audio track below. This converts the audio data into wave files.

Video objects can be moved and cut just like the other objects. However, it is not possible to fade in or out the objects or to use crossfades.

> ℹ️ *Note: Since* **Sequoia** *very limited capabilities for exporting video files, you typically end up with an audio file at the end of your edits, which you then combine with the video using a separate video editing program. In* **Sequoia** *the loaded video serves as a time reference and should therefore not be moved or cut at all if possible.*

Video objects do not have an object editor, double-clicking on the video object opens the video window instead. The video object's context menu displays the video's resolution and frame rate at the very top. This frame rate should match the frame rate (↗599) of the project. Only then the grid mode and the display of the project time with frames (SMPTE) in the grid bar, time display and video preview window will work correctly.

Video window

The video window is used to display the video. Open it by double-clicking the video object or from the menu **View**. It can be displayed floating or docked like the other windows.

Right-click in the video window to get some display options:

- With **Original aspect ratio** the window uses the original aspect ratio of the video, with **4:3** and **16:9** you can force a different aspect ratio.

- Select **Full Screen** for a full screen display.

- Use **Double**, **Original**, **Half**, or **Quarter video size** to adjust the video window to the appropriate size of the video. The function is available only when the video window is not docked.

Video Settings

Using the context menu of the video window and video object or menu **File** > **Properties of the project** you can open the video settings, for further video playback options.
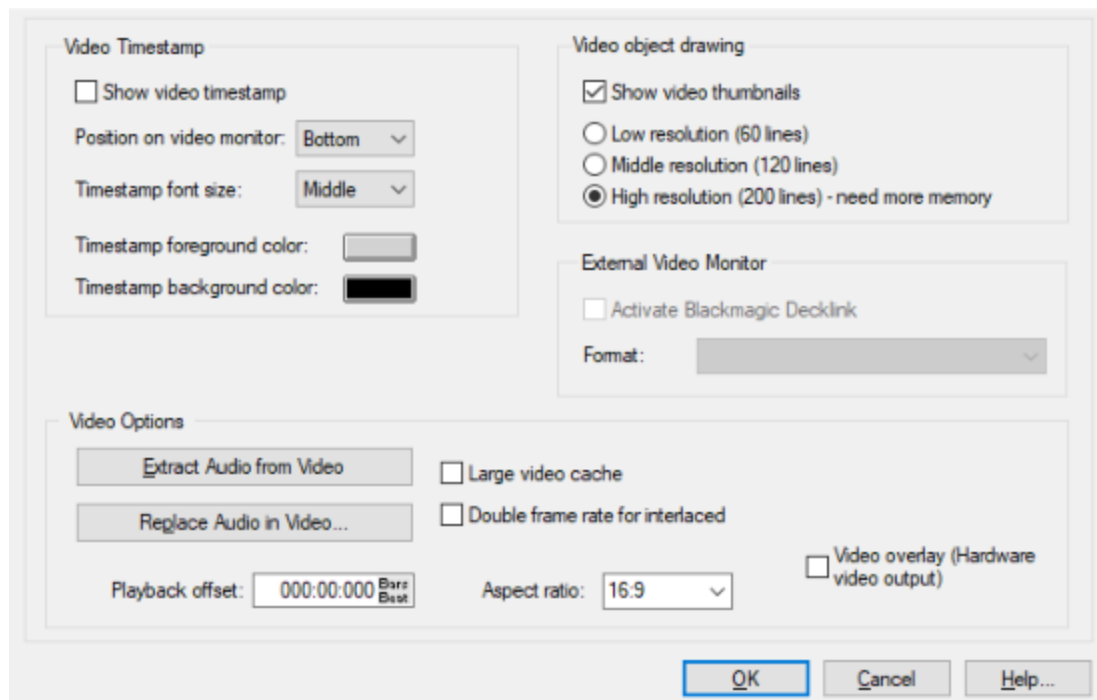
■ **Show video timestamp**: This option shows the timestamp display in the video window. You can display it in three sizes either at the top, in the center or at the bottom of the project window.

■ **Video object drawing**: The option **Show video thumbnails** displays thumbnails in the video track in different resolutions. In drawing mode 2 of the View options (↗716) (Tab key), only the first and last frames are displayed.

■ **External Video Monitor**: Enable **Blackmagic Decklink** allows video output to an external monitor via appropriate video hardware.

■ **Get Audio from Video**: This button allows you to extract audio from a video file and insert wave data into the current project.

■ **Replace Audio in Video**: This button executes the **Export video sound** function, which lets you replace an existing audio track in an video file with the project's audio.

■ **Playback offset**: Set a positive or negative playback offset in this field.

■ **Large video cache**: The video caching cache is increased, which increases the memory footprint, but also makes the program more responsive.

■ **Double frame rate for interlaced**: Here you can double the frame rate when playing back interlaced video to display the videos flicker-free.

■ **Aspect ratio**: Select an aspect ratio of either 4:3 or 16:9.

- **Current Video Frames Window**: An additional video window is opened to display the start and end frame of a selected range or object for range and object selections.

- **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). This speeds up the video output and should only be deactivated in case of problems.

Exporting Video Sound

Select Menu **File** > **Export video** > **Export Video Sound...** to replace to create a new video file from a video and the sound of the project.

Select the original video file at **video source**. At **audio length**, the length of the audio track in the video and the length of the project are specified. If these lengths differ, a warning is issued. On export, the longer component is then truncated.

Under Audio Format the format of the original video sound is displayed. The export of the project is done in this format.

Target file: Click the folder icon to set the new file name.

Pressing **OK** exports the master output of the project and inserts it into a new video file.

Impulse Response Extraction

Sequoia provides an assistant for extracting impulse responses from natural rooms or effect devices for use in the room simulator (↗292).

Method

The chirp method is used to generate the impulse responses, which works as follows:

1. A chirp is generated. A chirp is a sinusoidal signal that is varied over a wide frequency range.

2. This chirp is then played through loudspeakers in the room to be detected, and the room's "response" is recorded with microphones. Effects units can also be "sampled" in the same way, by playing the chirp through the effects unit and recording it.

3. The impulse response is then calculated from the original and the processed or recorded chirp.

Steps 1 and 3 are carried out in the dialog **Impulse response extraction**, which you open via menu **Object** > **Impulse response extraction...**.
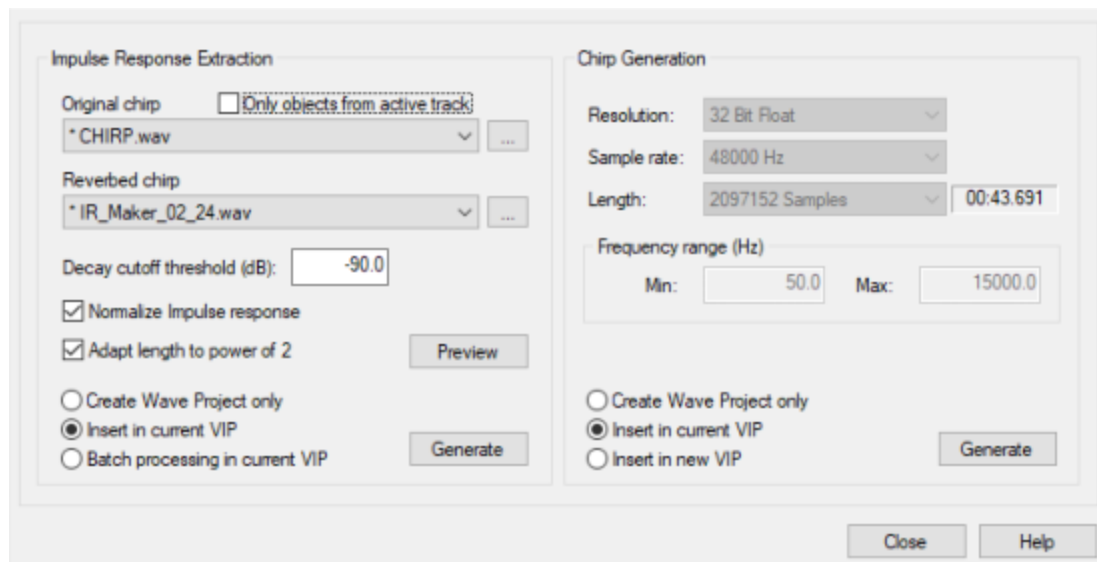
Chirp Generation

On the right-hand side of the dialog you will find 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 here. Note that the sample rates of the played chirp, the reverberated chirp, and the virtual project must match.

  > ℹ️ *Note on sample rate: If you want to work with 96 kHz or 192 kHz, keep in mind that when capturing natural impulse responses, the loudspeaker must be capable of reproducing very high frequencies in order to actually capture the extremely low ultrasonic components of the impulse response. The microphones used also define limits here, so that in practice it is usually sufficient to work with 44.1 kHz or 48 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 constant noises like hissing, traffic noise, or non-linear distortions in the result is reduced. On the other hand, the CPU load increases when processing the impulse response.

  > ℹ️ *Tip: The chirp should be at least as long as the duration of the reverberation.*

- **Frequency range (Hz)**: Specify the start and end frequency of the chirp here. The frequency range depends on the frequency response of the loudspeaker. You should therefore avoid frequencies that are too low or too high for the speaker to reproduce.

The following options determine how the generated chirp will be used:

- **Create wave project only**: The chirp is written to a wave project only.

- **Insert into current VIP**: The wave project is inserted into the current virtual project.

- **Insert into new VIP**: The wave project is inserted into a new virtual project. There you can immediately play the chirp and simultaneously re-record the reverberated room in the same virtual project.

Click on **Generate** to generate the chirp. After generating the chirp, the dialog can be closed again.

Reverberation of the Generated Chirp

Natural Rooms

To capture the impulse response, the chirp signal is reproduced monophonically in the room using a loudspeaker or loudspeaker combination. The loudspeaker(s) should be placed in the room where the acoustic sound sources such as instruments or singers are located - in an opera house, for example, on the stage or in the orchestra pit.

Individual loudspeakers have a different directional behavior than natural sound sources, they emit less sound energy to the rear. Therefore, it is better to use speaker combinations to radiate the chirp. A proven loudspeaker combination is to have one loudspeaker directed to the front - towards the audience - and a second loudspeaker directed to the back.

> ℹ️ *Note: For the best possible impulse response, always record the reverberation until it has completely decayed. Also avoid any impulsive noise during decay.*

Digital Reverb Devices

When processing the chirp signal with a reverb unit, you should always set the un-reverberated, dry portion ("Dry") to zero, so that the original chirp is not mixed in.

> ℹ️ *Note: With the impulse response method you can only simulate linear and time-invariant systems. 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 that feature parameters that are modulated by LFOs. Distortion or compressors have a non-linear behavior and can also not be reproduced. The reverb programs of multi-effects units often use other effects besides the actual reverb, for example chorus effects to cover resonances. Also in these cases, an exact reproduction of the reverb is not possible. The strength of the impulse response method, on the other hand, is the reproduction of natural spaces in high quality.*

> 💡 *Use the **External FX Plug-in** (↗225) for convenient integration of the effects unit!*

Simultaneous Playback and Recording of the Chirp in Sequoia

1. Load the chirp into a track of a project.
2. Activate recording in another track and route corresponding input and output devices on these tracks.
3. Disable monitoring (e.g. right-click on **MON** in the Transport Console and select **No Audio Monitoring (Peakmeter Only)**), as other monitoring settings may cause acoustic feedback.
4. If not already done, open the window **Visualization** in the mode **Peakmeter**, select the recording track and in the menu of the visualization window **Track Visualization (current track)**.
5. 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.
6. After the optimal levels have been set, record the reverberated chirp.
7. Play the reverberated chirp and check the recording in the visualization window in the spectrogram view for nonlinear distortion, spurious pulses, or aliasing.

Calculation of the Impulse Response from Original and Reverberated Chirp

Now reopen the dialog **Impulse response extraction** in the menu **Object**. The dialog elements for calculating the impulse response are featured on the left hand side of the dialog.

- **Original Chirp**: Select the audio file or object that contains the original unprocessed chirp here. Via the **...** button you reach a file selection dialog.

    > ℹ *If the audio file of the original chirp is not available, you can select the **[Manual]** option and set the parameters used to generate the chirp on the right side of the dialog. Note that if the set parameters differ from those of the original chirp used, usable results will no longer be obtained.*

- **Reverberated Chirp**: Here you select the audio file or object that contains the reverberated chirp. In the list field all objects present in the project are listed, with **Only objects from active track** only objects of the selected track are listed. The letter "T" (for track) and the track numbers are placed in front of the object names in the list.

- **Decay cutoff threshold**: Here you can specify the threshold of the signal value in dB, below which the impulse response will be cut off.

- **Adjust length to power of two**: This option fades the impulse response so that the length in samples is slightly less than a power of two. This is used to optimize the performance when using the impulse response in the room simulator.

- **Create wave project only**: The impulse response is saved as an audio file only.

- **Insert in current VIP**: An object with the impulse response in the current project is additionally created.
- **Batch processing in current VIP**: This allows to calculate several impulse responses one after the other. The precondition for this is that the same original chirp was always used. Arrange the reverberated chirps under the object of the original chirp one track at a time. The impulse responses are calculated from all objects one after the other. The impulse responses are inserted behind the objects with the reverberated chirps, respectively.

  > ℹ️ *Individual objects can be excluded from batch processing via the **M** button of the tracks.*

- **Preview**: Calculates and plays the impulse response.
- **Generate**: Calculates and saves the impulse response.

Chirp Residuals in the Impulse Response

It may happen that the calculated impulse responses contain residues of the chirp signal used. Typical causes for this are explained below and remedies are described.

Aliasing

When using older digital reverb devices that work with lower internal sampling rates, aliasing may occur. In the spectrogram representation of the recorded chirp, the chirp mirrored in the frequency domain can be easily recognized.

To remedy this, experiment with different sample rates and see if the aliasing component is reduced when you use digital inputs and outputs instead of analog.

Otherwise, only use of a very long chirp will help. The interfering signal then extends in the impulse response over a narrow frequency range (e.g. 12000 Hz - 12100 Hz) and can be removed there with the FFT filter (↗274).

Faulty Impulses during Recording, Reverberation or Signal Transmission

Impulses or impulse-like acoustic sounds also produce chirp residue in the calculated impulse response. The process reacts very sensitively to these noises. Even pulses with a level of less than -50 dB have a disturbing effect.

Causes of faulty pulses:

- Clicks when playing the chirp or recording the reverberated chirp, e.g. when the chirp is not played from the beginning.
- Discontinuity due to premature termination of the recording, e.g. due to an incompletely recorded reverb tail after the end of the chirp.

- Impulsive sounds during recording of natural spaces such as falling objects, footsteps, chairs moving, etc.

- Faulty cables, connectors or similar.

To remedy this, avoid impulsive noises during playback and recording. To determine the cause of the interfering pulses, listen carefully to the reverberated chirp. If the disturbing pulses sound like the recorded room or effect device, the faulty pulses have occurred during playback. If the noise sounds "dry", the faulty impulses occurred during recording. In the spectrogram display of the visualization, these impulses may be recognized as vertical lines.

Different Characteristics of the DA-Converter for the Playback of the Chirp and the AD-Converter for the Recording of the 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 pop noise at the beginning of the impulse response sounds "chirped" in this case.

To remedy this, always use the same device for recording and playback.

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 does not normally need to be longer than one minute. This does not apply if chirp residues appear in the result. Details can be found in the corresponding section above. In practice, a length of 47 sec has proven effective in rooms.

- A bandwidth of 50-15000 Hz is normally sufficient.

- 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 multichannel impulse responses for surround room simulation, true multichannel recording is not absolutely necessary. The impulse responses for the different positions can also be recorded one after the other.

To synchronize the recordings, record the individual positions in stereo, using a reference microphone for the second stereo channel that always remains in the same 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. Then you can obtain the surround impulse responses by exporting all the stereo recordings together, muting the channel of the reference microphone in each track

Natural Rooms

■ Slight analog overloads of the speakers or amplifiers used to reproduce the chirp are not problematic, at least when longer chirps are used.

■ Avoid digital distortion in the AD/DA converter.

■ The louder the chirps are played, the better the signal-to-noise ratio will be in the measurement.

■ The longer the chirps are, the better constant noises such as traffic or hissing will be removed from the resulting impulse response. You can also use longer chirps (> 1min) to extract impulse responses with very good signal-to-noise ratio at lower playback volume.

■ Be sure to avoid impulsive noise during recording.

■ 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.

Long chirps

Computing impulse responses from longer chirps is very computationally expensive and requires a lot of memory. 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 bar behaves non-linearly during the calculation, it progresses slower and slower as the number of operations increases towards the end of the calculation.

If you are planning a larger number of calculations, it is recommended to use batch processing in the virtual project.

Auto Ducking

Auto Ducking is a dialog-based tool for creating volume curves for the automated fade-in and fade-out of background music to spoken content. This modifies the volume automation curve of one or more destination tracks based on the volume of the audio material of one or more source tracks. To do this, proceed as follows:

1. Create a montage from speech parts and music.

2. Place the audio on different tracks, separated by speech and music.

3. Open the Auto Ducking dialog (Menu **Edit** > **Auto Ducking...**).

4. Specify which tracks (**Duck**) should be affected by the volume of other tracks (**Source**). In this example, the background music and video are affected by two voice tracks.

5. Click **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.
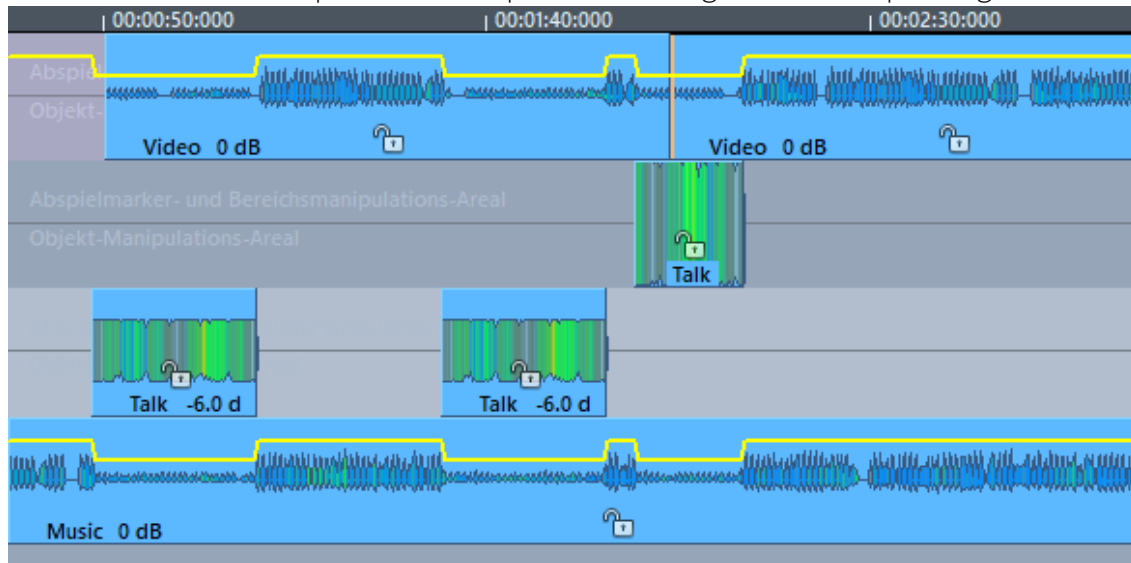
The following **Options** are available:

Strength: Value in dB by which the tracks are lowered. Negative values allow increases.

Fade-in length: Duration of the fade.

Fade-in offset: Time offset of the fade-in. With 0%, the volume change starts at the beginning of the source material. With 100%, the fade is performed completely before the source material.

Independent Fade-out If active, the length and offset can be set for fade ins/fade outs independently.

Fade-out length Length of the fade at the end of a volume adjustment.

Fade-out-Offset: Offset value for fade outs. With 0%, the volume change starts in the source material and ends at the end of the source material. At 100% the fade will happen entirely after the source material.

Threshold: Value in dB that the level of the source audio must exceed to trigger ducking.

Min.Interval: Minimum time that the source material may fall below the threshold without stopping ducking in this section. This is useful for example to straddle short pauses in speech.

Only marked ranges: Auto ducking will only be used in the selected range.

Absolute (not relative): When applying ducking, the volume curve is recalculated and existing curve points are deleted. If this option is not active, a relative change takes place, i.e. already existing curve points are adjusted by the value of the change. Note that when ducking is applied multiple times in relative mode, the decreases add up.

Duck on object borders (audio independent): Ducking is done independently of the audio material and is performed on the object edges of the source tracks.

Transferring Edits

If you have made a multitrack recording with a large number of tracks and want to edit them, you can, in order to work more performantly, make the editing with only one track and then apply these edits to the other tracks using the **Transfer Edits** function.

The function relies on the fact that, on the one hand, when setting up a project for recording, the recorded files are named uniformly according to a specific naming scheme (either unique track names or track numbers as a part of the name) and, on the other hand, that the objects of a multitrack recording are always arranged exactly over each other on the tracks.

Here's how you do it:

1. Create your recording project.
2. Right-click in a track name field at the Track head to open the **Track options** dialog.
3. In the lower part of the **recording** dialog, click **All tracks...** and select a naming scheme for the recording files. Select a scheme that includes either the track number or the track name. (Not both, or it will not work later!) If you use the track name, the labels should be unique and the track names should not be changed later.
4. Make your recording.
5. Select all objects and press the keyboard shortcut **Ctrl+U** to ungroup them.
6. Select a reference track from the recording that you want to use for the edit, and delete the objects on all other tracks.
7. Carry out your edits in the remaining track. All forms of virtual editing are possible, including transitions and object effects.
8. Once the editing is complete, you restore the other tracks, and transfer your edits from the reference track in the process. To do this, open the **Transfer Edits...** dialog from the **Edit** menu.
   Set the options in the dialog:

   - **Source track**: This is the finished edited reference track whose edits are to be transferred.

   - **Destination tracks**: These are the tracks for which audio material is available and which are to be cut according to the reference track. e.g. track 2...4.

   - **Create missing tracks**: If not already existing, new tracks will be created for the desired number of tracks.

   - **Ignore locked tracks**: When transferring, locked tracks are omitted and no objects are created on those tracks.

   - **Vertical grouping**: The objects lying on top of each other are grouped again, so the default state after a multi-track recording is restored.

- **Marked range only**: This limits the edit transfer to the selected part of the project.
- **Filename**: Select here the naming scheme according to which the files of the project were named during recording:

(file name)(track number)	You can specify a file name in the field below, to which the number of the corresponding track is then automatically added.
...(track number)...	The files differ only in the track number. Select this option if a scheme with track number was selected when recording.
...(track name)...	Select this option if a scheme including the track name was selected when recording.
Track record file name	The name for the record file is the one that was set up in the corresponding track in the track settings.

In the file list, the Source Track file is displayed at the top, marked with an S and its track number. The other tracks are listed underneath, each with the identification D for the destination, the track number and the associated file. If the file does not exist, the file will be marked with **(NOT FOUND)**.
If more tracks are affected than can be viewed, you can review all tracks by clicking the **Test** button. If an error occurs, the first missing file will be shown. You should then check your details and the existing files and their names in the project folder.
Click **OK** to apply all edits and all properties of the reference objects, such as object volume, resampling, and effects, to the other tracks.

Auto-Conforming

Auto-conforming lets you apply edits from a video edit to a multitrack recording made in the same shooting situation.

Suppose a concert performance was simultaneously recorded with Sequoia and filmed with a video camera. After that, the video was edited and other material was inserted.
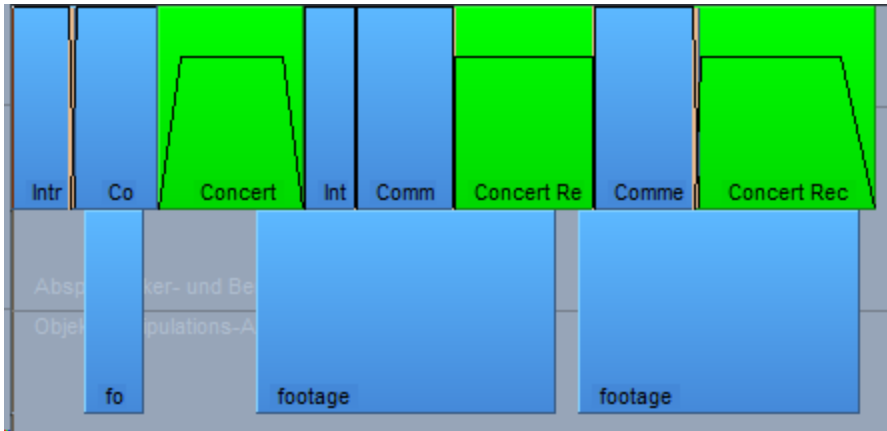
Example of a reference EDL project (created here in Sequoia, but the depiction is similar to any video editing program)

If you now want to replace the original sound of the camera with that of the multitrack recording in the video, you would have to select the corresponding sections individually from the recording, export them and load them into the video project at the right places.

Auto-conforming helps by analyzing a cut list of the video in EDL format and transferring the contained cuts that affect this audio recording 1:1 to all tracks of the audio project.
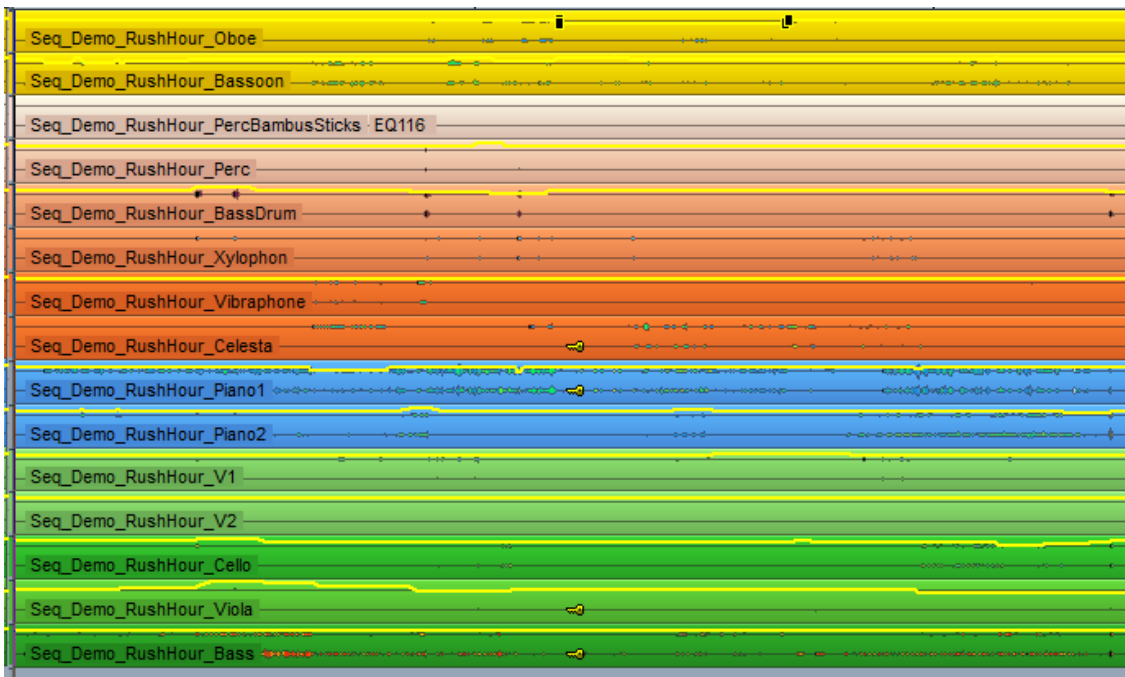
Project of the corresponding live recording

You will receive a modified version of the audio recording project, containing only the sections used in the video at the correct positions.
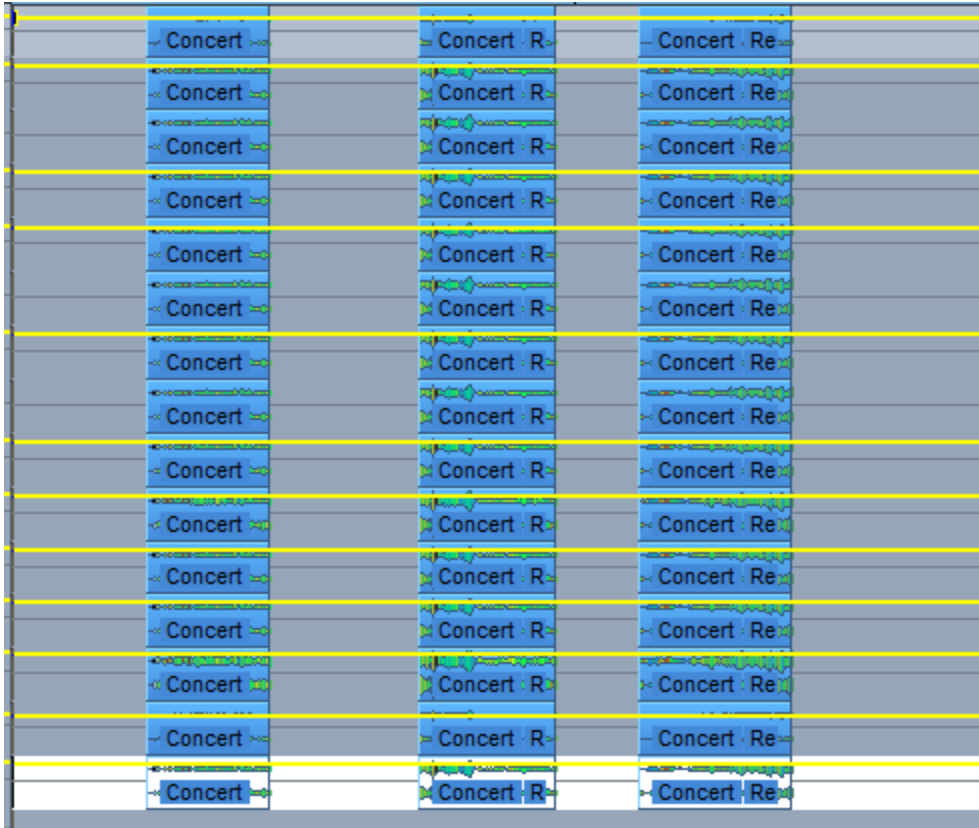
Result of the auto-conforming

You can now export this project and, after removing the original sound of the video recordings in the video project, use it as a new sound track.
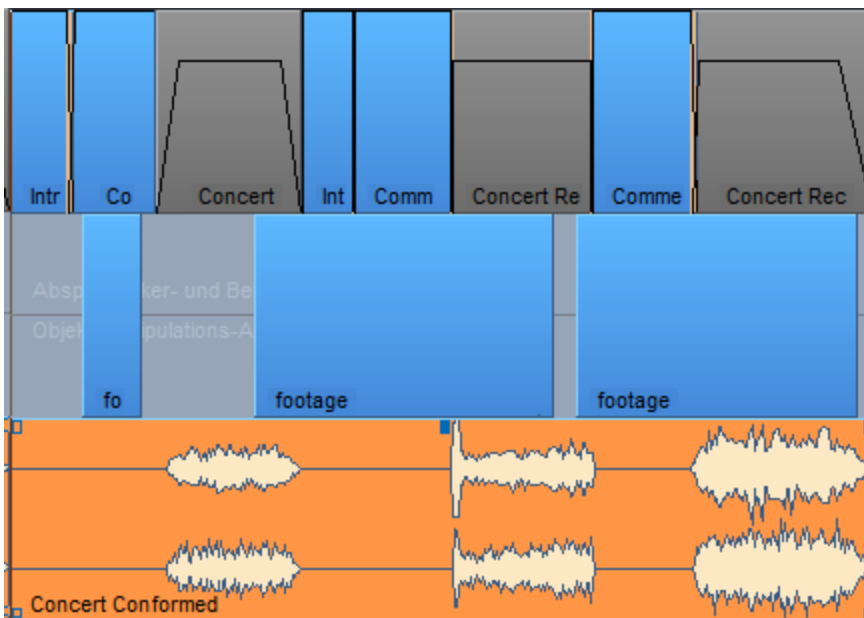
⚠ *Attention: In order for the auto-conforming to work, the video recording and the audio recording must be exactly synchronized in time.*

Here's how you do it:

1. Open the project with the multitrack audio recording and select **Auto-Conforming…** from the **Edit** menu. The dialog Auto-Conforming will then be opened.
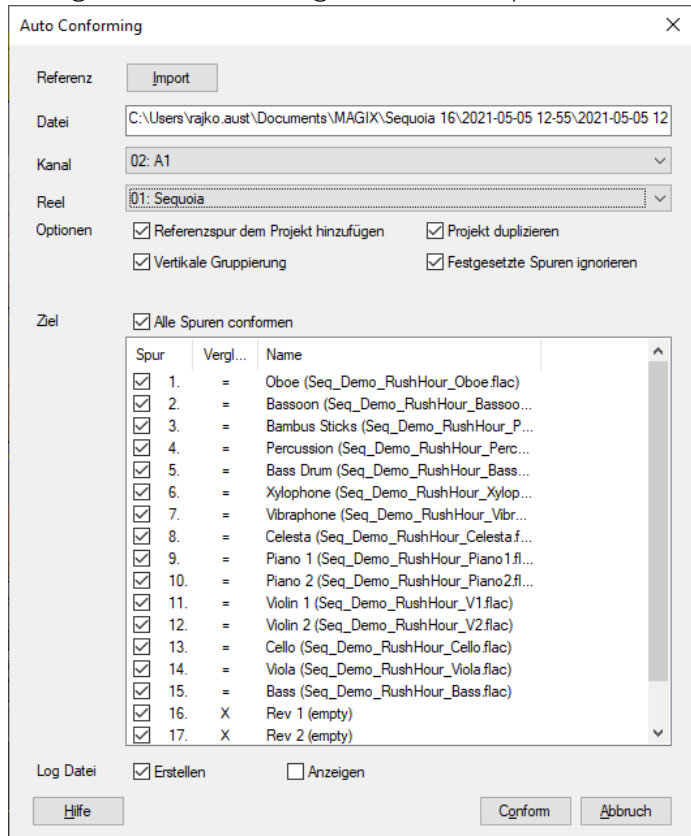
2. With **Import** you load the reference file, i.e. the project of the edited video, in EDL (CMX3600), AAF or Sequoia VIP format.
   You then have the option to select the frame rate if it differs from the standard frame rate (30fps).

3. At **channel**, select the track that contains the reel(s) of video and original audio that you want to replace.

4. Go to **Reel** and select the video file that contains the original sound.

   ℹ *In EDL format, the reel name can only contain 8 characters. If the channel (track) still contains material from other reels, ensure that the files are named uniquely before editing.*

It may also happen that the video footage used is from different reels, if several consecutive takes of the recording were used. Then select the option **All reels with different timecodes**.

⚠ *The reels of the different takes must be on one track and the timecodes of the different takes must not overlap!*

5. At **Target** the target tracks are now listed and the objects on the tracks are compared with the length of the reel. Conforming only makes sense for tracks where the length is the same (indicated by a = sign). Use the checkboxes to deactivate individual tracks in order to exclude them from conforming. The **Conform all tracks** option above selects all tracks.

6. With **Conform** the editing is executed and the Auto-Conforming dialog is closed.

Options

Add reference track to the project: For control purposes, you can optionally add the reference track from the EDL as a track to your project. If the corresponding reels (audio or video files) are not available locally, the object references in them will go nowhere, but you can still see the position of the objects.

Duplicate project: By default, a copy of the project is created for the conformed version. If you deactivate this option, the conforming is executed directly in the active project.

Vertical grouping: By default, the objects lying on top of each other in the individual cuts are grouped together; you can deactivate this here.

Ignore locked tracks: By default, locked tracks are excluded from conforming, you can deactivate this here.

Log file

If the option **Create log file** is active, a log file is generated during conforming and stored in the project folder of the target project. The settings of the dialog (reference file, channel, reels, content of the target display) are stored in this for later review. If you also activate **Open log file**, the log file is opened immediately in the text editor after conforming.

Macro Functionality

With macros you have the possibility to combine frequently needed consecutive commands as a macro and execute them together.

> ■ *With the control commands used in the macros (referred to as "macro commands" in the following), Sequoia can also be operated via text input, e.g. search for specific object or track names, filter the displayed tracks by name, zoom, scroll and much more!*
>
> ■ *If you are using an EUCON controller, you can assign softkeys to the macros in the EUControl app. In order for the macros to be available in the EUcontrol app, it must first be restarted.*

Recording Macros

To record macros, click the magnifying glass icon on the command search bar (↗52) and select **Record Macro** from the menu or select Menu **File** > **Program Preferences** > **Record Macro**.

For this function, you can also add a dedicated icon to the toolbar (↗54)
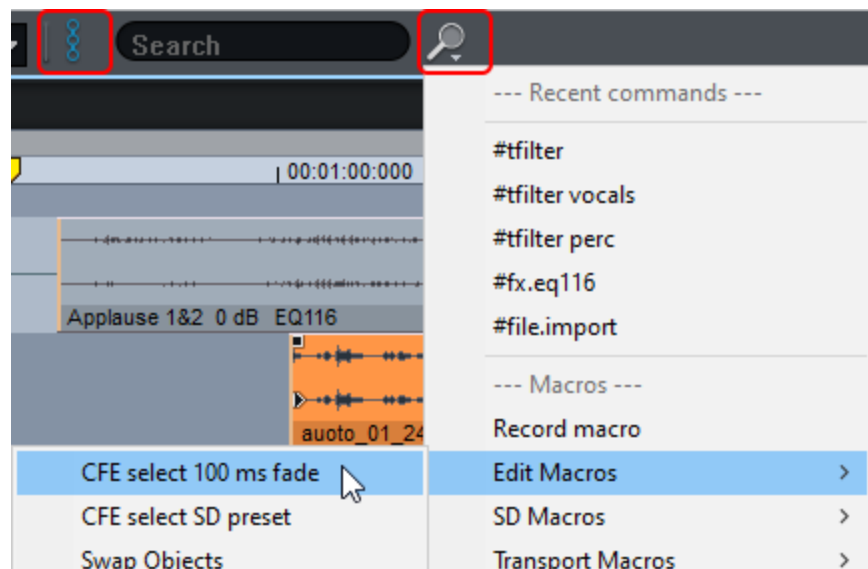
1. Execute the commands consecutively that you want to record together as a macro.

   ⓘ **Example:** *Place the play cursor 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.*

2. If you have executed your command sequence, chose **Record macro** again.

3. Now give the newly created macro a name and save it.

   ⓘ **Note**: *Only simple command sequences without settings dialogs can be recorded in the way described. 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 commands available can be found in the file* `MacroCommand.list` *in the program folder.*

Executing Macros

Macros can be executed using the following methods:

- via the context menu of the command search bar

```
--- Macros ---
Record macro
Edit Macros          >
SD Macros            >
Transport Macros     >
split_duplicate
```
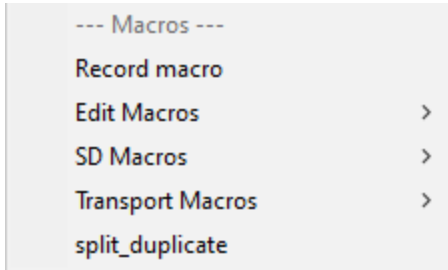
- directly in the command search bar with the command #run

    `#run MacroName`

    (in our example: `#run split_duplicate`)

- In the Macro menu of keyboard settings/menu settings you can also assign a keyboard shortcut (↗709) to any macro.

- In the toolbars (↗55) you can assign your own buttons to the macros.

- When playing over specially named project markers (see below).

Creating and Editing Macro Files in a Text Editor
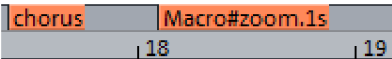
Macro files are simple text files. To create a macro file, open a text editor, copy and paste the desired command sequence from the `MacroCommand.list` file and save the file in the .macro format in the c:\ProgramData\MAGIX\Sequoia 17 > "Macros" folder.

ℹ *The "AppData" folder is not visible unless the the option **Hidden Items** is selected on the "View" tab of the Windows Explorer.*

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

Executing Macros/Macro Commands at Project Markers

To execute macros at a specific playback position, you can use project markers.

1. Place the play cursor at the desired position in the project and select Menu **Play/Rec** > **Marker** > **Marker with name...**.

2. Name the macro marker `Macro#MacroCommand`, e.g. `Macro#zoom.1s`.

   ```
   chorus      Macro#zoom.1s
           18              19
   ```

3. This causes the corresponding macro command to be executed each time the play cursor reaches the marker during playback. In this example: The zoom factor, i.e. the screen section displayed, changes to one second.

4. Entire macro files with a sequence of macro commands can be executed instead of individual macro commands. To do this, name the marker in the form `Macro#run filename.macro`.

Executing Macro commands in the 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, without arguments)
- #mfind MarkerName
- #ofind ObjectName
- #rfind RangeName
- #timeedit.startpos ms -1000

> ℹ️ *A complete list with short descriptions of the commands available can be found in the file* `MacroCommand.list` *in the program folder.*

Aliases for Macro Commands

Besides the macros in the folder c:\ProgramData\Magix\Sequoia 17 there is also a special file `alias.list`.

In this file you can assign your own names for certain macro 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

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

Multi-user Operation/Administration

If a **Sequoia** workstation is to be used by several users, the administration function can be used to ensure that each user can work with their own user-specific settings.

Each user has their own independent set of preferences that are loaded when they log in:

- Visible and hidden menu entries
- Keyboard shortcut
- Color settings, skins, default folder paths

- Toolbars and icons
- VIP display modes
- Lock options, crossfade editor properties, undo settings, dithering, etc.
- Crossfade presets

> ℹ️ **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. However this means that they are no longer displayed in the preset selection lists.*

The user settings can also be managed via a network drive and can thus be made available for several different workstations.

Set up multi-user operation

To set up Sequoia for multi-user operation, chose Menu **File** > **Program Settings** > **Administration** or in the System Options at **Load and Save Settings** the button **User Administration**.

If the user administration does not yet contain any entries, you must first create an `Admin` user and assign a password. Next, the dialog for user management will open.

In this dialog you can create or delete users and change the password of a user. Sequoia shows when each user was last logged in, whether there are access restrictions to paths or whether the settings can be permanently changed by the user himself or not.

Create new user

To set up a new user, click the button **New user** and enter the username and a password (optional). If you want to adopt the settings of an existing user, first select this user and then click on **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 properties

User access rights You can assign access rights for the respective user. You can specify some of drives and folders for which write accesses (recording, export...) are allowed. All other folders can then not be written to. Conversely, you can lock individual folders or drives against write access.

Fixed settings: The button **Fixed settings** allows you to lock the program settings for a user. For the affected user, this means that the settings can be changed during the program runtime, but after each restart of the program, the settings that were specified when the user was created will apply again.
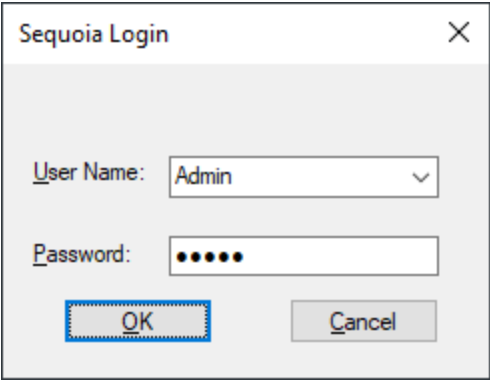
Change user password

If multi-user operation is set up, a logged-in user can change his password via **Menu File** > **Program Settings** > **Change User Password**.

ℹ *The administrator can change the passwords of any local user via the administration function; in multi-user operation via a network drive, he must use the **AdminCreator tool** for this purpose (see below).*

Activate/deactivate multi-user mode

Once the administration dialog has been opened and an administrator password has been assigned, Sequoia is in multi-user mode and each user must log in to Sequoia with a valid username/password when starting the program.

With the option **Use network login** user settings from a network drive can also be used for the login process (see below).

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 settings (↗732) is specified in the program settings at **Save and load settings**, the login screen is still displayed when the program is started. This makes it possible to log in as a network user with a specific user profile or locally without a specific user name.

> ⓘ *You cannot change your identity while the program is running. You will need to exit and restart the program to log in under a different identity.*

If you delete the user `Admin`, Sequoia will be completely reset to single-user operation.

> ⚠ **Caution**: *All user settings will be lost!*

Set up multi-user operation on a network folder

You can also manage user settings centrally on a network folder to make them available to different workstations on the network. For this purpose, the network folder must contain the settings of the users as well as their login data in a `sequoia_admin.ini`.

Here's how you do it:

1. Let each user store a Setting Container (↗730) with their individual settings, collect these containers and name them after the respective users (e.g. Meier.INZ, Schulze.INZ...).
2. Place all these containers in the network folder.
3. Unzip the file `RemoteSetupTools.zip` from the program folder to the network folder.
4. To create the new users in the `sequoia_admin.ini` use the command line program `AdminCreator.exe` (see below) in the program folder. For new users, a blank password is created, which the user can customize via menu **File** > **Program Settings** > **Change User Password**. With `AdminCreator.exe` you can also assign passwords to users instantly.

On each Sequoia workstation that is to use network user management, the network folder must be set at **Network Path for Settings** in the section **Load and Save Settings**.
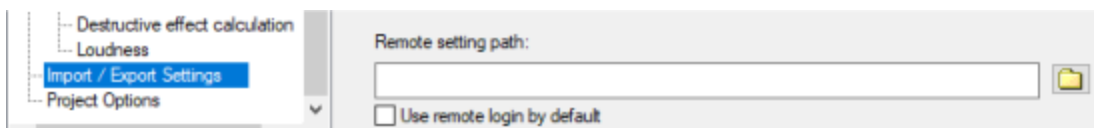
Activate the option **Network login as default** to use the network login by default at Sequoia startup.

When a user logs in via the network login, their settings are copied from the network folder to a temporary local folder.

Template usage (project templates, crossfade templates)

- User-specific templates are now stored in `fx-preset/user_name/templates`.

- If user-specific templates are present, only these will be displayed to the user when creating new projects. If those templates don't exist then the templates of the local admin will be displayed.

- 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 lists and work paths for remote users are saved separately for each remote user in the `audio.ini` file of the admin (in other sections)

- The current recent files list is kept when importing INZ containers. It is only lost if the `audio.ini` is overwritten by the admin, e.g. when loading an INZ container.

AdminCreator tool

Unzip the content of the file `RemoteSetupTools.zip` into the network folder and start the AdminCreator tool with

`AdminCreator.exe /T:"c:\ini" /U:"UserName" /P:"UserPassword" /N /R /I`

- `/T:""` = Path for `sequoia_admin.ini` (remote), if `AdminCreator.exe` is not started directly in the network folder.

- `/U:""` = user name
- `/P:""` = password

If username and password are specified, only the password for the specified user will be changed, no users will be added or removed.

- `/l` = All existing users in the `sequoia_admin.ini` will be deleted. If `/N` has not been set, then all settings containers available on the network drive will be added again as users.

- `/N` = no new user will be added to `sequoia_admin.ini`, even in case a new settings container is present. (this makes sense in combination with option `/R`)

- `/R` = Users that are contained in `sequoia_admin.ini` but for which no setting container exists are removed from `sequoia_admin.ini`.

SAVING, IMPORTING AND EXPORTING
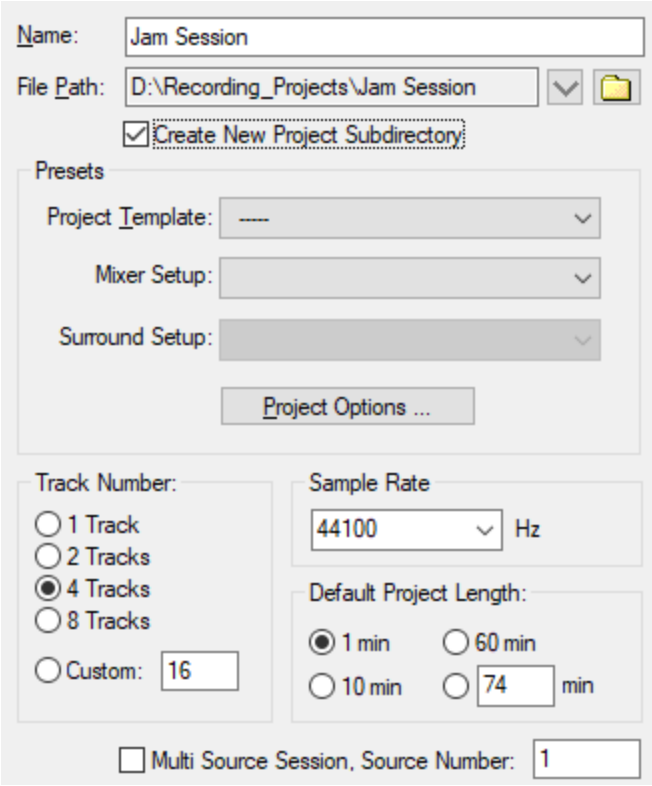
This chapter covers all topics related to loading and saving project files, project file properties and settings, and importing and exporting media files.

Creating New Projects

To create a new project, go to menu **File** > **New virtual project...** (keyboard shortcut **E**) or click this button in the upper toolbar.

The **New virtual project** dialog opens, where you define the most important properties of the new project. Most of these properties can be changed later.

> ℹ️ *All settings made here (except for the name of the project) will apply to future new projects, even after the program is closed.*

- **Name**: Enter a name for the new project here. By default, the current date is set as the name.

- **File path**: Here you can define in which folder you want to store your project. By default, the new project is created in the default folder for projects, see Program Preferences > Program > General (). The folder

icon can be used to browse to another location, the arrow button can be used to select from the last used paths.

- **Create new project subfolder**: By default, a subfolder is created with the project name in the file path.

- **Project template**: In this list box you can choose from supplied and your own project templates. These templates are actually ordinary projects. Therefore, you can create your own project templates by saving a project with the command Menu **File** > **More** > **Save project as template**. A new project using a project template gets all its properties, including busses, track states like mute or record, routings, instrument and effect plug-ins, but no objects.

  > **ℹ** *When a project template is selected, the track number, mixer, and surround setup settings are not available.*

- **Mixer setup**: In this list box, you can choose from supplied and own custom mixer presets. These presets include the number and type of tracks and busses, as well as input/output and effect routing. For more information, see Mixer setup (↗646).
  If you have selected a mixer preset with surround master, you can select a surround format in the **Surround Setup** list box below it.

- **Project Options**: The dialog corresponds to the General Project Options dialog (↗643), where you can make settings such as snap and grid settings, BPM, auto crossfade… for each loaded project. However, the project options here apply to any newly created project.

  > **ꞏ꞉Q̣ꞏ** *To set project properties as default for all new projects at a later time, activate the option **Save as default** in the dialog **Project options**.*

- **Number of tracks**: The start track count of the virtual project is defined here. You can add more tracks with Menu **Track** > **Insert New Tracks** (↗128).

- **Sample Rate**: Here you can set the sample rate of the virtual project.

- **Default project length**: Here you can set the original project length. The length is automatically adjusted when loading audio files or recording and can also change when zooming out.

- **Multi Source Session, Number of sources**: Here you can enter the number of desired source projects of a multi-source session. Sequoia then creates a destination project and the selected number of source projects.

Project Properties

All project-specific settings dialogs and some special dialogs of the program are combined in one large dialog, from which you can also reach the other settings dialogs via the tree structure on the right. The quickest way to open it is to use the keyboard shortcut **I** or also menu **File** > **Project Properties** > **Project options…**.
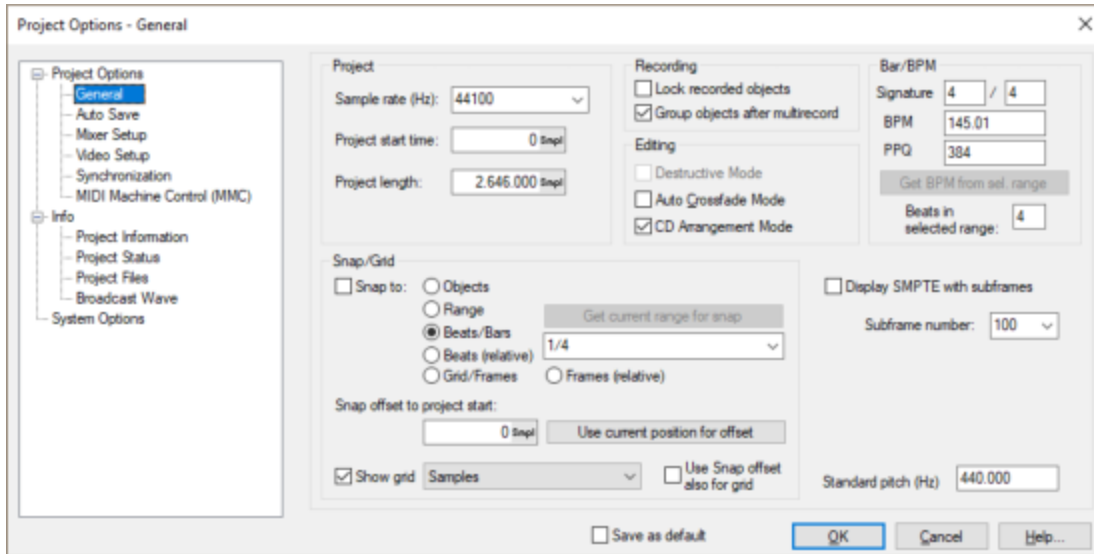
Use the bottom entry **System options** in the tree to switch to the tree structure with the Program preferences (⬈688).

General Project Options

In this dialog you make the most important settings for the current project. This includes global specification of beat and tempo in BPM, recording and editing options, and snap and grid settings.

You open the general project options via menu **File** > **Project properties** > **Project options...** (keyboard shortcut: **I**).
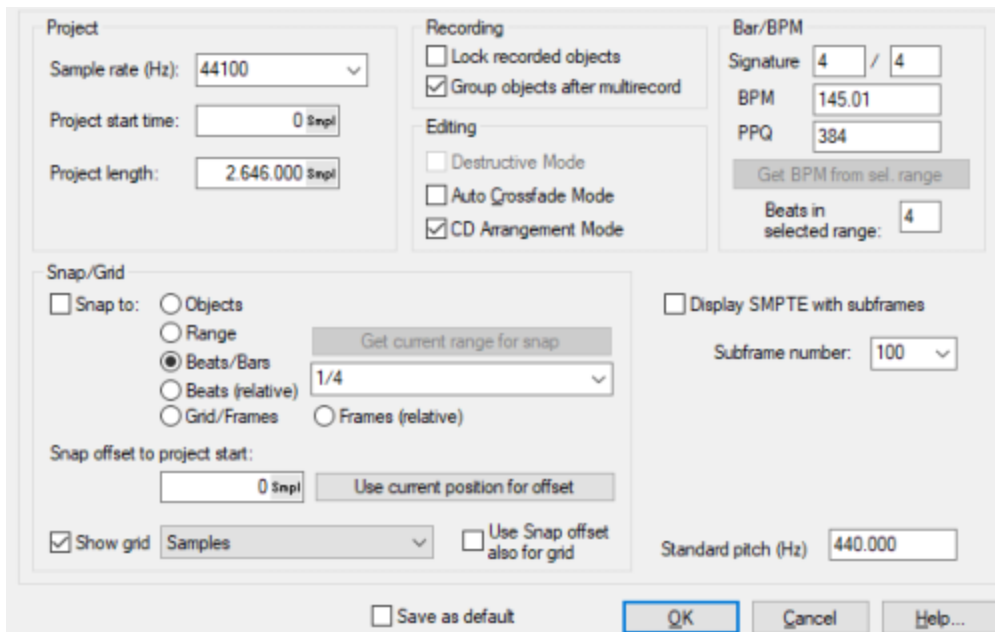
Save as default: If you activate this option before closing the dialog with **OK**, the settings in the dialog will be the default values used in all newly created projects. This also happens if you open this dialog from the dialog New project (↗641).

Project

- **Sample rate (Hz)**: This displays the sample rate used in the project.
  If you change this value and the project already contains audio and MIDI objects, you will be asked whether the objects should be adjusted to the modified sample rate. The adjustment is default on, since it is not useful only in rare exceptional cases.

- **Project start time**: Here you can determine the project start time. This is useful when loading audio files that should be placed at the original time positions (timestamps) of the recording. If a recorder with free running timecode (clock) was used, this would create a gap of many hours in front of the audio, making zooming and navigation difficult. You can compensate for this gap by specifying a project start time. If you change the start time, you will be asked whether you want to leave the objects in the project at the current time positions or whether the time positions should be adjusted. With an adjustment, the objects remain visually at the same time position; without an adjustment, all objects are moved together with the timeline.

- **Project length**: The project length is displayed here in bars and beats.

  - ᯓ *If the* general program preference *(↗706)* **No change of project size when zooming** *is active, you can extend your project here. If this option is not active, the project will automatically extend when you zoom out further. In this case you can use the* **Project Length** *setting here to shorten it again.*

Recording/Editing

Here you can make some settings for recording and editing:

- **Lock recorded objects**: Recorded objects are protected against unintentional moving.

- **Group objects after multirecord:** Objects that belong together will be grouped after a multitrack recording.

- **Destructive editing mode** (Wave projects only): Switches between destructive editing and real-time audio editing for wave projects (↗570).

  - ᯓ *This option can also be set directly in the menu* **File** > **Project properties**

- **Auto crossfade mode (↗169)/ CD arrangement mode (↗583)**: The respective modes are activated when the project is loaded. For more information, read the corresponding linked help topics.

Bar/BPM

Set here the global **time signature** (numerator/denominator) and the global tempo in beats per minute (**BPM**). **PPQ** is the timer resolution (Peaks Per Quarter) and determines how many ticks a quarter note is divided into.

-💡- *Tempo and time signature can change during the course of the project, read the chapter* Project Tempo *(↗354) for more information on this.*

You can have the global tempo determined by selecting a range in the project. Select a range in the project by hearing, using the waveforms, and enter the number of beats that comprise your selection in the field next to **Beats in selected range**. Then click **Get BPM from sel. range**, to set the tempo in BPM based on the selected range.

Snap/Grid

Snap to: Here you can switch the grid on and off. Various snap types can be selected:

- ■ **Objects**: Objects snap to the edges of other objects and to markers when moved.

- ■ **Beat**: Musical time measures are used as step size for the grid. In the menu next to the option you can select the unit for snap.

- ■ **Beat (relative)**: A beat grid is also used as a basis, but an object retains its relative distance to the next snap point when it is moved.

- ■ With **Snap to range** the grid is similar to the beat grid, except that the snap width is defined by selecting a range before activating the grid. The length of this range then corresponds to one bar, and the snap width can be divided according to the **Q** value as with the beat snap.

- ■ **Frames**: A frame length is used as step size. The exact frame length depends on the set frame rate (SMPTE). Thus, the grid corresponds to the spacing of the frames in a video.

- ■ **Frame grid (relative)**: The step size is also frame, as with the relative beat snap, the relative distance to the snap point is preserved when moving.

-💡- *For more information on the function of the snap, see* Snap *(↗79)*

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 play cursor as the grid's zero position.

Show grid: When the option is turned on, the grid is displayed, according to the unit of measurement selected in the menu next to it.

Use snap offset also for grid: The snap offset is also used as offset for the grid.

Show SMPTE with sub-frames: The SMPTE position is additionally indicated with sub-frames for all e-time specifications in the program. At **Sub-Frame Count**, specify the number of subdivisions for a frame.

Default Pitch: In this field you can change the default pitch for the concert pitch a' (440 Hz) if you want to use an alternative tuning for the internal tuner (↗568).

Mixer Setup

The **Mixer Setup** dialog lets you quickly assign the desired number of tracks, busses, or masters to a project, automatically assign inputs and outputs to tracks, and initialize track panorama for mono or stereo recording tracks.

The settings of the dialog can be saved and loaded as a preset, whereby it can be flexibly determined which mixer settings are to be changed by loading the preset and which are not.

All changes to mixer, routing and tracks in the dialog are not applied until the dialog is closed with **OK**.

In addition to opening the mixer setup via the **Project Properties** dialog and via menu **File** > **Project Properties** > **Mixer Setup...** (keyboard shortcut: **Ctrl + Shift + M**), you can also open it in the mixer itself with the **Setup** button in the upper right corner.
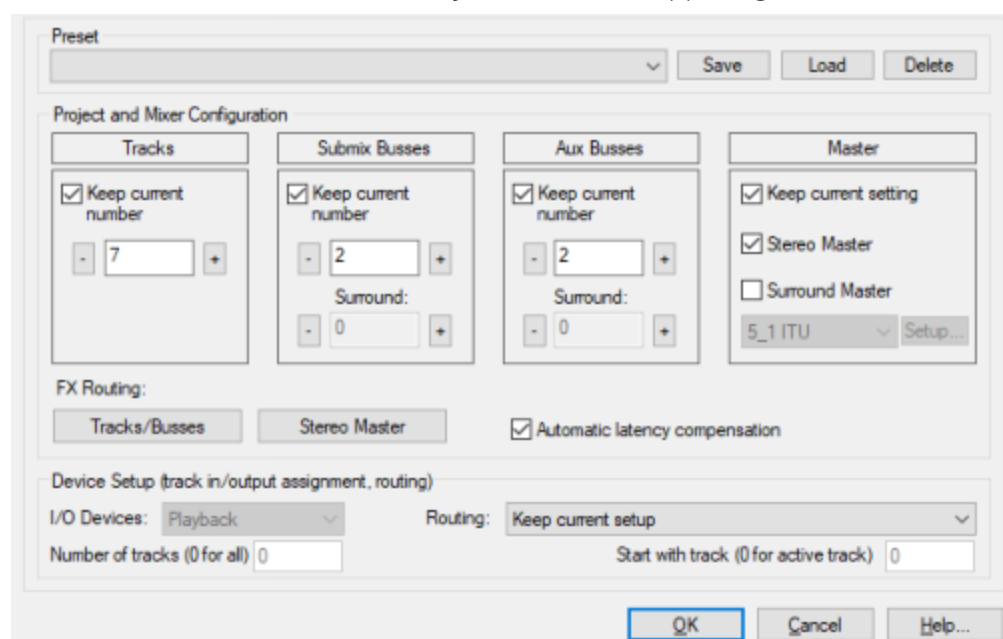
Project and Mixer Configuration

Here you set the number of tracks, AUX busses and submix busses and configure the masters as well as the FX routing for all tracks and the stereo master.

Tracks, submix busses, AUX busses: The option **Keep current number** works both when saving and loading presets. It is initially activated for tracks, buses and AUX buses. Only if you change one of these numbers, it will be deactivated and only these changes will be included when saving a preset. For example, if you load the "4 Busses + 4 AUXes" preset, the number of tracks will not be changed, because the number of tracks is not included in the preset.

If a mixer preset is loaded that changes the number of tracks, AUX or submix busses, you can set the checkboxes respectively to prevent this. So if you only need four additional submix busses, you can still load the "4 Busses + 4 AUXes" preset and then activate the **Keep current number** option on the AUX busses.

Surround adds the desired number of surround submix busses and surround AUX busses respectively. The option only becomes active when a surround master exists or is to be created.

Master: Here you can provide the project with a stereo master and/or surround master.The option **Keep current setting** works the same way as for the track numbers. When selecting a surround master, you can also select a surround format from the menu by defining the number of surround channels and their assignment to the output devices. For more information, see Surround settings (↗513).

FX Routing: This opens the FX Routing dialog for the current track and the stereo master. Changes here are not applied to the mixer presets.

Automatic latency compensation: Automatic latency compensation for plug-ins is preset to on and should only be deactivated in exceptional cases.

Device Setup

In this section you can assign the input and output devices for several tracks simultaneously and make automatic panorama settings for using mono or stereo tracks.

I/O Devices specifies whether the routings should be applied to the input devices, output devices, or both.

Track count (0 for all) / **Start with track (0 for active track)** determines to which tracks the routing changes are applied. For example, if you set the number of tracks to 4 and start with track 8, the selected routing preset will be applied to tracks 8-11.

In the menu **Routing** different automatic routing and panorama settings can be selected:

- **Keep current setup**: Similar to the other sections, there is also a flag for keeping the current settings.

- **Assign all tracks to stereo master**: The tracks and busses are routed to the master. With **L/R Panning** the track panorama is alternately set to left and right. Use this option if you want to record stereo sources as mono pairs.

- **Assign all tracks to available stereo/mono devices**: The tracks are assigned to the available output or input devices (↗696) in ascending order according to the option selected at **I/O Devices**. When routing to mono devices, the track panorama is set alternately to left and right.

- **Assign tracks to surround channels**: According to the selected surround format for the surround master, the first tracks are routed to the surround master and panned to the individual surround channels.

- **Use the setup contained in the preset**: When saving a preset, all device assignments are saved, but when loading this preset, only tracks and masters are created and the default assignment is used. If this option is selected, the device routing contained in the preset is also applied.

Information

Information about the current project is displayed in these dialogs.
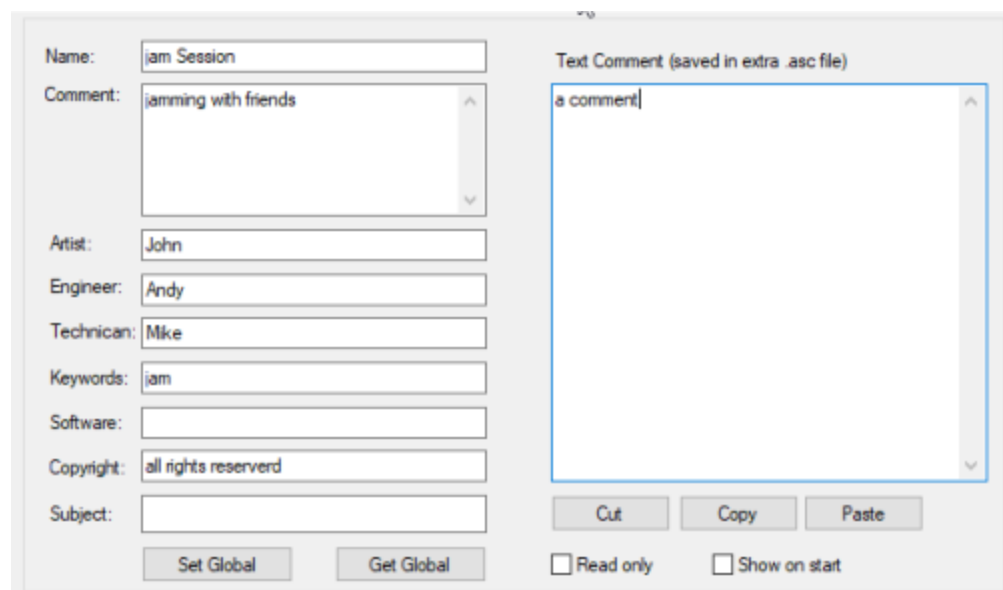
Project Information

Here you can enter metadata about the project that will be stored in the project file, such as project comments, copyright information, and artists and sound engineers involved.

The content of the field **Text comment** on the right is also displayed in the Infomanager at **Project**.

> ⚠ *The text comment is saved alongside the project file (.vip) in a separate text file with the same name and the file extension *.asc.*

Use the **Write protection** option to lock the text comment against accidental changes. If you set the **Show comment when opening** option, the Infomanager (↗337) opens to display the comment when you open the project.

If you want to provide entries in this dialog with the same information more often, fill in the corresponding fields and click on **Set Global**. Next time you can then fill in these fields by clicking on **Get Global**.

Project Status
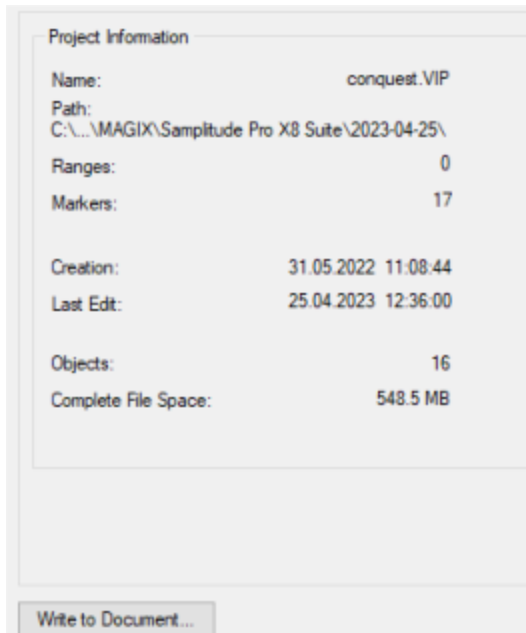
This dialog displays statistical information about the project: project name, path, number of ranges, markers, objects and takes, the date of creation and the last change. In the menu under the button **Clean up...** there are several functions for optimizing the project and deleting unnecessary data.

> ℹ️ *This does not delete any recorded audio data, for this there is the function* Delete unused samples*Remove unused samples (↗684) in the menu* **File** > **Clean up & backup**.

- **Remove ARA**: This function removes all ARA plug-in instances from the project.
- **Remove obsolete take entries**: When working in the take lanes and deleting objects, take entries are sometimes created that no longer have any function. For better performance and to reduce memory requirements, you can use this command to remove these entries.
- **Remove all take entries**: You can also remove all unused take entries once comping is complete in order to optimize the project.

Project Files

The **Project Files dialog** lists all audio files used in the project. Files located outside the project folder are listed separately.
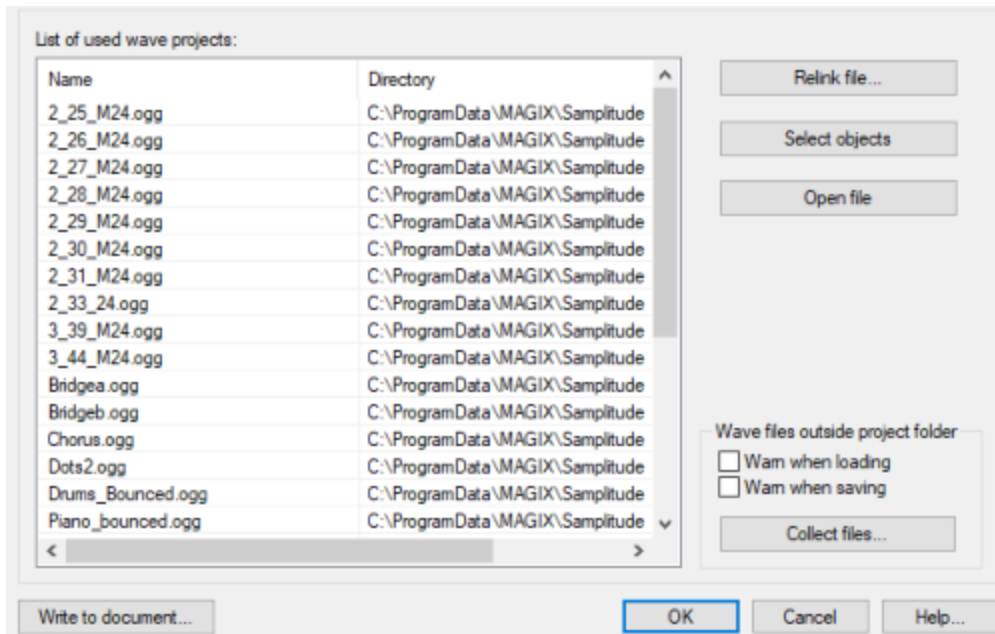
Click on a file to select it. With **Re-link File...** you can swap the audio file in the project. Select a new file, all objects that used the old file will be changed to use the new file. Use **Select objects** to select all objects that refer to a file. Use **Open file** to open the selected file as a wave project (↗569).

At **Wave files outside the project folder** you can activate warnings that are displayed when saving or loading the project as soon as audio files of the project are outside the project folder. With **Collect files...** you open the dialog Collect project files (↗681) to copy the files collectively into the project folder.

Click on **Write to document** to open the information from **Project status** and **Project files** as a text file in the system text editor.

Broadcast Wave Manager

The Brodcast Wave Extension allows you to store information about an audio file as metadata in an extension of the wave format known as a "chunk" (Broadcast Wave Format, or BWF for short) in the file itself. This metadata can basically be applied in a proprietary manner, but observing the respective EBU and SMPTE guidelines is recommended.

Wave project: A list of the wave files contained in your project is displayed on the left. To add information to or view information about a file, click the file. The associated BWF information is then displayed on the right side of the dialog. For newly recorded audio material, you can specify metadata to be written to the Broadcast Wave Extension.

This metadata can be read or extracted later and used for further purposes - like managing the audio files in databases.

The entry **Preset for new audio files** contains all the information that should be used within the project for newly created audio files.

With **Save** you write the BWF data to the selected file, **Save All** writes all changes made to the metadata in the dialog to the files.

The following metadata can be saved as BWF data:

- **File description**: This is a freely usable text field (maximum 256 ASCII characters).

- **Originator**: This field (maximum 32 characters) contains information about the origin of the file, for example the name of the engineer.

- **Originator Reference**: This field (maximum 32 characters) is defined 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.

- **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 edited in BWF Manager, this value is independent of the date of the file on disk.

- **Time**: This specifies the time when the file is created. As with the date, this is automatically created from the file properties but can be edited retroactively.

- **ISRC**: Enter the ISRC code 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. See also the notes below in Importing and Exporting ISRC to and from CD Tracks.

- **Timestamp**: The timestamp saved in the BWF extension is displayed here. This is the project timestamp when the file was recorded. (Elsewhere in Sequoia, this is also referred to as the "original position" of the object.) For other applications, this timestamp can provide information on the time on the day of recording. With the button **Edit...** you can edit the timestamp or take it over from the object position. You can also assign the timestamp from the object positions to all audio files or or undo the timestamp assignment.

- **Unique Material Identifier (UMID)**: The specifications for UMID are defined 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. The use of UMID is not absolutely necessary for a valid BWF.

- **Loudness**: With this button you can have the Loudness (↗559) of the audio file entered into the BWF metadata. Select **Automatically update loudness metadata** to have this information automatically updated when changes are made to the audio in the wave file.

- **Coding History**: In addition to the information about the format of the file (A: encoding, e.g. PCM; F: sample rate; W: bit depth; B: bit rate; this value is only used for non-transparently encoded material such as MPEG or MP3; M: channel count), each entry of this field contains a value T. This is a string without a comma in which, for example, the serial number of the recording analog tape machine, codecs, dither types, AD converters, or special signal processing applied to the file such as DeNoising can be entered.
  When a file is recorded in Sequoia, an entry is made. If this file was then edited again, e.g. by bit depth reduction or MPEG encoding, another entry is added.
  Use of the coding history is regulated in the EBU Recommendation R98-1999.

Advanced Options

There are some additional options at **Extended Options**:

- **Automatic update of loudness data**: Here you can automatically update the loudness metadata in BWF files. This corresponds to the option under the button **Loudness**.

- **Save BWF graphic data in wave files**: If you activate this option, Sequoia creates a "peak chunk" that stores the graphic information as metadata in the file. Normally, the graphics data in Sequoia is stored in separate peak files (.h0 files).

- **Read BWF graphic data and use it in** Sequoia: If you enable this option, the included "peak chunk" will be used instead of the .h0 file.

Notes on Importing and Exporting ISRC from and to CD Track Indices

To copy ISRC from wave files to CD track indices, load one or more wave files in succession onto the *first* track and select **CD/DVD** > **Indices** > **Set indices at object edges** from the menu. Only with this procedure the ISRC are taken over from the wave files into the CD track indices, but not when setting CD track indices manually. You can see the ISRC of the imported files in the **ISRC** column of the Marker Manager.

To take over the ISRC entry to the imported WAV files when importing CD tracks, activate the option **Import ISRC and pause indices** in the CD Import dialog (↗574). To see the ISRC entries already in the import dialog, click the **Read ISRC** button.

To transfer the ISRC you have previously set in CD Track Indexes to the wave files, select the settings **Split at markers** and **Each CD track to a file** in the **Export dialog** at **Source**. Because each track requires its own ISRC, it is not possible to enter only one ISRC for all included files.

Loading and Saving Projects

> ℹ️ **Note**: *To distinguish from wave projects, projects in which references to the wave projects are arranged as objects were called virtual projects (*.vip) or VIPs for short. In its early days, Sequoia worked with "wave projects" (*.hdp), where the actual samples were stored in a separate sample format; only later wave files were used for this purpose. Since "wave file" can now be used identically to "wave project" except for insignificant details, for the sake of brevity we can simply speak of projects or project files for virtual projects.*

Loading Projects

Load saved projects with menu **File** > **Open** > **Virtual project (*.vip)...**, keyboard shortcut **O** or this button on the top toolbar .

> 💡 *Projects can also be loaded from the* File Manager *(↗318), where you have full-text search, favorite folders and more.*

The last 16 files that were opened are listed for quick opening in the menu **File** > **Open recent files**. This menu can also be opened with the keyboard shortcut **Ctrl + Backspace**. To select the file, use the arrow keys and the Enter key; you can also load the first 9 files by pressing the corresponding number.

Via menu **File** > **More** > **Load project without VST plug-ins…** you can load projects that use VST plug-ins that are not installed on the computer or are not compatible with Sequoia.

Saving Projects

In the menu **File** you will find the common commands for saving your project:

Save (keyboard shortcut: **Ctrl + S**): The active project is saved. If it is a newly created project that has not yet been saved, the name and save folder can be selected in a file selection dialog. By default, the name of the project and the project subfolder are used.

> ℹ️ *By default,* new virtual projects *(⬈641) are created in a project subfolder of the same name, with the current date as the default name. If you assign a different name when you save for the first time, the project subfolder will not be renamed as well, so you may not find the project again as quickly. You can manually rename the project subfolder afterwards, but we recommend that you give the project a proper name already when you create it.*

If you want to keep intermediate states of your project saved in separate files, select either **Save as…** (keyboard shortcut: **Shift + S**) and use this command to save the project under a new name. This will keep the last saved state in the file with the old name and you will now continue working on the file with the new name.

Or you can use **Save project copy…** (keyboard shortcut: **Ctrl + Alt + S**) to save a copy of the project under a different name. By default, the project name is extended by the current date and time. After saving the copy, you continue working on the current project with the original name.

If you just want to give the project a different name, but you want to continue working with the same project, use **Rename Project…** in the menu **File**.

> 💡 *An asterisk (*) at the file name in the project docker tells you that a project has been modified and needs to be saved.*

Auto Save

Sequoia can automatically save project files as a backup at regular intervals for safety. In the dialog **Automatic Save**, which you open via menu **File** > **Project Options** > **Automatic Save…**, you can configure this function.

In the list you can see the backup files available for the project. If you select a backup by clicking on it, you can open it in a separate project window by clicking on the button **View backup project**. Use the **Restore backup** button to restore the current project to the state of the selected backup. This means that the current project is discarded and the backup project is opened under that name and is now considered the current project. If there are any unsaved changes in the project before, this last state is additionally saved under the project name with the addition _OLD.

ℹ️ *This can become important: You will probably want to restore a backup when you notice an error and want to revert to a version before the error. But at the same time, if you want to keep the latest changes, you can find them in the _OLD version of the project.*

Activate **Auto save** with the checkbox **Active**. The options are as follows:

- **Save to original**: No separate backup files are created, the project is only saved automatically at specified saving intervals. If the **Save to subfolder** option is active, the previous version of the file will be moved to the **Backup** subfolder, overwriting the second to last version. You will then always have two versions of the project with the same name, the current version and the version before that in the backup folder.

- **Create backup file (*_bak.VIP)**: Creates a backup file and updates it regularly. If you select the option **Save to subfolder**, the backup file will be saved and updated in the subfolder **Backup**. So you always have a current version of the project and a backup version that you can return to with **Restore backup**. this is the default option.

- **Create max. … backup files (*_bak0.VIP)**: Multiple backup files are saved up to the specified maximum number (*_bak0…*_bak9). When the maximum number is reached, the oldest backup is deleted and the other backups are renamed so that a current backup can be added. So you always have a current version of the

project and at most the specified number of backup versions to which you can return with **Restore backup**.

- ■ **No Auto save during playback and recording (prevent drop-outs)**: Automatic saving during playback and recording is set to inactive to avoid recording errors.

- ■ **Auto save after finished recording**: When this option is activated, the project is automatically saved directly after a recording has been completed.

- ■ **Save interval**: Here you define the time interval at which automatic storage takes place. The default setting is 10 minutes.

Edit Root VIP

When you export a project to a wave file in Sequoia, the information from which project it was created is stored in the file. If the file is later used in another project, you can use the function Menu **Object** >**Edit** > **Edit Root VIP...** to reopen this VIP for editing.

Example: You have compiled a CD from several tracks previously created in Sequoia, each of which has been saved to individual wave files using the Export function (↗661). They are now sitting as audio objects in the arranger window of your new project, and you notice that you still have to change something in one piece. **Edit root VIP...** opens the original project in which you now make the desired changes. When you save, the modified project is re-exported and also updated in the project where you are currently compiling your CD.

Append Project

Use this function to append a project to another project. The objects of a project (for wave projects: the audio data) are copied directly to the end of the other project.

To execute this command:

1. First load both projects.
2. Switch to the project at the end of which you want to attach the other project.
3. Select Menu **File** > **More** > **Append Project**.
4. Then switch to the project to be appended. Confirm the operation by selecting **Append** in the query dialog.

ⓘ *For virtual projects, only the objects from the appended project are copied; the mixer settings and automations of the appended project are not copied.*

Loading Audio Files

Loading Audio Files

To load an audio file into the current project you have the following options:

- **Drag and drop one or more files from Windows Explorer** into the project at the desired location.

  > ⓘ *If you do not drag the file into the project, but to the* Project docker *(↗82), you'll open the audio file as a wave project in a separate window to edit it* destructively *(↗569).*

- Use the **file manager** (↗318). This is the most convenient option, because there you will have an automatic preview function, full text search, favorites folder and others.

- Select menu **File** > **Import** > **Load audio file** (keyboard shortcut: **W**) or click this icon on the upper toolbar 🎛️ .

  A file selection dialog opens. The small playback control at the bottom of the dialog lets you preview the file before loading it. With the **Autoplay** option enabled, the preview starts automatically as soon as you select the file.

  > ⓘ *The pre-listening function uses the default playback device from* Sequoia *(see Loading options).*

  To load multiple files at once, expand your selection to include individual files with **Ctrl + mouse click**, for a set of files **Shift + mouse click**.

  With the option **Load file as wave project** you open the audio file as a wave project in a separate window to edit it destructively (↗569).

  The option **Load files only into clipstore** does not include the files in the project, but only creates an entry in the Clip Manager (↗311) (clipstore).

The Loading options (↗321) will appear first, where you can specify whether the file should be copied to the project folder, how to deal with multiple files, and more. Use **Do not show this window again** to prevent this dialog from being displayed every time.

Importing Audio Files

The following audio formats are loaded from Sequoia directly, i.e. objects are created in the project that reference these files: Wave files (.wav), MP3/MPEG files (.mp3, .mpg, .mus), Quicktime files (.aif), MS Audio files (.asf, .wma), Ogg Vorbis (.ogg), FLAC (.flac) and playlists (.m3u, .cue).

Some compressed file formats cannot be loaded directly. These files must be imported and converted to Wave format in the process. In this case, a file selection dialog will open automatically, where you can specify the name and location of the created file. By default, this is the project folder. The created object refers to this file.

In terms of performance and stability, it may be appropriate to convert files to Wave format, even if Sequoia is able to process the files in their original format. In this case, load the desired file with menu **File** > **Import** > **Import audio...** (keyboard shortcut: **Ctrl + I**).

Even if you prefer to save all audio files loaded into the project as a copy in the project folder right from the start, you can use this menu option when loading.

> ℹ️ *In the Loading options (see below) you can specify at **File management** that audio files are always copied and/or converted even during normal loading.*

Transferring Metadata from Audio Files

Metadata is information about the content of audio files, such as track name, album or artist. This information is included in many compressed audio formats such as MP3 (ID3 tags), OGG or FLAC. To be able to edit this data in Sequoia and use it for later exports or when burning to CD for CD-Text, the data must be imported into **CD Track Indices**. Read the notes in the section Indices (↗579)!

Options for Loading Audio Files

In this dialog, you specify various options for how audio files should be handled when loaded into a project.

As long as these options have not been set as permanent with the checkbox **Do not show dialog again, settings always apply**, the dialog will appear every time audio files are loaded. However, you can always open it using the **Options** button (gear icon) in the File Manager or the **Options** button in the **Load Audio File** dialog.
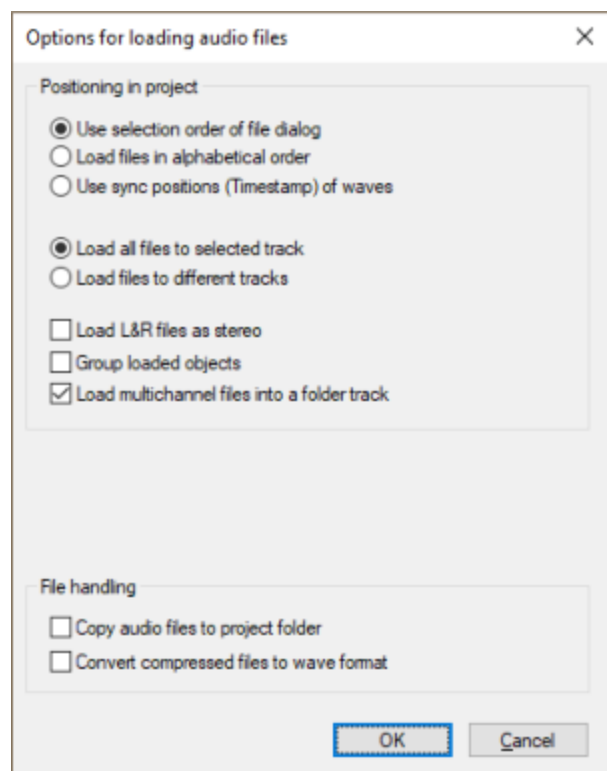
For the following descriptions, the dialog is shown here as it appears when you load multiple files. When loading a single file, some options are omitted.

Positioning in project

- **Use selection order of file dialog**: If this option is selected, Sequoia remembers the order in which the files were selected and then arranges them in that order.

- **Load files in alphabetical order**: With this option, Sequoia arranges the selected files in alphabetical order in the VIP.

- **Use sync positions (timestamp) of waves**: Broadcast wave files containing time stamps are positioned precisely at this position in the VIP.

- **Load all files consecutively on the selected track**: The files are loaded one after the other on the selected track.

- **Load all files one below the other**: The files are loaded from the selected track one below the other into the next tracks. If necessary, additional tracks are created.

- **Load L&R files as stereo**: Identically named audio files whose names end with _L and _R are loaded as stereo objects.

- **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 track (↗135) with the required number of tracks is created and routed to the surround master (that is created if necessary). The tracks in the folder track get the surround panning according to the detected surround format. If this option is not active, the individual channels are loaded onto the selected track and the tracks below it.

Other (only when loading via Menu File > **Load audio file**)

- **File preview with Windows Media Player Control**: Activate the option if the file preview should not be done via the standard playback device but via an embedded Windows Media Player. This uses the system output of Windows. This has the advantage of always having a playback option regardless of the actual use of the audio outputs in Sequoia, but the Windows MME driver used for this often cannot be used simultaneously with ASIO.

File handling:

- **Copy audio files to project folder**: The files are copied to the project folder.

- **Convert compressed files to wave format**: Compressed audio formats such as MP3 can be loaded and played directly from Sequoia. However, this increases the CPU load. Therefore enable this option to convert all files to Wave format on loading.

Audio Files with Different Sample Rate

If you load audio files whose sample rate differs from the project sample rate, you will be asked how to handle such files in the project.

The default option **Object resampling** applies real-time resampling to the object created during loading so that it plays back at the correct speed when played back in the project. Note that this will prevent some other effects from being applied:

- Musical Tempo Adjustment
- AudioWarp
- Elastic Audio (pitch automation)

So if you plan to use these effects with this object, choose the first option **Convert file**. In this case, a copy of the audio file is created, which in the process is given the appropriate sample rate by offline resampling.

The option **Adjust VIP sample rate** adjusts the sample rate of the project to the sample rate of the audio file loaded. This will cause all objects already in the project to be played at the wrong speed afterwards..

If **No change** is selected, the file is used with the (wrong) sample rate of the project and played at the wrong speed and pitch.

Select **Do not show this message again for the current VIP** if you want your selection to always apply to the current project, with **Don't show this message again** the selection also applies to all future projects.

Loading Audio Files as Raw Data

The command Menu **File** > **Import** > **Import as Dump...** lets you import audio files in unknown formats or corrupted audio files where the "header", the part of the file that contains the information about how the sample data is stored in the file, cannot be interpreted, is corrupted, or is not present at all.

Select the file in the file selection dialog, then a dialog will open where you can specify by yourself in which format (bit resolution, byte order, mono or stereo and the sample rate) the sample data is in the file.

With **Start position of data** you can, if the length of the header is known, skip this part during import, default this is the header length of wave files. For other files, you may need to experiment. But you can also enter 0 and import all the data and edit the file after import.

Exporting Audio

With menu **File** > **Export...** (keyboard shortcut: **Ctrl + E**) you save your project to one or more audio files. (This is also referred to as "rendering" or "trackbouncing").

In the **Export**In this dialog all aspects of an export like used tracks, time selection, file formats... can be set very flexibly. dialog, all aspects of an export such as tracks used, time sections, file formats... can be set very flexibly That's a lot of parameters, so all the settings made in the dialog can be saved together as a preset for later use, so you don't have to start from scratch every time.

The dialog is divided into several sections that can be opened and closed individually. In the collapsed state, a summary of the selected settings is displayed in the heading.
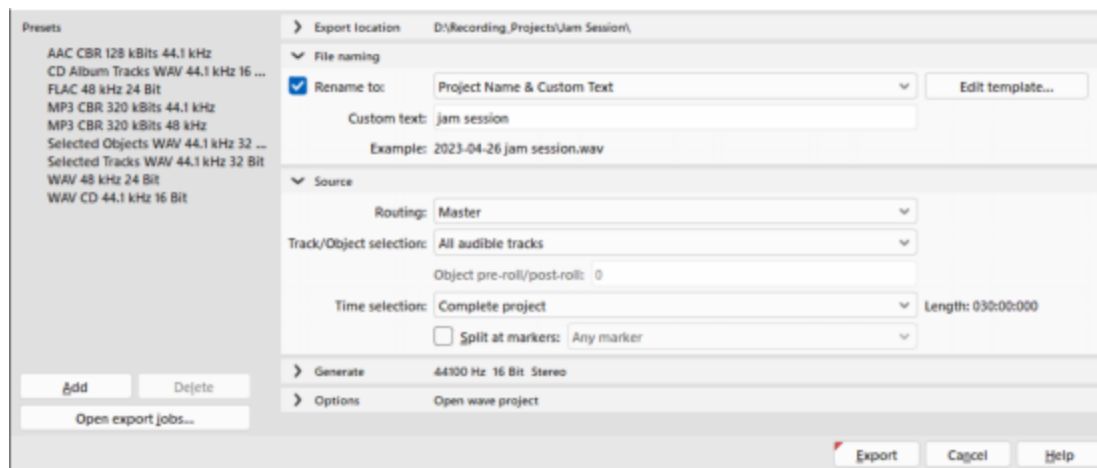
Export to

Here you specify where the exported files will be saved.

In the menu **Export to:** you can select the project folder (preset) or various default folders (Music, Desktop, Documents). You can use the folder button 📁 to select other folders in the file system (**Custom folder**). Click on the arrow button to the right of the text field to select a folder from the most recently used export folders. When choosing **Choose folder later** you specify the folder just when you run the export function.

At **Put in subfolder** you can optionally specify an additional subfolder. With **Edit template...** you can define the name of the subfolder as with the file names (see below) using variables from project properties.

Existing files: Here you specify how to handle it if files with the same name already exist at the selected location, e.g. if you run an export again with the same settings. If you select the option **Choose new name** you will be asked for a new file name in these cases, if you select **Overwrite** existing files will be overwritten without warning. **Skip** skips the file. **Ask what to do** leaves this choice to you in individual cases.

File naming

Here you specify how the exported files will be named. The project name is used by default. If several files are created during export, e.g. by splitting at markers or when exporting individual tracks or objects, the file name is generated according to the following rules:

- **Single track export**: \<track number\> \<track name\>

- **Export of individual objects**: \<object name\> + \<track number\>, for identical object names in a track additionally + \<consecutive number\>.

ℹ️ *If **Chose file name later** (see below), the name entered + consecutive number is used for tracks and objects.*

- **Split at markers**: \<MarkerNumber\>_\<MarkerName\>.
  The project name is not used for the files, but used for the name of an optional playlist (see Options).

Activate **Rename to** for further naming options. In the menu, you will find various combinations of date, project name and a **custom text**, which you can enter in the field below the menu. Use the option **Select file name later** to enter the name in an input field for each export process.

With **Edit template...** you can create your own templates. Click the buttons with the variables to compose the naming scheme. You can also insert any other characters such as spaces or underscores. Characters that Windows interprets as path components, such as **** or **:**, are not allowed. The \<CustomText\> variable refers to the text you can enter at **Custom text**. Click on **Save as...** to give the template a name and add it to the **Rename to** menu.

For all of these options, you can see which file name results from the current project with the selected settings at **Example**.

Source

Here you can specify which audio material from the project should be used for export and how.

Routing

This menu is used to specify the outputs whose output is to be written to the files.

- **Master**: (preset) All tracks that are routed directly or indirectly via submix buses to the master output of the mixer are used.

- **All outputs (ignore device routing)**: All tracks are used and rendered by the master, even if they are routed to physical outputs other than the master.

- **All outputs (use device routing)**: All unmuted tracks are exported according to their routing to the individual hardware output devices. Depending on the settings at **Format**, individual mono files or stereo files for each physical output are generated, or an interleaved multi-channel wave file with the corresponding number of channels.

  ℹ️ *The other options under **Track/Object selection** for exporting individual tracks or objects are not available if this routing option is selected.*

If a surround master is present, there are the additional options

- **Surround Master**: The surround master is exported, see Surround Export (↗531).

- **Surround + Stereo Master**: The surround master and the stereo master are both exported.

Track/Object selection

In this menu you determine whether you want to combine the output of all tracks or only selected tracks into one file, create separate files per track, or export individual objects:

- **All audible tracks**: All unmuted tracks are mixed. This is the default option.

- **Selected tracks only**: All selected tracks are mixed.

- **All audible tracks individually (Multi-track export)**: All unmuted tracks and busses are exported individually to files.

- **All selected tracks individually (Multi-track export)**: All unmuted and selected tracks and busses are exported individually.

The individual tracks are always rendered through the entire effects chain, including sub-mix busses and master effects. When sending to AUX busses, the track portion of the AUX bus is also added. In other words, as if the individual tracks were set to solo one after the other, see Global solo mode **Solo Safe**Global Solo Modes (↗505). This behavior can be changed at Options so that the track effects, master effects and the portion from the buses can be deactivated separately.

- **Export 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 libraries).

- **Glue selected objects together**: All selected objects, including object effects, are exported to one new file per track. All objects between the first and the last selected object of a track are glued together.

- **Glue selected objects until silence**: All selected objects, including object effects, are exported to new files. The objects in each track are glued together until the next silence.

When exporting objects, there is an additional setting **Object pre-/post-roll**, which allows you to include additional samples before and after the object boundaries in the file for safety.

> ℹ️ *Notes:*
>
> - *If you select **Surround Master** or **All Outputs (Use device routing)** for **Routing** (see above), exporting individual tracks or objects is not available.*
>
> - *If a selected track is a Submix bus, all tracks routed to this bus will also be exported. This is done recursively, i.e. if these tracks are in turn buses, their sources are also exported. This permits fast mixing of individual groups.*

Time Selection

In this menu you select the time period of the project for export:

- **Selected range only**: Only the range selected in the project will be exported.

  ℹ *Note: If no range is selected, the complete project will be exported.*

- **Selected range markers**: If several range markers have been selected in the range marker track or in the range manager, these ranges are exported separately.

- **From the beginning of the project to the last object**: (default) The time selection goes from the beginning of the project to the end of the last object plus the reverb time (↗702).

- **Complete project**: The entire project is used, i.e. including the silence behind the last object until the end of the project.

- **Complete CD**:Time selection goes from the first CD track index to the CD end index.

  ℹ *Note: If there is no CD end index, the complete project will be exported.*

Split at Markers

ℹ *This option is only available with the track selection **All audible tracks**.*

Enable the option if you want to export to multiple files:

- **Each CD track into one file**: Individual files are created, ranging from one CD track index to the next.

- **Each CD track to pause marker into a file**: Individual files are created, ranging from one CD track index to the next pause index. The sections between the pause index and the next CD track index are not exported.

  ℹ *Select these options if you want the track information from the CD track indexes to be entered into the metadata (ID3 tags) of the exported files.*

- **All markers**: Individual files are created, each ranging from one marker (all marker types) to the next.

Generate

Here you specify the format to export to and what to do with the created audio files

Format
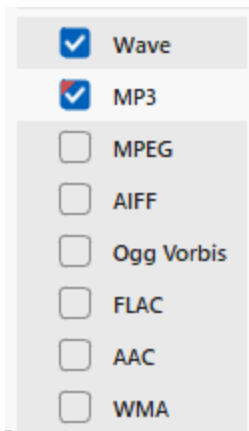
Select the file formats for export here. You can export to several of the available formats at once by enabling the corresponding option. You can select a single format with **double-click**. Click the format name to set the options for that format.

Sample rate and **Bit depth** can be selected for all formats in the corresponding menus. The options that are not available for a particular format will be grayed out.

You can use the option **Use current project sample rate** to create presets that are independent of the specific sample rate of the project.

Dithering: You can define your own dithering settings for each export, independently of the global settings. You can bypass dithering altogether (**No dithering**), apply the default dithering **Triang.**, or dither according to the system options. The button shows in brackets (e.g. **Triang.** or **POW-r 1**) the currently set dithering algorithm. Dithering Options opens the System-Dithering Settings (↗723).

Resampling quality: With this button you open the program settings for the resampling quality (↗726) during export.

With **Codec Settings...** you open the settings specific to the respective file format:

Wave

Wave (*.wav files) is the standard format for storing audio files on Windows computers and the recording format used by Sequoia. It allows bit depths up to 32bit (float) and any number of audio channels. Use this format for all files that should be further processed and should be in maximum quality.

Codec: Leave the setting at **uncompressed**, unless you want to use special compressing wave codecs, which you can select from this list. These codecs are from the early days of computer-based audio processing and are now only used in very rare special cases such as mobile telephony. When selecting these codecs, you can choose from several preset channels/sample rates/bite depth defaults that override the settings for these parameters in the Export dialog.

MP3

MPEG II Layer 3, abbreviated as MP3 (*.mp3 file) is a widely used file format for audio files that uses lossy compression.

Encoder quality: You can choose between a faster encoding and one with high quality. If time is not very tight, you can keep **High Quality**.

Bitrate: The bitrate of the compression can be set from 16 kBit/s to 320 kBit/s. At the highest setting of 320kBit/s, there are no longer any audible differences to uncompressed audio. However, if you plan to process the files later,

we recommend using uncompressed formats, since applying effects may cause artifacts resulting from the previous compression.

Variable bitrate: The bitrate is dynamically adjusted to the audio material, a lower bitrate is used in quieter places, a higher bitrate in more complex places. Therefore you can no longer specify a fixed bitrate, instead there is a setting for **quality**. A constant bitrate provides streaming capability and maximum compatibility; a variable bitrate provides better audio quality at the same file sizes. Not all playback programs can process VBR correctly, some will result in problems during title length display or when seeking.

Info text editor: MP3 files can contain metadata (title, artist, genre...). When exporting with **Split at markers**, the metadata from the CD track indexes is transferred to the files. This button opens the CD-Text/MP3-ID Editor (↗581) for editing the metadata.

MPEG
MPEG (*.mpg, *.mp2, *.mus, *.wav) files, more precisely MPEG-1 Audio Layer 2 is a lossy compressed audio file format. It is a precursor to the popular MPEG 1 Audio Layer 3 format, now known as MP3 for short. While MP3 is essentially used for Internet streaming and portable audio playback, MP2 is still used for digital broadcasting (DAB, DVB) and occasionally for DVDs.

Bitrate: The bitrate of the compression can be set from 64kBit/s to 384 kBit/s.

Stereo options: In addition to stereo and mono, there is also Joint Stereo, where common components of the channels are stored only once, allowing better compression at the expense of channel separation.

File format: Here you can select the container format used for the compressed data.

AIFF
AIFF (*.aif files), Audio Interchange File Format, is the standard audio file format on the Mac platform. It has nearly the same properties (lossless, bit depths up to 32bit float) as Wave. Use this format for data exchange with MAC audio software users.

> ⓘ *No special codecs are selectable, so the Codec settings button is not available.*

Ogg Vorbis
Ogg Vorbis (*.ogg files) is a royalty-free open source audio codec with very good sound characteristics in comparatively small files. It works, similar to MP3, with lossy compression. Not all hardware players can handle this format.

Bitrate: The bitrate of the compression can be set from 46 kBit/s to 500 kBit/s (for mono up to 240 kBit/s).

Variable bitrate: Constant bitrate provides streaming capability and maximum compatibility, variable bitrate provides better audio quality at the same file sizes. The selected bitrate is then only a guideline, the bitrate changes dynamically depending on the audio material.

Info text editor: OGG files can contain metadata (title, artist, genre...). When exporting with **Split at markers**, the metadata from the CD track indexes is transferred to the files. This button opens the CD-Text/MP3-ID Editor (↗581) for editing the metadata.

FLAC

FLAC (*.flac files) is the abbreviation for "Free Lossless Audio Codec". This is a royalty-free open source audio codec that lets you compress your audio to about 50% of the original size. Unlike other compression methods like MP3 or OGG, the full sound quality is kept intact with FLAC. Lossless compression algorithms similar to the zip or rar archives used for file transfers are used.

> 💡 *FLAC is the only compressed audio format that allows a bit depth of 24bit. It is therefore best suited for space-saving archiving of raw material from 24bit recordings.*

The **compression rate** can be set between 0 and 8. This does not affect the quality of the generated files (the compression is lossless), but only the speed of the export: the higher the compression is, the longer it takes.

Info text editor: FLAC files can contain metadata (title, artist, genre…). When exporting with **Split at markers**, the metadata from the CD track indexes is transferred to the files. This button opens the CD-Text/MP3-ID Editor (↗581) for editing the metadata.

AAC

AAC (*.mp4 files), Advanced Audio Coding is a lossy compressed audio file format primarily used in video and audio streaming and portable audio players.

Type: You can choose between different so-called profiles here. In most cases, for maximum compatibility with hardware players and streaming providers, you should keep the AAC LC (Low Complexity) profile. Only at very low bit rates and for special requirements is the use of HE-AAC/HE-AAC v2 (also referred to as AAC+ or AAC+ v2) worthwhile.

Bitrate mode: When selecting **Constant Bitrate** you can select a fixed bitrate between 64 and 480kBit/s at **Bitrate**, furthermore you can select different guide values for variable bitrates, furthermore you can find settings for different broadcast standards like DAB+.

Quality: Choose between different quality levels here, higher quality also increases the time required for encoding.

> ℹ️ **Note**: *For AAC export, the sample rate depends 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 AAC at 128kbit/s, you get a 44.1 kHz sample rate file. If, on the other hand, you export the same file at 256 kbit/s, you get a 48 kHz file.*

Info text editor: AAC files can contain metadata (title, artist, genre…). When exporting with **Split at markers**, the metadata from the CD track indexes is transferred to the files. This button opens the CD-Text/MP3-ID Editor (↗581) for editing the metadata.

WMA

Windows Media (*.wma files) is a Microsoft audio/video format optimized for streaming purposes. In the dialog, different, rather historical profiles can be selected for streaming in low bitrates. To create additional profiles you need the Windows Media Encoder 9 from Microsoft.

Info text editor: WMA files can contain metadata (title, artist, genre...). When exporting with **Split at markers**, the metadata from the CD track indexes is transferred to the files. This button opens the CD-Text/MP3-ID Editor (↗581) for editing the metadata.

The result of these settings is displayed at **Resulting Codec**.

Channels

Here you define how the stereo channels are treated during export. For some file formats only stereo is possible, then this menu is grayed out.

- **Stereo**: Both channels are output to a stereo file.

- **Left & Right**: The stereo channels are output separately in two files, **filename_L** and **filename_R**.

  > 💡 *The load option **Load L&R files as stereo** allows such file pairs to be automatically treated as stereo files when loaded by loading them into one stereo object.*

- **Left channel only**: The left channel is output to a mono file.

- **Right channel only**: The right channel is output to a mono file.

- **Mono mix**: The left and right channels are mixed together according to the formula **(L+R)/2** and output to a mono file.

- **Mono**: Select this setting if the project contains mono track pairs (panned alternately all the way to the left and right). Mono is formed according to the formula **L+R** with level reduction (panning law) of -6dB. This ensures that existing center-panned mono signals are also exported with the correct level.

- **Automatic**: Stereo tracks are exported stereo, mono tracks are exported mono.

Options

Realtime export: This option allows you to perform the export in real time.The project is played back so that the output signals of connected hardware effects (↗225) are included in the export.

Include master effects: Deactivate this option if the master effects should not be included in the export. As individual tracks are also exported through the entire effects chain including the master, you should deactivate this option if you want to export individual tracks without master effects as stems, e.g. for a remix.

Include track effects: Deactivate this option if the track effects are not to be included in the export.

Include AUX and submix buses: Deactivate this option to not include the influence of tracks that are routed to submix busses or that send to AUX busses when exporting them.

> ℹ️ *This corresponds to the behavior of older versions of* **Sequoia**.

If you export individual objects or glue them together (see above at **Track/object selection**), the track effects are automatically deactivated. If you also select the options **Create new objects** and **Replace object(s)** as **action after Export** (see below), the master effects and the AUX and submix bus effects are also deactivated so that the effects are not used twice.

Loudness and level analysis: This option allows you to display the loudness and the maximum level in dB after the export process.

Export marker file: If there are CD track indexes in the project, you can use this option to create a cue file (*.cue) that contains the track information. If the option **Split at markers** is activated at **Source**, a playlist file (*.m3u) is created instead.

Embed markers in audio file: All markers in the project are written to the audio files as audio markers. (only when exporting to Wave format)

Action after export: In this menu you can define what should happen to the created audio files after the export.

- ■ **Create new audio file**: The audio file or files will only be created.

- ■ **Open new wave project**: The audio file or files will be created and opened as wave projects in new windows.

- ■ **Create new objects in VIP**: The audio file or audio files are created and inserted as new objects in new tracks in the project.

- ■ **Replace object(s) in VIP**: The objects involved in the export are removed and the created files are inserted as new objects on the involved tracks.

- ■ **Create new VIP**: A new project is created and the new audio material is inserted.

Presets

The complete settings of the export dialog can be saved to presets.

Presets

AAC CBR 128 kBits 44.1 kHz
CD Album Tracks WAV 44.1 kHz 16 Bit
FLAC 48 kHz 24 Bit
MP3 CBR 320 kBits 44.1 kHz
MP3 CBR 320 kBits 48 kHz
Selected Objects WAV 44.1 kHz 32 Bit
Selected Tracks WAV 44.1 kHz 32 Bit
WAV 48 kHz 24 Bit
WAV CD 44.1 kHz 16 Bit

Add   Delete
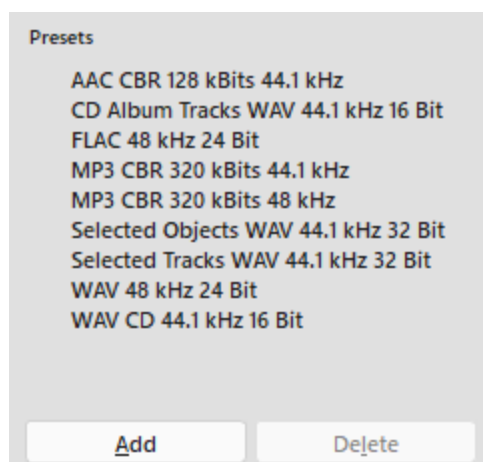
- ■ Click **Add** to add the current settings of the dialog as a preset.

- ■ To load a preset, select it. The export dialog takes over the settings of the preset.

- ■ To change a preset, load it, make the desired changes, right-click on it and select **Update with current settings**.

- ■ To rename a preset, double-click it and enter the new name.

- ■ To delete a preset, select it and choose **Delete**.

Each preset is included in the menu **File** > **Export with preset**, so you can perform an export with the settings of a preset without opening the export dialog.

🔅 *With menu **File** > **Export with previous settings** you repeat an export with the last selected settings.*

Use **Open export list...** to open the export list (↗671), in which you can perform multiple exports with several presets in one go.

Export List

With the export list, several projects can be exported one after the other using different export presets. For example, you can output different versions of a project as both stereo and surround mixes, or export different projects together.

ℹ *Note: For exporting a project to several different file formats (e.g. Wave and MP3) you can already define several file formats within an export preset in the section **Create**, for this you do not necessarily need the export list. However, the export list is useful when you need several files of the same file format with different settings (e.g. MP3s with different bitrates).*

You can access the dialog **Export list** via the menu **File** > **Export list...** or via the button **Open export list...** in the export dialog.
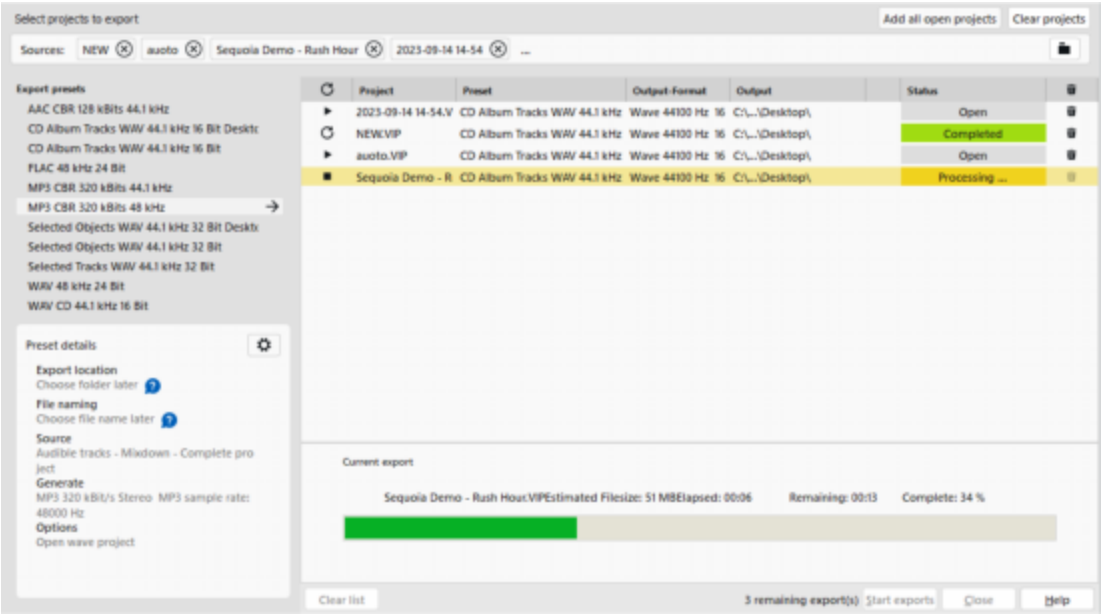
1. At **Sources**, click at **Select projects to export**. You can now define the projects for the export in the menu:
   - ■ Use **Add current project**/**Add all open projects** to add the current or all open projects as a source.
   - ■ With **Select project from hard disk...** you can add further projects via a file selection dialog.

- There is also a list of recently used projects in the menu, from which you can select projects for export by clicking on them. This list can be filtered by entering name components.

You can remove a project from the selection using the close field at the name (**x**). Use the recycle bin icon 🗑 on the right to remove all selected sources.

2. The list on the left shows all export presets. Move the mouse over the preset name to display a summary of the preset's settings at the bottom left of the dialog. To view the complete settings for this preset in an export settings dialog (↗661), click on the preset and then click on the gear icon at the bottom of the summary.

   ℹ️ *You can also make changes to the settings in this dialog and save these changes with* **Save preset**. *You have the choice of overwriting the existing preset or saving the changed preset under a new name.*

   Use **Export dialog...** to switch to simple export.

3. Click the arrow icon to the right of the export preset name to insert an export job with that preset in the export list for each source project.

4. You can now add more jobs with other presets.
   Individual jobs can be removed from the list by clicking on the trash can icon behind the job, and all jobs can be removed by clicking on the trash can icon 🗑 above the list.

5. When you have finished creating the export list, start the exports with **Start exports**.

   ⚠️ *Attention: The export list is only processed without interruption if no presets are used that require user input. This occurs if you have specified the options* **Choose file name later** *or* **Choose folder later** *in the preset for the output folder or the file name. Such presets are marked with a warning symbol ⚠️ in the summary and in the list. Click on the icon next to an export job to specify the storage location for this export. Such jobs are then marked with a check symbol ✓ . You can click on this icon to change the path specification again before executing the job.*

   Individual export jobs can be started with the play icon at the list entry ▶ .

6. The export jobs remain in the list even after the export until the end of the program session. So you can close the list and continue working on the projects involved. You can open the list again later and carry out individual or all exports again. To keep track, you can reset the status of a job using the repeat icon ↻ in front of the individual job or at the top in the column header.

Other File Formats

In addition to loading and saving project files and importing and exporting audio files, Sequoia supports a number of other formats.

MIDI Files

> 💡 *MIDI files can be loaded more conveniently like all other media files and projects from the* File Manager (↗318), *where you have an automatic preview function, full text search, favorite folders and more.*
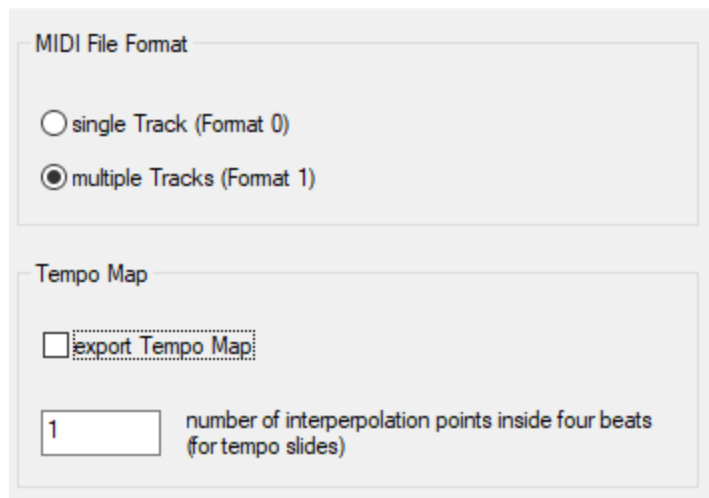
Loading MIDI Files

Load standard MIDI files (*.mid) with menu **File** > **Import** > **Load MIDI file...**.

For more information on MIDI import, please also refer to the chapter **MIDI in** Sequoia (↗417).

Exporting MIDI Files

With menu **File** > **MIDI Export** > **Export MIDI File...** you export the entire MIDI data of a project into a MIDI file in Standard MIDI File (SMF) format. All changes of the MIDI playback by the track (MIDI section in the track header) and in the MIDI object editor, like transposition or realtime quantization, are taken over into the MIDI file.

Select the desired **MIDI file format**: format 0 exports all MIDI data to one track, format 1 saves several tracks in the MIDI file corresponding to the tracks of the project. The track names are adopted.

With **Export Tempo Map** the tempo map is written to the MIDI file. Since the MIDI file format does not support the continuous tempo progressions from Sequoia, such progressions must be simulated in the MIDI file by many individual tempo entries. You define the accuracy for this by specifying the **Number of interpolation points per bar**.

The export is always done with the PPQ resolution of the current VIP. The project markers are also exported in the process.

Exporting MIDI Objects

With Menu **File** > **Export MIDI...** in the MIDI Editor (↗423) you can export the MIDI data of a single MIDI object to a file.

With menu **File** > **MIDI Export** > **Export MIDI objects as individual MIDI files...** you export the MIDI data of selected objects into single MIDI files. If multiple objects are selected, the object names of the MIDI objects are always used as file names.

> ℹ️ *Note that when exporting individual MIDI objects via these functions, changes to MIDI playback by the track (MIDI section in the track header) and in the MIDI object editor, such as transposition or real-time quantization, are not adopted. Drum maps (↗438) of the tracks are transferred, however, that is, the pitches are changed according to the maps.*

MIDI Mixdown

Use Menu **File** > **MIDI Export** > **MIDI Mixdown...** to combine selected MIDI objects into a single new MIDI object. This applies all track and object MIDI effects, just like MIDI export. You can choose to mixdown the selected objects in a selected range or in the whole project.

The new MIDI object is inserted on a new track above the selected track. The objects involved - unlike the equivalent function Range Mixdown (↗127) for audio - are not removed from the project, you have to delete them manually.

> ⚠️ *You should only merge MIDI objects from tracks that control the same instrument, since this assignment is track-dependent and will be lost when you mix down to a new track. You must then assign the instrument to the new track again (↗374).*

Sequoia **Sessions**

In a **Session** all opened projects can be saved together with their respective window positions. For complex projects that include multiple project files, you can use a session to load all the required projects and their arrangement at once when you resume your work later.

> ℹ️ *The use of Sessions is particularly useful for the* Source Destination Cut *(↗201) with multiple source projects. At* multi-synchronous cut *(↗205), a session is automatically created with the project files involved*

- Save a session with menu **File** > **More** > **Save Session...**.

- Open a session with menu **File** > **Open** > **Session (*.samx)...**.
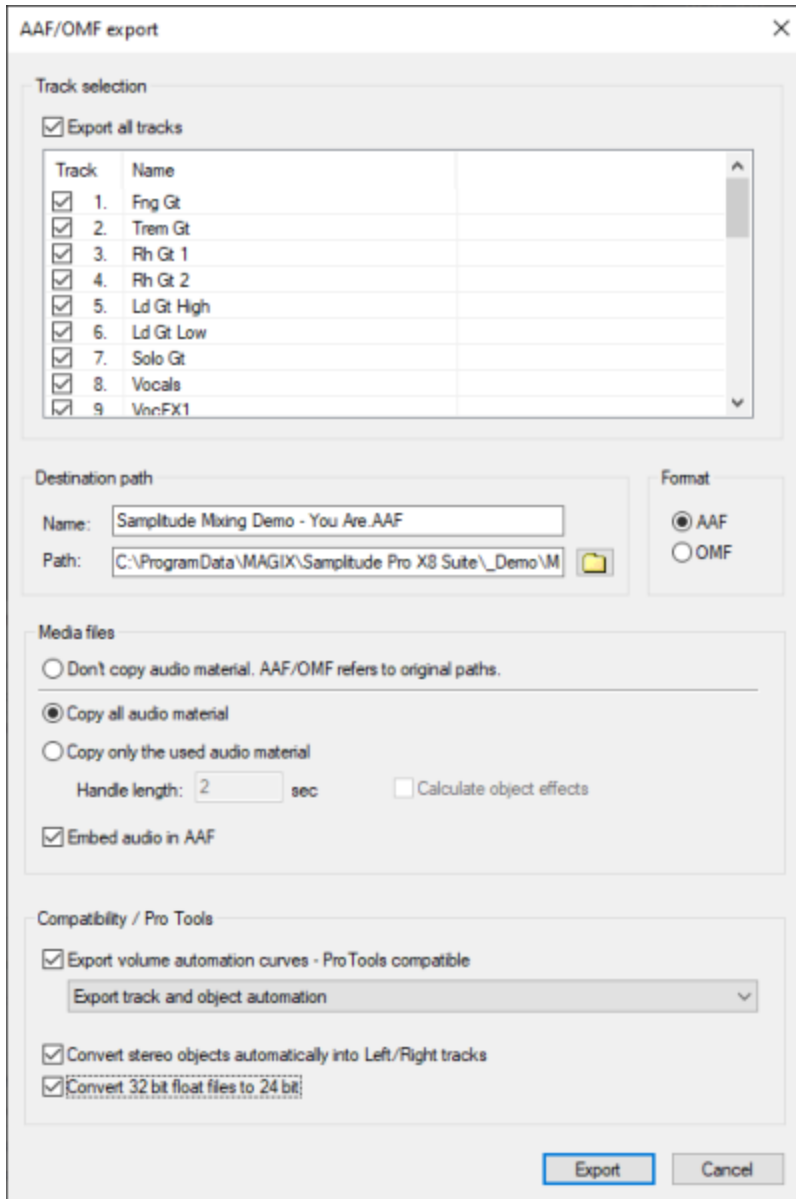
> ℹ️ *You can open sessions in *.sam and *.samx formats, always saved in the newer *.samx format.*

AAF/OMF

AAF is a set of specifications for project interchange (.aaf) files. Media files can be embedded or referenced by link. AAF files can have envelope information, static level information or both. **OMF** is the precursor of the AAF format that could be used for project exchange with very old media applications but it loses track names, markers and automation information during export. It also supports mono files only.

Exporting Project as AAF/OMF

Export the active project with menu **File** > **Advanced export** > **Export project as AAF/OMF…**.

This Picture shows the most compatible export settings for transferring a project to any DAW

The AAF Export transfers the following content:

- Object position/wave offset
- Object fade in/out - linear fades only
- Object crossfade - linear crossfades only
- Object volume

    ℹ️ *Note: ProTools clips all object volume settings over 12dB to a maximum of +12dB.*

- Object pan (not imported into ProTools)
- Object volume automation
- Track pan automation (not imported into ProTools)
- Track names
- Track Volume
- Track pan (not imported into ProTools)
- Track volume automation
- Track pan automation (not imported into ProTools)
- VIP markers (marker names not imported into ProTools)
- Timecode offset
- Timecode format (24, 25, 30 fps)
- Processing timestamps in audio files

OMF Export transfers the following content:

- Object position/wave offset
- Object fade in/out - linear fades only
- Object crossfade - linear crossfades only
- Object volume
- Track names

ℹ️ *Note: During AAF/OMF export, media files will always be converted to WAV files. You can't export video files. Corresponding objects are not transferred as well.*

Track selection: Select the tracks of the project to be exported. With **Export All Tracks** you select all tracks with one click.

Name/Destination path: Specify here where you want to export the AAF/OMF file and under which name.

Format: Select here if you want to export the file in AAF format or OMF format.

Media files:

- **Don't copy audio material. 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.

- **Copy all audio material**: If you select this option, all media files (including those that are only referenced in the Clipstore) will be placed next to the AAF/OMF into the target folder.

- **Copy only the used audio material**: If you select this option, only the parts of the media files that are actually used in the VIP are copied into the target folder.

  - **Handle length**: Set an additional length of audio to the exported audio files. This is useful if you want to adjust the crossfades of the objects in the new project.

  - **Calculate object effects**: All effects applied to objects are included in the new audio files.

- **Embed audio in AAF**: If you select this option when exporting, the complete project will be saved as one file containing both the media and the metadata.

> ℹ️ **Note**: *Copy the audio material or embed it in the AAF file if you want to give the entire project along with the associated media files to another studio as AAF/OMF.*

Compatibility/ ProTools

- **Export volume automation curves - Pro Tools compatible**:

  - **Off**: Object Volume is recognized as ProTools clip gain. The automation curves of the track volume and object volume are ignored.

  - **Export track automation only**: If you select this option, only the track volume automation will be exported. The object volume automation curves are ignored, but the static object volume are recognized as ProTools Clip Gains.

  - **Export object automation only**: If you select this option, only the object volume automation is exported into Track volume automation to be imported into ProTools. The static object volumes are recognized as ProTools Clip Gains. Track volume automation is ignored by ProTools.

  - **Export track and object automation:** If you select this option, track volume automation, object volume automation and static object all will be combined into the volume automation curve for Pro Tools. The static object volumes are thus not reflected as ProTools Clip Gains .

- **Convert stereo objects automatically into Left/Right tracks**: Stereo objects are not supported by Pro Tools. With this option active, those objects are converted into left/right mono objects and placed on two mono

tracks panned accordingly.

- **Convert 32bit float files to 24 bit**: As AAF uses only 24bit audio files, 32bit float files are converted to 24bit with this option on.

ⓘ *Note: Timestamps saved with ProTools in Frames for improved compatibility.*

Import AAF/OMF

Import AAF files with menu **File** > **Import** > **Import AAF/OMF...**.
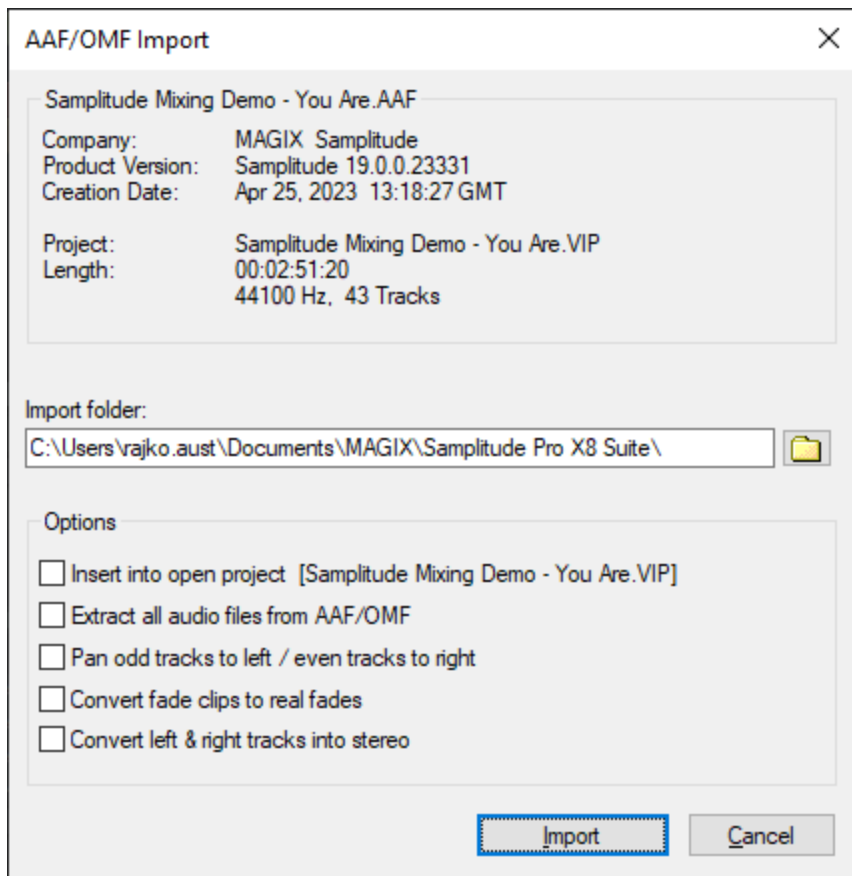
The following content is imported from **AAF** files:

- Objects: Position/Offset, Fade-in/-out (linear only), Crossfades (linear only), Volume, Pan, Volume automation, Pan automation
- Tracks: Name, Volume, Pan, Volume automation, Pan automation
- Project markers
- Timecode: Offset, Format (24,25,30fps)
- Timestamps of the audio files

The following content is imported from **OMF** files:

- Objects: Position/Wave Offset, Fade-in/-out (linear only), Crossfades (linear only), Volume
- Tracks: Name

Options:

- **Insert into open project**: Normally a new Sequoia project is created from the AAF/OMF file. This option instead inserts the content into the active project as new tracks below the existing tracks.

- **Extract all audio files from the AAF/OMF**: If the media data is embedded in the AAF/OMF file, it will be extracted during import and saved as wave files in the project folder.

- **Track Panorama - Odd/even track to left/right**: It is assumed that the tracks of the imported project are present as alternating left and right mono tracks and the panorama of the tracks in the created project is adjusted accordingly.

- **Convert fade clips to real fades**: Pro Tools uses a special format for clips with fades when saving AAF, where the fades are rendered into separate audio files. In this way, Pro Tools implements special fade curve shapes that the AAF format does not support. Such fade objects therefore also create separate objects for all fades and crossfades when imported into Sequoia. This option converts such fade clips into normal objects with fades. Activate the option if you import AAF files that come from Pro Tools and you do not need these special fade curves. This way you can reduce the number of objects and create a clearer project.

- **Convert left/right tracks to stereo**: If the tracks of the imported project exist as alternating left and right mono tracks, stereo tracks are created for superimposed clips, on which stereo objects (↗164) are located.

Edit Lists (*.edl)

An Edit list is a text file and contains information about the used audio files, video files, object boundaries, object volumes, markers, volume as well as panorama curves of a project. It can be used to transfer Sequoia projects to or from other programs.

It does not contain the program specific features of the project like object effects, AUX routings or fade curve shapes, but already much more details than for example an export and import of pure single tracks ("stems").

Save the active project as an Sequoia Edit list with menu **File** > **Advanced Export** > **Export project as EDL...**.

At **File type** you can select the CMX 3600 compatible format in addition to the Sequoia own format **Sequoia EDL**. With the **Compatibility** option, you can create cut lists that are compatible with Premiere Pro, DaVinci Resolve, Media Composer, and Pyramix.

Open Edit lists with menu **File** > **Open** > **Edit list (*.edl)...**.

A new virtual project that resembles the EDL is created. The media files used should be in the same location as at the time of EDL export. If not, there will be an error message. You can then either cancel the import or use **Search** to

navigate to a media file that was not found via a file selection dialog. The remaining media files will then be found automatically if they are located in the same folder.

CD Projects

Via menu **File** > **Open** you can load **TOC files** and **DDP Images** which describe the content of an Audio CD:

■ **Table Of Contents (*.toc)**: Table of Contents (*.toc) files describe the contents of an audio CD: the wave project used and the position of the track markers and track names. When loading a TOC file, a new virtual project is created, the audio file is loaded and the track markers are inserted.
TOC files for your own CD projects can be saved in the **Create CD dialog** (↗584).

■ **CUE files (.cue)**: Load CUE files via menu **File** > **Import** > **Load audio file...** into an existing virtual project. CUE files together with a wave file also contain the complete data you need to create an audio CD Wave files with a .cue file can also be used by other programs to burn wave files to CDs. To create an additional .cue file, activate the option (↗669)**Export markers into file** during file export and export the project as a single large file, i.e. with the option **Split at markers** (↗665) inactive.

■ **DDP Import File**: DDP is an exchange format that can be used to transfer an audio CD as binary data to a pressing plant. When loading a DDP image, a new project is created with the associated track markers and audio data. All files belonging to a DDP image should be located in one folder for this purpose. For more information, read the section DDP Export (↗588)!

Objects

Objects can also be saved separately. An object file (*.obj) contains all the settings of the object editor (↗183) such as fades, effects, object start and end, but not the actual audio data. The audio file referenced by the object must be in the same location when the object is reloaded.

■ Save a selected object with menu **File** > **More** > **Save object..**

■ Import an object into the current project at the play cursor position on the selected track with menu **File** > **Open** > **Session (*.samx)...**.

Obsolete Formats

Via menu **File** > **Open** files can be opened in further formats, which originate from the early days of Sequoia:

■ **RAM wave project (*.rap)**: This format is only used for loading very old projects. RAM wave projects are loaded directly into RAM.

- **HD Wave Project (*.hdp)**: HD wave project is also a "historical" term that, as opposed to RAM wave projects, was used for wave projects where the audio data is played back from the hard disk. Every wave file automatically becomes an HD wave project, because the .hdp file is automatically created as soon as a wave file is loaded or recorded into Sequoia. However, with this menu option here you can also load very old HD Wave projects where the sample data was saved in a proprietary file format before *.wav was used for it.

In the menu **File** > **Advanced Export** > **Convert Mono/Stereo** there are still commands for converting **LR-Waves**. These were special HD Wave projects, where the sample data of a stereo recording is in two separate mono wave files.

- **LR-Wave > 2 Mono**: Divides one LR-Wave project into two independent mono wave projects.
- **LR-Wave > 1 Mono**: Mixes the channels of a LR-Wave project together into a mono wave project and lowers the level by - 6dB to avoid overloads.
- **2 Mono > LR-Wave**: Connects two mono wave projects to one LR-Wave project or stereo wave file.
- **1 Mono > LR-Wave**: Converts a mono wave project to an LR-Wave project with two identical channels.

Clean up & Backup

In the submenu **Clean up & Backup** in the menu **File** you will find the functions to collect complete projects with all the required files in the project folder, delete any data in it that is no longer needed and save the complete project to a folder or burn it to disc.

Collecting Project Files

In the course of working on a project, besides the files directly recorded in the project folder, media files from various other folders may have been used in the project. To archive projects or share them with others, you can use the function **Collect project files** to ensure that all media files required by the project are located in the project folder.

With Menu **File** > **Clean up & Backup** > **Collect Project Files...** you open a dialog that lists all files that are outside the project folder.
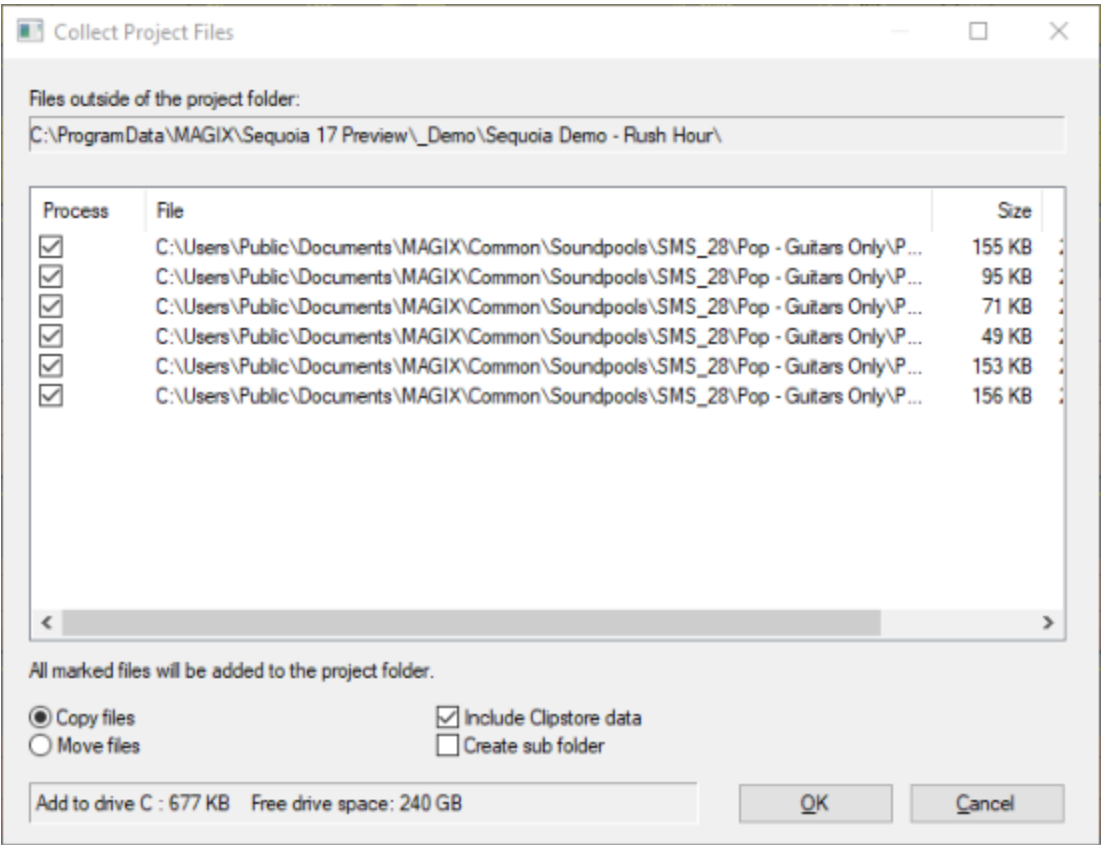
Use **Copy files** or **Move files** to select whether to keep the files in their original location or move them to the project folder. By default, all files are selected to be copied to the project folder. You can exclude individual files from processing by deselecting them at **Process**.

So if there are files that you want to use later for other projects (e.g. files from sample libraries) and files that are only used within the project, execute the function twice, for each of the files with the appropriate setting.

If you enable the **Create subfolders** option, subfolders will be created in the project folder for the files copied to the project folder.

If you enable the **Include Clipstore data** option, the files in the Clipstore (↗311) are also included.

Deleting Unneeded Files

To save hard disk space, the following functions can be used to delete data that is no longer needed in the project:

Delete virtual project (VIP)

This function allows you to delete projects completely, including the media files they use. Select **File** > **Clean up & Backup** >**Delete Virtual Project (VIP)...**.
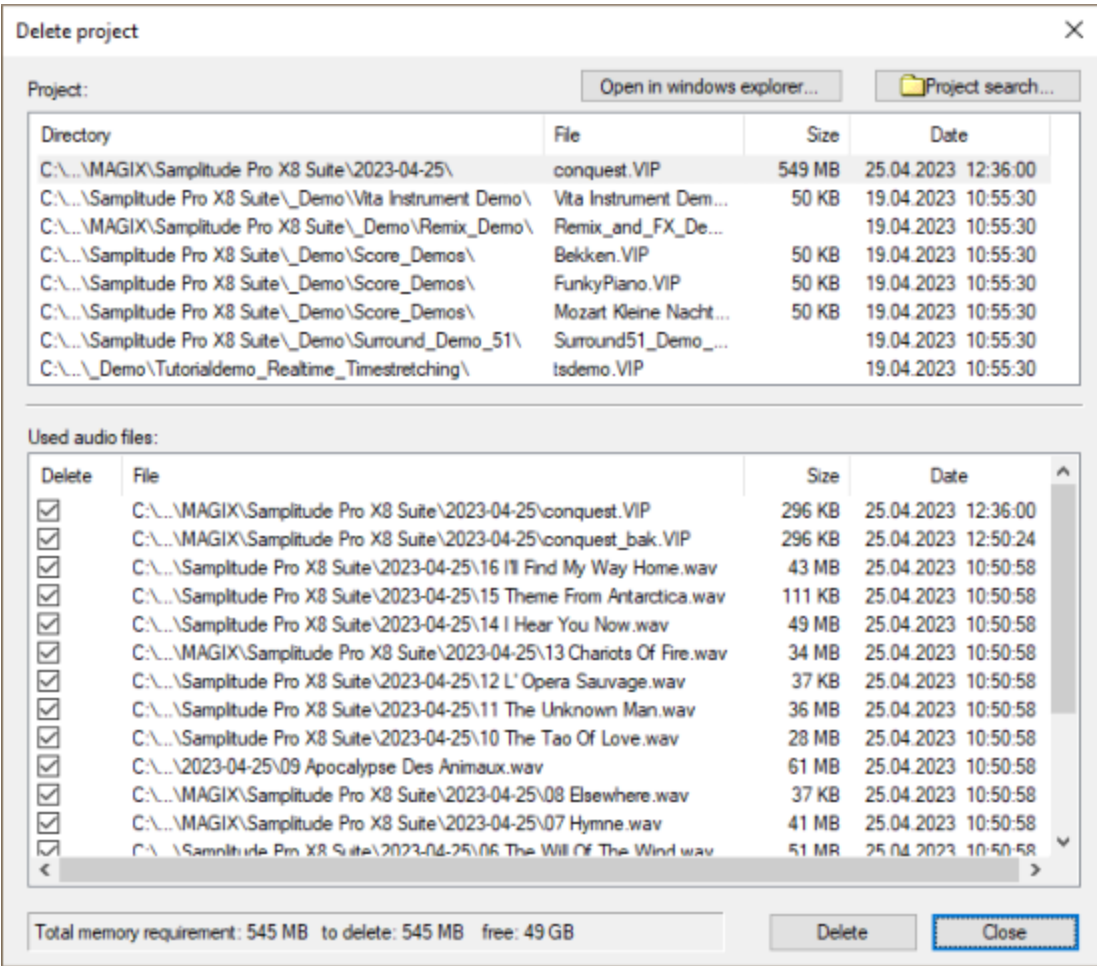
The upper section of the dialog lists all projects (*.vip) located in the default folder for projects (↗706) (including subfolders). Select the project to delete from the list. With **Open in Windows Explorer** you open the project folder of the selected project in Windows Explorer. Projects that are located outside the default folder can be located via **Find Project...**.

In the lower section of the dialog all used files of the project are listed and can be excluded from deletion by unchecking the checkbox in the column **Delete**. If you now click on **Delete** at the bottom, all files that will be deleted will be listed once again. Confirm the final deletion with **Delete files**.

⚠ **Attention**: *There is no additional security prompt, the files are now deleted immediately.*

The **Delete Virtual Projects** dialog remains open so that you can delete more projects.

Delete wave project

This function allows you to delete individual wave projects from the hard disk. All files belonging to a wave project (*.wav, *.hdp, *.h2...) will be deleted together.

Remove unused samples

This function deletes all unused sections from the audio files of the active virtual project, i.e. the sections in the audio files that are not used by objects in the project and therefore are never played. The objects in the virtual project are adjusted accordingly.

⚠️ *This function permanently deletes data from the hard disk and has no "Undo" option. Therefore, use it wisely.*

If other virtual projects use the same audio files, to be on the safe side, open those projects as well before performing this function to prevent possible data loss.

This function can be used to save hard disk space, but afterwards length corrections of the objects are only possible to a limited extent, because the audio data outside the object boundaries have been removed. For this reason, you can specify a safety reserve for each object at **Save additional samples for each object**. This reserve is additionally left in the audio material before and after the object boundaries. The default value is 22050 samples , which corresponds to 500ms at 44.1 kHz sample rate.

Select **File** > **Clean up & Backup** >**Remove Unused Samples...**.

The dialog lists all audio files used by the virtual project. The **Size** column shows the total amount of disk space used by the audio files, and next to it, in the **Unused** column, how much of it is not used in the virtual project. Use the checkbox in the column **Process** to select the files to be processed. Files that contain unused audio data are already selected. Uncheck the ones you want to keep completely.

> ⚠️ ***Attention***: *The list can also contain files that are currently no longer used in the VIP, but still appear as entries in the **Undo history**. These files contain 100% unused samples and would therefore be deleted completely. This can be intentional: For example, if you have completely discarded a recording session, you can thus also delete these unused files. However, if you had opened audio material from other sessions or your own sample library and did not use them later, such files are now also selected for deletion. Therefore, before using the function, clear the* Undo history *(↗553) and close all unused audio files.*

> ℹ️ *The same function is also executed when archiving a complete project with menu **File** > **Clean up & Backup** > **Save complete VIP in...**, if you activate the option **Copy only samples used in VIP** there. Then only the audio data that the archived project actually needs will be copied to the new folder.*

Delete Freeze Data

Use the function **Freeze data** to delete unused data that was created when freezing tracks (↗139) or objects (↗163) but is no longer needed after "unfreezing".

Archiving Projects

You can save a project with all used files to a separate folder or burn it to disc.

Save complete VIP in

The complete project with all used media files will be saved to the specified folder. Use this function for backing up projects or if you want to continue editing a project on another computer with Sequoia.
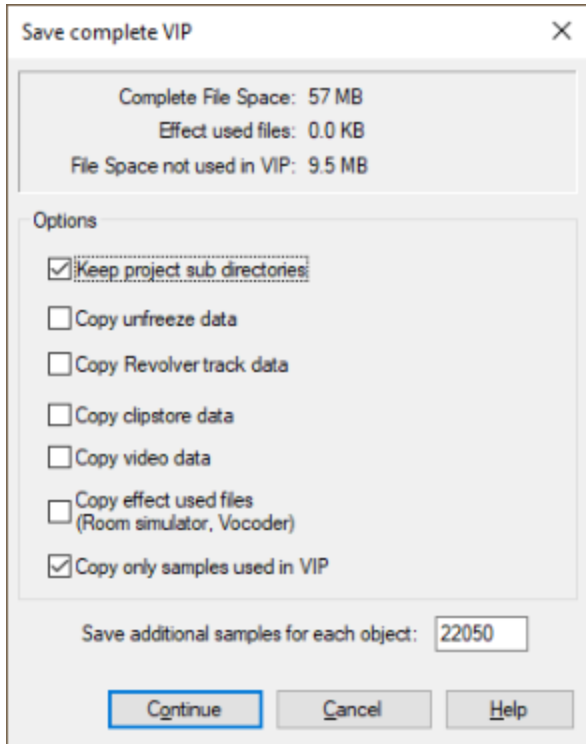
Setting the following options will copy more files to the new folder:

- **Keep project subfolders**: If there are subfolders in the project folder, they will also be created in the new folder.

- **Copy unfreeze data**: All data will be copied to the new folder that is needed to continue editing and undoing object or track freezes.

- **Copy revolver track data**: All Revolver tracks (↗137) (track variants) are copied as well.

- **Copy Clipstore data**: Copies all files from the clip manager with.

- **Copy video data**: Copies associated video files.

- **Copy files used by effects**: Files used by effects like room simulator or vocoder are copied as well.

If you activate the option **Copy only samples used in VIP**, only the parts of the audio files that are also used by objects in the project will be copied. With this function you can save memory space. However, by doing so, you accept that the object boundaries in the new project can no longer be moved outward, since all audio data outside the object boundaries is no longer available. Therefore you can use **Save additional samples for each object** to specify a safety range in samples that should additionally remain before and after the object boundaries in the audio data, in order to have a reserve if object fade-ins/fade-outs should be changed afterwards. The default value is 22050 samples , which corresponds to 500ms at 44.1 kHz sample rate.

Burning Project Backups to CD/DVD

This function allows you to burn complete projects onto multiple CDs or DVDs. The separate burning program **MAGIX Speed BurnR** is used for this purpose.

> ℹ️ *Of course, you can also burn the project folder to disc using the Windows burn function. However, you must then use the function **Collect project files** to ensure that all the files used are also in the project folder. In addition, using **MAGIX Speed BurnR** has the advantage of being able to burn project backups to multiple discs.*

PROGRAM PREFERENCES

All settings dialogs of the program are combined in one large dialog, from which you can also reach any other settings dialog via the tree structure on the right. The fastest way to open it is to use the keyboard shortcut **Y** or also menu **File** > **Program preferences** > **System Options...**.
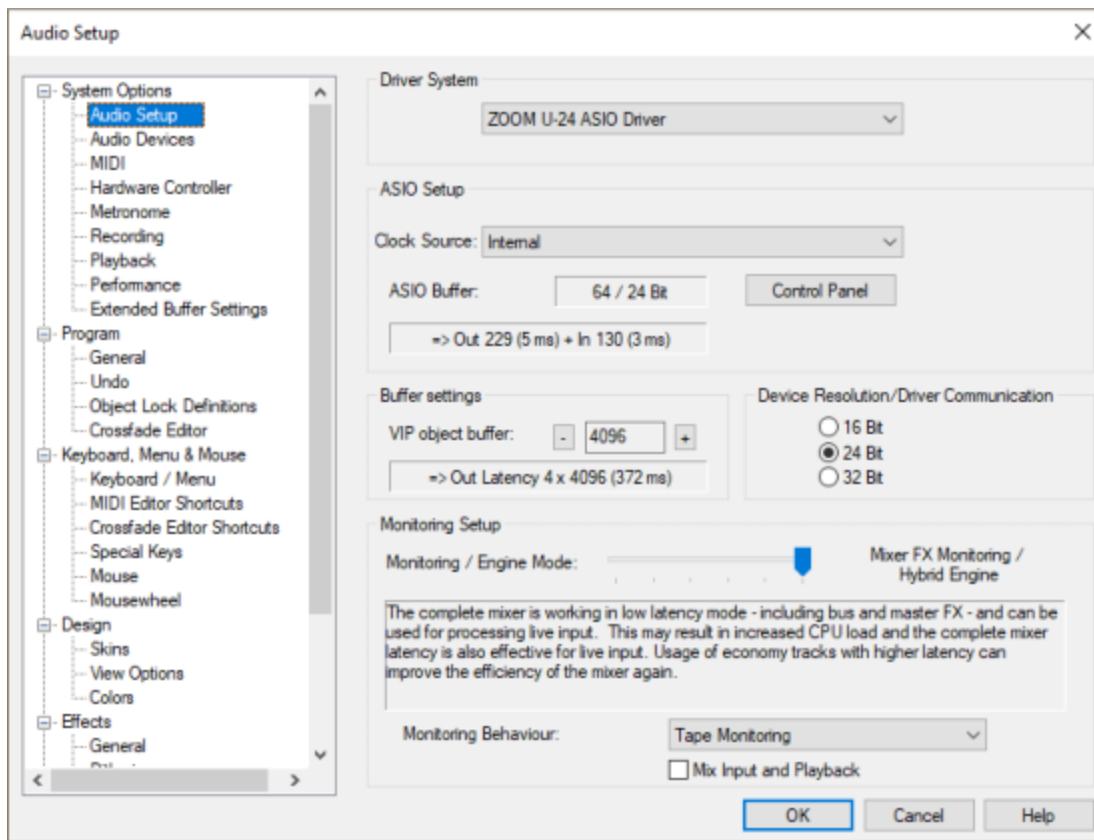
ℹ *In the menu **File** > **Program preferences** you will find a whole series of menu items that allow you to directly access a specific settings dialog (e.g. **Recording**, **Playback**, **View options** etc.). In many other menus, too, there are entries for certain settings dialogs that match the corresponding context, such as for the dialog **View options** in the menu **View**. For the sake of clarity, these many ways to get to the settings are not all mentioned individually here.*

Via the lowest entry **Project options** in the tree you switch to the tree structure with the project-specific settings.

> ℹ️ *Note: From version 17, unlike in older program versions of Sequoia, the changes to the program settings are saved immediately and not only when closing the program. This has the advantage that these changes are preserved even if the program terminates irregularly, for example if it crashes or freezes. However, if the crash was caused by these very changes, in rare cases this could cause the program to crash again every time it is launched. For this reason, the previous program settings are stored in backup files* `(Sequoia_64_backup.ini, Sequoia_64_audio_backup.ini...)`. *To restore the previous program settings, proceed as follows:*
>
> 1. *Exit the program.*
> 2. *Go to the program folder in Explorer and double-click on the link* `_Programdata.lnk` *to go to the program data folder* `(c:\ProgramData\MAGIX\Sequoia 17\)`
> 3. *Rename all* `*_backup.ini` *to* `*.ini` *files, thereby overwriting the current settings back to the backed up old settings.*
> 4. *Restart the program, the old settings are valid again.*

Resetting Program Settings to Default Settings

To reset all program settings to the default settings, go to the menu **Help** > **Reset program settings to default**. The current settings are saved in the file `BackupSetting.INZ` and can be restored via the Start selection dialog (↗31) or the **Load and save settings function** (↗730).

> ℹ️ *To apply the changed settings, restart Sequoia.*

System Options

Audio Settings

Here you make all settings for driver system, buffers, driver communication and monitoring.

Driver system: For the communication between Sequoia and your sound card you have the choice between the driver systems ASIO, MME and WDM. To take advantage of all the benefits of the program, we strongly recommend using ASIO.

MME is the old "Windows" multimedia driver system with the greatest compatibility, somewhat newer is the **WDM** driver system. These driver systems are suitable for performance-critical recordings with a very high number of tracks; there is then greater security against drop-outs compared to ASIO drivers. However, the following applies to both driver models: There is no possibility of monitoring, unless it is provided by the sound card on the hardware side. Multitrack recordings with multiple sound cards cannot be synchronized.

ASIO: Use a sound card model fitted with ASIO drivers if possible. This has some advantages over the MME/WDM driver system:
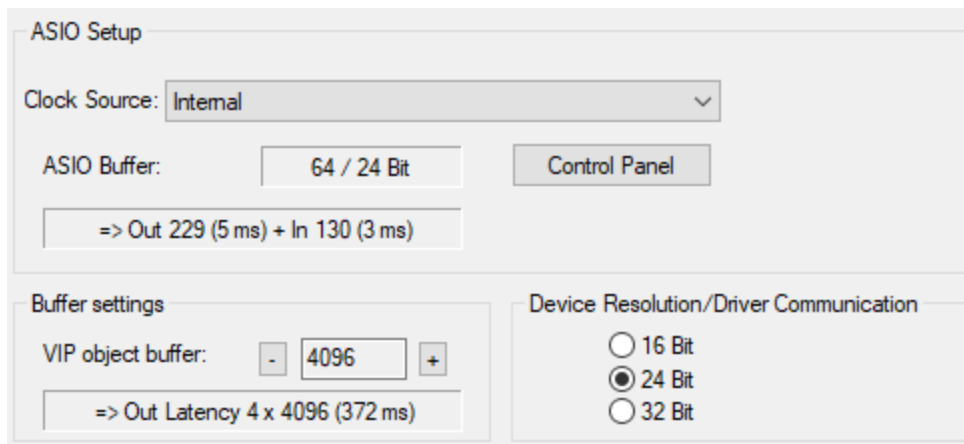
- There is less latency (input/output delay). This gives you the ability to use software monitoring of inputs and VST instruments, and to record software instruments live.
- ASIO is suitable for multitrack recording with multiple sound cards using the same ASIO driver. The sound

cards are synchronized by the ASIO driver.

■ Advanced hardware monitoring options are also available by using ASIO Direct Monitoring.

💡 *For sound cards without an own ASIO driver, but which support WDM, you can use a universal ASIO driver like the included MAGIX Low Latency driver or FlexASIO. These drivers also allow multiple sound cards to be addressed with one driver, but experience has shown that this is not very stable and does not perform very well.*

ASIO Settings

Driver System: Select the driver to use from the list of ASIO devices installed on the system. Use the **Control Panel** button to open the driver's settings dialog. Next to **ASIO Buffer** in the display field you can see the buffer size and bit resolution set in the driver. The output and input latencies are also displayed.

The entries in the menu **Clock Source** are provided by the sound card driver. There some sound cards offer the possibility to synchronize the sound card to an external clock.

Buffer settings: At **VIP object buffer** the buffer size for the Economy Engine is set. In the preset Monitoring/Engine mode **Mixer FX Monitoring/Hybrid Engine** it is used only for playback of objects and economy tracks (see below), its size does not matter for recording and monitoring latency. Only when changing the playback position or starting the playback, smaller buffer sizes result in a faster reaction of the program. In the other monitoring modes, processing of track effects also takes place in this engine, so the buffer size plays a role in the playback delay (latency) there.

The VIP object buffer size should be larger than the ASIO buffer size and between 1024 samples and 8096 samples.

Device Resolution/Driver Communication: Here you can select the bit resolution for addressing 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 adopts the bit resolution set in the ASIO driver. The setting determines whether the data received from the sound card is converted to a different bit resolution, performing a dithering (↗723). Often 32 bit data transmission is dithered to 24 bit or 16 bit.

The 32-bit setting is particularly recommended if no dithering is desired or if the input is not via the sound card's A/D converters but directly digital from a DSP card.

With the MME driver system, however, **Sequoia** opens the driver with the bit depth set under Devices/Driver Communication. If the device cannot process the set bit resolution, a correspondingly lower resolution (with dithering) is generated and the MME driver is used with this resolution.

Monitoring Setup

Monitoring refers to listening to the recorded input signals, for example via headphones during recording in the studio. Recording with monitoring including track effects or playing software instruments via MIDI requires processing audio data with low latencies.
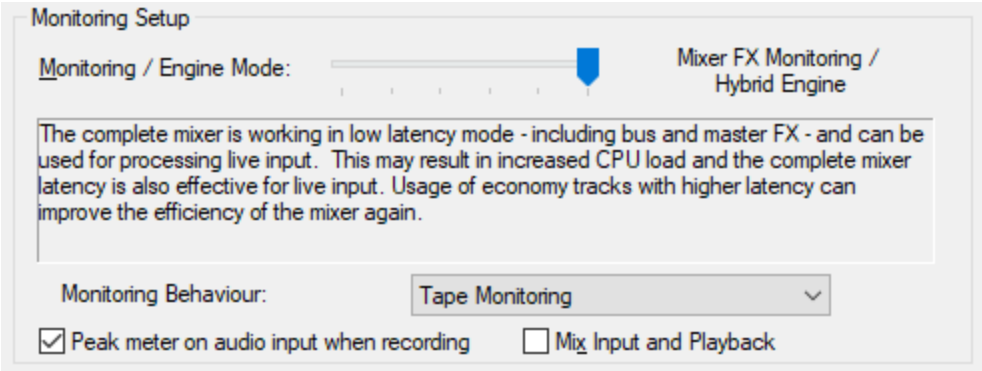
As long as there are no performance problems, you can leave the setting at the default **Mixer FX Monitoring/Hybrid Engine**. When the computer is pushed to its performance limits by many tracks, effects, and software instruments, you can choose various other monitoring/engine modes from the slider that will lighten the load on the computer while limiting those real-time capabilities. Also when using a driver model other than ASIO and if the sound card offers monitoring on the hardware side, it may be necessary to change the monitoring mode.

> ℹ️ *For background information on an audio engine in general and the Hybrid Engine in particular, read the section* Hybrid Audio Engine *(↗695)!*

Explanations for the selected monitoring modes are displayed in the text field below the slider, and a table also opens showing which engines are used for the various audio paths and how the currently selected engine mode affects the latencies.

Monitoring Modes

- **No audio monitoring (peak meter only)**: Display of input levels, but no monitoring. This monitoring setting is only useful if you use an external mixer for audio monitoring.

Input signals sent to tracks and VST instruments played via MIDI input cannot be monitored. Objects of the tracks, VSTi controlled via MIDI objects, busses and masters are played back via VIP object buffers with higher latency.

■ **Hardware Monitoring**: Audio monitoring through the sound card. This is the only possible monitoring type with MME drivers. Some sound cards also offer hardware monitoring under ASIO and can then directly process functions like mute/solo, volume and pan. You cannot apply effects to the input signal.
Input signals applied to tracks cannot be monitored via Sequoia, but only directly via the sound card if it offers the corresponding function.VST instruments played via MIDI input can be monitored with low latency. Objects of the tracks, VSTi controlled via MIDI objects, busses and masters are played back via VIP object buffers with higher latency.

■ **Software Monitoring/Economy Engine**: (only with ASIO drivers) Audio monitoring while taking the set levels of the recording tracks into account and when recording software instruments. No effects are applied to input signals.
Input signals applied to tracks and VST instruments played via MIDI input can be monitored with low latency. Objects of the tracks, VSTi controlled via MIDI objects, busses and masters are played back via VIP object buffers with higher latency.

■ **Track FX Monitoring**: (only with ASIO drivers) Audio monitoring including the track effects of the recording tracks. However, no bus or master effects are included.
Input signals applied to tracks and VST instruments played via MIDI input can be monitored with effects in low latency. Objects of the tracks, VSTi controlled via MIDI objects, busses and masters are played back via VIP object buffers with higher 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.
Input signals applied to tracks cannot be monitored via Sequoia, but only directly via the sound card if it offers the corresponding function. Busses and masters and VSTi played via MIDI input or controlled via MIDI objects can be monitored with low latency using effects. Objects of the tracks are played back via VIP object buffers with higher latency.

■ **Mixer FX Monitoring/Hybrid Engine**: The Hybrid Audio Engine allows audio monitoring through the entire mixer, while also calculating track playback in the mixer with short latency.
Input signals of tracks and the output of busses and masters and VSTi played via MIDI Input and controlled via MIDI objects can be monitored with effects at low latency. Objects of the tracks are played back via VIP object buffers with higher latency.

ℹ *Playback with higher latency does not mean that there will be delays between the tracks, because the latencies are known in the program and are automatically compensated, so internally the playback starts earlier by the corresponding time offset. This also allows the use of effects that add latency independently of the engine latency (e.g. dynamic effects with preview or FFT-based effects).*

The following applies to all monitoring modes: The effects of the recorded track are not recorded. However, depending on the monitoring mode used, you will hear the recorded signal with effects during recording and playback.

If the CPU load on your system becomes too high in **Mixer FX Monitoring / Hybrid Engine** mode, select a monitoring/engine mode that best suits your needs while not overloading your system.

If, for example, you want to do without effects monitoring for input signals on principle and completely, software **Monitoring / Economy Engine** could be the right monitoring mode for you.

If you cannot do without monitoring track effects of the incoming input signals, but can do without bus or master effects in the monitor path, then we recommend the monitoring mode **Track FX Monitoring**.

If you monitor the track inputs via your sound card, choose between the monitoring modes **Hardware Monitoring** or **Hardware Monitoring / Hybrid Engine**, depending on the performance of your system.

ℹ **Note**: *If you use "TotalMix" from RME in combination with a hardware monitoring mode, then set the panning law in "TotalMix" to -6dB. This ensures that the recording levels in Sequoia correspond to the monitoring levels in "TotalMix".*

Economy Tracks

If you have selected the Hybrid Engine as the monitoring setting, you can take individual tracks out of the Low Latency Engine and have track effects or software instruments calculated with the larger buffer size for VIP objects during playback. This will reduce the load on your system's processor. However, this increases the latency during playback.

This latency is automatically compensated during playback, so that there is no time offset even for tracks that are calculated in different engines. But you can no longer record to such tracks with monitoring, and you can no longer play software instruments via live MIDI input.

To switch a track to "Economy", select the track and choose from the menu **Track** > **Track type** > **Economy track**.

The speaker button of an economy track is displayed with a green frame.

Monitoring Behavior

Tape monitoring is the preset monitoring behavior that works like a tape machine: In stopped state and during recording the input signal is played back, during playback the content of the track. During a recording, the input signal is played back, except for punch-in recordings (↗102), in which case the track content is output outside the punch markers and the input signal is output inside the markers.

Manual monitoring: For each track, manually specify whether monitoring is active for the track using the speaker button in the Track head, Track editor, or Mixer. This behavior is only available when the ASIO driver system is used.

Mix Input and Playback: If this option is active, the track's input signal is also monitored during playback when track monitoring is activated.

Peak meter on audio input when recording: This option (active by default) ensures that the peak meters of the tracks that are armed for recording automatically display the level at the audio input instead of the track level.

In addition to the **System Options** > **Audio Setup**, you can also switch the monitoring settings and switching behavior by right-clicking on the **Mon** button of the Transport control. There you will find two more recording options for MIDI:

- **Automatic MIDI record switch on selected track(s)**: This option ensures that MIDI tracks are always ready for recording as soon as they are selected.

- **Automatic MIDI monitoring (Thru) on arming**: If this option is active, monitoring is automatically switched on for each MIDI track that you activate for recording, i.e. if you play a MIDI keyboard that is recorded on this track, you will hear the output signal of the software instrument for this track.

Basics - Hybrid Audio Engine

In the **Audio Engine**, the core of every DAW, mixes the audio data that is present at the inputs and played back from the hard disk, calculates the effects that go with it, and outputs the result via the outputs. The processing takes place block by block: The data is read in small portions into a memory area (buffer), calculations are performed with the data of the buffer and transferred to other buffers, which are then provided to the sound card for output. While one buffer is being filled with new input data, another buffer is being summed with another, effects calculations are taking place on yet another buffer, another buffer is being copied to an output buffer, another output buffer is being read by the sound card for playback. These read, write and compute processes must be synchronized, because if one sub-function cannot provide its results to the next in time, there will be dropouts in recording or playback. In this context, the size of the buffers has a significant influence on the behavior of the audio engine: Large buffers are less sensitive to short load peaks in the system, because the copy operations between the buffers occur less frequently and the parts of the processing chain have more time to catch up with the delays they create. The disadvantage of large buffers is that the engine can respond to changes in parameters by the user (e.g. the volume on a mixer channel or an effect parameter) with a greater delay (latency), since these changes can only take effect with the next buffer.

Sequoia uses the **Hybrid Audio Engine** by default. This engine combines two different engines with each other (hence "hybrid"): For real-time functions like monitoring, i.e. the immediate playback of the received audio signal with effects, and for the playback of software instruments played live via MIDI, a low-latency engine with low delay (latency) and small buffers (ASIO buffer size of the sound card) is used. For the playback of recorded audio data, a second engine (Economy Engine) can be used, which works with much larger buffers (VIP object buffers) and is resource-saving and allows the integration of computationally intensive effects.

The Hybrid Engine thus enables a combination of Low Latency Engine and Economy Engine for track and effect calculation. For example, you can play back many CPU-intensive VST instruments in so-called "Economy" tracks (see below), while only the VST instrument that you are currently playing live is calculated in the Low Latency Engine.

Audio Devices

In this dialog you can select the devices to be used in Sequoia from all installed input and output devices.

Activate the devices you want to use in Sequoia with the checkboxes in front. You can change the order and name with which the devices appear in selection menus. Select a device by clicking on it and use the arrow buttons to

move it and the **Rename** button to give it an individual name. The **Reset** button activates all devices. Under ASIO only the first 4 stereo channels are activated by default. If you also hold down the **Shift** key when clicking **Reset**, all inputs/outputs are activated. The button **Device Info** opens the settings window of the sound card driver.

MIDI Settings

MIDI inputs and MIDI outputs: In the lists all MIDI devices present in the system are listed. Devices that you do not want to use in Sequoia can be deactivated with the checkboxes. You can change the order and name with which the devices appear in the selection menus. Select a device by clicking on it and use the arrow buttons to move it and the **Rename** button to give it an individual name. This may be necessary to distinguish MIDI devices that work with the USB standard driver without their own driver and for which all the same default name USB MIDI Device is displayed.

Default input/default output: The devices set here are used for previewing MIDI files in the file manager and file selection dialogs, and as default input and output devices for new tracks.

Recording offset: Here you can enter an offset value in milliseconds by which the recorded MIDI events are shifted back to compensate for a possible latency in MIDI processing.

Retrospective MIDI recording: This activates the Retrospective MIDI recording feature. All MIDI inputs are stored in the background even without extra activated recording for a certain time, which can be set at **Buffer**. Via menu **Play/Rec** > **Retroactive MIDI recording** a MIDI object is created in the selected track with this data.

 Audio/MIDI sync: With **Force Sync** sets the amount with which Sequoia synchronizes the MIDI tracks with the audio tracks. On fast systems this setting should be set to 100% in order to get the closest synchronization possible between MIDI and audio. If your system experiences difficulties synchronizing the MIDI tracks with the audio tracks, select a lower force sync 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: Set the default behavior of the volume fader in new MIDI tracks here. See the section Control behavior of the volume fader (↗486).

Activate SysEx input: Activate/deactivate here the receiving of System Exclusive data from external MIDI devices.

Use system time for MIDI timestamps : If this option is activated, the MIDI device driver's time stamp will be ignored. This can help if the driver provides a timestamp that is not synchronized to the audio or is completely wrong. In case of MIDI recording problems with MIDI devices, the option may help.

Link VIP Q and MIDI Editor Q: This option causes a linking of the quantization settings of VIP and MIDI Editor, so that a change of the grid and quantization settings in the VIP is applied to the quantization settings of the MIDI Editor and vice versa.

Note On Chasing: "Note On Chasing" means that MIDI notes are played even if their start is before the current playback start position, but they should still be sounding at the start time of playback, in other words, if the NoteOn event is before the start time, and the corresponding NoteOff event is behind it.

Send "All Notes Off" to VST instruments: By default, the MIDI panic function (↗477) for ending hanging notes does not send individual NoteOff events for all notes on all channels to VST instruments , but sends an "All Notes Off" MIDI message. However, this command is ignored by some VST instruments. If you deactivate this option, individual NoteOff events are also sent to the VST instruments for all notes on all channels, which then takes a little longer.

Soft split for MIDI objects: see Editing MIDI objects (↗419)

Deactivate all MIDI functions: If you do not use MIDI in your project, you can use this option to deactivate all MIDI functions in order to increase program performance.

Hardware Controllers

In this dialog all settings for the operation of Sequoia with hardware controllers are made. For more information, please refer to the separate document `Hardware Controller.pdf` in the program folder.

Metronome Options

ℹ️ *For detailed information on metronome settings, see the section **Tempo Editing > Metronome** (↗367).*

Recording

In this dialog, various further settings for the recording are made. It can also be accessed via the button **Advanced Options...** directly from the dialog Recording Options (↗95). Some options from this dialog can also be found here (**Format**, **Record offset in samples**, **Read CD/DAT marker**...). Only the additional options are described below:

Options:

- **Save each take in a new file**: If active, a new file is created for each take of a recording. New takes created manually (**+** key at **Take** in the recording window) and multiple passes of a loop recording are always saved as takes in the same file.

- **Prepare all tracks for track punch record**: If you enable this option, all assigned input devices will also be opened during playback, so you can start recording on all tracks when you enable recording for a track.

- **Treat audio inputs exclusively**: This option (active by default) ensures that recording can only be activated in one track at a time if the same input is assigned to multiple tracks. This avoids recording identical audio data on multiple tracks.

- **Pre-Recording (ASIO and Wave Format only)**: If active, this function adds audio material that was present at the input before recording was started to the beginning of the current recording when you activate recording from the stop state or from playback. Preset is 5 seconds, but up to 120 seconds are possible. Move the object start of the recorded objects to the left to make this material visible.

- **Post-Recording**: If active, up to 2 seconds of additional audio material is recorded in the background even after recording has stopped (default: 0.5 seconds) If you move the end of the object to the right after recording, you can make the additional audio material visible.

Extended options:

- **Show small recording window during recording/punch-in recording**: Use these options to display a small, non-modal recording window with the most important recording controls during direct recording or punch-in recording.

- **Confirmation after recording**: With the options in the menu you can define the behavior when stopping the recording.

  - **No confirmation dialog**: The recording is finished and the recorded material is taken over into the project.

  - **Confirmation after recording**: (default) The recording is stopped and you are asked whether you want to accept the recording or not. If you select **Delete**, you will also be asked whether the recorded audio should also be deleted or not.

  - **Confirmation at record stop, recording continues**: The recording continues at first, and a dialog appears in which you can choose whether you want to accept the recording or delete it; in both cases, the recording is then stopped. Press **Continue recording** to close the dialog and continue recording.

  - **Confirmation at record stop, recording continues (protected)**: The recording continues at first, and the same dialog appears, but the buttons **Delete recording** and **Apply recording** are not available yet. The dialog can be closed immediately by **Continue recording** and closes automatically after 30

seconds. To stop the recording, you must first uncheck the **Prevent record stop** checkbox to be able to click on the two buttons and thus stop the recording. You can use this option as a safety measure against accidental recording stop.

- **ASIO drop-out warnings**: Specify whether a warning should be issued when ASIO drop-outs occur. Drop-outs are short gaps in recorded audio that occur when a buffer could not be written in time. If **Show warnings** is selected, a red warning is displayed in the status display when drop-outs occur. At **Show warnings and set markers** markers are also set at the locations in the project where the drop-outs occurred.

- **Update object during recording**: If active, the objects created by recording in the project are updated during recording, so they grow as the recording progresses, allowing immediate visual control of the recorded audio.

- **Record position autoscroll**: If you have activated this option, the visible section is scrolled along with the record position during the recording.

- **Warning if remaining recording time is less than … minutes**: If a timed recording was started, a warning is issued if the remaining recording time until the end of the recording falls below the specified time.

- **Set cursor to record end position - next recording starts at this position**: If you have activated this option, the play cursor will be set to the end of the recording after the recording, the next playback or recording will then start from this position.

- **Recording button in the recording options can stop recording**: If active, the **Record** button in the **Record Options** dialog (keyboard shortcut: **Shift + R**) can both start and stop the recording. By default, the option is off and the recording can only be stopped with the stop button.

- **Recording button stops playback after standard recording**: If active, you can stop both recording and playback while recording is in progress using the **Recording** button on the transport console.

- **Select recorded objects**: After finished recording, all recorded objects are selected, if several takes were recorded, the objects with the last take.

Take Handling:

- **Set take markers in VIP timeline**: If active, new takes are indicated by markers in the project.

- **3-digit take number**: If active, the take numbers are formed with three digits instead of two, starting with 001, thus avoiding complications when more than 99 takes are generated.

- **Short object names**: If active, only the take number is used as object name for the recorded objects.

- **Initialize comment with take name**: Comments that can be inserted during recording using the **Comment** button in the recording options are given the current take name in the input field.

■ **No new object on false start (FS) key**: No new object is created after pressing the FS (false start) key, only a new take with the same name is created in the same object.

Playback

The playback options control the behavior during playback. For more information, see the **Play project** (↗83)

Keyboard shortcut: **P**

Stop at position: if active, the play cursor remains at the current position when playback is stopped.

Space bar (play/stop) also works in background: The space bar can also be used for play and stop in Sequoia when you are working with another application in the foreground.

Esc key stops playback and recording: If active, playback can also be stopped with the **Esc** key. This still works even if the program interface is otherwise unresponsive because the computer is overloaded.

Pre-roll time/Post-roll time (s): See Play cut (↗86) and Crossfade editor (↗177)

Maximum reverberation time for objects without fade-out: When audio effects such as reverb or echo are applied to objects, the resulting audio becomes longer than the original object. With this value you define for how long after the object end this audio data will be calculated. Values up to 60 seconds are possible.

 Cue mode, see Cue mode (↗87).

With **Autoscroll** the graphical display constantly moves even before the playback marker leaves the visible section. This way you always keep the overview. Activate the autoscroll mode with the option **Active**. With **Active for zoom <1s** autoscrolling can be activated separately for very high zoom levels of less than one second, because this leads to very fast movement in the project display, which is often not desirable.

You can choose between **page** and **soft autoscroll mode**. When scrolling page by page, the section changes as soon as the play cursor moves out of the displayed section. With soft scrolling, the play cursor always stays in the center of the section while the project scrolls under it. At smaller buffer sizes (↗689) (e.g. < 4096 samples), scrolling will be softer.

> **ⓘ** *Notes*:
>
> - *Note: When using a lot of tracks, autoscroll may cause computer overload, where dropouts may occur during playback. Deactivate Autoscroll mode if this happens.*
>
> - *In playback mode* Continuous playback while editing *(↗88) Autoscroll is disabled.*

Pause autoscroll after clicking in the arranger: When this option is active, autoscroll is temporarily disabled after clicking into the arranger (lower half of a track) until playback stops, to make it easier to edit the project while playback is in progress. It will then be active again when playback is resumed.

Performance

The settings at **Performance** enable system performance optimizations that may not be compatible with certain sound cards or plug-ins in individual cases and are therefore optional.

> **ⓘ** *In general, if you enable all options, you will get the maximum system performance. If all options are deactivated, maximum compatibility will be achieved.*

```
☑ Multi CPU Support, maximum number of CPUs   [5        ]  / 6 (system maximum)
      Use more than one CPU for audio processing for improved performance.
      Disadvantage: may cause incompatibilities with some plug-ins which rely on a strict
      track processing order.
      Reduce CPU number if you experience performance issues.

☑ ASIO Priority Boost
      Advantage: More reliable recording and playback.
      Disadvantage: may cause incompatibilities with some audio drivers.


☐ Deactivate muted tracks for ASIO
      Advantage: less CPU load (similar to economy tracks)
      Disadvantage: slows down mute button response time


☐ Deactivate FX on empty or silent tracks for ASIO
      Advantage: less CPU load (similar to economy tracks)
      Disadvantage: may cause incompatibilities with some plug-ins and irregular CPU load.


☐ Switch off hybrid engine for bouncing/export
      Advantage: All effects will be calculated with the faster economy engine.
      Disadvantage: For some plug-ins changing the buffer size may cause incompatibilities
      and problems with latency compensation.

Note:

      - Activating all options gives maximum performance.
      - Deactivating all options gives maximum compatiblity.
```

- **Multi CPU Support**: Sequoia supports the use of multi-core CPUs so that different tasks are distributed to different cores. If you have chosen **MME** or **WDM** as driver system, audio processing runs mainly on the first core. Disk I/O operations, graphics and video playback are calculated on the other cores. At **ASIO** the tracks of the mixer with their effects are distributed to the first cores. Object effects, graphics and video playback are distributed to other available cores. By default, Sequoia always uses all cores of the CPU. In case of compatibility problems with certain plug-ins, you can reduce the number of cores used.

- **ASIO priority boost**: Among all threads of the application, special priority is granted to the ASIO thread. This option increases the reliability of audio recording and playback and is active by default and normally does not need to be changed. Only a few audio drivers are not compatible with this option and require disabling.

- **Deactivate muted tracks for ASIO**: With this option you reduce the CPU load of your system. However, there are short delays when switching off mute.

- **Deactivate FX on empty or silent tracks for ASIO**: With this option you reduce the CPU load of your system even more. However, this feature can cause unsteady CPU usage and compatibility issues with some plug-ins.

ⓘ *The* Recording of track outputs *(⤴104) is not possible if this option is active.*

■ **Switch off Hybrid Engine for mixdown/export**: If you activate this option, all effects will be calculated with the faster Economy Engine during an export process. However, this may cause latency compensation errors with some plug-ins.

Extended Buffer Settings

In this dialog you can optimize the buffer settings for projects, disk caching and plug-in processing. If there are no problems during recording and playback, you do not need to make any changes here.

Program Buffer/HD Performance: Here you can save and load presets for buffer settings. Various presets are included, e.g. for scrubbing.

Program Buffer (in stereo samples)

■ **VIP object buffer**: Since error-free playback is usually more important than fast reaction times, this value should be increased when playing back many tracks. When playing and editing virtual projects, only this setting is important.

■ **HD/Scrub buffers**: These buffers are used when playing wave projects. Also test smaller values here to achieve faster response times.

■ **Test buffer**: This buffer is only used for real-time previewing when effects are applied offline.

■ **Buffer count**: Specify here how many buffers are to be used. More buffers increase reliability, but also increase memory requirements. This results is longer response times. You can read the current buffer utilization during playback in the status display at the bottom right. A buffer number between 4 and 6 is recommended.

Hard Disk Performance/TrackSpeed

TrackSpeed technology lets you significantly increase the number of simultaneously playable hard disk tracks 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 work with small audio buffers (e.g. 8000 samples or less) even when using many tracks, without performance suffering. This enables very short reaction times to user interventions, e.g. at the mixer and at the object handles.

There are settings available which can be used to customize TrackSpeed to your individual requirements:
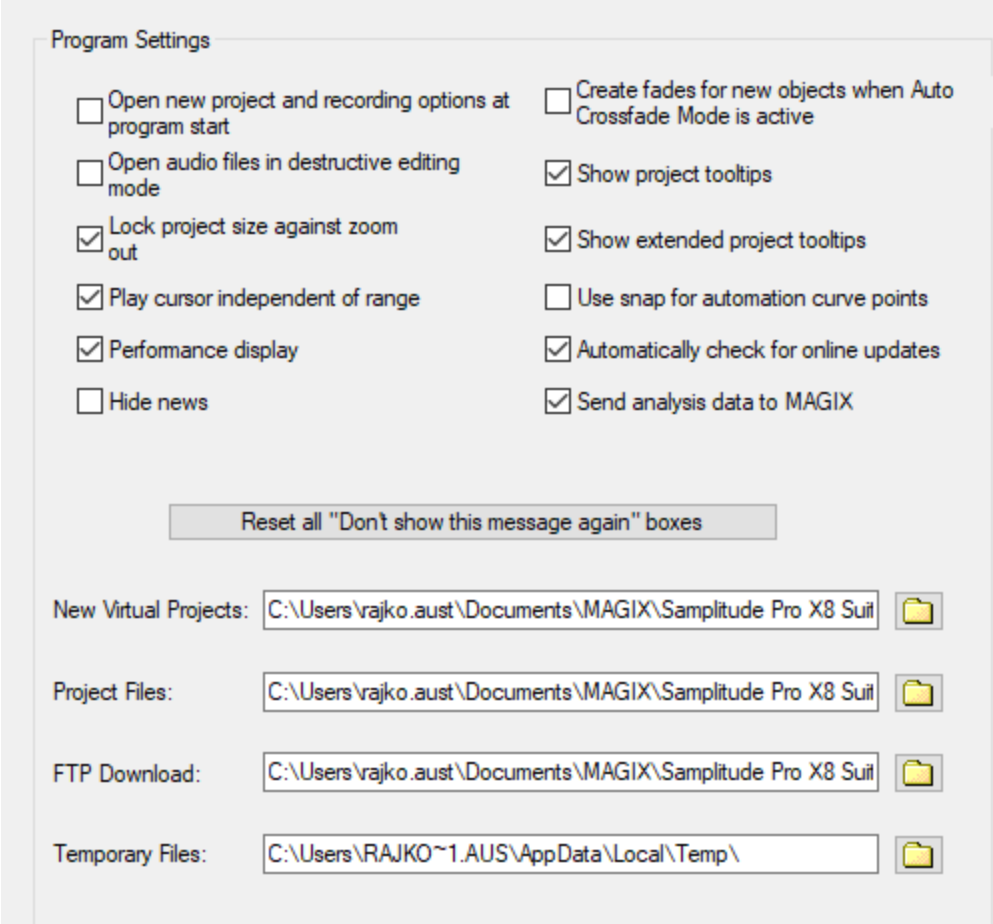
■ **Activate TrackSpeed**: Check this box to turn on the TrackSpeed functionality.

■ **Preload file cache while stopped**: If this option is enabled, the file cache is preloaded each time the play cursor is positioned. You can recognize this by the message "Cache preloaded" in the status display at the bottom right. After that, the next play start is immediate because all the audio data you need is already loaded.

■ **Preload before start**: This value specifies how much % of the cache is loaded at play start. Larger values delay the start, but increase the playback stability with a large number of tracks.

Program

General

This dialog contains various general options for the program's behavior.

Program settings:

■ **Open project and recording options at program start**: If active, the Start Selection is not displayed when Sequoia is opened, but an empty project is created and the Recording options dialog (↗95) is displayed.

■ **Open audio files in destructive editing mode**: If active, audio files loaded as wave projects separately from virtual projects are also opened in destructive wave editing mode (↗570).

- **Lock project size against zoom out**: If active, a project is not automatically lengthened when you zoom out further than the maximum section size.

- **Play cursor independent of range**: If inactive, the play cursor is set to the start of the range when ranges are selected, whereas moving the playback marker cancels a range selection.

  > ⚠️ *Attention: Disable this option only if you really need this range and play cursor behavior that has been changed since many years. Playback and range selection will then no longer work as usual and as described* in these instructions (↗83).

- **Performance display**: Activates the DSP performance display in the lower left corner.

- **Create fades for new objects when auto-crossfade mode is active**: When loading audio files into a project, even if auto-crossfade mode (↗169) is active, the objects are created without automatic fades, since it is assumed that the files already contain fades and are to be used unchanged. If you still want fades, enable this option.

- **Show project tooltips**: If active, additional information is displayed when the mouse pointer rests over a control.

- **Show extended project tooltips**: If enabled, detailed tooltips are displayed.

- **Use snap for automation curve points**: If active, automation points also snap when moved and snap (↗79) is active.

- **Override docking with the Ctrl key**: When moving undocked windows, docking can be prevented by holding down the Ctrl key at the same time. Deactivate this option to reverse the behavior: Docking then only takes place if the Ctrl key is pressed when dragging windows.

- **Automatically check for updates online**: If active, Sequoia checks for program updates when the program is started. If your computer does not have Internet access, you can disable this option.

- **Reset all "Don't show this message again" boxes**: Many notification dialogs can be turned off by selecting the "Don't show this message again" option before closing them. Press this button to reactivate all notification dialogs.

Below you can set preset paths for project files:

- **New virtual projects**: Under the path selected here, all new virtual projects as well as the recorded audio files and those created during import are saved.

- **Project files**: All other files created by Sequoia that cannot be assigned to a specific project are stored under the path specified here.

- **FTP download**: All files downloaded via the integrated FTP client will be saved under this path.

- **Temporary files**: This preset path is directed to the standard temporary files folder. This folder should be on a hard drive or partition with sufficient free storage space.

Undo

This dialog is used to configure the undo function for virtual projects and wave projects. The function can be activated or deactivated separately for the respective edits. In addition, you can define the number of stored undo steps in each case. A value of 20 means that the last 20 changes in the project can be rolled back.

Always have temporary undo files created as HD Wave projects for RAM Wave projects: If active, for such projects the changes are saved in temporary HD-Wave projects to save RAM.

Object Lock Definitions

Here you determine which functions should be prevented by locking objects or tracks. You can select the following options:

Locking Objects

- **Disable moving**: Objects can no longer be moved horizontally. This avoids unintentional offsets between individual tracks in multitrack recordings.

- **Disable vertical movement**: The objects can no longer be moved between the tracks.

- **Disable volume changes**: The volume handles of the objects are disabled.

- **Disable fade-in/-out**: The fade handles of the objects are disabled.

- **Disable length changes**: The length handles of the objects are deactivated.

- **Disable deleting**: Locked objects cannot be deleted.

- **Lock automation**: The object automation is locked against editing.

- **Disable ripple when objects are locked**: These options control how objects are rippled during range-oriented edit functions (for example, Delete with ripple) and when moving and deleting objects in the **Link...** object modes (↗155) when locked objects are on the tracks.

    - **All objects**: Ripple is prevented for all objects in the track.

    - **All objects behind locked objects**: Ripple is prevented for all unlocked objects in the track that are behind a locked object.

    - **Enable ripple**: Ripple is allowed for all objects that are not locked, even if there are locked objects on the track.

Locking Tracks

Tracks are locked in the track head by activating the lock symbol.

Disable 4 point cut: This deactivates the source destination cut function for locked tracks. The edits may no longer be changed.

Disable crossfade editing: This disables crossfade editing for locked tracks - the crossfades can no longer be changed.

Crossfade Editor

This page contains the **settings for the Crossfade editor** (↗181).

ⓘ *For detailed information on the crossfade editor, see the section* **Crossfade editor** *(↗170).*

SD Cut / MuSyC

ⓘ *For detailed information on MuSyC settings, refer to the chapter* Source-Destination-Editing > MuSyC - Multi-Synchronous Cut > MuSyC System Settings *(↗213).*

ⓘ *For detailed information on the source-destination cut settings, see the chapter* Source-destination cut *(↗204).*

Keyboard, Menu & Mouse

Keyboard / Menu

In this dialog you can assign keyboard shortcuts for all menu items in the main menu of **Sequoia** and change the existing assignments. You can also hide rarely used menu items to make the menus clearer.
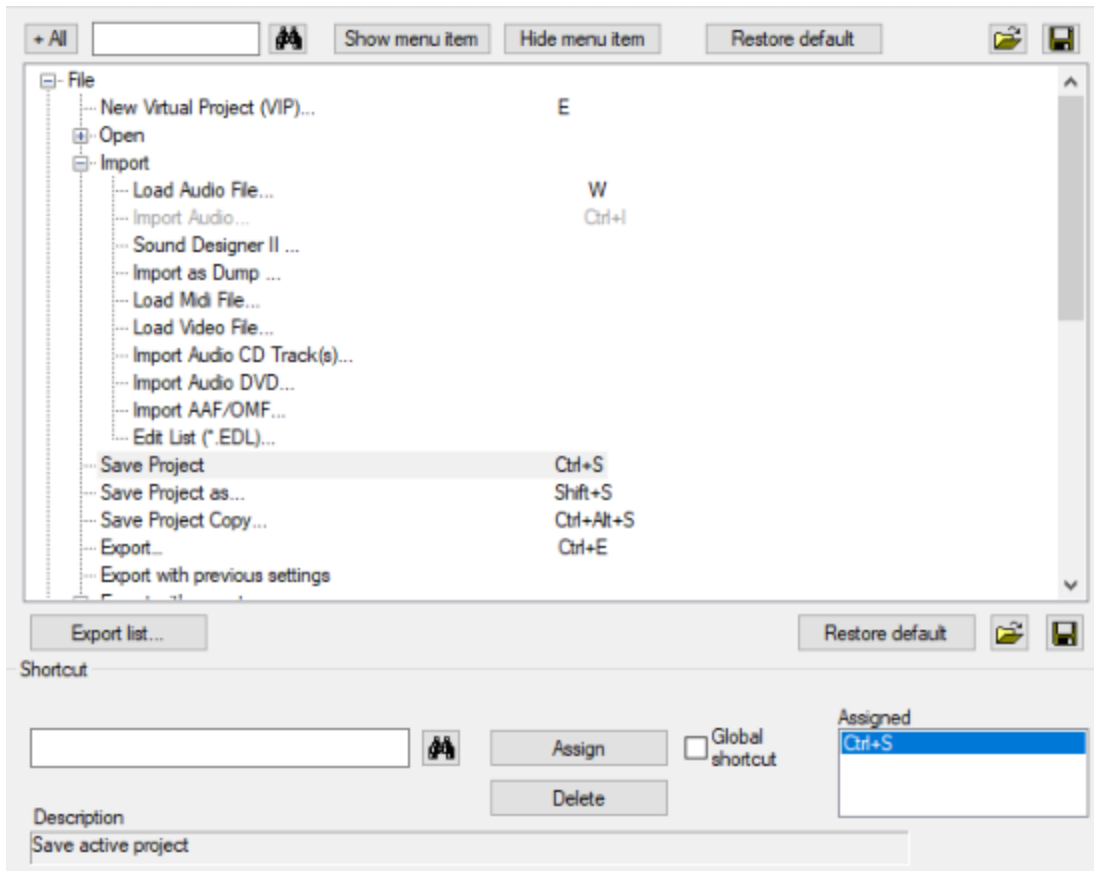
For this purpose, the menu is displayed in a tree structure. Submenus can be expanded by clicking on a **+** icon, with **+All** the tree is expanded completely. You can also search for a specific menu item by entering a search term in the input field above and then clicking on the binoculars icon next to it.
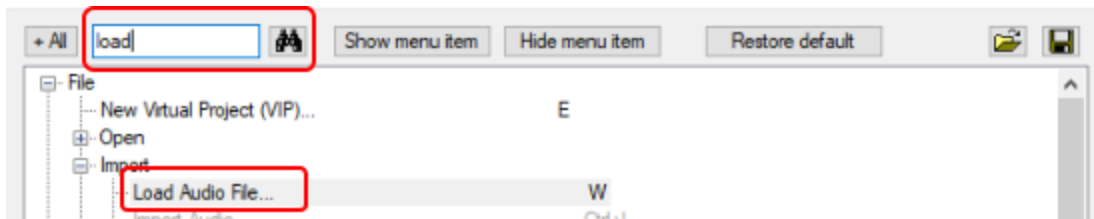
Editing Keyboard Shortcuts
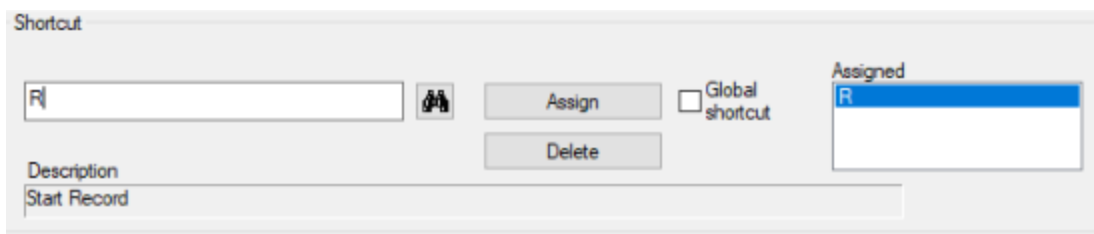
1. Select a menu item for which you want to create or change a keyboard shortcut.

2. If keyboard shortcuts are already defined for the menu item, they are listed at **Assigned** on the right in the **Shortcut** area.
   To find out which menu item is assigned to a particular keyboard shortcut, click in the input field at **Shortcut**, press the desired keys and click the binoculars icon next to them. If an assignment exists, the corresponding menu item is selected in the tree structure.

3. To set a new keyboard shortcut, click in the input field at **Shortcut** and press the desired keys. You can use combinations of any key with the modifiers **Shift**, **Alt** and **Ctrl**.

4. Click **Assign** to set the keyboard shortcut for the selected menu item.
   If you enable the **Global Shortcut** option, the keyboard shortcut will work throughout the program, even if a window is focused for which separate keyboard shortcuts apply (see below).

5. If the key combination is already assigned to another program function, the existing assignment is displayed in a message dialog. Click **Cancel** to cancel the assignment and keep the existing shortcut or **Replace** to permanently reassign the shortcut.

6. To remove a keyboard shortcut assignment, click the **Delete** button.

The **Export list** button lets you save the complete list of current keyboard shortcuts as a text or CSV file, or just display it in a separate window.

Showing and Hiding Menu Items

Select the menu item you want to hide. **Hide menu item**" removes the menu item from the menu. It will then be grayed out in the tree.
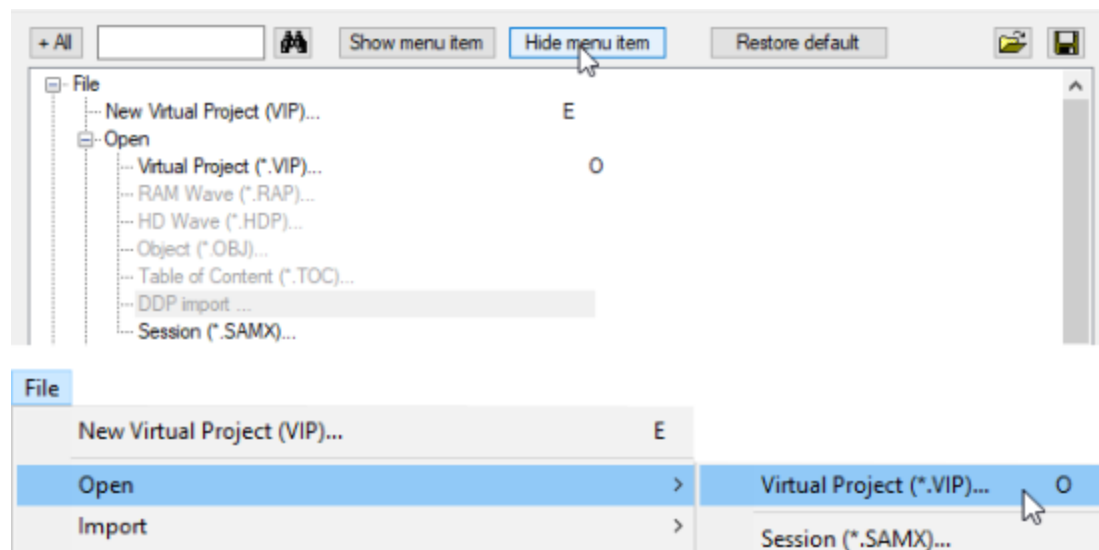
Note that after this, the menu item can also no longer be accessed via the assigned keyboard shortcut .

With **Show menu item** you make the hidden menu item visible again. **Restore default** restores the preset state in which all menu items are displayed.

Save/Load: Click the **Save** button to save your keyboard and menu settings. Use the **Load** button to load saved settings.

MIDI Editor/Crossfade Editor/Spectral Cleaning keys

Some windows in Sequoia have their own menus that can be controlled by separate keyboard shortcuts. In these dialogs these keyboard shortcuts are defined, similarly to those of the main menu

Special Keys

In the window **Special keys** you define the keys for temporary switching (↗78) of mouse and object modes, with which you can temporarily switch to other mouse and object modes.

Object modes

■ **Temporary key for Link one track/all tracks to the right/left**: Activates the function **Link one track/all tracks to the right/left**, as long as the assigned key is pressed.

■ **Temporary key for moving object content**: Special button for moving the object content with the mouse.

■ **Temporary toggle for snap**: Activates the snap as long as the assigned key is pressed. When the snap is active, the snap is temporarily deactivated.

Mouse modes:

■ **Temporary key for Object mode/Curve mode/Draw automation mode/Zoom mode/Cut mode**: Activates the respective mouse mode (↗70), as long as the assigned key is pressed.

■ **Temporary key 1/2 for scrub mode** (Scrubbing mode (↗90)): Activates scrubbing mode as long as the assigned key is pressed.

> ⓘ *There are 2 key assignments for this, because in the default setting the key **0/Einfg** on the numeric keypad is used for this, the function of the key is thus independent of whether **NumLock** is active or not.*

■ **Temporary toggle for Scrub mode (↗90)**: Switches the Scrub mode permanently until the next time playback stops.

Other keys:

■ **Temporary key for writing automations (↗345)**: Activates automation recording as long as the assigned key combination is pressed.

- **Temporary switch for mono to stereo**: When two files selected in the file manager are loaded into a project via drag & drop, a stereo object is created from both files if this button is held before dragging.

- **Activate Talkback**: Here you can set a button for the talkback in the monitoring section. The monitoring section has its own fixed keyboard shortcuts. But this one works even if the monitoring window is not selected.

- **Multi Tap Sequence Key**: Pressing this key separately several times activates special functions. Currently, the button doubles the pre-roll time in the Crossfade Editor. If you press (by default) Ctrl and then the space bar, playback is started with double the pre-roll time, with 2 x Ctrl and space bar with four times the pre-roll time, and so on.

Mouse

At **Mouse** you can set special options for mouse operation to ensure compatibility with older versions.

- **Disable range zoom with double click**: Preset on. Deactivate the option if you want to zoom into the project by double-clicking in an empty place of the project.

- **Zoom lasso allows vertical zoom without "shift"**: Preset off. When a selection rectangle is drawn in the project in Zoom mouse mode, the default setting is to zoom into the selection horizontally only. For additional vertical zoom you also need to press the **Shift** key. With this option active, you can zoom in horizontally and vertically without using the **Shift** key.

- **Disable zoom by vertical mouse dragging in timeline**: Preset off. Disables the zoom function by dragging the mouse vertically (↗108).

- **Backward compatible use of "Shift" + click to mute/solo/record (instead of Alt + Shift + click)**: Switches modifier when clicking **Solo**, **Mute** and **Record** to switch **Exclusive** from **Alt + Shift** to **Shift**, according to behavior in older versions.

- **Knob characteristics same as for faders**: Preset off. When this option is enabled, knobs (pots) can be adjusted like sliders by dragging them up and down.

- **2nd click required to move object**: This option requires a second click to moving objects. The first click selects objects and prevents unintentional moving.

- **Movement delay**: Preset on (200 ms). When selecting an object by clicking on it, you may accidentally move the object. If the movement delay is activated, Sequoia waits for the specified period of time before a movement can be executed.

- **Minimum range length**: Preset off (100 ms). To protect against accidental range selection of very few ranges, ranges smaller than the specified length are prevented from being selected when the option is active.

Mouse wheel
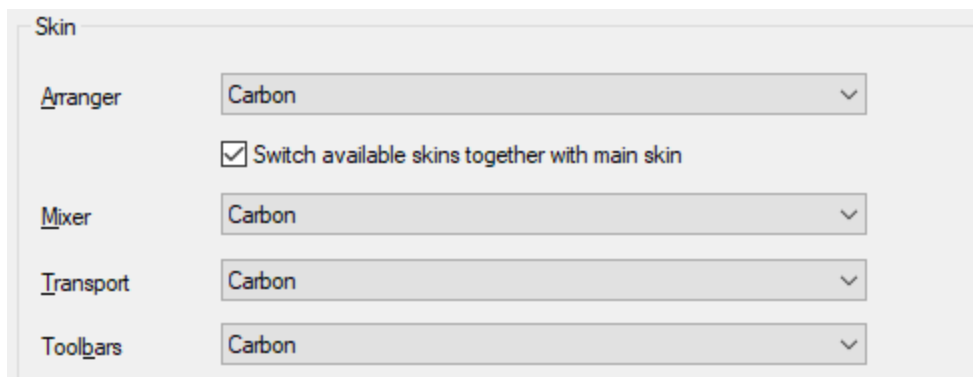
At **Mouse wheel** you can change the function of the mouse wheel for zooming and scrolling in the project. Set which modifier (Alt, Ctrl, Shift) triggers which action in combination with the mouse wheel. Actions can also be defined for the middle mouse button.

> ⓘ **Note**: *To make it easier for program beginners to switch to Sequoia, the mouse wheel in VIP scrolls vertically (tracks) rather than horizontally (time position) by default, in line with the standard behavior of many other DAWs. At **Reset** you can choose to restore the new (vertical scrolling) or the old default behavior (horizontal scrolling).*

Design

Skin

In this dialog you can choose between different skins for the arranger, the toolbars, the mixer and the transport console. By loading another skin, you can change the color and graphic design of the Sequoia program interface.

You can also choose between the different skins by right-clicking on the drag icon in the upper left corner of a docker's title bar.

Colors

This is where you adjust all the colors used in the program. These color settings can be optionally loaded and saved as presets.
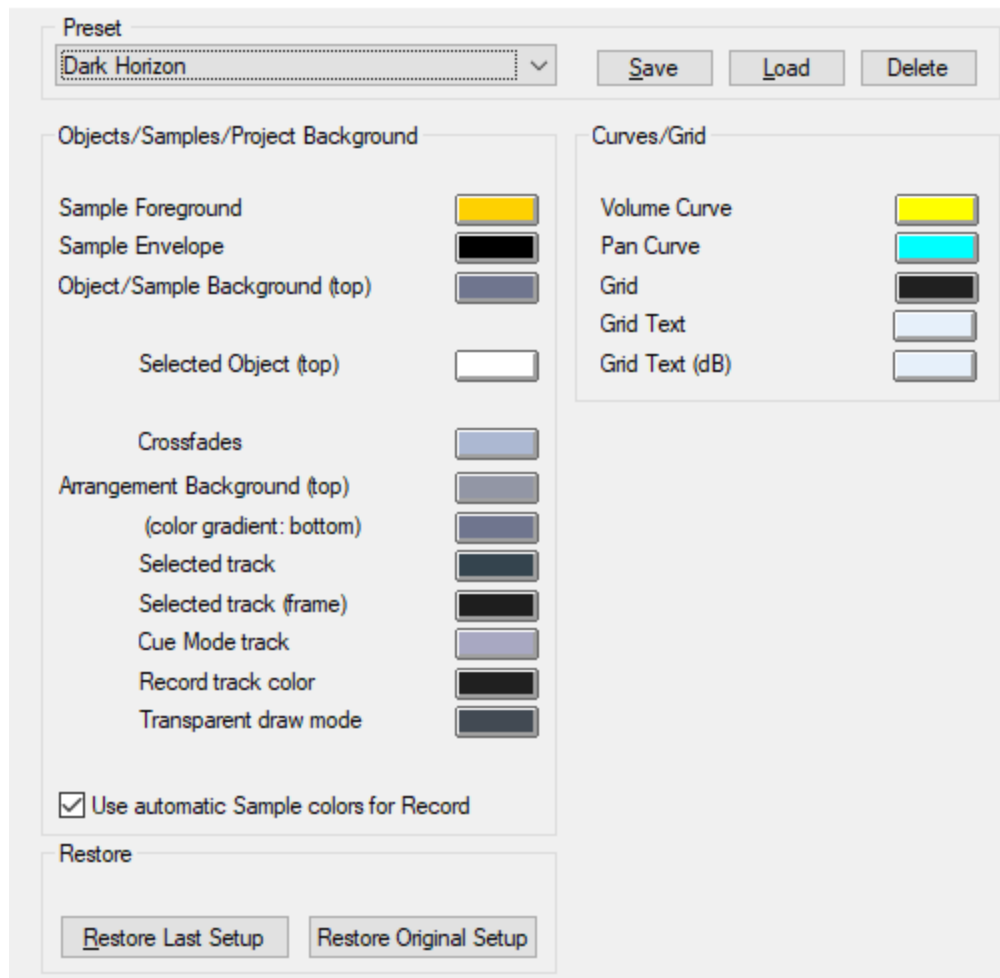
Use automatic sample colors for record: A separate random color is chosen for each recorded object per track.

Restore

Restore last setup: The last status of the color setup, before the dialog was opened, is restored.

Restore original setup: Here you can reset the color settings to those of the previous or original color state. Click the button to see a menu containing the various options.

- **Previous state**: The previous color setup will be restored.

- **Original state**: Resets all colors to their default setting.

- **Object colors**: This option resets all object colors. This may additionally be necessary if you import a project from another computer that had different color settings or if automatic sample color assignment during recording causes objects to deviate from the default object color.

- **Track colors**: The track colors are 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.

Project Display

You can configure the display of the project and the objects in it in great detail. To do this, use the section **Designs** > **Project view** in the **Program preferences**. You can reach the dialog via menu **View** > **VIP display mode** > **View options...** or keyboard shortcut **Shift + Tab**.

In the drop-down menu at the top, you can switch between four different complete sets of options, the **drawing modes**. You can quickly switch between drawing mode 1 and 2 with the **Tab** key without having to open the dialog.

Waveform display
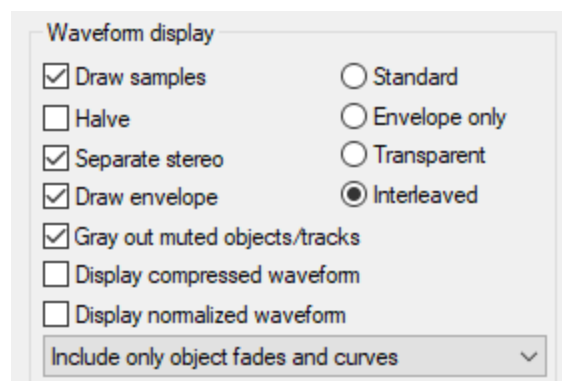
- **Draw samples:** Here you can turn the wave form display on and off. When deactivated, volume or pan curves are more clearly visible.

    > ⓘ **Note**: *In drawing mode 2, the waveform display is disabled by default.*

- **Halve:** The display of the samples is switched to half waveforms.
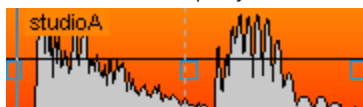
- **Separate stereo**: For stereo files, the channels are displayed one above the other.
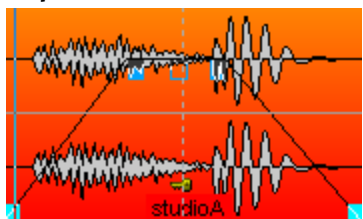
- **Draw envelope**: The envelopes of the objects are drawn.
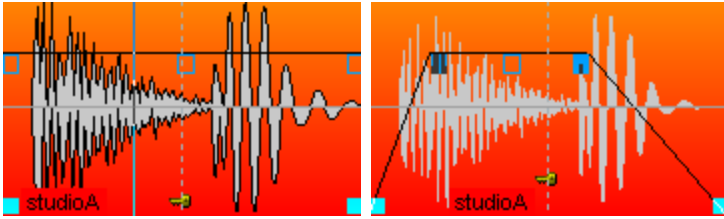
- **Gray-out muted objects/tracks:** Muted objects and tracks will be shown gray.
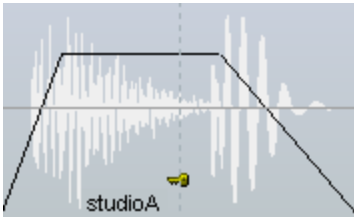
- **Display compressed waveform**: In compressed waveform display, the assignment of levels to the height of the waveform is not proportional, but compressed in such a way that differences at low levels are displayed larger than at high levels. This makes it easier to see the transition between low level signals and the noise floor.
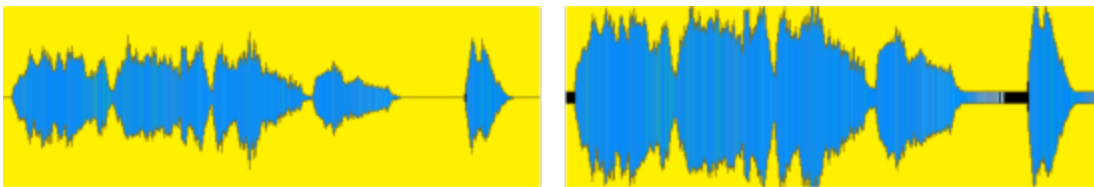
Image of the waveform without and with compressed display at the same zoom level

- **Display normalized waveform**: If you have activated this function, the waveform of each track is zoomed vertically so that the sample with the highest value exactly fills the track height. Manual vertical zooming of the waveform using the keyboard shortcuts **Ctrl + up/down arrow** or the buttons for **Zoom Wave In** and **Zoom Wave Out** are disabled.

   This option can be switched via this button on the lower toolbar (↗53).

  > ⚠ *Attention: In this display mode, the volumes of objects can no longer be compared visually, as they always look the same regardless of the object volume setting. Use this view mode when you want to cut objects with very different levels and do not want to constantly change the zoom level of the waveform.*

-
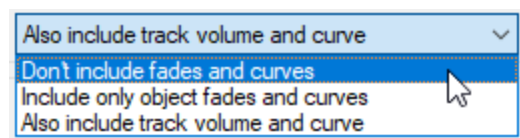
  In this menu, you select how the waveform display is scaled by the object fade-in/out and by the volume curves (track/object). In the preset mode **Include only object fades and curves**, the waveform is only scaled

by the object fades and, if applicable, by object volume curves. With **Also include track volume and curve** the track volume or its automation curves are also included. This corresponds to the previous option **Scale with fades/curves**. This scaling makes, for example, the decay of the sound during a fade-out visible as well as volume changes by a volume curve. Even with crossfades, this mode provides a good visual evaluation of the resulting audio signal.
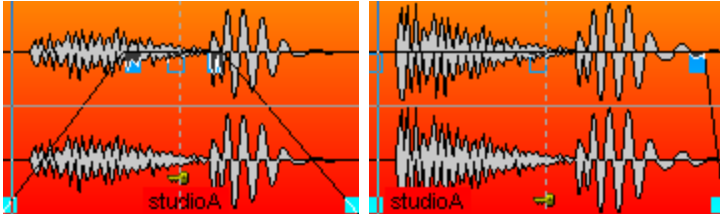
On the right side of this section there are options to control the display of crossfades:

- **Standard**: The wave form of the second object is drawn over the wave form of the first one:
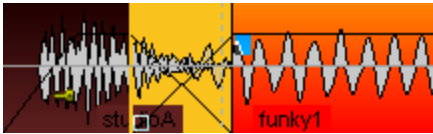
- **Envelope only**: This drawing mode displays only the envelope of the waveform, so that with crossfades the volume course of both objects becomes very well visible.
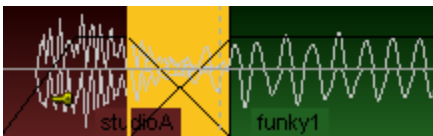
- **Transparent**: In this drawing mode, the waveforms of the crossfade are superimposed in color. For this, the waveforms appear against a dark background. It is best to combine this mode with the **alternating red/blue** option for the **waveform color** (see below) so that the the crossfaded regions are optimally visible.
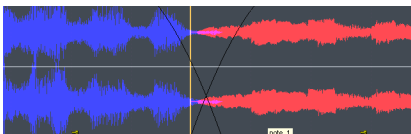
- **Interleaved:** In this mode, within a crossfade, a sample of the left object and a sample of the right object are always drawn alternately. Especially if the two neighboring objects have different colors, the fade area can be well visually estimated.
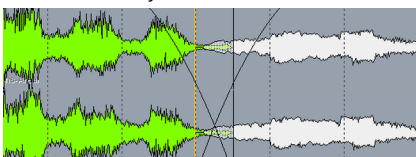
Objects
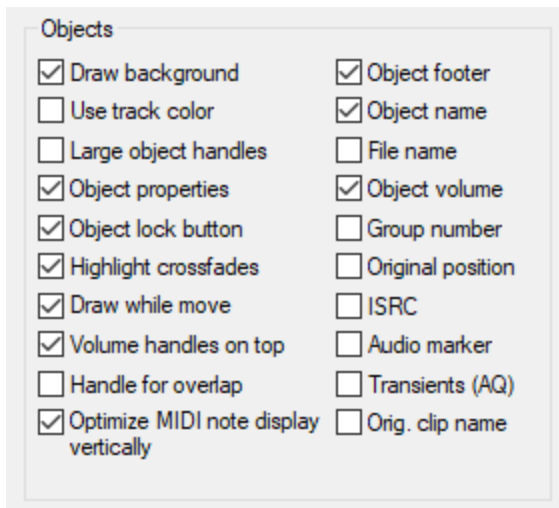
- **Draw background**: Activates the background colors of the objects. Every object can be assigned its own color (see Object Color/Name (↗160) or in the **Object Editor** under **Color**).

- **Use track color**: The track color is used as the background color of the objects.

- **Large object handles**: The object handles are displayed larger.

- **Object properties**: Display object settings such as EQ, dynamics, pan, effects, or plug-ins as abbreviations on the object.

- **Object lock symbol**: Display lock symbol for Locking objects (↗154).

- **Highlight Crossfades**: The crossfades between objects are highlighted.

- **Draw while moving**: The user interface is updated while moving. This option can be deactivated for computers with low processing capabilities.

- **Handle for overlap**: This handle allows you to set the overlap of crossfades without having to open the crossfade editor or object editor.
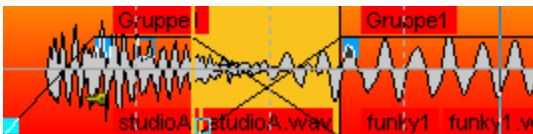
- **Volume handles on top:** If active, the volume handle of an object is always displayed at the top regardless of the object volume setting, otherwise it is moved when the volume is adjusted.

- **Vertically optimize MIDI note display**: In MIDI objects, notes are arranged vertically so that the lowest and highest notes present in the object are placed at the top and bottom of the object to make best use of the

available space.

- **Object volume**: Numerical display of the object volume in dB.

- **Object footer**: The texts displayed on the object are shown in a special area below the waveform

- **Object name**: The object names are displayed.

- **File name**: The file names are displayed.

- **Group number**: Objects can be grouped in VIPs. The groups are numbered. When viewing these group numbers, you get a quick overview of which objects belong to a group.

- **Original position**: Display of the original recording position related to the beginning of the project in the set format. By comparing the value in the object with the position on the timeline or the timecode in the video, it is easier to resolve deviations in the synchronization.

- **ISRC**: This option displays the ISRCs in the objects. If the ISRC for an object's file is not specified in the Broadcast Wave File (BWF) entry (↗653), the ISRC of a CD Track Index at the object is displayed, if available. For this the object must be on the first track.

- **Audio markers**: This option allows you to display the markers from the audio files in the associated objects. See Audio Markers (↗116).

- **Transients (AQ)**: Display transients in objects that you have previously created by the Audio Quantization (↗612).

- **Orig. clip name**: At objects created by loading a clip (↗311), the clip name is displayed.

VIP
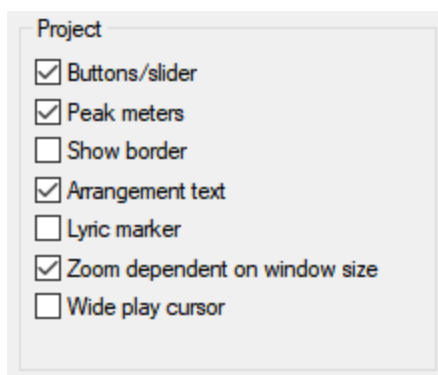
Various VIP components (buttons/sliders, peak meters, border and arrangement text) can be activated/deactivated here.

- **Buttons/slider** displays the track head with all mixer controls, i.e. the channel faders, solo buttons, etc. If you always have the Track Editor (↗59) open, you won't have to use these functionally identical controls here. You can also leave the track head switched on and only hide the **peak meter** for the track head.

- **Show Border**: The tracks are given a border to better separate them from each other.

- **Arrangement text**: The arrangement text is used to indicate the division of the track into the two editing areas in Universal mouse mode.

- **Lyric Marker**: Display of the lyrics marker (↗472)

- **Zoom dependent on window size**: The vertical zoom (track height) adjusts as the window size changes so that the number of tracks displayed remains the same.

- **Wide play cursor:** By default, the play cursor is displayed 3 pixels wide for better visibility. If you disable the option, it will be only 1 pixel wide.

Waveform colors
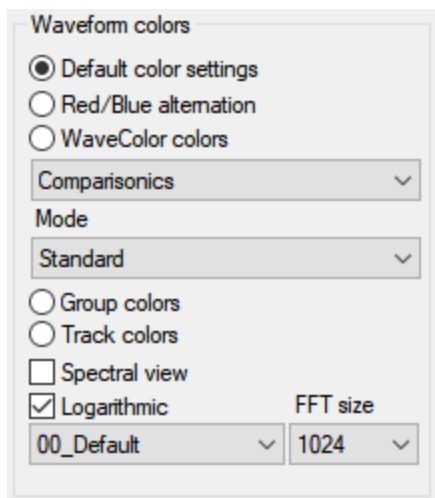
- **Default color settings**: The waveform contains the color that is preset in the color settings or defined in the Object Editor or **Object** > **Object Color/Name** menu.

  > ℹ️ *For more information about object colors, see* Renaming and Coloring Objects (↗161)

- **Red/Blue alternation:** This display mode always colors the waveform of neighboring objects alternately red and blue. This way, crossfades can be optimally viewed in the "Transparent" and "Interleaved" drawing modes.

- **WaveColor colors**: 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 below: The **default** setting applies intense rainbow colors to the audio signal. The **Comparisonics** preset provides coloring similar to the Comparisonics color display of older versions of Sequoia. **Black & White - Pitch** and **Black & White - Noise Factor** displays only the pitch or noise component in grayscale.

Based on WaveColor colors, a special audio search method is also available in the menu **Object** > **WaveColor Audio Search** (↗610), which can be used to find regions with the same or similar audio material in wave files.

■ **Group colors**: Object groups receive a randomly created individual group color. This allows a quick overview of the objects belonging to the different groups.

■ **Track colors**: When a track has been assigned a color, all objects on that track will automatically receive that color. Accordingly, the waveform color also changes when you move an object to another track. On tracks without color assignment, objects have the default color.

■ **Spectral view**: The audio material is displayed in the spectral view by a spectrogram that shows the frequency components over time. The level of the frequencies is visualized by the color or the brightness in the display.
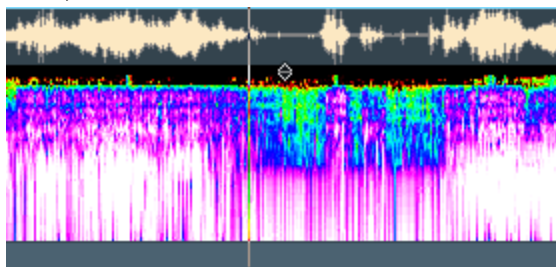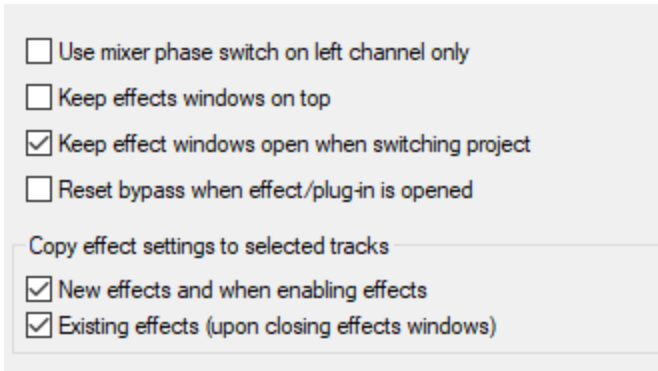
In the menus under the option you can select the color palette and the FFT size. The smaller you choose this size, the more accurate the display, at the expense of performance. Therefore, for a sluggish display, increase the FFT size. The **Logarithmic** option uses a different mapping of levels to the palette's brightness values, allowing for improved rendering with certain audio material.

Effects

General

Here you can change the default behavior of effects and effect windows:

- **Use mixer phase switch only on left channel**: By default, the phase button in Mixer and Track editor inverts the phases of both stereo channels. You can change this behavior here to use this switch to correct the classic inverted phase error that occur with incorrect wiring and that often affect only one stereo channel.

- **Keep effects windows on top**: Effects windows are displayed above all other windows in the system and remain visible even if you bring other programs to the foreground. However, when minimizing the Sequoia program window, they are also minimized.

- **Keep effect windows open when switching project**: When you switch to another project, open effect windows are closed. If the option is active, they remain open.

- **Reset bypass when effect/plug-in is opened**: If the option is enabled, deactivated effects will be activated as soon as you open their windows.

- **Copy effect settings to selected tracks**: see Apply effect to multiple channels simultaneously (↗489).

Dithering

In this dialog you can set the type of dithering and make further settings for dithering.

Dithering Mode:

- **No dithering, mathematical rounding**: In this mode the conversion of the sample values is done by rounding without dithering.

- **Dithering with linearly distributed noise**: In this mode, the conversion with dithering is performed with noise in which all amplitude values occur on average with equal frequency.

- **Dithering with triangular spread noise (standard dithering)**: In this mode, audio data at 32-bit float is converted via dithering with a noise where the amplitude values are triangularly distributed. Values in the medium range appear more frequently, and maximum or minimum values appear less frequently. This type of dithering typically produces more subtle results than linear dithering. The noise is not modulated by the signal here, so a decaying signal is immersed in a constant noise signal.

- **Dithering depth in bits**: The strength of the noise can be adjusted by the parameter **Dithering depth in bits**. This allows you to specify how many bits of the resulting 16 bits should be affected by dithering. In most cases, values between 0.5 and 2 will produce good results. Always increase the value until no more noise effects are audible.

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.

> ℹ️ *Noise Shaping minimizes the errors generated by quantization by shifting the quantization noise in the spectrum to above 10 kHz, the range to which the human ear is less sensitive.*

Which dithering mode sounds the best depends mainly on the audio signal.

Use POW-r dithering when playing back for master outputs only, use standard dithering for other outputs: If this option is selected, 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 distributed noise**.

Force 16-bit resolution for output (CD mastering quality): If you enable this option, dithering is performed during audio playback even if the audio output supports higher bit resolutions. This allows you to listen to the project as it will sound on CD in the end.

Smart dithering for export/CD burning (switch off dithering dependent on audio material)

- **For wave files**: This option is set as the default. In this case, dithering will only occur on 16-bit exports and when burning CDs if the bit-depth is not 16 bits. 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 switching off dithering (Autoblack)**: Here you set the threshold below which no dithering noise is added.

What is dithering?

Sequoia processes audio signals internally with an accuracy of 32bit float. When converting to audio data in a lower bit resolution, for example when exporting to 16bit audio files or burning to CD, the audio signal must be quantized. Instead of a very precise value, whose accuracy is also independent of its actual value, the sample value is mapped between its maximum value and 0 to a fixed range of integer values. This works very well for normal levels and without an audible difference. At very low levels, however, this results in sound distortions that occur when, during

conversion, at the lowest bit, it must be decided by rounding how the sample value is to be encoded. A very quiet, but especially in the decay phase of recordings well audible so-called "quantization noise" is created.

ⓘ *Quantization noise can be heard very clearly, for example, in children's toys with sound output, in which samples with very low bit depths of 8 or 4 bits are used.*

To mask this effect, dithering adds normal noise (random values) to the audio signal at a very low level. As a result, when quantizing, very quiet signals smoothly transition to "natural" noise as they decay, sounding much more pleasant and unobtrusive.

ⓘ *You can specify individual dithering settings for each export operation. For more information, see the section* **Export** *(↗666)!*

Resampling/Freeze options

Resampling quality: Here you can set which quality settings should be used for resampling (changing the sample rate) at different points in the program. The higher the quality level, the higher the computational load (for real-time application) or the waiting time (for offline resampling).

- **Record resampling**: Real-time resampling when recording at a sample rate that differs from the project sample rate and during chase-lock sync (correction of sample rate fluctuations by a reference source).

- **Playback resampling**: Real-time resampling at Scrubbing (↗90) and Varispeed (↗89) and at chase-lock sync.

- **Object resampling for new objects**: Real-time resampling as an object effect applied to objects created when loading audio files with a sample rate different from the project sample rate.

- **Offline resampling**: Destructive resampling of audio files applied when exporting audio files, importing and burning CDs, and importing audio files with a sample rate different from the project sample rate with conversion to wave files.

  ⓘ *Offline resampling uses the improved SOX algorithm; for options, see* Adjust sample rate (offline) (↗298).

Time Stretching/Pitchshifting options: Here you can set the default algorithm (↗297) for time stretching/pitchshifting to be used for new objects.

Freeze options:

- **Keep mono if possible**: When gluing objects together (↗165) mono objects are created again from mono objects if possible.

- **Use additional samples for object freeze**: If effects need a certain amount of lead time to sound as desired, this option allows additional samples to be included in the freeze calculation before the object start.

- **Object freeze without object volume**: Wen freezing objects (↗163) the object volume is not included in the calculation, but is retained as the object volume at the object.
- **Format**(16 bit, 24 bit, 32 bit): Here you can specify the format with which the freeze files are calculated.

VST/ReWire

- **VST folders**: List of search paths for VST plug-in effects and VST instruments. With **Add...** and **Remove** you can add and remove search paths.
- **Scan:** Sequoia performs a VST scan for a selected folder. All plug-ins are checked for their compatibility in Sequoia.

> ℹ️ *More information regarding this dialog and general information on installing VST plug-ins can be found under* Installation of VST Plug-Ins *(↗370).*

- **Scan user and system VST folders automatically for new plug-ins**: With this option you can set the program to scan automatically at every program start, but especially if you have a lot of plug-ins installed, this will increase the startup time.

- **Activate ReWire**: If this option is activated, rewire-compatible client applications can be integrated into Sequoia as synthesizers.

  > **i** *Additional information about ReWire can be found in the* ReWire *(↗381) chapter.*

- **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 buffer size is unlimited and is determined automatically.
  - **-1**: The ASIO buffer size is used.
  - **-2**: Default setting. The ASIO buffer size is used in the Hybrid Engine and the VIP buffer size in the Economy Engine.

  The VST buffer size is by default equal to the 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)` preset minimizes latency with UAD/Powercore plug-ins.

- **VST Automation Resolution**: In the Hybrid Engine, the ASIO buffer size is used. To achieve a better resolution of the automation also in the "Economy Engine", which works with VIP buffers, you can enter a correspondingly lower value here.

- **Start all object-based plug-ins on playback**: The advantage of object effects is that effects generate computational load only while the object in question is playing. However, some plug-ins cause delays at startup. This option therefore allows you to specify that the plug-ins on objects are always started when project playback is started, regardless of whether the object in question is being played.

Automation

Automation Release Time (ms): This sets the release time in ms for the write automation modes. When an existing automation curve is updated, i.e. new values are received for this curve, the already existing values of this curve are prevented from being used. As soon as no more new values are received, the value moves back to the original value of the curve over this period of time.

Arranger update interval (ms): This field sets the display update interval when recording automations.

VST automation resolution (samples): This is the time constant for sending automation values to VST plug-ins. When automation data is recorded, values are captured if they have changed within this time window. During playback, the values are sent to the VST plug-in at this interval.

> **ℹ Note**: *VST2 does not support sample-exact automation, only parameter changes at buffer limits. To increase the automation resolution, enter low sample values 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.

MIDI Resolution (ms): Time constant for sending the MIDI controller values. In this time interval the MIDI controller values are sent.

Destructive Effect Calculation

The Destructive Effects (↗227) options specify the form in which edited audio is saved when destructive effects are applied, if the original material is to be preserved for the Undo function.

Loudness

In this dialog window you can specify the basic settings for loudness normalization compliant with EBU R128/ ITU-R BS.1771. These values are used in the Loudness Adjustment (↗279) dialog and in the Loudness meter (↗559). Read there for more information!

Loading and Saving Settings

In this section you can save and load all program settings, project templates and other settings together. Thus, it is possible to save the complete program settings on a mobile storage medium and transfer them to other Sequoia workstations.
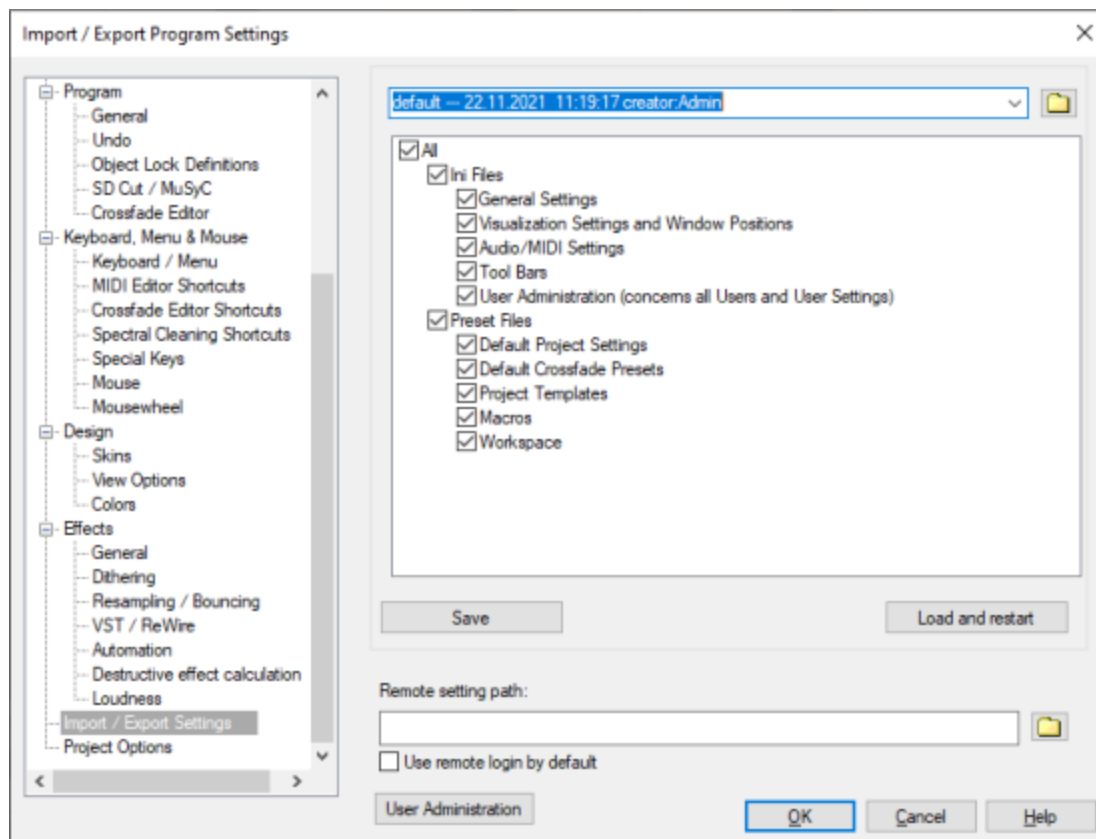
Saving Settings

Use the **Save** button to save your current project and program settings in a **Setting Container file (*.INZ)**. This always saves all program settings. By selecting the individual options in the tree structure when saving, you merely specify which options are pre-selected when you later select the file for loading. So they are a reminder for you that a setting container was intended only for a certain subset of settings.

From now on, the saved file will appear as a preset entry in the list box at the top.

Sequoia settings are provided for the following dialogs:

- General Settings (↗706)
- Project display (↗716) and window positions
- Audio (↗689)/MIDI (↗697) Options
- Toolbars (↗50)
- User administration (↗635) (users and user-specific settings in multi-user mode)
- Default project settings (↗643)
- Default crossfade templates (↗170)
- Project templates (↗642)

Loading Settings

Select a Settings Container from the list box above or click the folder button to load a *.INZ file from any folder.

Then specify in the tree structure which settings and templates should actually be loaded from the selected container file. Now press **Load and restart** to activate the settings. Sequoia then restarts. In addition, a backup container is created that contains the last settings before loading.

> ℹ️ *Tip: The settings can also be loaded at program start from the* start selection dialog (↗31)!
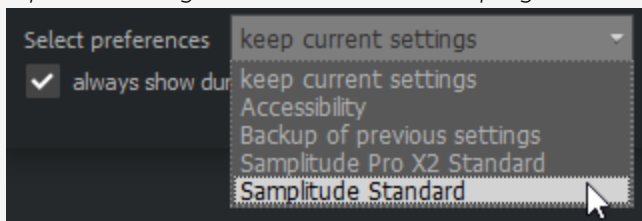
The Settings containers are stored in the program data folder C:\ProgramData\Magix\Sequoia 17.

There, in the **Customize** subfolder, you will also find a number of "INI patches" that do not contain complete sets of configuration files (*.ini), but only enable or disable 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.

> ℹ️ ***Note**: In multi-user mode (↗636), loading Setting Containers is not possible for users with the Fixed Settings (↗637) property and for users logged in via network login.*

Remote Settings Path

As a logged in administrator (↗635) you define here the network path in which the settings for multiple users can be managed. Read more in Setting up multi-user operation on network drive.
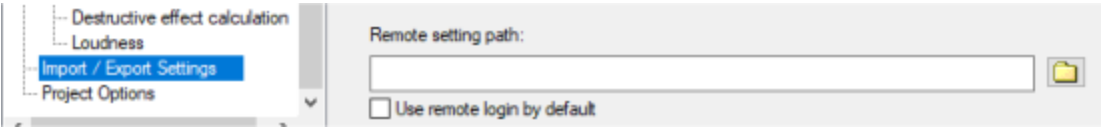
With the **Use remote login by default**option , login defaults to the login data for users stored on the specified network drive.

With **User administration**: open as a logged in administrator into the user administration, for all others the button is without function.

External Tools

This dialog is used to configure the transfer of objects to other audio programs for editing.
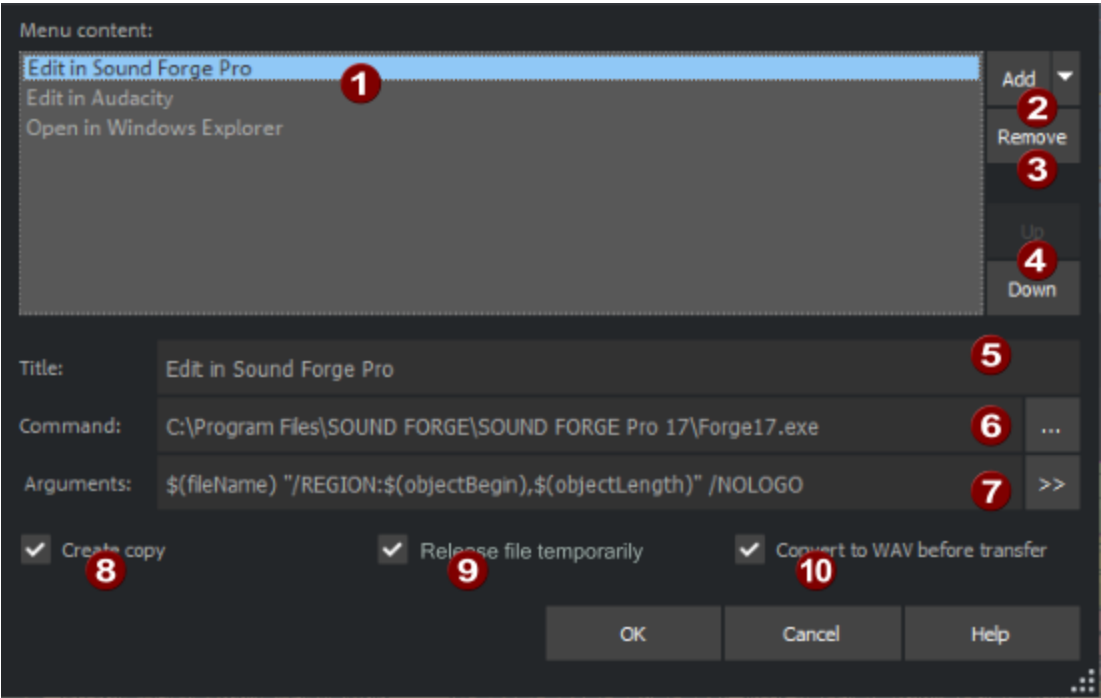
1 **Menu content**: All entries in this list create an entry in the context menu of an object. This entry can then be used to transfer the object to the corresponding external editor and edit it there.

② **Add**: Click here to create a new list entry. A file selection dialog opens where you can select the executable file in the program folder. If SOUND FORGE or SpectraLayers are installed, you can use the menu next to the button to immediately create suitable entries with all the correct arguments for these programs.

> ℹ️ *The entry **Open in Windows Explorer** opens an Explorer window at the location of the object's audio file and selects it.*

③ **Remove**: Deletes the selected entry. After deleting an entry and closing the dialog with **OK**, the entry is no longer displayed in the context menu of objects.

④ **Up/Down**: This can be used to change the order of the menu items.

⑤ **Title**: Enter text for the menu entry here.

⑥ **Command**: Here the name of the program with path is displayed. Use the **...** Button to open the file selection dialog, which allows you to select another file.

⑦ **Arguments**: Additional arguments for the external program can be specified here. In the menu next to it **>>** you can find predefined variables to insert into this command line to pass object parameters, such as object start and end, to the external program if the program supports it.

⑧ **Create copy**: This option, as with other destructive effect edits (↗227) of objects, ensures that the edits are made on a copy of the audio data to allow undoing.

⑨ **Release file temporarily**: This option is enabled by default and ensures that the file from Sequoia is released during editing in the external program so that changes can be written to this file. If no change is intended by the external program (such as when viewing in Explorer), this option can be disabled. This allows the project to be played back in Sequoia even after the external tool has been started.

⑩ **Convert to Wave before transfer**: Since some editors cannot open MP3, FLAC or video files, files in these formats are converted to Wave before opening the external editor and saved under a different 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.

> 💡 *Keyboard shortcuts can also be assigned to the context menu items. The corresponding entries in the tree structure at **Keyboard shortcuts and Edit menu** (↗709) are located at the very bottom at Menu **File** > **Program Preferences**.*

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